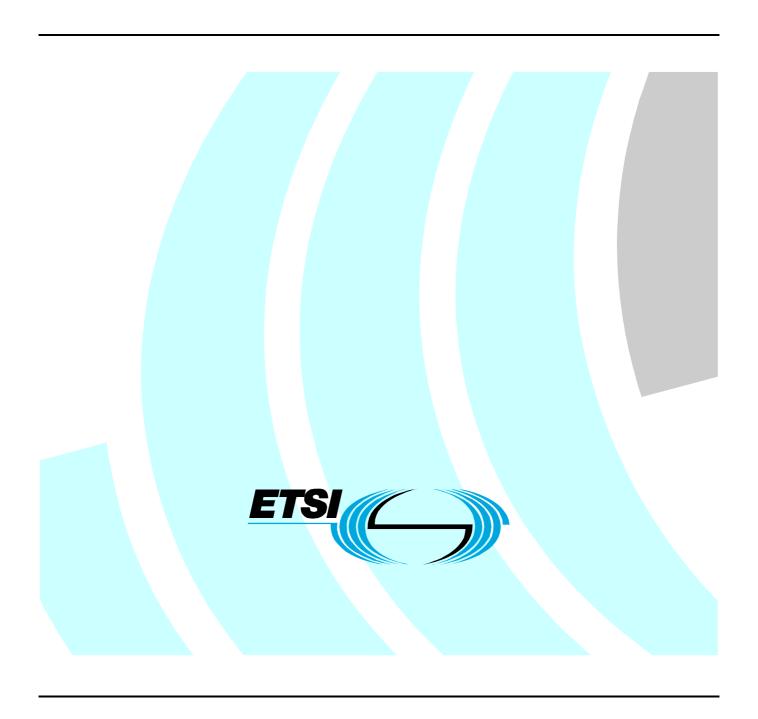
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Technical Specification

Technical Committee for IMS Network Testing (INT); Interworking between the 3GPP Cs domain with BICC or ISUP as signalling protocol and external SIP-I networks; Part 2: Test Suite Structure and Test Purposes (TSS&TP)



Reference

DTS/INT-00008

Keywords

BICC, ISUP, SIP, testing, TSS&TP

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Foreword

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

The present document is part 2 of a multi-part deliverable covering the Interworking between the 3GPP Cs domain with BICC or ISUP as signalling protocol and external SIP-I networks, as identified below:

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

1 Scope

The present document specifies the network Test Suite Structure and Test Purposes (TSS and TP) for Interworking between the 3GPP Cs domain with BICC or ISUP as signalling protocol and external SIP-I networks) described in the ITU-T Recommendation Q.1912.5 [1] and TS 129 164 [10].

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

2 References

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

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

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

2.1 Normative references

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

· ·	
[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 (1998): "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".
[10]	ETSI TS 129 164: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Interworking between the 3GPP Cs domain with BICC or ISUP as signalling protocol and external SIP-I networks (3GPP TS 29.164 version 8.0.0 Release 8)".

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

2.2 Informative references

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

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". [i.3] ITU-T Recommendation Q.731.7: "Malicious call identification (MCID)". [i.4] ITU-T Recommendation Q.732: "Call diversion services". 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)". [i.7] ITU-T Recommendation Q.734: "Stage 3 description for multiparty supplementary services using Signalling System No. 7: Conference calling". [i.8] ITU-T Recommendation Q.734.2: "Three-party service". [i.9] ITU-T Recommendation Q.735: "Closed user group (CUG)". [i.10] ITU-T Recommendation Q.737: "User-to-user signalling (UUS)". ITU-T Recommendation Q.784: "ISUP basic call test specification". [i.11] [i.12] ITU-T Recommendation Q.764: "Signalling System No. 7 - ISDN User Part signalling procedures". [i.13] ITU-T Recommendation T.38: "Procedures for real-time Group 3 facsimile communication over IP networks". [i.14] ITU-T Recommendation Q.1902.4: "Bearer independent call control protocol (Capability Set 2): Basic call procedures". IETF RFC 4040: "RTP Payload Format for a 64 kbit/s Transparent Call". [i.15] [i.16] ETSI EN 300 403-1: "Integrated Services Digital Network (ISDN); Digital Subscriber Signalling System No. one (DSS1) protocol; Signalling network layer for circuit-mode basic call control; Part 1: Protocol specification [ITU-T Recommendation Q.931 (1993), modified]".

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

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

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

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

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

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

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

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

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

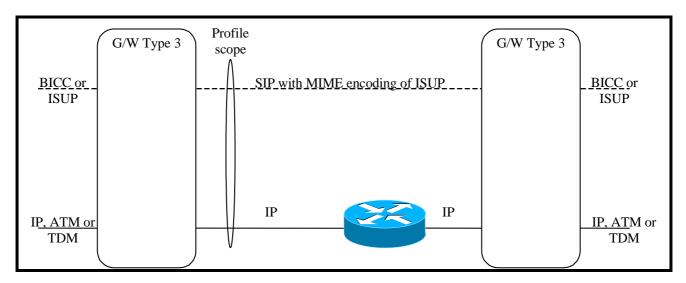


Figure 1: Profile scope for SIP with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateways

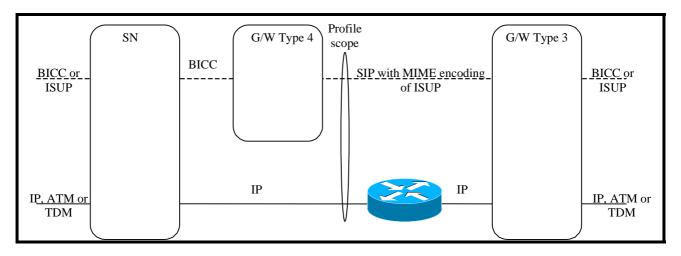


Figure 2: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 3 and 4 gateways

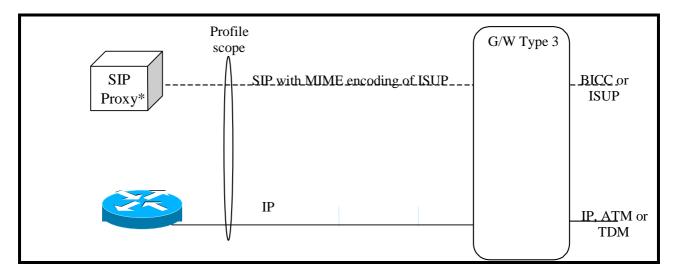


Figure 3: Profile scope for SIP with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateways

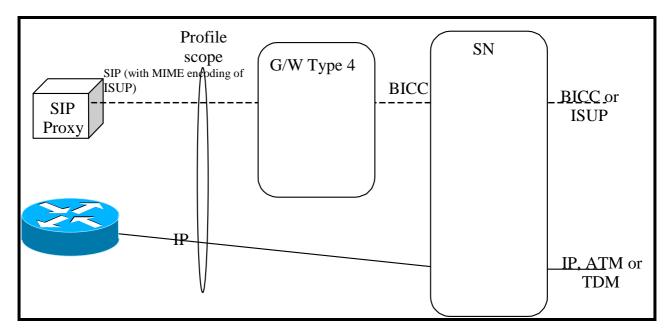


Figure 4: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 4 gateway

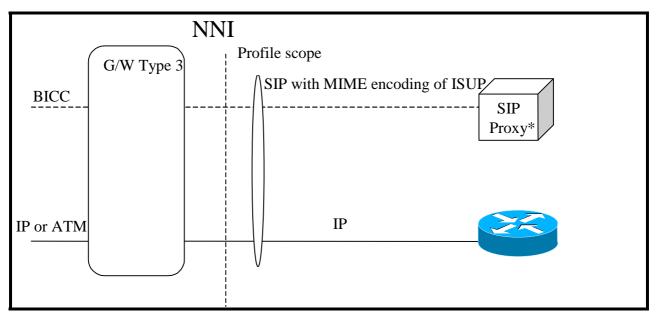


Figure 5: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateway

3.2 Abbreviations

ACM

Three-Party

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

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
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

COT COnTinuity message
CPG Call Progress Message
CPS Calling Party's Category
CTNb ConnecTed Number
CUG Closed User Group
CW Call Waiting

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 FunctionMHSMessage Handling System

MIME Multi-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 **SUSPEND** SUS SUT System Under Test Transmission Medium Requirement **TMR** 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 User-to User message **USR** 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	TP110xxx
	message	
	Receipt of ReSet Circuit message (RSC), circuit Group ReSet	TP111xxx
	message (GRS) or Circuit Group Blocking message (CGB)	
	with the indication hardware failure oriented	
	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 message (GRS) or Circuit Group Blocking message (CGB) with the indication hardware failure oriented	TP309xxx
	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 - PROGRESS Message	TP603xxx
	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: RF0	C 3261 [4]]			SUP reference:	
	Q.1912.5 [1], c					5 [1], clause 6.1.2 (i,1)	
TSS reference	SIP-ISUP/Basic call/Sendir	ng of the I	nitial Add	ress Me	essage (IAM)	
SIP selection	NOT PICS 4/4 AND NOT P	PICS 4/5					
criteria							
ISUP selection	NOT PICS 1/6						
criteria							
Test purpose	Ensure that if the SUT upor	n receipt o	of the first	INVITE	E with su	ufficient digits, with a SDP	
	offer:						
	 the SUT shall delete μ 	-law (PCN	/IU), if pre	sent, fr	om the i	media description that it will	
	send back in the SDP	answer;					
	 the SUT shall immedia 	itely send	out the IA	۱M.			
SIP parameter	SIP INVITE: Audio RTP/AV	/P 0 8					
values	200 OK: Audio RTP/AVP 8						
ISUP parameter values	IAM USI: A-law or absent						
Comments	SIP-I		SL	ΙΤ		ISUP	
	INVITE(IAM)	→			→	IAM	
	180 Ringing(ACM)	+			+	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
	ACK	→					
			Conver	sation			
	BYE(REL)	→			→	REL	
	200 OK BYE(RLC)	+			+	RLC	

TP101002	SIP reference: RFC 3	3261 [4]			ISUP reference:	
				Q.191	2.5 [1], clause 6.1.2 (i,2ai)	
TSS reference	SIP-ISUP/Basic call/Sending	of the Init	ial Add			
SIP selection	PICS 4/4 AND PICS 4/5			,		
criteria						
ISUP selection	PICS 1/4 AND NOT PICS 1/6	AND PIC	S 4/1			
criteria						
Test purpose					n sufficient digits, with a SDP	
	offer 100rel extensions and p					
	 the SUT shall delete μ-la 	w (PCMU), if pre	sent, from t	he media description that it wi	ill
	send back in the SDP an	iswer;				
					de with the coding of the Natu	ıre
	of Connection Indicators		r: "CO	T to be exp	ected".	
SIP parameter	SIP INVITE: Audio RTP/AVP	8 0				
values	200 OK: Audio RTP/AVP 8					
ISUP parameter				ted, USI: A	-law or absent	
values	<u> </u>	continuity			I	
Comments	SIP-I		SL		ISUP	
	INVITE(IAM)	→		-	▶ IAM	
	183 Session Progress	+				
	PRACK	→				
	200 OK PRACK	+				
	UPDATE	→		-	▶ COT	
	200 OK UPDATE	+				
		1	econditi	ons met		
	180 Ringing(ACM)	+			E ACM	
	200 OK INVITE(ANM)	+		•	- ANM	
	ACK	→				
			Conver			
	BYE(REL)	→			▶ REL	
	200 OK BYE(RLC)	←		•	RLC	

TP101003	SIP reference: RFC 3	3261 [4]		0 191	ISUP reference: 2.5 [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	OI THE III	itiai / taa	icoo inicooaç	<i>yo (ii tivi)</i>		
criteria	1 100 4/4 / (100 4/0						
ISUP selection	PICS 1/4 AND NOT PICS 1/6	S AND PI	CS 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 preconditions extensions in the SIP Require header: the SUT shall delete μ-law (PCMU), if present, from the media description that it will send back in the SDP answer; 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". 						
SIP parameter	SIP INVITE: Audio RTP/AVP	•					
values	200 OK: Audio RTP/AVP 8						
ISUP parameter	IAM Continuity Indicator: C	COT to b	e expec	ted, USI: A-	law or absent		
values	COT Continuity Indicator: c	ontinuit	y				
Comments	SIP-I		SL	JT	ISUP		
	INVITE(IAM)	→			▶ IAM		
	183 Session Progress	+					
	PRACK	→					
	200 OK PRACK	+					
	UPDATE	→		1	COT		
	200 OK UPDATE	+					
		P	reconditi	ons met			
	180 Ringing(ACM)	+		•	ACM		
	200 OK INVITE(ANM)	+		•	- ANM		
	ACK	→					
			Conver	sation			
	BYE(REL)	→		-	REL		
	200 OK BYE(RLC)	+		•	RLC		

TP101004	SIP reference: RFC 3261 [4]				12.5	SUP reference: [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	PICS 1/5 AND NOT PICS 1/6	2 AND DI	CC 4/4				
criteria	PICS 1/5 AND NOT PICS 1/6	S AND PI	CS 4/ I				
Test purpose	 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: the SUT shall delete μ-law (PCMU), if present, from the media description that it will send back in the SDP answer; 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". 						
SIP parameter	SIP INVITE: Audio RTP/AVP						
values	200 OK: Audio RTP/AVP 8						
ISUP parameter						his circuit or continuity	
values						circuit, USI: A-law or absent	
	·	continuit	•	success	ful		
Comments	SIP-I		SU	ΙΤ		ISUP	
	INVITE(IAM)	→			→	IAM	
	183 Session Progress	→					
	PRACK	+					
	200 OK PRACK	→					
	UPDATE	→			→	COT	
	200 OK UPDATE	+					
			reconditi	ons met			
	180 Ringing(ACM)	+			←	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
	ACK	→					
			Convers	sation			
	BYE(REL)	→			→	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	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		., .					
Test purpose	 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: the SUT shall delete μ-law (PCMU), if present, from the media description that it will send back in the SDP answer; 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". 							
SIP parameter	SIP INVITE: Audio RTP/AVP							
values	200 OK: Audio RTP/AVP 8							
ISUP parameter						this circuit or continuity		
values		check perfo				circuit, USI: A-law or absent		
Comments	SIP-I		SU	T		ISUP		
	INVITE(IAM)	→			→	IAM		
	183 Session Progress	→						
	PRACK	+						
	200 OK PRACK	→						
	UPDATE	→			→	COT		
	200 OK UPDATE	+						
		Pred	ondition	ons met				
	180 Ringing(ACM)	(+	ACM		
	200 OK INVITE(ANM)	+			←	ANM		
	ACK	→						
		Co	onvers	sation				
	BYE(REL)	→			→	REL		
	200 OK BYE(RLC)	←			+	RLC		

TP101006	SIP reference: RFC 3	3261 [4]	_	-	SUP reference:
						5 [1], clause 6.1.2 (i,2b)
TSS reference	SIP-ISUP/Basic call/Sending	of the I	nitial Addı	ress Mess	age (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					
			ИU), if pre	sent, from	the r	media description that it will
	send back in the SDP an	,				
	 the IAM shall be deferred 		II precond	litions hav	e bee	en 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						
Comments	SIP-I		SU	IT		ISUP
	INVITE(IAM)	→				
	183 Session Progress	←				
	PRACK	→				
	200 OK PRACK	+				
	UPDATE	→			→	IAM
	200 OK UPDATE	+				
			Preconditi	ons met		
	180 Ringing(ACM)	+			+	ACM
	200 OK INVITE(ANM)	+			←	ANM
	ACK	^				
			Convers	sation		
	BYE(REL)	→			→	REL
	200 OK BYE(RLC)	+			+	RLC

TP101007	SIP reference: RFC	3261 [4]	0.19	_	SUP reference: 5 [1], clause 6.1.2 (i,2b)		
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	9 00 .			<u>.gc</u>	()		
criteria								
ISUP selection	NOT PICS 1/6 AND PICS 4	/1						
criteria				INDUTE:		(C.)		
Test purpose	Ensure that if the SUT upon							
	offer 100rel extensions and							
	• the SUT shall delete μ-l send back in the SDP a		vio), ii pre	sent, from	tne	media description that it will		
		,	II process	litiana have	. h.a.	on mot		
SIP parameter	the IAM shall be deferred SIP INVITE: Audio RTP/AVI		ii preconc	illions nave	bee	en met.		
values	200 OK: Audio RTP/AVP 8	- 0 0						
ISUP parameter	IAM USI: A-law or absent							
values	IAW 031. A-law 01 absent							
Comments	SIP-I		SL	JT		ISUP		
	INVITE(IAM)	→						
	183 Session Progress	+						
	PRACK	→						
	200 OK PRACK	+						
	UPDATE	→			→	IAM		
	200 OK UPDATE	+						
			Preconditi	ons met				
	180 Ringing(ACM)	+			←	ACM		
	200 OK INVITE(ANM)	+			←	ANM		
	ACK	→						
			Conver	sation				
	BYE(REL)	→			→	REL		
	200 OK BYE(RLC)	+			←	RLC		

TP101008	SIP reference: RFC 3	3261 [4]	Į;	SUP reference:
			Q.1912.	5 [1], clause 6.1.2 (i,1)
TSS reference	SIP-ISUP/Basic call/Sending	of the Initial Add	lress Message (IAM)
SIP selection	NOT PICS 4/4 AND NOT 4/5			
criteria				
ISUP selection	PICS 1/6			
criteria				
Test purpose	Ensure that if the SUT upon r	eceipt of the firs	t INVITE with su	ifficient digits, with a SDP
	offer:			
) and μ-law (PCMU) were
	present in the offer of the	media descripti	on, that it will se	end it back in the SDP
	answer;			
	 the SUT shall immediate 	ly send out the I	AM.	
SIP parameter	SIP INVITE: Audio RTP/AVP	0 8		
values	200 OK: Audio RTP/AVP 0			
ISUP parameter	IAM USI: μ-law			
values				
Comments	SIP-I	SI	JT	ISUP
	INVITE(IAM)	→	→	IAM
	180 Ringing(ACM)	+	+	ACM
	200 OK INVITE(ANM)	+	+	ANM
	ACK	→		
		Conve	rsation	
	BYE(REL)	→	→	REL
	200 OK BYE(RLC)	+	+	RLC

TP101009	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 I	nitial Add	ress Mess	sage	(IAM)
SIP selection	PICS 4/4 AND PICS 4/5					
criteria						
ISUP selection	PICS 1/4 AND PICS 1/6 AND	PICS 4	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 preconditions extensions in the SIP Supported header: 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: 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". 					
SIP parameter	SIP INVITE: Audio RTP/AVP					
values	200 OK: Audio RTP/AVP 0					
ISUP parameter	IAM USI: μ-law; Nature of Co	nnectio	n Indicato	rs param	eter:	"COT to be expected" COT;
values	Continuity Indicator: continui			•		•
Comments	SIP-I		SU	ΙΤ		ISUP
	INVITE(IAM)	→			1	IAM
	183 Session Progress	+				
	PRACK	→				
	200 OK PRACK	←				
	UPDATE	→			1	COT
	200 OK UPDATE	←				
		I	Preconditi	ions met		
	180 Ringing(ACM)	←			+	ACM
	200 OK INVITE(ANM)	+			+	ANM
	ACK	→				
			Conver	sation		
	BYE(REL)	→			1	REL
	200 OK BYE(RLC)	+			+	RLC

TP101010	SIP reference: RFC	3261 [4	1			ISUP reference:	
		•	•	Q.	1912.	5 [1], clause 6.1.2 (i,2ai)	
TSS reference	SIP-ISUP/Basic call/Sending	of the I	nitial Add	ress Mes	ssage	(IAM)	
SIP selection	PICS 4/4 AND PICS 4/5						
criteria							
ISUP selection	PICS 1/4 AND PICS 1/6 AND	PICS 4	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						
						A) and μ-law (PCMU) were	
	present in the offer of the	e media	description	on, that i	t will s	end it back in the SDP	
	answer;						
						with the coding of the Nature	
	of Connection Indicators	•	eter: " CO	Γto be e	expect	ted".	
SIP parameter	SIP INVITE: Audio RTP/AVP	0 8					
values	200 OK: Audio RTP/AVP 0						
ISUP parameter			n Indicato	ors paran	neter:	"COT to be expected" COT;	
values	Continuity Indicator: continu	ity				I	
Comments	SIP-I		SL	JT		ISUP	
	INVITE(IAM)	→			→	IAM	
	183 Session Progress	+					
	PRACK	→					
	200 OK PRACK	+					
	UPDATE	→			→	СОТ	
	200 OK UPDATE	+					
			Preconditi	ions met			
	180 Ringing(ACM)	+			+	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
	ACK	→					
			Conver	sation			
	BYE(REL)	→			→	REL	
	200 OK BYE(RLC)	←			+	RLC	

TP101011	SIP reference: RFC 3	3261 [4]		O 1912	ISUP reference: 5 [1], clause 6.1.2 (i,2aii)		
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)						
SIP selection	PICS 4/4 AND PICS 4/5	or aro mila	ai 7 iaai	occ meccage	(ii iiii)		
criteria							
ISUP selection	PICS 1/5 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 preconditions extensions in the SIP Supported header: 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; 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". 						
SIP parameter	SIP INVITE: Audio RTP/AVP						
values	200 OK: Audio RTP/AVP 0						
ISUP parameter	IAM: USI: μ-law; Continuity ch	neck indica	ator "co	ntinuity che	ck required on this circuit"		
values	or continuity check pe				-		
	COT Continuity Indicator: cor	ntinuity ch	neck si	uccessful			
Comments	SIP-I		SU		ISUP		
	INVITE(IAM)	→		→	IAM		
	183 Session Progress	+					
	PRACK	→					
	200 OK PRACK	+					
	UPDATE	→		→	COT		
	200 OK UPDATE	+					
		Pre	conditi	ons met			
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	→					
			convers	sation			
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	←		+	RLC		

TP101012	SIP reference: RFC 3			ISUP 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 PICS 1/6 AND	PICS 1/5 AND PICS 1/6 AND PICS 4/1						
Test purpose	 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: 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; 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". 							
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 perfo	ormed on previ	ous circuit	ck required on this circuit"				
Commonto	COT Continuity Indicator: cor			LIGUID				
Comments	SIP-I INVITE(IAM)	→	UT →	ISUP IAM				
	183 Session Progress	-	7	IAW				
	PRACK	→						
	200 OK PRACK	(
	UPDATE	→	→	СОТ				
	200 OK UPDATE	Process	itions met					
	180 Ringing(ACM)	+ Precond	tions met	ACM				
	200 OK INVITE(ANM)	-		ANM				
	ACK	→		/ALVIVI				
	7,010	-	ersation					
	BYE(REL)	→	→	REL				
	200 OK BYE(RLC)	+	+	RLC				

TP101013	SIP reference: RF	C 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	 offer 100rel extensions and the SUT shall delete A present in the offer of the answer; 	present in the offer of the media description, that it will send it back in the SDP						
SIP parameter	SIP INVITE: Audio RTP/AV							
values	200 OK: Audio RTP/AVP 0							
ISUP parameter	IAM USI: μ-law							
values	·							
Comments	SIP-I		SU	Т	ISUP			
	INVITE(IAM)	→						
	183 Session Progress	+						
	PRACK	→						
	200 OK PRACK	+						
	UPDATE	→		→	IAM			
	200 OK UPDATE	+						
		F	Preconditi	ons met				
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	→						
			Convers	sation				
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP101014	SIP reference: RFC 3261 [4]			ISUP reference:				
				Q.1912	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	PICS 4/4 AND PICS 4/5						
criteria								
ISUP selection criteria	PICS 1/6 AND PICS 4/1							
Test purpose	 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: 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; the IAM shall be deferred until all preconditions have been met. 							
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	Т	ISUP			
	INVITE(IAM)	→						
	183 Session Progress	←						
	PRACK	→						
	200 OK PRACK	←						
	UPDATE	→		→	IAM			
	200 OK UPDATE	←						
		L .	reconditi	ons met				
	180 Ringing(ACM)	←		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	→						
			Convers	sation				
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	←		←	RLC			

TP101015	SIP reference: RFC	3261 [4]		ISUP reference: 1], clauses 6.1.3.2, 6.1.3.3, 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	 Ensure that the SUT on receipt of an INVITE message sends an IAM message, where: the Calling party's category is generated from the Calling Party's Category present in the encapsulated IAM; the Nature of Connection Indicators (NCI) is generated by the MGCF using the Nature of Connection Indicators received in the encapsulated IAM; 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. 							
SIP parameter values								
ISUP parameter values								
Comments	SIP-I	SI	JT	ISUP				
	INVITE(IAM)	→	→	IAM				
	180 Ringing(ACM)	+	+	ACM				
	200 OK INVITE(ANM)	+	+	ANM				
	ACK	→						
		Conve	rsation					
	BYE(REL)	→	→	REL				
	200 OK BYE(RLC)	+	+	RLC				

TP101015A	SIP reference: RF	C 3261 [4]	ES 283 02	SUP reference: 7 [11], clause 7.2.3.1.2.5 4 [10], clause 6.2.4.1.3.2		
TSS reference	SIP-ISUP/Basic call/Sendi	ng of the Initial Add	dress message (IAM)		
SIP selection criteria	Based on table 1A					
ISUP selection criteria						
Test purpose	Mapping of SDP into the TMR					
		in table 1 with the ' I the media descrip	'a =" "b =" and "rotion does not ma			
	to TMR_VALUE derived from the USI is set in parallel to		cription. The Info	rmation transfer capability in		
SIP Parameter values	INVITE; a_b_m_LINE_VAL	LUE				
ISUP Parameter values	IAM; TMR: ISUP_TMR					
Comments	SIP	S	UT	ISUP		
	INVITE (IAM)	→	→	IAM		
	180 Ringing (ACM)	(+	ACM		
	Ringing tone					
	200 OK INVITE (ANM)	+	ANM			
	ACK	→				
		Conve	rsation			
	BYE (REL)	→	→	REL		
	200 OK BYE (RLC)	←	+	RLC		

Table 1A

	_			s for test purposes TP10	1015A	
			a_b_m	_LINE_VALUE		
		m= line		b= line	a= line	TMR_VALUE
test purposes	<media></media>	<transport></transport>	<fmt-list></fmt-list>	<modifier>:<bandwidt h-value=""></bandwidt></modifier>	rtpmap: <dynamic-pt> <encoding name="">/<clock rate="">[/encoding parameters></clock></encoding></dynamic-pt>	TMR codes
VA_01	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap: <dynamic-pt> PCMU/8000</dynamic-pt>	"3,1 KHz audio"
VA_02	audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A	"3,1 KHz audio"
VA_03	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap: <dynamic-pt> PCMA/8000</dynamic-pt>	"3,1 KHz audio"
VA_04	audio	RTP/AVP	Dynamic PT	AS: 64 kbit/s	rtpmap: <dynamic-pt> CLEARMODE/8000 (see note 2)</dynamic-pt>	"64 kbit/s unrestricted"
VA_05	image	udptl	t38	N/A or up to 64 kbit/s	Based on ITU-T T.38 [i.13]	"3,1 KHz audio"
VA_06	image	tcptl	t38	N/A or up to 64 kbit/s	Based on ITU-T T.38 [i.13]	"3,1 KHz audio"
VA_07	audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A	"3,1 KHz audio"

- NOTE 1: In this table the codec G.711 is used only as an example. Other codecs are possible.
- NOTE 2: CLEARMODE is specified in RFC 4040 [i.15].

 NOTE 3: If the b=line indicates a bandwidth greater than 64 kbit/s then the call may use compression techniques or reject the call with a 415 response indicating that only one media stream of 64 kbit/s is supported.
- NOTE 4: <bandwidth value> for <modifier> of AS is in units of kbit/s.

P101016	SIP reference: RF0	C 3261 [4]]		Į:	SUP reference:		
		Q.1912.5 [1						
TSS reference	SIP-ISUP/Basic call/Sendir	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	Idle state	on receip	t of an IN	VITE	message with an		
	encapsulated IAM message							
	The TMR and USI shall be	taken from	m the enc	apsulated	ISUF) _:		
	 sends an IAM message from the encapsulated 		e Transm	ission Me	dium	Requirement (TMR) taken		
SIP parameter	SIP INVITE							
values								
ISUP parameter values	IAM; USI; ISDN_BC_ITR; 7	ΓMR						
Comments	SIP-I		SU	T		ISUP		
	INVITE(IAM)	→			→	IAM		
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
	ACK →							
			Convers	sation				
	BYE(REL)	→			→	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		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]			ISUP reference: Q.1912.5 [1], clause 6.1.3.5			
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)						
SIP selection	NOT PICS 4/4 and NOT PI			-			
criteria							
ISUP selection criteria							
Test purpose	Ensure that the SUT in the encapsulated IAM messag sends an IAM messag	e the HLC sh	all be taken	from the	encapsulated ISUP:		
SIP parameter	INVITE ;						
values	·						
ISUP parameter values	IAM; Access transport pa	rameter HL	C: HLC_VAL	UE; USI			
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		→	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK →						
		C	conversation				
	BYE(REL)	→		→	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: Q.1912.5 [1], clause 6.1.3.9				
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)							
SIP selection criteria		NOT PICS 4/4 and NOT PICS 4/5						
ISUP selection criteria	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 paramete	r value						
Comments	SIP-I		SU	Т		ISUP		
	INVITE(IAM)	→			→	IAM		
	180 Ringing(ACM)	+			←	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
	ACK							
			Conver	sation				
	BYE(REL)	→			→	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP101019	SIP reference: RF0	C 3261 [4]		ISUP reference:			
				2.5 [1], clause 6.1.3.1			
TSS reference	SIP-ISUP/Basic call/Sendir			(IAM)			
SIP selection	PICS 1/9 AND NOT PICS 4	1/4 and NOT PICS	5 4/5				
criteria							
ISUP selection	NOT PICS 1/7						
criteria							
Test purpose	contained in the user info c	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:					
	Analyse the information contained in received URI with user=phone, and if it is in the format: +CC NDC SN where CC is the country code of the network in which the next hop terminates, then set Nature of Address indicator to "National (significant) number", remove "+CC" 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: NDC SN.						
SIP parameter							
values							
ISUP parameter	IAM: Called party number						
values				lious			
Comments	SIP-I		UT	ISUP			
	INVITE(IAM))	→	IAM			
	180 Ringing(ACM)	+	-	ACM			
	200 OK INVITE(ANM)	+	+	ANM			
	ACK	→					
			rsation				
	BYE(REL)	→	→	REL			
	200 OK BYE(RLC)	←	+	RLC			

TP101020	SIP reference: RFC	3261 [4]		ISUP reference:				
T00 (Q.1912.5 [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	5 4/5					
criteria	7.00 / /							
ISUP selection	PICS 1/7							
criteria								
Test purpose	Ensure that the SUT on rece			a Called party number				
	contained in the user info co							
	Send an IAM Message with		number coded a	s follows:				
	 Nature of address indicates 							
		contained in rec	eived URI with	user=phone, and if it is in the				
	format:	S	1 64					
				network in which the next hop				
				rnational number", remove				
	"+" and use the remaining							
				etwork number not allowed.				
	 Numbering plan Indica 	•	elephony) nun	nbering plan.				
	 Address Signals CC ND 	C SN.						
SIP parameter								
values								
ISUP parameter	IAM: Called party number							
values								
Comments	SIP-I	S	UT	ISUP				
	INVITE(IAM)	→	→	IAM				
	180 Ringing(ACM)	+	←	ACM				
	200 OK INVITE(ANM)	+	+	ANM				
	ACK	→						
		Conve	rsation					
	BYE(REL)	→	→	REL				
	200 OK BYE(RLC)	+	+	RLC				

TP101021	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	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)						
SIP selection	NOT PICS 4/4 AND NOT PIC	CS 4/5 AI	ND PICS	1/9				
criteria								
ISUP selection								
criteria								
Test purpose	a-Law, then independent fr	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 RT	P/AVP 0	8			·		
values	Answer: m=audio 4712 RT	P/AVP 8	0					
ISUP parameter								
values								
Comments	SIP-I		SU	Τ		ISUP		
	INVITE(IAM)	→			→	IAM		
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
	ACK	ACK →						
			Convers	sation				
	BYE(REL)	→			→	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP101022	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 Add	ress Message	(IAM)
SIP selection	PICS 4/4 AND PICS 4/5 AND			
criteria				
ISUP selection	PICS 1/4 AND NOT PICS 1/6	AND PICS 4/1		
criteria				
Test purpose	Ensure that the SUT on recei	pt of an INVITE	message with a	SDP offer for µ-Law and
	a-Law 100rel extensions and	preconditions ex	ktensions in the	SIP Supported header, then
	independent from the receive	ved order of pro	eference:	
				with the coding of the Nature
	of Connection Indicators	parameter: "CO	T to be expect	ed";
	 the G.711 a-law codec sł 	nall be returned	in the SDP ans	wer 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	IAM Continuity Indicator: C	OT to be exped	cted, USI: A-lav	v or absent
values	COT Continuity Indicator: c	ontinuity		
Comments	SIP-I	Sl	JT	ISUP
	INVITE(IAM)	→	→	IAM
	183 Session Progress	+		
	PRACK	→		
	200 OK PRACK	+		
	UPDATE	→	→	COT
	200 OK UPDATE	+		
	180 Ringing(ACM)	+	+	ACM
	200 OK INVITE(ANM)	+	+	ANM
	ACK	→		
		Conver	sation	
	BYE(REL)	→	→	REL
	200 OK BYE(RLC)	+	+	RLC

TP101023	SIP reference: RFC 3	3261 [4]			I	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	PICS 1/4 AND NOT PICS 1/6	AND PIC	3 4/1				
criteria							
Test purpose	Ensure that the SUT on recei						
	a-Law 100rel extensions and					SIP Require header, then	
	independent from the recei						
						with the coding of the Nature	
	of Connection Indicators						
	 the G.711 a-law codec sl 	hall be retu	rned i	n the SD	P ans	wer as preferred codec.	
SIP parameter	Offer: m=audio 4711 RTF						
values	Answer: m=audio 4712 RTF						
ISUP parameter	1		exped	ted, US	I: A-lav	v or absent	
values		ontinuity					
Comments	SIP-I		SL	<u>IT</u>		ISUP	
	INVITE(IAM)	→			→	IAM	
	183 Session Progress	+					
	PRACK	→					
	200 OK PRACK	+					
	UPDATE	→			→	COT	
	200 OK UPDATE	(
	180 Ringing(ACM)	(←	ACM	
	200 OK INVITE(ANM)	(←	ANM	
	ACK	→					
		С	onver	sation			
	BYE(REL)	→			→	REL	
	200 OK BYE(RLC)	+	•		+	RLC	

TP101024	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 1/5 AND NOT PICS 1/6	AND PICS 4/1					
criteria							
Test purpose	Ensure that the SUT on recei						
	a-Law 100rel extensions and			SIP Supported header, then			
	independent from the recei						
	the IAM shall be sent out in disease "senting its above."						
	indicator "continuity che performed on previous		tnis circuit or	continuity cneck			
	1 . 0 - 44		in the CDD and	war as professed and a			
SIP parameter	offer: m=audio 4711 RTF		III LITE SUP ATIST	wer as preferred codec.			
values	Answer: m=audio 4711 RTF						
ISUP parameter			k required on t	his circuit or continuity			
values				circuit, USI: A-law or absent			
values		ontinuity chec		circuit, ooi. A-law of absent			
Comments	SIP-I		JT	ISUP			
	INVITE(IAM)	→	→	IAM			
	183 Session Progress	+					
	PRACK	→					
	200 OK PRACK	+					
	UPDATE	→	→	СОТ			
	200 OK UPDATE	+					
	180 Ringing(ACM)	+	+	ACM			
	200 OK INVITE(ANM)	+	+	ANM			
	ACK	→					
		Conve	rsation				
	BYE(REL)	→	→	REL			
	200 OK BYE(RLC)	+	(RLC			

TP101025	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					
criteria							
ISUP selection	PICS 1/5 AND NOT PICS 1/6	AND PICS 4/1					
criteria							
Test purpose	Ensure that the SUT on recei						
				the SIP Require header, then			
	independent from the recei						
				de with the Continuity check			
	indicator "continuity che		n this circuit'	or "continuity check			
	performed on previous						
			in the SDP a	nswer as preferred codec.			
SIP parameter	Offer: m=audio 4711 RTF						
values	Answer: m=audio 4712 RTF						
ISUP parameter				n this circuit or continuity			
values		neck perform continuity che		us circuit, USI: A-law or absent			
Comments	, , , , , , , , , , , , , , , , , , , ,			ISUP			
Comments	SIP-I	→	SUT =				
	INVITE(IAM)	-	7	IAM			
	183 Session Progress	-					
	PRACK	7					
	200 OK PRACK	→		N COT			
	UPDATE	-		COT			
	200 OK UPDATE	-	•	- ACNA			
	180 Ringing(ACM)			7.0			
	200 OK INVITE(ANM) ACK	←		ANM			
	ACK	- 1					
	DVE(DEL)		ersation	N DEI			
	BYE(REL)	→					
	200 OK BYE(RLC)	7	•	RLC			

TP101026	SIP reference: RFC 3	3261 [4]	1	FN 383	ISUP reference: 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	PICS 4/4 AND PICS 4/5 AND			Tood Moodage	, (n (ivi)			
criteria	1100 1/1/1121100 1/071112	00	., 0					
ISUP selection	PICS 1/5 AND NOT PICS 1/6	AND N	IOT 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: the shall be deferred until all preconditions have been met; the G.711 a-law codec shall be returned in the SDP answer as preferred codec.							
SIP parameter	Offer: m=audio 4711 RTF				•			
values	Answer: m=audio 4712 RTF	P/AVP 8	3 0					
ISUP parameter								
values								
Comments	SIP-I		SU	IT	ISUP			
	INVITE(IAM)	→						
	183 Session Progress	+						
	PRACK	→						
	200 OK PRACK	+						
	UPDATE	→		→	IAM			
	200 OK UPDATE	+						
	180 Ringing(ACM)	+		←	ACM			
	200 OK INVITE(ANM)	+		←	ANM			
	ACK	→						
			Conver					
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		←	RLC			

TP101027	SIP reference: RFC	3261 [4]				SUP reference:	
				EN 3	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 AND	D PICS 1/	9				
criteria							
ISUP selection	PICS 1/5 AND NOT PICS 1/6	6 AND NO	OT PICS	3 4/1			
criteria							
Test purpose	Ensure that the SUT on rece						
	a-Law 100rel extensions and				n the	SIP Require header, then	
	independent from the recei		-				
	 the shall be deferred unt 						
	the G.711 a-law codec s			n the SDF	ans	wer as preferred codec.	
SIP parameter	Offer: m=audio 4711 RT		-				
values	Answer: m=audio 4712 RT	P/AVP 8	0				
ISUP parameter							
values		1 1				I	
Comments	SIP-I		SU	JT		ISUP	
	INVITE(IAM)	→					
	183 Session Progress	+					
	PRACK	→					
	200 OK PRACK	+					
	UPDATE	→			→	IAM	
	200 OK UPDATE	+					
	180 Ringing(ACM)	←			←	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
	ACK	→					
			Convers	sation			
	BYE(REL)	→			→	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 In	itial Addr	ess Mess	sage (IAM)
SIP selection	NOT PICS 4/4 AND NOT PIC	S 4/5 AI	ND 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 RTF	P/AVP 8				
values	Answer: m=audio 4711 RTF	P/AVP 8				
ISUP parameter values						
Comments	SIP-I		SU	Т		ISUP
	INVITE(IAM)	→			→	IAM
	180 Ringing(ACM)	+			+	ACM
	200 OK INVITE(ANM)	+			+	ANM
	ACK	→				
			Convers	sation		
	BYE(REL)	→			→	REL
	200 OK BYE(RLC)	+			+	RLC

TP101029	SIP reference: RFC	3261 [4]			I	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 AN	ID PICS 1	1/9					
criteria								
ISUP selection	PICS 1/4 AND NOT PICS 1.	/6 AND P	ICS 4/1					
criteria								
Test purpose						SDP offer for a-Law and no		
						SIP Supported header, then		
	independent the normal o							
						with the coding of the Nature		
	of Connection Indicator							
	the G.711 a-law codec			n the SD	P ans	ver.		
SIP parameter		Offer: m=audio 4711 RTP/AVP 8						
values	Answer: m=audio 4711 R							
ISUP parameter	IAM Continuity Indicator:			ted, USI	: A-lav	v or absent		
values	COT Continuity Indicator:	continuit		_	1	liou in		
Comments	SIP-I		SU			ISUP		
	INVITE(IAM)	→			→	IAM		
	183 Session Progress	+						
	PRACK	→						
	200 OK PRACK	+						
	UPDATE	→			→	COT		
	200 OK UPDATE	+						
			Preconditi	ons met	1 -			
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
	ACK	→						
			Convers	sation				
	BYE(REL)	→			→	REL		
	200 OK BYE(RLC)	+			←	RLC		

TP101030	SIP reference: RFC 3	3261 [4]			I	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	PICS 1/4 AND NOT PICS 1/6	AND P	ICS 4/1				
criteria							
Test purpose	Ensure that the SUT on recei						
	μ-Law 100rel extensions and					SIP Require header, then	
	independent the normal off						
						with the coding of the Nature	
	of Connection Indicators						
	the G.711 a-law codec sl			n the SDF	ans	wer.	
SIP parameter	Offer: m=audio 4711 RTF						
values	Answer: m=audio 4711 RTF						
ISUP parameter	1		•	ted, USI:	A-lav	v or absent	
values	,	continui	•	-		LOUID	
Comments	SIP-I		SU	ı		ISUP	
	INVITE(IAM)	→			→	IAM	
	183 Session Progress	+					
	PRACK	→					
	200 OK PRACK	(007	
	UPDATE	→			→	СОТ	
	200 OK UPDATE	+	1945				
	100 D: : (1011)		reconditi	ons met		10014	
	180 Ringing(ACM)	+			<u>+</u>	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
	ACK	→					
	D)(E(DEL)	_	Conver	sation		551	
	BYE(REL)	→			<u>→</u>	REL	
	200 OK BYE(RLC)	+			←	RLC	

TP101031	SIP reference: RFC	3261 [4]		FN 38	ISUP reference	-		
TSS reference	EN 383 001 [2], clause 6.1.3.5.2.2 SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)							
SIP selection	PICS 4/4 AND PICS 4/5 AN			COO MICOCA	30 (17 (1V1)			
criteria	1100 1/1/1100 1/0/111	D 1 100 17	Ü					
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 for a-Law and no μ-Law 100rel extensions and preconditions extensions in the SIP Supported header, then independent the normal offer answer procedures apply: 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"; the G.711 a-law codec shall be returned in the SDP answer. 							
SIP parameter	Offer: m=audio 4711 RT	TP/AVP 8						
values	Answer: m=audio 4711 R7							
ISUP parameter	IAM Continuity Indicator:	continuit	ty check	required o	n this circuit or c	ontinuity		
values	COT Continuity Indicator:			l on previo successfu	u s circuit , USI: A- I	law or absent		
Comments	SIP-I		SU		ISUP			
	INVITE(IAM)	→		-	IAM			
	183 Session Progress	+						
	PRACK	→						
	200 OK PRACK	+						
	UPDATE	→		-	COT			
	200 OK UPDATE	+						
		Р	reconditi	ons met				
	180 Ringing(ACM)	+		•	ACM			
	200 OK INVITE(ANM)	+		•	- ANM			
	ACK	→				-		
			Convers	sation		-		
	BYE(REL)	→		•	REL	-		
	200 OK BYE(RLC)	+		•	RLC	·		

TP101032	SIP reference: RFC	3261 [4]				ISUP reference:
				EN	383 O	01 [2], clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/Sending	of the Ir	nitial Addı	ress Mes	sage	(IAM)
SIP selection	PICS 4/4 AND PICS 4/5 AND					
criteria						
ISUP selection	PICS 1/5 AND NOT PICS 1/0	6 AND P	ICS 4/1			
criteria						
Test purpose						a SDP offer for a-Law and no
	μ-Law 100rel extensions and					SIP Require header, then
	independent the normal of					
						with the Continuity check
	indicator "continuity ch			tnis circ	uit" o	r continuity check
	 performed on previous the G.711 a-law codec s 			a tha CDI	.	wor
SIP parameter	the G.711 a-law codec sOffer: m=audio 4711 RT			ii tile SDI	ans	wer.
values	Answer: m=audio 4711 RT					
ISUP parameter				require	d on t	this circuit or continuity
values						circuit, USI: A-law or absent
			ity check			
Comments	SIP-I		SU	ΙΤ		ISUP
	INVITE(IAM)	→			→	IAM
	183 Session Progress	+				
	PRACK	→				
	200 OK PRACK	+				
	UPDATE	→			→	COT
	200 OK UPDATE	+				
			Preconditi	ons met		
	180 Ringing(ACM)	+			+	ACM
	200 OK INVITE(ANM)	+			+	ANM
	ACK	→				
		<u> </u>	Conver	sation		
	BYE(REL)	→			→	REL
	200 OK BYE(RLC)	+			←	RLC

TP101033	SIP reference: RFC	3261 [4]				SUP reference:		
	EN 383 001 [2], clause 6.1.3.5.2.2							
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 AN	ID PICS 1	/9					
criteria								
ISUP selection	NOT PICS 1/6 AND NOT PI	ICS 4/1						
criteria								
Test purpose						SDP offer for a-Law and no		
						SIP Supported header, then		
	independent the normal o		•		-			
	the IAM shall be deferred.							
	the G.711 a-law codec			n the SDP	ans	wer.		
SIP parameter	Offer: m=audio 4711 R							
values	Answer: m=audio 4711 R	TP/AVP 8	8					
ISUP parameter								
values	loip i	1 1		-		lious		
Comments	SIP-I		SL)		ISUP		
	INVITE(IAM)	→						
	183 Session Progress	-						
	PRACK	→						
	200 OK PRACK	+						
	UPDATE	→			<u>→</u>	IAM		
	200 OK UPDATE	+						
	180 Ringing(ACM)	+			<u>+</u>	ACM		
	200 OK INVITE(ANM)	+			-	ANM		
	ACK	→						
			Conver					
	BYE(REL)	→			→	REL		
	200 OK BYE(RLC)	←			-	RLC		

TP101034	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	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	CS 4/1					
criteria							
Test purpose					a SDP offer for a-Law and no		
	μ-Law 100rel extensions and						
	independent the normal of						
	 the IAM shall be deferred 						
	the G.711 a-law codec s	shall be r	eturned i	n the SDP ans	swer.		
SIP parameter	Offer: m=audio 4711 R7	ΓΡ/AVP 8					
values	Answer: m=audio 4711 R7	TP/AVP 8					
ISUP parameter							
values							
Comments	SIP-I		SU	JT	ISUP		
	INVITE(IAM)	→					
	183 Session Progress	+					
	PRACK	→					
	200 OK PRACK	←					
	UPDATE	→		→	IAM		
	200 OK UPDATE	←					
	180 Ringing(ACM)	←		←	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	→					
			Conver	sation			
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	+	•	+	RLC		

TP101035	SIP reference: RFC	3261 [4]	-	SUP reference:			
T00 (EN 383 001 [2], clause 6.1.3.5.2.2					
TSS reference	SIP-ISUP/Basic call/Sending			IAM)			
SIP selection	NOT PICS 4/4 AND NOT PI	CS 4/5 AND PICS	S 1/9				
criteria							
ISUP selection	PICS 1/7						
criteria							
Test purpose	Ensure that the SUT on rece	ipt of an INVITE	message with a	SDP offer m line without			
	a-law codec:		-				
	• the u-law codec shall b	e rejected.					
SIP parameter	Offer: m=audio 4711 RT	P/AVP 0					
values	m=audio 4712 RT	P/AVP 8					
	Answer: m=audio 0 RTP/A	VP 0					
ISUP parameter							
values							
Comments	SIP-I	SU	JT	ISUP			
	INVITE(IAM)	→	→	IAM			
	180 Ringing(ACM)	+	+	ACM			
	200 OK INVITE(ANM)	+	+	ANM			
	ACK	→					
		Conver	sation				
	BYE(REL)	→	→	REL			
	200 OK BYE(RLC)	+	+	RLC			

TP101036	SIP reference: RFC 3261 [4]			ISUP reference:
				001 [2], clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/Sending		dress 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 recei			
				s in the SIP Supported header:
				with the coding of the Nature
	of Connection Indicators		T to be expec	ted";
	 the u-law codec shall b 	•		
SIP parameter	Offer: m=audio 4711 RTF			
values	m=audio 4712 RTF			
	Answer: m=audio 0 RTP/A\			
ISUP parameter	,	COT to be expe	cted, USI: A-la	w or absent
values	,	ontinuity		
Comments	SIP-I		UT	ISUP
	INVITE(IAM)	→	→	IAM
	183 Session Progress	+		
	PRACK	→		
	200 OK PRACK	+		
	UPDATE	→	→	COT
	200 OK UPDATE	+		
			tions met	
	180 Ringing(ACM)	+	←	ACM
	200 OK INVITE(ANM)	+	+	ANM
	ACK	→		
			rsation	
	BYE(REL)	→	→	REL
	200 OK BYE(RLC)	-	←	RLC

TP101037	SIP reference: RFC 3261 [4]			IS	SUP reference:
					1 [2], clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/Sending	of the Initial A	ddress Messa	age (l	AM)
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 recei				
	a-law codec 100rel extension				
					vith the coding of the Nature
	of Connection Indicators		OT to be exp	oecte	ed";
	 the u-law codec shall b 				
SIP parameter	Offer: m=audio 4711 RTF				
values	m=audio 4712 RTF	,			
	Answer: m=audio 0 RTP/A\				
ISUP parameter		OT to be exp	ected , USI: A	4-law	or absent
values	<u> </u>	ontinuity			
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	(
	PRACK	→			
	200 OK PRACK	+			
	UPDATE	→		→	COT
	200 OK UPDATE	+			
			ditions met		
	180 Ringing(ACM)	-			ACM
	200 OK INVITE(ANM)	-		←	ANM
	ACK	→			
			ersation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		-	RLC

TP101038	SIP reference: RFC	3261 [4]		EN 38	-	SUP reference: 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 AN				-3- (,
criteria	DIGG 4/5 AND NOT DIGG 4	(0. ANID DI	00.4/4			
ISUP selection criteria	PICS 1/5 AND NOT PICS 1/	6 AND PI	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"; the u-law codec shall be rejected. 					
SIP parameter	Offer: m=audio 4711 R3					
values	m=audio 4712 R7	ΓΡ/AVP 8				
	Answer: m=audio 0 RTP/A	AVP 0				
ISUP parameter	IAM Continuity Indicator:	continuit	y check	required o	on t	his circuit or continuity
values	,					circuit, USI: A-law or absent
	COT Continuity Indicator:	continuit	y check	successf	ul	
Comments	SIP-I		SU	T		ISUP
	INVITE(IAM)	→			→	IAM
	183 Session Progress	+				
	PRACK	→				
	200 OK PRACK	+				
	UPDATE	→			→	COT
	200 OK UPDATE	+				
		Р	reconditi	ons met		
	180 Ringing(ACM)	←			←	ACM
	200 OK INVITE(ANM)	←			←	ANM
	ACK	→				
			Convers	sation		
	BYE(REL)	→			→	REL
	200 OK BYE(RLC)	+			←	RLC

TP101039	SIP reference: RFC 3	3261 [4]			ISUP reference:
T00 ((1) 1 20 1 4			001 [2], clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/Sending		daress Mes	ssage	(IAM)
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND	PICS 1/9			
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 m line without a-law codec 100rel extensions and preconditions extensions in the SIP Require 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"; the u-law codec shall be rejected.				
SIP parameter	Offer: m=audio 4711 RTF	P/AVP 0			
values	m=audio 4712 RTF Answer: m=audio 0 RTP/A\				
ISUP parameter	IAM Continuity Indicator: c	ontinuity che	ck require	d on t	his circuit or continuity
values	c		ed on pre	vious	circuit, USI: A-law or absent
Comments	SIP-I		SUT	1	ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	-		† <u>-</u>	7
	PRACK	→			
	200 OK PRACK	+			
	UPDATE	→		→	COT
	200 OK UPDATE	+			
		Precon	ditions met		
	180 Ringing(ACM)	-		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
	ACK	→			
		Conv	ersation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		+	RLC

TP101040	SIP reference: RFC 3	3261 [4]			ISUP reference:
				EN 383	001 [2], clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/Sending			ess Message	e (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	Ensure that the SUT on recei				
					ns in the SIP Supported header:
	the IAM shall be deferred			itions have b	een met;
	the u-law codec shall b	•	ed.		
SIP parameter	Offer: m=audio 4711 RTI				
values	m=audio 4712 RTF	-			
	Answer: m=audio 0 RTP/A	VP 0			
ISUP parameter					
values					1
Comments	SIP-I		SU	T	ISUP
	INVITE(IAM)	→			
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	+			
	UPDATE	→		→	IAM
	200 OK UPDATE	+			
		<u> </u>	reconditi		
	180 Ringing(ACM)	+		+	,
	200 OK INVITE(ANM)	+		+	ANM
	ACK	→			
			Convers		
	BYE(REL)	→		→	
	200 OK BYE(RLC)	←		+	RLC

TP101041	SIP reference: RFC 3261 [4]				_	SUP reference:	
						01 [2], clause 6.1.3.5.2.2	
TSS reference	SIP-ISUP/Basic call/Sendin	ng of the In	itial Add	ress Messa	ge (IAM)	
SIP selection	PICS 4/4 AND PICS 4/5 AN	ND PICS 1/	9				
criteria							
ISUP selection	NOT PICS 1/6 AND NOT P	PICS 4/1					
criteria							
Test purpose		Ensure that the SUT on receipt of an INVITE message with a SDP offer m line without					
						in the SIP Require header:	
	 the IAM shall be deferr 			litions have	bee	en met;	
	 the u-law codec shall 		ed.				
SIP parameter	Offer: m=audio 4711 R	TP/AVP 0					
values	m=audio 4712 R	TP/AVP 8					
	Answer: m=audio 0 RTP/	AVP 0					
ISUP parameter							
values							
Comments	SIP-I		SL	JT		ISUP	
	INVITE(IAM)	→					
	183 Session Progress	←					
	PRACK	→					
	200 OK PRACK	←					
	UPDATE	→			→	IAM	
	200 OK UPDATE	+					
		Pi	reconditi	ions met			
	180 Ringing(ACM)	+			(ACM	
	200 OK INVITE(ANM)	+			←	ANM	
	ACK	→					
			Conver	sation			
	BYE(REL)	→			→	REL	
	200 OK BYE(RLC)	+	•	,	(RLC	

TP101042	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 Ad	dress Mess	sage (IAM)			
SIP selection	NOT PICS 4/4 AND NOT PICS 4/5	AND PIC	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 or						
	 the call is refused with a 415 	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)	→					
	415 Unsupported media type	+	·				
	ACK	→	·				

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	e Initial Ad	dress Mes	sage (IAM)				
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS	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							
	one media streams 100rel extensions and preconditions extensions in the SIP Supported							
	header and based on operator p	-						
	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)	→						
	415 Unsupported media type	+						
	ACK	→	•					

TP101044	SIP reference: RFC 3261	[4]		ISUP reference:				
				383 001 [2], clause 6.1.3.5.2.2				
TSS reference	SIP-ISUP/Basic call/Sending of the	Initial A	Address Mes	sage (IAM)				
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS	3 1/9 AN	ID PICS 4/19	9				
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 100rel extens	ions an	d preconditio	ns extensions in the SIP Require				
	header and based on operator po	olicy the	en:	·				
	• the call is refused with a 415	Unsup	ported 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)	→						
	415 Unsupported media type	+						
	ACK	→						

TP101045	SIP reference: RFC 3	3261 [4]	l	SUP reference:
				01 [2], clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/Sending			
SIP selection	NOT PICS 4/4 AND NOT PIC	S 4/5 AND PICS	3 1/9 and not	PICS 4/19
criteria				
ISUP selection				
criteria				
Test purpose	Ensure that the SUT on receip			SDP offer with more than
	one media streams and bas			
	• if the SDP offer contains			
			udio streams sh	all be considered; the other
	streams shall be rejected			
	• if the SDP offer contains			ms, the IWU shall only
	consider one, and reject		S.	
SIP parameter	Offer: m=audio 4711 RTF			
values	m= audio 4712 RT			
	m= video 4713 RTI	P/AVP 31		
	Answer: m=audio 4711 RTF			
	m=audio 0 RTP/AV			
	m=video 0 RTP/AV	<u>/P 31 </u>		
ISUP parameter				
values	OID I	01	· -	lioup
Comments	SIP-I	SI		ISUP
	INVITE(IAM)	→	→	IAM
	180 Ringing(ACM)	←	+	ACM
	200 OK INVITE(ANM)	←	←	ANM
	ACK	→		
		Conver		
	BYE(REL)	→	→	REL
	200 OK BYE(RLC)	←	←	RLC

TP101046	SIP reference: RFC	ISUP reference:						
					01 [2], clause 6.1.3.5.2.2			
TSS reference	SIP-ISUP/Basic call/Sending	of the Initia	I Address Mes	sage	(IAM)			
SIP selection	NOT PICS 4/4 AND NOT PIC	CS 4/5 AND	PICS 1/9 ANI	TON C	Γ PICS 4/19			
criteria								
ISUP selection	PICS 1/4 AND NOT PICS 1/6	S AND PICS	S 4/1					
criteria —								
Test purpose	Ensure that the SUT on recei							
		one media streams 100rel extensions and preconditions extensions in the SIP Supported header and based on operator policy then:						
				\ _: _i _	with the ending of the Neture			
	 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							
	if the SDP offer contains	several au	dio type media	strea	ms, the IWU shall only			
	consider one, and reject				•			
SIP parameter	Offer: m=audio 4711 RT	P/AVP 8						
values	m= audio 4712 RT							
	m= video 4713 RT	P/AVP 31						
		D/A\/D 0						
	Answer: m=audio 4711 RT							
	m=audio 0 RTP/A\ m=video 0 RTP/A\							
ISUP parameter			expected, USI	· Δ_ 2\	w or absent			
values		continuity	expected, Ooi	i. A-ia	w or absent			
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	183 Session Progress	+		-	,			
	PRACK	→						
	200 OK PRACK	+						
	UPDATE	→		→	COT			
	200 OK UPDATE	+						
		Precon	ditions met					
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	→						
			onversation					
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	←		←	RLC			

TP101047	SIP reference: RFC 3261 [4] ISUP reference:								
					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 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 Require								
	header and based on opera	tor policy	then:		·				
	 the IAM shall be sent out of Connection Indicators 	t immediat	ely on the BIC	C side	with the coding of the Nature				
					streams and one or more				
					nall be considered; the other				
	streams shall be rejected								
	if the SDP offer contains	several a	udio type med	ia strea	ms, the IWU shall only				
	consider one, and reject		streams.						
SIP parameter	Offer: m=audio 4711 RTF								
values	m= audio 4712 RT								
	m= video 4713 RTP/AVP 31								
	Answer: m=audio 4711 RTP/AVP 8								
	m=audio 0 RTP/A\								
	m=video 0 RTP/A\								
ISUP parameter			expected, US	SI: A-la	w or absent				
values		continuity							
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	183 Session Progress	+							
	PRACK	→							
	200 OK PRACK	+							
	UPDATE	→		→	COT				
	200 OK UPDATE	+							
			nditions met						
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK	→							
	DVE(DEL)		Conversation		DEL				
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		←	RLC				

TP101048	SIP reference: RFC 3	261 [4]		ISUP reference:			
			EN 383 0	01 [2], clause 6.1.3.5.2.2			
	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 PICS 4/1						
criteria							
	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams100rel extensions and preconditions extensions in the SIP Supported 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 streams. Offer: m=audio 4711 RTP/AVP 8 m= audio 4712 RTP/AVP 8 m= video 4713 RTP/AVP 31						
ISUP parameter values	Ċ	P 8 P 31 ontinuity chec heck performe	d on previous	this circuit or continuity circuit, USI: A-law or absent			
		ontinuity chec		lia.			
	SIP-I		JT	ISUP			
	INVITE(IAM)	→	→	IAM			
	183 Session Progress PRACK	←					
I	200 OK PRACK						
	UPDATE	→	→	COT			
	200 OK UPDATE	+					
ľ	200 OK OF BATE	Precondition	s met				
	180 Ringing(ACM)	←	(ACM			
	200 OK INVITE(ANM)	+	+	ANM			
	ACK	→					
		Conve	rsation				
	BYE(REL)	→	→	REL			
	200 OK BYE(RLC)	+	+	RLC			

TP101049	SIP reference: RFC	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	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 PICS 4/1								
criteria									
Test purpose SIP parameter values	 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 streams. Offer: m=audio 4711 RTP/AVP 8 m= audio 4712 RTP/AVP 8 								
ISUP parameter values	Answer: m=audio 4711 RTI m=audio 0 RTP/A\ m=video 0 RTP/A\ IAM Continuity Indicator:	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							
Comments	SIP-I	continuity che	BUT	ISUP					
Comments	INVITE(IAM)	→	→	IAM					
	183 Session Progress	,		IZAIVI					
	PRACK	→							
	200 OK PRACK	-							
	UPDATE	→	→	СОТ					
	200 OK UPDATE	+							
		Precondition	ns met						
	180 Ringing(ACM)	+	+	ACM					
	200 OK INVITE(ANM)	+	+	ANM					
	ACK	→							
			ersation						
	BYE(REL)	→	→	REL					
	200 OK BYE(RLC)	-	+	RLC					

TP101050	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	1/9 AND N	OT PICS	4/19		
criteria							
ISUP selection	NOT PICS 1/6 AND NOT PIC	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 Supported header and based on operator policy then: the IAM shall be deferred until all preconditions have been met;						
	non-audio type media st streams shall be rejected	 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 					
SIP parameter	Offer: m=audio 4711 RT			· .			
values	m= audio 4711 R1		-				
	m= video 4713 RT		-				
	Answer: m=audio 4711 RT m=audio 0 RTP/A' m=video 0 RTP/A'	VP 8	3				
ISUP parameter values							
Comments	SIP-I		SU	Т		ISUP	
	INVITE(IAM)	→					
	183 Session Progress	+					
	PRACK	→					
	200 OK PRACK	+					
	UPDATE	→			→	IAM	
	200 OK UPDATE	+					
			conditions	met			
	180 Ringing(ACM)	+			+	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
	ACK	→					
			Convers	ation			
	BYE(REL)	→			<u>→</u>	REL	
	200 OK BYE(RLC)	+			-	RLC	

TP101051	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 AN	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19						
criteria								
ISUP selection	NOT PICS 1/6 AND NOT P	'ICS 4/1						
criteria	Francis that the OUT on the	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than						
Test purpose	 one media streams 100rel extensions and preconditions extensions in the SIP Require 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							
SIP parameter values	Offer: m=audio 4711 R m= audio 4712 F m= video 4713 F Answer: m=audio 4711 R	RTP/AVP 8 RTP/AVP 3	3 31					
	m=audio 0 RTP/							
	m=video 0 RTP/							
ISUP parameter values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→						
	183 Session Progress	+						
	PRACK	→						
	200 OK PRACK	+						
	UPDATE	→		→	IAM			
	200 OK UPDATE	+						
			onditions me					
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	→	0					
	DVE(DEL)	+	Conversati		REL			
	BYE(REL)	→		→	RLC			
	200 OK BYE(RLC)				INLO			

5.2.1.2 Sending of the Subsequent Address Message (SAM)

TP102001	SIP reference: RFC 3	261 [4]		ISUP reference: 12.5 [1], clause 6.2 a)			
TSS reference	SIP-ISUP/Basic call/Sending of	SIP-ISUP/Basic call/Sending of the Subsequent Address Message (SAM)					
SIP selection	PICS 3/4						
criteria	DIGG 0/0						
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 greater 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 (F	PIXIT)					
Comments	SIP-I	S	UT	ISUP			
	INVITE	→	→	IAM			
	INVITE	→	→	SAM			
	INVITE	→	→	SAM			
	180 Ringing	+	+	ACM			
	200 OK INVITE	+	+	ANM			
	ACK	→					
		Conve	rsation				
	BYE(REL)	→	→	REL			
	200 OK BYE(RLC)	+	+	RLC			

TP102002	SIP reference: RFC 3	_	SUP reference:					
TSS reference	Q.1912.5 [1], clause 6.2 b) SIP-ISUP/Basic call/Sending of the Subsequent Address Message (SAM)							
		of the Subse	quent Adare	ss ivies	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 fewer than the number of digits already accumulated for the call: • then the SUT shall immediately send a 484 Address Incomplete response for this INVITE; • in this case no SAM is sent to BICC/ISUP procedures.							
SIP parameter values								
ISUP parameter								
values	<u> </u>				[
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	INVITE(IAM)	→						
	484 Address incomplete	(→	REL			
	ACK	→		+	RLC			

5.2.1.3 Sending of COT

TP103001	SIP reference: RFC	3261 [4]		_	SUP reference:
					Q.19	012.5 [1], clause 6.3
TSS reference	SIP-ISUP/Basic call/COT					
SIP selection	PICS 4/4 AND PICS 4/5					
criteria						
ISUP selection	PICS 1/4 AND PICS 4/1					
criteria						
Test purpose	Ensure that the when the SI					
		/ continu	ity proced	ures on th	e ou	tgoing BICC side have been
	successfully completed:					
	 the SUT shall send the C 			re the Cor	ntinui	ty Indicator in the COT
	message shall be set to	"Continu	ıity".			
SIP parameter						
values						
ISUP parameter	COT continuity indicator: Co	ontinuity				
values						
Comments	SIP-I		SL	Т		ISUP
	INVITE(IAM)	→			→	IAM
	183 Session Progress	←				
	PRACK	→				
	200 OK PRACK	+				
	UPDATE	→			→	СОТ
	200 OK UPDATE	+				
			•	•		
	180 Ringing(ACM)	+			+	ACM
	200 OK INVITE(ANM)	+			+	ANM
	ACK	→				
			Conver	sation		
	BYE(REL)	→			→	REL
	200 OK BYE(RLC)	+			←	RLC

TP103002	SIP reference: RF	C 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	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: C	Continuity ch	neck suc	cessful				
values								
Comments	SIP-I		SU		ISUP			
	INVITE(IAM)	→		→	IAM			
	183 Session Progress	+						
	PRACK	→						
	200 OK PRACK	←						
	UPDATE	→		→	COT			
	200 OK UPDATE	←						
		Pı	reconditi	ons met				
	180 Ringing(ACM)	←		←	ACM			
	200 OK INVITE(ANM)	←		←	ANM			
	ACK	→						
			Convers	sation				
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		+	RLC			

5.2.1.4 Receipt of the Address Complete Message (ACM)

TP104001	SIP reference: RFC 326	1 [4]		ISUP reference: Q.1912.5 [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 indi	cation					
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		1	IAM		
	183 Session Progress (ACM)	+		+	ACM(no indication)		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	→					
			Conversation				
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP104002	SIP reference: RF0	C 3261 [4]	ISUP reference: Q.1912.5 [1], clause 6.5 1)			
TSS reference	SIP-ISUP/Basic call/Receip	ot of the A	ddress compl	ete messa	ge (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 "subscriber free" where the ISUP indicator parameter set to ISUP_ID, the ISDN access indicator set to ISDN_ACCES_ID and the OBCI in-band information set to OBCI_INBAND then: • the 180 Ringing SIP response is sent. Ensure that the in-band information can be transmitted to the calling user; • the received ACM is encapsulated in the 180 Ringing.						
SIP parameter values		'		<u> </u>			
ISUP parameter values	ACM FCI: ISUP_ID, ISDN OBCI: OBCI_INBAN		S_ID,				
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		→	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	→					
			Conversation	n			
	BYE(REL)	→		→	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: Q.1912.5 [1], clause 6.6						
TSS reference	SIP-ISUP/Basic call/Receipt of the Call progress message (CPG).									
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	where the event information pathe 180 Ringing SIP respon	Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the event information parameter event indicator is set to "Alerting": the 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'	ı							
values	CPG; event information param			: Alertir	ng					
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	→		→	IAM					
	183 Session Progress (ACM)	+		+	ACM(no indication)					
	180 Ringing(CPG)	+		+	CPG(ALERTING)					
	200 OK INVITE(ANM)	+		+	ANM					
	ACK	→								
		Conversation								
	BYE(REL)	→		→	REL					
	200 OK BYE(RLC)	+		+	RLC					

TP105002	SIP reference: RFC 326	1 [4]		ISUP reference: Q.1912.5 [1], clause 6.6					
TSS reference	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 event information parameter event indicator 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 values	ACM: Called party status "no ind CPG; event information param		ent indicator: F	Progre	ess				
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	183 Session Progress (ACM)	+		+	ACM(no indication)				
	183 Session (CPG)	+		+	CPG(PROGRESS)				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK	→							
			Conversation						
	BYE(REL)	→		1	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP105003	SIP reference: RFC 326	1 [4]			ISUP reference:				
				Q.1	912.5 [1], clause 6.6				
TSS reference	SIP-ISUP/Basic call/Receipt of the Call progress message (CPG).								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Ensure that the SUT, having rec								
	where the event information pa			tor is	set to "in-band information or				
	an appropriate pattern is now	availabl	e":						
	 183 Session Progress response 	onse is s	ent from the I-	IWU;					
	 the received CPG is encaps 	sulated in	the 183 -sess	sion P	rogress.				
SIP parameter									
values									
ISUP parameter	ACM: Called party status "no inc								
values	CPG; event information param	neter eve	ent indicator: i	in-bar	nd-information or an appropriate				
	pattern is now available								
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	183 Session Progress (ACM)	+		+	ACM(no indication)				
	183 Session (CPG)	+		+	CPG (Inband Info available)				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK	→							
	Conversation								
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				

5.2.1.6 Receipt of the Answer message (ANM)

TP106001	SIP reference: RFC 3261 [4]		SUP reference: 012.5 [1], clause 6.7						
TSS reference	SIP-ISUP/Basic call/Receipt of the Answer message (ANM).								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	 sends a 200 OK INVITE; the received ANM is encapsulated in the The bearer path shall be connected in both or 	Ensure that the SUT, having received the ACM message, on receipt of an ANM message: sends a 200 OK INVITE; the received ANM is encapsulated in the 200 OK INVITE. The bearer path shall be connected in both directions when both of the following conditions							
	 the BICC outgoing bearer set-up proced Q.1902.4 [i.14]) is successfully complete 	 are satisfied: the BICC outgoing bearer set-up procedure, (see ITU-T Recommendation Q.1902.4 [i.14]) is successfully completed; and 							
	preconditions have been satisfied on the (if applicable). In addition, if BICC is performing the "Per-ca Outgoing bearer set-up procedure and the C bearer path shall be connected in both direct and the I-IWU determines (through the procepreconditions have been met for the session	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). 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							
SIP parameter values	200 OK INVITE with encapsulated ANM								
ISUP parameter values	ANM								
Comments		SUT	ISUP						
	INVITE(IAM) →	→	IAM						
	180 Ringing(ACM) ←	-	ACM						
	200 OK INVITE(ANM)	+	ANM						
	ACK →								
		ersation							
	BYE(REL) →	→	REL						
	200 OK BYE(RLC) ←	←	RLC						

5.2.1.7 Receipt of the Connect message (CON)

TP107001	SIP reference: RFC 3261 [4]		ISUP reference:						
				2.5 [1], clauses 6.4, 6.7						
TSS reference	SIP-ISUP/Basic call/Receipt of the CONNECT message (CON).									
SIP selection										
criteria										
ISUP selection										
criteria	000 (6)	LINDUTE E	4	OUT : (OON						
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;the received CON is encapsular	od in the 20		=						
	The bearer path shall be connected									
	are satisfied:	iii bolii allet	CHOIRS WITCH	both of the following conditions						
	 the BICC outgoing bearer set-u 	o procedure	(see ITU-T	Recommendation						
	Q.1902.4 [i.14]) is successfully			Trocommonation						
	the I-IWU determines (using the			RFC 3312 [5]) that sufficient						
	preconditions have been satisfic									
	(if applicable).			•						
	In addition, if BICC is performing the									
	Outgoing bearer set-up procedure a									
	bearer path shall be connected in bo									
	and the I-IWU determines (through t			n RFC 3312 [5]) that sufficient						
SIP parameter	preconditions have been met for the	session to p	proceed.							
values										
ISUP parameter										
values										
Comments	SIP-I	SUT		ISUP						
	INVITE(IAM) →		→	IAM						
	200 OK INVITE(CON) ←		+	CON						
	ACK →	→								
		Conversa								
	BYE(REL) →		→	REL						
	200 OK BYE(RLC) ←		←	RLC						

5.2.1.8 Receipt of the REL message

TP108001	SIP reference: RFC 3	261 [4]		ISUP reference:					
		Q.1912.5 [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		SU	T		ISUP			
	INVITE(IAM)	1		•	→	IAM			
	SIP_FAILURE_VA(REL)	4		•	+	REL			
	ACK	→			→	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						
VA_10	404 Address incomplete	incomplete")						
VA_17	500 Server Internal Error	Cause Value No. 29 ("facility rejected")						
VA_18	480 Temporarily unavailable	Cause Value No. 31 ("normal unspecified")						
V/_10	400 Temperarily unavailable	(Class default)						
VA_19	486 Busy here if Diagnostics indicator	Cause Value in the Class 010 (No circuit/channel available,						
	includes the (CCBS indicator = CCBS possible) else 480 Temporarily unavailable	Cause Value No. 34)						
VA_20	500 Server Internal Error	Cause Value in the Class 010 (resource unavailable, Cause						
VA_20	300 Gerver internal Entor	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_22	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)						
VA 27	500 Sorver Internal Error	(79 is class default) Cause Value No. 87 ("user not member of CUG")						
VA_27 VA_28	500 Server Internal Error	Cause Value No. 88 ("incompatible destination")						
_	500 Server Internal Error							
VA_29	500 Server Internal Error	Cause Value No. 90 ("Non-existent CUG")						
VA_30 VA_31	404 Not Found 500 Server Internal Error	Cause Value No. 91 ("invalid transit network selection") 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	ISUP reference: Q.1912.5 [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	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.									
values	SIP Statue-Code: SIP_FAILUR		1 17(11)							
ISUP parameter values	REL; cause value: CV_ISUP (F	PIXIT)								
Comments	SIP-I		SUT			ISUP				
	INVITE(IAM)	→		7)	IAM				
	183 Session Progress(ACM)	+		•	•	ACM(no indication)				
	SIP_FAILURE_VA(REL)	+		•		REL				
	ACK	→		7	•	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	3261 [4]	ISUP reference: Q.1912.5 [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 "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_FAILU	RE_VA	(PIXIT)						
ISUP parameter values	REL; cause value: CV_ISUP (PIXIT)								
Comments	SIP-I		SU	Τ		ISUP			
	INVITE(IAM)	→			→	IAM			
	180 Ringing(ACM)	+			+	ACM			
	SIP_FAILURE_VA(REL)	(+	REL			
	ACK	→			→	RLC			

TP108004	SIP reference: RFC 32			ı	SUP reference:				
	Q.1912.5 [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	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.								
values	SIP Statue-Code: SIP_FAILUR	// (1 12(11)						
ISUP parameter	REL; cause value: CV_ISUP (I	PIXIT)							
values									
Comments	SIP-I		SI	JT		ISUP			
	INVITE(IAM)	→			→	IAM			
	183 Session Progress(ACM)	+			+	ACM(no indication)			
	180 Ringing(CPG)	+			+	CPG(ALERTING)			
	SIP_FAILURE_VA(REL)	+			+	REL			
	ACK	→			→	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:				
T00 (Q.1912.5 [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, having received a ANM", a 200 OK message is sent, on receipt of an ISUP REL, where the cause value defined as CV_ISUP : • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; • the SUT shall send a BYE message with the encapsulated REL message.								
SIP parameter values									
ISUP parameter									
values									
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM) ← ANM								
	ACK	ACK →							
		Conv	versation						
	BYE(REL)	+		←	REL				
	200 OK BYE(RLC)	→		→	RLC				

TP108006	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [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 CON message, a 200 OK message is sent, on receipt of an ISUP REL, where the cause value defined as CV_ISUP : • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; • the SUT shall send a BYE message with the encapsulated REL message.								
SIP parameter values					-				
ISUP parameter values	REL; cause value: CV_ISUI	P (PIXIT))						
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	200 OK INVITE(CON)	←		←	CON				
	ACK	→	-						
			Conversati	on					
	BYE(REL)	+		+	REL				
	200 OK BYE(RLC)	→	-	→	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: RFC 3261 [4]			ISUP reference:			
TSS reference	Q.1912.5 [1], clause 6.11.2 SIP-ISUP/Basic call/Receipt of the Release message (REL)						
SIP selection criteria	PICS 4/21	or the re	Cicase III	coodge (i	(LL)		
ISUP selection criteria							
Test purpose	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 with cause value 23 the SUT shall: • the SUT immediately requests the redirection to the new destination according the ISUP/BICC procedures.						
SIP parameter values							
ISUP parameter values	REL; cause value: 23						
Comments	SIP-I		SU	ΙΤ		ISUP	
	INVITE(IAM)	→			→	IAM(Destination 1)	
					+	REL(new Destination)	
					→	RLC	
					→	IAM(Destination 2)	
	180 Ringing(ACM)	+			+	ACM	
	200 OK INVITE(ANM) ← ANM						
	ACK	→					
			Convers	sation	•		
	BYE(REL)	+			←	REL	
	200 OK BYE(RLC)	→			→	RLC	

5.2.1.9 Autonomous release at I-IWU

TP109001	SIP reference: RFC 3261 [4]				ISUP reference:			
TSS reference	Q.1912.5 [1], clause 6.11.3 SIP-ISUP/Basic call/Autonomous release at I-IWU							
SIP selection criteria								
ISUP selection criteria	PICS 4/6							
Test purpose	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	SIP-I SUT ISUP						
	INVITE(IAM)	→		→	IAM			
	480 Temporarily unavailable (REL)	+		+	RSC			
	ACK	→		→	RLC			

TP109002	SIP reference: RFC 3261 [4	<u> </u>		ISUP reference:				
		Q.1912.5 [1], clause 6.11.3						
TSS reference	SIP-ISUP/Basic call/Autonomous re	lease a	at I-IWU					
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that when the SUT receives unrecognized backward ISUP or BICC signalling information 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 comp	atibility	"Release ca	all"				
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM) ← ← ACM							
				+	Any unknown message			
	500 Server internal error(REL)	+		→	REL			
	ACK	→		+	RLC			

TP109003	SIP reference: RFC 326	1 [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	PICS 3/4							
Test purpose	Ensure that the SUT on receipt of insufficient digits received in an INVITE messages: • sends a 484 Address Incomplete message.							
SIP parameter values								
ISUP parameter values								
Comments	SIP-I		SUT	ISUP				
	INVITE(IAM) 484 Address incomplete ACK	→ ← →						

TP109004	SIP reference: RFC 3	261 [4]		ISUP reference:				
		Q.1912.5 [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 receip	ot of subseque	nt INVITE me	essage:				
	 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 SUT ISUP							
	INVITE(IAM)	→						
	INVITE(IAM)	→						
	484 Address incomplete	+						
	ACK	→						

TP109005	SIP reference: RFC 326	1 [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 SUT in congestionsends a 480 Temporarily Unit		•	message:					
SIP parameter values									
ISUP parameter values									
Comments	SIP-I		SUT	ISUP					
	INVITE(IAM)	→							
	480 Temporarily unavailable	+							
	ACK	→							

TP109006	SIP reference: RFC 3261 [4]		ISUP reference: Q.1912.5 [1], clause 6.11.3			
TSS reference	SIP-ISUP/Basic call/Autonomous re	lease at	I-IWU				
SIP selection criteria							
ISUP selection criteria							
Test purpose	Ensure that the call is released due parameters: sends 500 Server Internal Erro		ICC/ISUP	compa	ibility procedure for unknown		
SIP parameter values							
ISUP parameter values	Unknown parameter in ACM: Paran	neter cor	npatibility '	'Releas	se call"		
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM) → IAM						
	← ACM(any unknown parameter)						
	500 Server internal error(REL)	+		→	REL		
	ACK	→		+	RLC		

TP109007	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 6.11.3			
TSS reference	SIP-ISUP/Basic call/Autonomou	ıs relea	se at I-IWU				
SIP selection							
criteria							
ISUP selection criteria							
Test purpose	Ensure that the call is releasedsends 484 Address Incomp		expiry of T7 w	vithin the	BICC/ISUP procedures:		
SIP parameter values							
ISUP parameter values							
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM) → IAM						
	T7 expiry						
	484 Address incomplete	+		→	REL		
	ACK	→		+	RLC		

TP109008	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 6.11.3			
TSS reference	SIP-ISUP/Basic call/Autonomo	us relea	se 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		<u> </u>	·				
ISUP parameter values							
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		→	IAM		
	180 Ringing(ACM)	-		+	ACM		
	T9 expiry						
	480 Temporarily unavailable	-		→	REL		
	ACK	RLC					

5.2.1.10 Receipt of the Release message BYE / CANCEL

TP110001	SIP reference: RFC 3261 [4]			ISUP 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 receipt of SIP BYE, the SUT shall send an ISUP REL with the cause value # 16 to the ISUP side.							
SIP parameter values									
ISUP parameter values	REL: Cause value #16, Loo	cation "Network b	eyond ar	n interwo	rking point"				
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK	→							
		Conv	ersation						
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP110002	SIP reference: RFC	3261 [4]		ISUP reference: Q.1912.5 [1], clause 6.11.1				
TSS reference	SIP-ISUP/Basic call/Receipt	of the BYI	E-CANCEL m	essage				
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose		Ensure that the SUT on receipt of SIP CANCEL, the I-IWU shall send an ISUP REL with the cause value # 31 to the ISUP side.						
SIP parameter	CANCEL without encapsular	ted ISUP n	nessage					
values	·		· ·					
ISUP parameter	REL: Cause value #31, Loca	ation "Netw	ork beyond a	n interwo	orking point"			
values			-					
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	CANCEL	→		→	REL			
	200 OK CANCEL	+		+	RLC			
	487 Request Terminated	+						
	ACK	→						

TP110003	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 6.11.1				
TSS reference	SIP-ISUP/Basic call/Receipt	of the BYA	-CANCEL me	essage				
SIP selection criteria								
ISUP selection criteria								
Test purpose		Ensure that the SUT on receipt of SIP BYE, the I-IWU shall send an ISUP REL with the cause value # 31 to the ISUP side.						
SIP parameter values	BYE without encapsulated IS	BYE without encapsulated ISUP message						
ISUP parameter values	REL: Cause value #31, Loca	tion "Netw	ork beyond ar	n interwo	orking point"			
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	BYE	→		→	REL			
	200 OK BYE	+		+	RLC			
	487 Request Terminated	+						
	ACK	→						

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]	(ISUP reference: Q.1912.5 [1], clauses 6.11.4 and 5		
TSS reference	SIP-ISUP/Basic call/Receipt					
	(GRS) or Circuit group blocki	ing messaç	ge (CGB) 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	ceipt of a R	SC message	sends:		
	 a BYE message if the S^I 	UT has alre	eady received	an ACK fo	or the 200 OK INVITE message	
	which had it sent.		•		•	
SIP parameter						
values						
ISUP parameter						
values						
Comments	SIP-I		SUT		ISUP	
	INVITE(IAM)	→		→	IAM	
	180 Ringing(ACM)	←		+	ACM	
	200 OK INVITE(ANM)	←		+	ANM	
	ACK	→				
			Conversation)		
	BYE(REL)	+		+	RSC	
	200 OK BYE(RLC)	→		→	RLC	

TP111002	SIP reference: RFC 3	3261 [4]			IS	SUP reference:			
		Q.1912.5 [1], clauses 6.11.4 and 5							
TSS reference	SIP-ISUP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message								
						tion hardware failure oriented			
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose						nessage relating to the call has			
	already been received on rec								
	 a BYE message if the SI which had it sent. 	JT has alr	ready rece	eived an <i>i</i>	ACK fo	or the 200 OK INVITE message			
SIP parameter									
values									
ISUP parameter									
values						_			
Comments	SIP-I		SL	JT		ISUP			
	INVITE(IAM)	→			→	IAM			
	180 Ringing(ACM)	←			←	ACM			
	200 OK INVITE(ANM)	+			←	ANM			
	ACK	→							
			Conver	sation					
	BYE	+			+	GRS			
	200 OK BYE	→			→	GRA			

TP111003	SIP reference: RFC 320	61 [4]			-	SUP reference:			
TSS reference	Q.1912.5 [1], clause 6.11.4 SIP-ISUP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message								
	(GRS) or Circuit group blocking								
SIP selection	, and the state of								
criteria									
ISUP selection criteria									
Test purpose	already been received on receip Message Type Indicator coded	1 a B 1 E moddago ii tho dd i mad amaday roddivod am Mortior tho 200 dit ii ti i i i moddago							
SIP parameter values									
ISUP parameter values	Circuit Group Supervision Mess	age Ty	pe Indicat	or "hard	ware f	ailure oriented"			
Comments	SIP-I		SU	Τ		ISUP			
	INVITE(IAM)	→			→	IAM			
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
	ACK	→							
			Convers	ation					
	BYE	+			+	CGB(hardware failure)			
	200 OK BYE	→			→	CGBA			

TP111004	SIP reference: RFC	3261 [4]		ISUP reference:					
			Q.1912.5	[1], clauses 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 block	ing message (CG	B) with the indic	ation hardware failure oriented					
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	already been received on rec 200 OK INVITE if the SUT ha	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 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							
SIP parameter values									
ISUP parameter									
values									
Comments	SIP-I		SUT	ISUP					
	INVITE(IAM)	→	→	IAM					
	180 Ringing(ACM)	+	←	ACM					
	200 OK INVITE(ANM)	+	←	ANM					
			←	RSC					
	ACK	→	→	RLC					
	BYE(REL)	+							
	200 OK BYE(RLC)	→							

TP111005	SIP reference: RFC 3261 [4] ISUP reference:							
	Q.1912.5 [1], clauses 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 message	CGB) with the	indica	ation 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							
	 the SUT shall wait until it BYE. 	receives the	ACK for the 2	00 OK	INVITE before sending the			
SIP parameter								
values								
ISUP parameter values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
				+	GRS			
	ACK	→		→	GRA			
	BYE	←						
	200 OK BYE	→						

TP111006	SIP reference: RFC 32	261 [4]			SUP reference:			
				Q.191	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	g message (CGB) with t	the indica	tion hardware failure oriented			
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the SUT, when at already been received on rece							
	Message Type Indicator coded							
	200 OK INVITE if the SUT has							
					INVITE before sending the			
	BYE.	icceives the	NOIN IOI THE	200 OK	TIVITE before sending the			
SIP parameter								
values								
ISUP parameter	Circuit Group Supervision Mes	sage Type I	ndicator "ha	ardware f	ailure oriented"			
values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	←		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
				←	CGB(hardware failure)			
	ACK	→		→	CGBA			
	BYE	←						
	200 OK BYE	→						

TP111007	SIP reference: RFC 3261	[4]	Q.	ISUP reference: Q.1912.5 [1], clauses 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) → IAM									
	180 Ringing(ACM)	+		+	ACM					
	500 Server Internal Error(REL)	+		+	RSC					
	ACK → RLC									

TP111008	SIP reference: RFC 3261 [4] ISUP reference:								
	Q.1912.5 [1], clauses 6.11.4 and 5								
TSS reference	SIP-ISUP/Basic call/Receipt of I	Reset ci	ircuit me	ssage (R	SC), C	ircuit group reset message			
	(GRS) or Circuit group blocking	messag	ge (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/BICC message relating to the call has								
	already been received on receip	ot of a G	RS mes	sage sen	ds:				
	a 500 Server Internal Error	on the S	SIP side						
SIP parameter									
values									
ISUP parameter									
values									
Comments	SIP-I		S	JT		ISUP			
	INVITE(IAM)	→			→	IAM			
	180 Ringing(ACM)	+			+	ACM			
	500 Server Internal Error	+			+	GRS			
	ACK	→			→	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		S	JT		ISUP	
	INVITE(IAM)	→			→	IAM	
	180 Ringing(ACM)	+			+	ACM	
	500 Server Internal Error	+			+	CGB(hardware failure)	
	ACK	→			→	CGBA	

TP111010	SIP reference: RFC	3261 [4]	ISUP reference: Q.1912.5 [1], clauses 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 IAM	
	INVITE(IAM) 1 180 Ringing(ACM)	-	+	ACM	
	200 OK INVITE(ANM)	+	+	ANM	
	ACK	→			
	INVITE(IAM) 2	→	→	IAM	
	180 Ringing(ACM)	+	+	ACM	
	200 OK INVITE(ANM)	-	+	ANM	
	ACK	→			
	BYE 1	+	+	GRS	
	200 OK BYE	→	→	GRA	
	BYE 2	+			
	200 OK BYE	→			

TP111011	SIP reference: RFC	3261 [4]		ISUP reference:		
	Q.1912.5 [1], clauses 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	→	→	IAM		
	180 Ringing(ACM)	+	+	ACM		
	200 OK INVITE(ANM)	-	←	ANM		
	ACK	→				
	INVITE(IAM) 2	→	→	IAM		
	180 Ringing(ACM)	+	+	ACM		
	200 OK INVITE(ANM)	+	+	ANM		
	ACK	→				
	BYE 1	+	+	CGB(hardware failure)		
	200 OK BYE	→	→	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/receipt of a SUSPEND message with the suspend indicator set to "network initiated"					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Ensure that the SUT, on receipt of a SUSPEND message with the suspend indicator set to "network initiated": • is transferred in an INFO message.					
SIP parameter values						
ISUP parameter values	SUS; Suspend indicator: network initiated					
Comments	SIP-I INVITE(IAM) 180 Ringing(ACM) 200 OK INVITE(ANM) ACK INFO(SUS) 200 OK INFO	÷	SUT	÷ +	ISUP IAM ACM ANM SUS(network)	
	BYE(REL) 200 OK BYE(RLC)	→ ←		→	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 a SUSPEND message with the suspend indicator set to "network initiated"					
SIP selection criteria						
ISUP selection criteria	PICS 4/14					
Test purpose	Ensure that the SUT, on receipt of a SUSPEND message with the suspend indicator set to "network initiated": To is started; after To is expired, the call is released.					
SIP parameter values	INFO: encapsulated SUS					
ISUP parameter values	SUS; Suspend indicator: network initiated; REL: Cause value 102					
Comments	SIP-I INVITE(IAM)	→	SUT	→	ISUP IAM	
	180 Ringing(ACM) 200 OK INVITE(ANM)	+		+	ACM ANM	
	ACK → Conversation					
	INFO(SUS) 200 OK INFO	←		+	SUS(network)	
			T6 is started			
	BYE(REL)	+	T6 is expired	→	REL	
	200 OK BYE(RLC)	→		+	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	indicator set to "network init	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: r	network initiat	ed			
values						
Comments	SIP-I		SUT		ISUP	
	INVITE(IAM)	→		→	IAM	
	180 Ringing(ACM)	←		←	ACM	
	200 OK INVITE(ANM)	←		←	ANM	
	ACK	→				
			Conversation			
	INFO(SUS)	←		←	SUS(network)	
	200 OK INFO	→				
	INFO(RES)	+			RES(network)	
	200 OK INFO	→				
	BYE(REL)	→		→	REL	
	200 OK BYE(RLC)	+		+	RLC	

5.2.1.14 Receipt of Confusion message

TP114001	SIP reference: RFC 32	61 [4]		UP reference:	
				5 [1], clause A.1.1.3	
TSS reference	ISUP-SIP/ISUP Messages for s	pecial consid	eration / Confusio	n message	
SIP selection					
criteria					
ISUP selection criteria					
SIP parameter values	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					
Comments	SIP-I			ISUP	
	INVITE	→	→	IAM	
	183 Session Progress(CFN)	+	(CFN	
	180 Ringing(ACM)	+	(ACM	
	200 OK INVITE(ANM)	+	+	ANM	
	ACK	→			
		Con	nmunication		
	BYE(REL)	→	→	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 Messages	ISUP-SIP/ISUP Messages for special consideration / Receipt of a user part test message					
SIP selection	9						
criteria							
ISUP selection							
criteria							
Test purpose	Ensure that a call can be	successf	ully complet	ted if segr	nenta	tion applies in backward	
	direction.						
SIP parameter		ed ACM:	Backward c	all indicat	or abs	sent or set to "no additional	
values	information will be sent"						
	No action takes place on t						
ISUP parameter	ACM: optional forward cal	II indicato	r: additiona	I informat	ion wi	Il be sent in a segmentation	
values	message						
	SGM: optional parameters	3					
Comments	SIP-I		SU	Γ		ISUP	
	INVITE(IAM)	→			→	IAM	
	180 Ringing(ACM)	+			+	ACM	
					←	SGM	
	200 OK INVITE(ANM)	+			+	ANM	
	ACK →						
			Convers	ation			
	BYE(REL)	→		_	→	REL	
	200 OK BYE	+			+	RLC	

5.2.2 Interworking from ISUP to SIP-I

5.2.2.1 Sending of the INVITE message

TP301001	SIP reference: RFC 326	1 [4]	Q.191	ISUP reference: I2.5 [1], clause 7.1 1 a)		
TSS reference	ISUP-SIP/Basic call/Sending of t	he INVITE m	essage			
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	called party number and the se	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	sending con	plete indication			
values						
Comments	ISUP/BICC	,	SUT	SIP-I		
	IAM	→	→	INVITE(IAM)		
	ACM	+	+	180 Ringing(ACM)		
	ANM					
			→	ACK		
		Conv	ersation			
	REL	→	→	BYE(REL)		
	RLC	+	+	200 OK BYE(RLC)		

TP301002	SIP reference: RFC 326	1 [4]		IS	SUP reference:
			C	2.1912	.5 [1], clause 7.1 1 b)
TSS reference	ISUP-SIP/Basic call/Sending of	f the INVI	TE message		
SIP selection criteria					
ISUP selection criteria					
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.				
SIP parameter values					
ISUP parameter values	IAM; Called party number com	plete nur	nber		
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
	→ ACK				
		С	onversation	•	
	REL	→		→	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP301003	SIP reference: RFC 3261	[4]		IS	SUP reference:			
			Q.	1912	5 [1], clause 7.1 1 c)			
TSS reference	ISUP-SIP/Basic call/Sending of the	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 numbei	-					
Comments	ISUP/BICC		SUT		SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	(-	180 Ringing(ACM)			
	ANM	(-	200 OK INVITE(ANM)			
	→ ACK							
		Conv	ersation					
	REL	→		→	BYE(REL)			
	RLC	(-	200 OK BYE(RLC)			

TP301004	SIP reference: RFC 3261 [4	-]	(-	SUP reference: 2.5 [1], clause 7.1 1 d)
TSS reference	ISUP-SIP/Basic call/Sending of the I	NVITE me	essage		
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT in Idle state, on called party number where the end timer T _{OIW1} after the receipt of the late. • sends the INVITE message.	of addres	ss signallin	g is de	
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM →				
		T _{OIW}	1 expiry		
				→	INVITE(IAM)
	ACM ←			+	180 Ringing(ACM)
	ANM ←			+	200 OK INVITE(ANM)
				→	ACK
		Conv	ersation		
	REL →			→	BYE(REL)
	RLC +			←	200 OK BYE(RLC)

TP301005	SIP reference: RFC 3261	[4]		_	SUP reference: 2.5 [1], clause 7.1 A)	
TSS reference	ISUP-SIP/Basic call/Sending of the	ne IN				
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 complete called party number containing the Continuity Check indicator in the Nature of Connection Indicators parameter is set to indicate " continuity check not required ": • sends a INVITE message.					
SIP parameter						
values						
ISUP parameter						
values						
Comments	ISUP		SUT		SIP-I	
	IAM	←		→	INVITE(IAM)	
	ACM	+		+	180 Ringing(ACM)	
	ANM	+		+	200 OK INVITE(ANM)	
	→ ACK					
			Conversation	•		
	REL	→		→	BYE(REL)	
	RLC	(+	200 OK BYE(RLC)	

TP301006	SIP reference: RFC 3261 [4]	Į;	SUP reference:			
		Q.191	2.5 [1], clause 7.1 A)			
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE	message				
SIP selection criteria	NOT PICS 4/4 AND NOT PICS 4/5 AND N	OT PICS 4/15				
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 →					
	COT →	→	INVITE(IAM)			
	ACM ←	+	180 Ringing(ACM)			
	ANM ←	+	200 OK INVITE(ANM)			
	→ ACK					
	Co	onversation				
	REL →	→	BYE(REL)			
	RLC ←	+	200 OK BYE(RLC)			

TP301007	SIP reference: RFC 3261 [4]	Į;	SUP reference:			
		Q.191	2.5 [1], clause 7.1 A)			
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE n	nessage				
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	2.5 [1], clause 7.1 A)	
TSS reference	ISUP-SIP/Basic call/Sending of				
SIP selection	NOT PICS 4/4 AND NOT PICS	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	→			
	COT	→			

TP301009	SIP reference: RFC 3261	[4]	Q.1	ISUP reference: 912.5 [1], clause 7.1 A)		
TSS reference	ISUP-SIP/Basic call/Sending of the	e INVITE me		- · · · · · · · · · · · · · · · · · · ·		
SIP selection	NOT PICS 4/4 AND NOT PICS 4/5					
criteria						
ISUP selection criteria	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	UT	SIP-I		
	IAM -					
		T8 (expiry			
	1,22	-				
	RLC ÷	→				

TP301010	SIP reference: RFC 3261	[4]		ISUP reference:	
			Q.1	912.5 [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 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		•			
ISUP parameter values					
Comments	ISUP	3	SUT	SIP-I	
	IAM =	•	1	► INVITE(IAM)	
	ACM +		•	180 Ringing(ACM)	
	ANM ← 200 OK INVITE(ANM)				
	→ ACK				
		Conv	ersation		
	REL -)	-	▶ BYE(REL)	
	RLC •	-	•	200 OK BYE(RLC)	

TP301011	SIP reference: RFC 32	61 [4]			-	SUP reference:	
					2.191	2.5 [1], clause 7.1 B)	
TSS reference		ISUP-SIP/Basic call/Sending of the INVITE message					
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						
	offer or answer carrying the confirmation of a precondition being met is sent when the Continuity message with the Continuity Indicators parameter set to "continuity check successful" was received and the requested preconditions are met in the SIP network.						
SIP parameter			•				
values							
ISUP parameter							
values							
Comments	ISUP		S	SUT		SIP-I	
	IAM	→			→	INVITE(IAM)	
					4	183 Session Progress	
					1	PRACK	
					4	200 OK PRACK	
	COT(successful)	→			1	UPDATE	
					4	200 OK UPDATE	
	Preconditions met						
	ACM ← 180 Ringing(ACM)						
	ANM ← 200 OK INVITE(ANM)						
					1	ACK	
			Conve	ersation			
	REL	→			→	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 me	essage	
SIP selection	PICS 4/4 AND PICS 4/5 AND 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 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				
values				
ISUP parameter				
values				
Comments	ISUP		SUT	SIP-I
	IAM →		→	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 →		→	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 r	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	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	→		→	INVITE(IAM)	
				+	100 Trying	
	COT(unsuccessful) → CANCEL					
				+	200 OK CANCEL	
			`	+	487 Request Terminated	
				1	ACK	

TP301014	SIP reference: RFC 326	61 [4]		ISUP reference:		
			Q.1	1912.5 [1], clause 7.1 B)		
TSS reference	ISUP-SIP/Basic call/Sending of	the INVITE m	essage			
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PI	CS 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 ISUP Timer T8 expires. 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	→	-	→ INVITE(IAM)		
	← 100 Trying					
	T8 expires					
	REL(#47) ← CANCEL					
	RLC	→	•	€ 200 OK CANCEL		
			•	← 487 Request Terminated		
		-	•	→ ACK		

TP301015	SIP reference: RFC 3261 [4]		SUP reference:		
		Q.1912	2.5 [1], clause 7.1 C)		
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE m	essage			
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":				
	The sending of the INVITE is delayed un				
	 Continuity message, with the Con 	tinuity Indicators p	arameter set to "continuity"		
	shall be received;				
	 Bearer Set-up indication - for the 				
	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 ← 180 Ringing(ACM)				
	ANM ←	+	200 OK INVITE(ANM)		
		→	ACK		
	Con	versation			
	REL →	→	BYE(REL)		
	RLC ←	+	200 OK BYE(RLC)		

TP301016	SIP reference: RFC 326	1 [4]		ISUP reference: 12.5 [1], clause 7.1 C)	
TSS reference	ISUP-SIP/Basic call/Sending of the	ne INVITE m		[.],	
SIP selection criteria	NOT PICS 4/15				
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 COT(queequeful)	7 →			
	COT(successful) APM	7	→	INVITE(IAM)	
	ACM	7	+	180 Ringing(ACM)	
	ANM	-	-	200 OK INVITE(ANM)	
	AINIVI		→	ACK	
		Conv	ersation	NOIN	
	REL	→ COIN	/ersation →	BYE(REL)	
	RLC	-	+	200 OK BYE(RLC)	

TP301017	SIP reference: RFC 3261 [4]		ISUP reference:		
			Q.	1912.5 [1], clause 7.1 C)		
TSS reference	ISUP-SIP/Basic call/Sending of the I	NVITE me	essage			
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 mes	sage indicating "COT to be		
	expected":					
	 The sending of the INVITE delay 					
		the Conti	nuity Indicato	ors parameter set to "continuity"		
	shall be received;					
		ndication -	for the back	ward bearer set-up case was		
	received.					
SIP parameter						
values						
ISUP parameter						
values						
Comments	ISUP	S	UT	SIP-I		
	IAM →					
	COT(successful) →			→ INVITE(IAM)		
	ACM ← 180 Ringing(ACM)					
	ANM ← 200 OK INVITE(ANM)					
				→ ACK		
		Conve	ersation			
	REL →			→ BYE(REL)		
	RLC +			€ 200 OK BYE(RLC)		

TP301018	SIP reference: RFC 3261	[4]		SUP reference: 5 [1], clause 7.1 C) 2.4	
TSS reference	ISUP-SIP/Basic call/Sending of th	e INVITE m	essage		
SIP selection criteria	NOT PICS 4/15				
ISUP selection criteria	PICS 1/4				
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.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP	5	SUT	SIP-I	
	IAM ·	→			
	COT(successful)	→	→	INVITE(IAM)	
	ACM	(←	180 Ringing(ACM)	
	ANM	←	←	200 OK INVITE(ANM)	
			→	ACK	
		Conv	ersation		
	REL .	>	→	BYE(REL)	
	RLC	(+	200 OK BYE(RLC)	

TP301019	SIP reference: RFC 326	61 [4]		ISUP reference:		
	Q.1912.5 [1], clause 7.1 C)					
TSS reference	ISUP-SIP/Basic call/Sending of	the INVITE i	nessage			
SIP selection	NOT PICS 4/15					
criteria						
ISUP selection	PICS 1/4					
criteria						
Test purpose	Ensure that the SUT in Idle state expected":	e, on receipt	of an IAM mes	ssage indicating "COT to be		
	•					
		Continuity r	nessage was r	not received, i.e. the BICC timer T8		
	expires:					
	 send REL with Caus 	e Value 41 (temporary fai	lure) shall be sent on the BICC side		
	of the O-IWU.	`		,		
SIP parameter						
values						
ISUP parameter						
values						
Comments	ISUP		SUT	SIP-I		
	IAM	→				
	T8 expires					
	REL(#41)	-	·			
	RLC →					

TP301020	SIP reference: RFC 3261 [4]			ISUP reference:			
				912.5 [1], clause 7.1 D)			
TSS reference	ISUP-SIP/Basic call/Sending of the IN	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	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be 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 received; Bearer Set-up indication - for the forward bearer set-up case where the incoming Connect Type is "notification not required" was received.						
SIP parameter							
values							
ISUP parameter							
values							
Comments	ISUP	S	UT	SIP-I			
	IAM →		7	INVITE(IAM)			
			•				
			-	110.010			
			•	200 OK PRACK			
	COT(successful) →		1	0. 2 =			
			•	200 OK UPDATE			
		Preconditions met					
	ACM ←		•	10011119(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	VITE me	ssage		
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 4/15				
ISUP selection	DICC 4/4 AND DICC 4/0				
criteria	PICS 1/4 AND PICS 4/2				
Test purpose	Ensure that the SUT in Idle state, on re	ossint of	on IAM ma	20000	o indicating "COT to be
	 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 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. 				
SIP parameter					
values					
ISUP parameter					
values					
Comments	ISUP/BICC	S	JT		SIP-I
	IAM →			→	INVITE(IAM)
				+	183 Session Progress
				→	PRACK
				+	200 OK PRACK
	COT(successful) →			→	UPDATE
				+	200 OK UPDATE
	Preconditions met				
	ACM ←			+	180 Ringing(ACM)
	ANM ←			+	200 OK INVITE(ANM)
				1	ACK
		Conve	rsation		
	REL →			→	BYE(REL)
	RLC ←			+	200 OK BYE(RLC)

TP301022	SIP reference: RFC 3261 [4	.]		I;	SUP reference:			
	_		Q.1	1912.	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 concluexchange) confirmation of a precondition being 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	5	SUT		SIP-I			
	IAM →			→	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 →			→	BYE(REL)			
	RLC +			+	200 OK BYE(RLC)			

TP301023	SIP reference: RFC 3261 [4]		I;	SUP reference:			
			Q.1	1912.	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	The precondition signalling is concluexchange) confirmation of a precondition being satisfied when Continuity message, with the Correceived;	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					g			
values								
ISUP parameter								
values								
Comments	BICC	S	SUT		SIP-I			
	IAM →			→	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 →			→	BYE(REL)			
	RLC +			+	200 OK BYE(RLC)			

TP301024	SIP reference: RFC 326	61 [4]		19	SUP reference:			
	Q.1912.5 [1], clause 7.1 D)							
TSS reference	ISUP-SIP/Basic call/Sending of	the INVITE n			,			
SIP selection	PICS 4/4 AND PICS 4/5 AND PI							
criteria								
ISUP selection	PICS 1/4 AND PICS 4/2							
criteria								
Test purpose	The SUT in Idle state, on receip							
	in the Nature of Connection Indi							
	sends an INVITE message with							
	ensure that the SUT sends				xpires if the call has been			
	cleared before an early dia	logue has be	en establishe	ed.				
SIP parameter								
values								
ISUP parameter								
values								
Comments	BICC		SUT		SIP-I			
	IAM	→		→	INVITE(IAM)			
				+	100 Trying			
		T8	expires					
	REL(#47)	+		→	CANCEL			
	RLC	→		+	200 OK CANCEL			
			`	+	487 Request Terminated			
				1	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 indicator 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	Sl	JT		SIP-I			
	IAM →			→	INVITE(IAM)			
				+	100 Trying			
	internal resource reservation was unsuccessful							
	REL(#47) ←	REL(#47) ← CANCEL						
	RLC →			+	200 OK CANCEL			
				+	487 Request Terminated			
				→	ACK			

TP301026	SIP reference: RFC 3	261 [4]		-	SUP reference: 2.5 [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	Ensure that the SUT in the Idle state on receipt of a IAM message, with the Transmission Medium Requirement (TMR) parameter set to TMR_VALUE 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	Ē								
ISUP parameter values	IAM: TMR : ISUP_TMR									
Comments	ISUP/BICC		SUT		SIP-I					
	IAM	→		→	INVITE(IAM)					
	ACM	+		-	180 Ringing(ACM)					
	ANM	+		-	200 OK INVITE(ANM)					
		→ ACK								
		Conv	ersation							
	REL	→		→	BYE(REL)					
	RLC	+		-	200 OK BYE(RLC)					

TP301027	SIP reference: RFC		_	SUP reference: 2.5 [1], clause 7.1.1			
TSS reference	ISUP-SIP/Basic call/Sending	of the I	NVITE me	essage			
SIP selection criteria	Based on table 7						
ISUP selection criteria							
Test purpose	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	T		SIP-I	
	IAM	→			→	INVITE(IAM)	
	ACM	+			+	180 Ringing(ACM)	
	ANM	+			+	200 OK INVITE(ANM)	
					→	ACK	
			Convers	sation			
	REL	→			→	BYE(REL)	
	RLC	+		•	+	200 OK BYE(RLC)	

Table 5: Void

Table 6

	Values for test purposes TP301026									
	ISUP	SDP - a_b_m_LINE_VALUE								
	TMR parameter	TMR parameter m= line		n= 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></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

			Va	alues for test pu	rposes T	rposes TP301026, TP301027					
VA	VA ISUP						SDP - a_	b_m_LINE_VALUE			
		USI parameter		HLC IE in ATP	in ATP m= line		m= 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></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>		

TP301027A	SIP reference: RFC 3261	[4]	ISUP reference: Q.1912.5 [1], clause 7.1.2 TS 129 164 [10], clause 6.2.4.2				
TSS reference	ISUP-SIP/Basic call/Sending of the	INVITE m	essage				
SIP selection criteria							
ISUP selection criteria							
Test purpose	Ensure that the SUT is changing the in the USI from A-law to μ-law if trace outgoing IWU						
SIP parameter values	INVITE: IAM USI User information	layer 1 pro	tocol=µ-Law				
ISUP parameter values	IAM: USI User information layer 1	orotocol=A-	Law				
Comments	ISUP/BICC	SL	JT	SIP-I			
	IAM →		→	INVITE(IAM)			
	ACM ←		+	180 Ringing(ACM)			
	ANM ←		+	200 OK INVITE(ANM)			
			→	ACK			
		Conver	sation				
	REL →		→	BYE(REL)			
	RLC ←		+	200 OK BYE(RLC)			

TP301027B	SIP reference: RFC 3261	[4]	ISUP reference: Q.1912.5 [1], clause 7.1.2 TS 129 164 [10], clause 6.2.4.2				
TSS reference	ISUP-SIP/Basic call/Sending of the	ne INVITE m	essage				
SIP selection criteria							
ISUP selection criteria							
Test purpose	Ensure that the SUT is changing in the USI from μ -law to A -law if outgoing IWU						
SIP parameter values	INVITE: IAM USI User informatio	n layer 1 pro	tocol= A-Law				
ISUP parameter values	IAM: USI User information layer 1	l protocol= μ	-Law				
Comments	ISUP/BICC	SU	JT	SIP-I			
	IAM -	•	→	INVITE(IAM)			
	ACM ←	•	+	180 Ringing(ACM)			
	ANM ←	•	+	200 OK INVITE(ANM)			
			→	ACK			
		Conver	sation				
	REL -		→	BYE(REL)			
	RLC €	•	+	200 OK BYE(RLC)			

TP301028	SIP reference: RF	C 3261 [4	1	-	SUP reference: 2.5 [1], clause 7.1.2					
TSS reference	ISUP-SIP/Basic call/Sendi	ISUP-SIP/Basic call/Sending of the INVITE message								
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	Called Party Number para	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter of the IAM: • to the addr-spec component of the To header field in the INVITE message.								
SIP parameter	INVITE: To:	i i								
values										
ISUP parameter										
values										
Comments	ISUP/BICC		SUT		SIP-I					
	IAM	→		→	INVITE(IAM)					
	ACM	+		+	180 Ringing(ACM)					
	ANM	+		+	200 OK INVITE(ANM)					
				→	ACK					
			Conversation							
	REL	→		→	BYE(REL)					
	RLC	+		+	200 OK BYE(RLC)					

TP301029	SIP reference: RFC 3261	[4]	I	SUP reference:					
			Q.191	12.5 [1], clause 7.1.2					
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message								
SIP selection criteria									
ISUP selection criteria									
Test purpose	Called Party Number parameter of to the addr-spec component	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter of the IAM: to the addr-spec component of the To header field which shall include the "user=phone" URI parameter if the To header field contains a sip: URI.							
SIP parameter values	INVITE: To: sip:; user=phone								
ISUP parameter values									
Comments	ISUP/BICC	SL	JT	SIP-I					
	IAM -	•	→	INVITE(IAM)					
	ACM ←		+	180 Ringing(ACM)					
	ANM	•	+	200 OK INVITE(ANM)					
			→	ACK					
		Conver	sation						
	REL -	•	→	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	ISUP-SIP/Basic call/Sending of the INVITE message						
SIP selection criteria	NOT PICS 1/9							
ISUP selection criteria								
Test purpose	Called Party Number parame	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter of the IAM and the and the followed SAM: to the addr-spec component of the To header field.						
SIP parameter values	INVITE: To:							
ISUP parameter values								
Comments	ISUP/BICC		SU	Т		SIP-I		
	IAM	→						
	SAM	→						
	SAM	→			→	INVITE(IAM)		
	ACM	+			+	180 Ringing(ACM)		
	ANM	+			+	200 OK INVITE(ANM)		
				_	→	ACK		
			Convers	ation				
	REL	→		_	→	BYE(REL)		
	RLC	+			+	200 OK BYE(RLC)		

TP301031	SIP reference: RFC 32	61 [4]		ISUP reference: 12.5 [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 in the Called Party Number parameter contained in the Called Party address information of the IAM and followed SAM: to the addr-spec component of the To header field which shall include the "user=phone" URI parameter if the To header field contains a sip: URI.					
SIP parameter values	INVITE: To: sip:; user=phor	ne				
ISUP parameter values						
Comments	ISUP/BICC		SUT	SIP-I		
	IAM	→				
	SAM	→				
	SAM	→	→	INVITE(IAM)		
	ACM	(+	180 Ringing(ACM)		
	ANM	(+	200 OK INVITE(ANM)		
			→	ACK		
		Conv	ersation			
	REL	→	→	BYE(REL)		
	RLC	+	+	200 OK BYE(RLC)		

TP301032	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.1.4			
TSS reference	ISUP-SIP/Basic call/Sendi	ing of the Initia	al Address Me	ssage	(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	→		→	INVITE(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
				→	ACK		
		С	onversation				
	REL	→		→	BYE(REL)		
	RLC	+		←	200 OK BYE(RLC)		

TP301033	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.1.2				
TSS reference	ISUP-SIP/Basic call/Sending	of the I	NVITE me			• •/		
SIP selection criteria	PICS 1/9							
ISUP selection criteria	PICS 1/8	PICS 1/8						
Test purpose SIP parameter values	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:							
ISUP parameter values								
Comments	ISUP/BICC		SU ⁻	Г		SIP-I		
	IAM	→		-3	>	INVITE(IAM)		
	ACM	+		€	-	180 Ringing(ACM)		
	ANM ← 200 OK INVITE(ANN → ACK							
			Convers	ation				
	REL	→		7	•	BYE(REL)		
	RLC							

TP301034	SIP reference: RFC	3261 [4]		-	SUP reference: 12.5 [1], clause 7.1.2	
TSS reference	ISUP-SIP/Basic call/Sending	of the INVITE		4	1.1,	
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: URI, tag, parame	eters				
ISUP parameter values						
Comments	ISUP/BICC	5	SUT		SIP-I	
	IAM	→		1	INVITE(IAM)	
	ACM	+		+	180 Ringing(ACM)	
	ANM	+		+	200 OK INVITE(ANM)	
				→	ACK	
		Conve	ersation			
	REL	→		→	BYE(REL)	
	RLC	-		+	200 OK BYE(RLC)	

TP301035	SIP reference: RFC 326	1 [4]	ISUP reference: Q.1912.5 [1], clause 7.1.2			
TSS reference	ISUP-SIP/Basic call/Sending of t	he INVITE m	essage			
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: URI, tag, parameters	S				
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 =	• <u> </u>	→	BYE(REL)		
	RLC	-	+	200 OK BYE(RLC)		

TP301036	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	NOT PICS 1/8					
criteria						
Test purpose SIP parameter	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 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.					
values	INVITE: To:					
ISUP parameter values						
Comments	ISUP/BICC	SUT	SIP-I			
	IAM →					
	SAM →					
	SAM →	→	INVITE(IAM)			
	ACM ←	+	180 Ringing(ACM)			
	ANM ←	+	200 OK INVITE(ANM)			
		→	ACK			
	Conv	ersation				
	REL →	→	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]		-	ISUP reference: 912.5 [1], clause 7.2		
TSS reference	ISUP-SIP/Basic call/R	eceipt of SAM aft	er INVITE I				
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	→		→	INVITE(IAM)		
	SAM	→					
	ACM	+		+	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
				→	ACK		
		(Conversatio	n			
	REL	→		→	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 →			→	INVITE 1 (IAM)
	SAM →			→	INVITE 2 (IAM)
				+	484 Address Incomplete (1)
				→	ACK
	SAM →			→	INVITE 3 (IAM)
				+	484 Address Incomplete (2)
				→	ACK
	ACM ←			+	180 Ringing (3) (ACM)
	ANM ←			+	200 OK INVITE (3) (ANM)
				→	ACK
		Conv	ersation		
	REL →			→	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	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". Sends the INVITE after the receipt of the Continuity message with the Continuity Indicators parameter "continuity check successful". 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	→			
	SAM	→			
	COT	→	→	INVITE1(IAM)	
	SAM	→	→	INVITE2(IAM)	
			+	484 Address Incomplete (1)	
			→	ACK	
	ACM	+	+	180 Ringing (2) (ACM)	
	ANM	+	+	200 OK INVITE (2) (ANM)	
			→	ACK	
		Conv	ersation		
	REL	→	→	BYE(REL)	
	RLC	+	+	200 OK BYE(RLC)	

TP302004	SIP reference: RFC 3261	[4]		ISUP reference:	
	Q.1912.5 [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				
	Ensure that the SLIT in Idle state	on receipt o	f an IAM massac	so containing the Continuity	
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 performed on previous circuit". Sends the INVITE after the receipt of the Continuity message with the Continuity Indicators parameter "continuity check successful". 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	→			
	SAM	→			
	COT	→	→	INVITE 1 (IAM)	
	SAM	→	→	INVITE 2 (IAM)	
			+	484 Address Incomplete (1)	
			→	ACK	
	ACM	-	+	180 Ringing (2) (ACM)	
	ANM	-	+	200 OK INVITÉ(ANM)	
			→	ACK	
		Conv	ersation		
	REL	→	→	BYE(REL)	
	RLC	-	+	200 OK BYE(RLC)	

TP302005	SIP reference: RFC 3261 [4]	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	has been	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 ISUP parameter	Ensure that the SUT in Idle state 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 ISI 1) Stop timer TOIW3 (if it is runi 2) TOIW2 shall be restarted.	Connection sending of I ISUP time	Indicators parar NVITE is delaye T8 expires.	meter which is set to "continuity				
values								
Comments	ISUP/BICC		SUT	SIP-I				
	IAM	IAM →						
	SAM →							
			expires					
	REL	+						
	RLC	→						

TP302007	SIP refe	erence	: RFC 3261	[4]		ISUP reference: Q.1912.5 [1], clause 7.2.1		
TSS reference	ISLIP-SIP/Basio	call/R	eceint of S	ΔMa	fter invit	e has been sent		
SIP selection	PICS 3/2 AND	PICS 4	/5 AND PIC	:S 4/	15	e nas been sent		
criteria	1.100 0/27/1101 1100 1/101							
ISUP selection	PICS 1/5 AND PICS 4/2							
criteria								
Test purpose	Check indicator check required Sends an INVIT Indicators paral preconditions a On receipt of a 1) Stop timer 7 2) TOIW2 shal a) the Rec receive b) a new I INVITE c) the new that hav resource in ques	Issure that the SUT in Idle state, on receipt of an IAM message containing the Continuity neck indicator in the Nature of Connection Indicators parameter which is set "continuity neck indicator in the Nature of Connection Indicators parameter which is set "continuity neck required on this circuit". In the SIP message after the reception of the Continuity message with the Continuity dicators parameter set to "continuity check successful" and after the requested econditions 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	INVITE2: Request URI contains digits from the IAM and digits from SAM x and SAM y. The IAM is also contained							
values ISUP parameter	IAW IS also con	IVI 15 AISO COITAINEO						
values								
Comments								
Comments	ISUP/BICC		SUT		SIP-I			
	IAM	→	001	→	INVITE	1(IAM)		
	SAM x	→				()		
				+	183 Se	ession Progress without encapsulated ACM		
	COT	→		→	UPDA ⁻			
				+	200 Oł	(UPDATE		
	SAM y	→		→	INVITE	2 (IAM and digits from SAM X + SAM Y)		
	•			←	484 Ac	Idress Incomplete (1)		
				→	ACK			
	ACM	←		←	180 Ri	nging2 (ACM)		
	ANM	+		+	200 Oł	(INVITE(ANM)		
				→	ACK			
		С	onversatio	n				
	REL	→		→	BYE(R	EL)		
	RLC	+		←	200 Oł	(BYE(RLC)		

TP302008	SIP refe	rence	: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.2.1		
TSS reference	ISUP-SIP/Basic	call/R	eceint of SAM	after	invite			
SIP selection criteria	PICS 3/2 AND PICS 4/5 AND PICS 4/15							
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 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 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	original IAM. INVITE2: Request URI contains digits from the IAM and digits from SAM x and SAM y. The							
values	IAM is also contained							
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	+_	SUT	_	SIP-			
	IAM	→		→	IINVI	TE1(IAM)		
	SAM x	7	+	L	100	Consign Draggeon without an approvided A CAA		
	COT	→	+	<u>←</u>		Session Progress without encapsulated ACM DATE		
	СОТ	7	+	7		OK UPDATE		
	SAM	→		<u>▼</u>				
	SAIVI	7				TE2 (IAM and digits from SAM X + SAM Y)		
				Ć		Address Incomplete (1)		
	A CNA	1		_	ACK			
	ACM	+		<u>+</u>		Ringing2 (ACM)		
	ANM	+		<u> </u>		OK INVITE(ANM)		
				→	ACK	\		
		 _	Conversation			(2-1)		
	REL	→		→		(REL)		
	RLC	+		+	200	OK BYE(RLC)		

TP302009	SIP reference: RFC 3261 [4] ISUP reference:							
	10115 015/5 : 11/5 : : : : : : : : : : : : : : : : : : :			912.5 [1], clause 7.2.1				
TSS reference	ISUP-SIP/Basic call/Receipt of SA	M after invi	te has been ser	nt				
SIP selection	PICS 3/2 AND NOT PICS 4/15							
criteria	PIOC 4/4 AND NOT BIOC 4/9							
ISUP selection criteria	PICS 1/4 AND NOT PICS 4/2							
Test purpose	Ensure that the SUT in Idle state, expected".	•						
	The sending of the INVITE is dela							
	 Continuity message, with the received; 	Continuity I	ndicators param	neter set to " continuity " shall be				
	Type is "notification not require On receipt of a SAM from the BIC	red" was red C the SUT s	eived.	ase where the incoming Connect				
	 Stop timer TOIW3 (if it is runni TOIW2 shall be restarted and 		all invoke the fol	llowing 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.	, w 114 v 11 L c	ile interworked	nom the parameters of the				
SIP parameter	J. J							
values								
ISUP parameter								
values Comments	ISUP/BICC		SUT	SIP-I				
Comments		→	501	Oii -i				
		<u>→</u>						
		→	→	INVITE(IAM)				
		→	→					
		(+	\ /				
		((
			→					
		Conv	ersation					
		→	→	BYE(REL)				
	RLC	-	+	200 OK BYE(RLC)				

TP302010	SIP reference: RFC 3261 [4]			ISUP reference:					
				1912.5 [1], clause 7.2.1					
TSS reference	ISUP-SIP/Basic call/Receipt of SAM a	fter invite	has been s	ent					
SIP selection	PICS 3/2 AND NOT PICS 4/15								
criteria									
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								
		itinuity In	dicators para	meter set to " continuity " shall be					
	received;								
	APM with Action indicator set to "								
	or without bearer control tunnellin			g Connect Type is "notification					
	required", and for the fast set-up (
	On receipt of a SAM from the BICC that 1) Stop timer TOIW3 (if it is running);		iaii.						
	2) TOIW2 shall be restarted and the S		l invoke the t	following procedures:					
	a) the Request-URI and the To h	eader fie	ld of the new	/ INVITE shall contain all digits					
	received so far for this call;	loudor no	ia or the new	THE OTHER CONTENT OF GIG					
		Call-ID ar	d From head	der (including tag) as the previous					
	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								
		NVITE ar	e interworke	d from the parameters of the					
oin .	original IAM.								
SIP parameter									
values ISUP parameter									
values									
Comments	ISUP/BICC	S	UT	SIP-I					
	IAM →	 	0.	0.11					
	SAM x →								
	COT →			→ INVITE(IAM)					
	SAM y →			→ INVITE(IAM)					
	ACM	1		€ 180 Ringing(ACM)					
	ANM ←		1.	€ 200 OK INVITE(ANM)					
			1.	→ ACK					
		Conve	rsation						
	REL →			→ BYE(REL)					
	RLC ←			€ 200 OK BYE(RLC)					

TP302011	SIP reference: RFC 3261	[4]		ISUP reference:					
				12.5 [1], clause 7.2.1					
TSS reference	ISUP-SIP/Basic call/Receipt of SA	M after invi	te has been sent	<u> </u>					
SIP selection criteria	PICS 3/2 AND NOT PICS 4/15								
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; Bearer Set-up Connect indication - for the backward bearer set-up case 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; 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					
Commicing		→ `	-						
		>							
	G	,	→	INVITE(IAM)					
		>	→	INVITE(IAM)					
	S y	,	-	180 Ringing(ACM)					
		<u>`</u>	+	200 OK INVITE(ANM)					
		-	→	ACK					
		Conv	ersation						
	REL -	<u>→ </u>	→	BYE(REL)					
		<u>-</u>	+	200 OK BYE(RLC)					

TP302012	SIP reference: RFC 320	61 [4]		-	SUP reference:			
TCC votovonos	ICUD CID/Dania anll/Danaint of	CAM -44- ::			12.5 [1], clause 7.2.1			
TSS reference SIP selection	ISUP-SIP/Basic call/Receipt of SPICS 3/2 AND NOT PICS 4/15	SAM atter i	nvite has beei	n sent				
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; 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	→						
	SAM x	→						
	COT	→		→	INVITE(IAM)			
	SAM y	→		→	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
				→	ACK			
		Co	nversation					
	REL	→		→	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP302013	SIP refe	erenc	e: RFC 3261 [4]		ISUP reference:		
					Q.1912.5 [1], clause 7.2.1		
TSS reference					invite has been sent		
SIP selection criteria	PICS 3/2 AND PICS 4/5 AND 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". Sends the INVITE message. The events: Continuity message, with the Continuity Indicators parameter set to "continuity" was received; Bearer Set-up indication - for the forward bearer set-up case where the incoming Connect Type is "notification not required" was received; are indicating the successful completion of bearer set-up. 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; 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 con	taine	d				
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	→		→	INVITE1(IAM)		
	SAM x	7		_	183 Session Progress without encapsulated ACM		
	COT	_		←	UPDATE		
	001	→		7 ←	200 OK UPDATE		
	SAM y	→		→	INVITE2 (IAM and digits from SAM X + SAM Y)		
	OAIVI y	7	1	7	484 Address Incomplete (1)		
				→	ACK		
	ACM	+		/	180 Ringing2(ACM)		
	ANM	+		\	200 OK INVITE(ANM)		
	, at vivi	<u> </u>		`	ACK		
			Conversation				
	REL	→		→	BYE(REL)		
	RLC	/		+	200 OK BYE(RLC)		
	IVEO			_	1200 ON DIE(NEO)		

TP302014	SIP refere	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	invit	
SIP selection	PICS 3/2 AND PI					
criteria						
ISUP selection	PICS 1/4 AND PI	CS 4	/2			
criteria						
Test purpose	Ensure that the S expected".	UT ii	n Idle state, on	rece	ipt of	an IAM message indicating "COT to be
	Sends the INVITE	me	ssage The eve	nte:		
					uity Ir	dicators parameter set to "continuity" was
	or without be	arer	control tunnellii	ng) v	vhere	ed" - for the forward bearer set-up cases (with, ethe incoming Connect Type is "notification
	are indicating the	succ		ion c	of bea	arer set-up.
	On receipt of a SA 1) Stop timer TO	IW3	(if it is running)	,		
	a) the Reque	est-U				Il invoke the following procedures: eld of the new INVITE shall contain all digits
			,	Call	ID o	ad From hooder (including tog) as the provious
	INVITE is			Call-	·ID ai	nd From header (including tag) as the previous
				a ne	2w SI	OP offer. The O-IWU may re-use any resources
						is call. This re-use of existing reserved
						precondition attributes for the SDP parameters
	in questio					F
		onte	nts of the new	NVI	TE a	re interworked from the parameters of the
SIP parameter			I contains digits	fror	m the	IAM and digits from SAM x and SAM y. The
values	IAM is also contain		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 satis	sfied			10.0	
	ISUP/BICC		SUT		SIP	
	IAM	<u> </u>		→	INV	ITE1(IAM)
	SAM	→			400	
	COT			<u> </u>		Session Progress without encapsulated ACM
	СОТ	→		<u>→</u>		DATE
	CAM	_		<u>+</u>		OK UPDATE
	SAM	→		<u>→</u>		(ITE2 (IAM with digits from SAM X + SAM Y)
				<u>←</u>	484 ACI	Address Incomplete (1)
	ACM	+		7		National Control of the Control of t
	ANM	-		-		OK INVITE(ANM)
	VINIAI	7 -		<u>~</u>	ACI	
	-		Conversation	7	AUI	1
	REL	→	Conversation	→	DVΙ	E(REL)
	RLC	7		7		CKEL) OK BYE(RLC)
	NLO		<u> </u>	~	∠∪∪	ON DIE(NLO)

TP302015	SIP refere	ence	: RFC 3261 [4]		ISUP reference:	
TSS reference	Q.1912.5 [1], clause 7.2.1 ISUP-SIP/Basic call/Receipt of SAM after 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	expected". Sends the INVITE Continuity management of a Send of a	Is the INVITE message. The events: Continuity message, with the Continuity Indicators parameter set to "continuity" was				
CID novemeter	original IA				sees the LANA and distinct frame CANA are and CANA at The	
SIP parameter values	INVITE2: Reques		i contains digits	s tror	rom the IAM and digits from SAM x and SAM y. The	
ISUP parameter	IAIVI IS AISO CONTA	nea				
values						
Comments	The O-IWU shoul	d init	iate the precon	ditio	ion signalling procedure using the SDP Offer in the	
	exchange) the co the confirmation of message are sati	nfirm of a p	ation of a prece recondition bei	ondit	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	→	SIP-I	
	IAM	<u>→</u>	-	7	INVITE1(IAM)	
	SAM	7			4.02 Coopies Progress with suit an assemble to A.C.A.4	
	COT	_		<u>+</u>	U	
	СОТ	→		<u>→</u>		
	CAM	_		+		
	SAM	→		<u>→</u>	(
	ACM			<u>→</u>		
	ACM ANM	<u>←</u>		-	3 3 \ /	
	AINIVI	_				
			Conversation	→	ACK	
	DEL	_	Conversation	_	DVE(DEL)	
	REL	<u>→</u>		<u>→</u>		
	RLC	←	1	+	200 OK BYE(RLC)	

TP302016	SIP refer	ence	: RFC 3261 [4]		ISUP referen	
TSS reference	Q.1912.5 [1], clause 7.2.1 ISUP-SIP/Basic call/Receipt of SAM after invite has been sent					
SIP selection	PICS 3/2 AND PI				vite rias been sent	
criteria				/13		
ISUP selection	PICS 1/4 AND PI	CS 4	/2			
criteria						
Test purpose	be expected". Sends the INVITE Continuity mareceived; BNC set-up are indicating the On receipt of a S. Stop timer TC TOIW2 shall be an exerced by a new INVITE is continued in question.	nds the INVITE message. The events: Continuity message, with the Continuity Indicators parameter set to "continuity" was				
	original IA				•	
SIP parameter			I contains digits	s fror	the IAM and digits from SAM x	and SAM y. The
values	IAM is also conta	ined				
ISUP parameter						
values	T. 0 04/11 1			1		200000000000000000000000000000000000000
Comments	INVITE. The precedence exchange) the continuation of message are sati	ondit nfirm of a p	ion signalling is ation of a precorecondition bei	s cor ondit	signalling procedure using the Suded upon sending (within an Son being met. The SDP Offer or is sent when the conditions to	SDP offer-answer Answer carrying
	ISUP/BICC	_	SUT	_	SIP-I	
	IAM	<u>→</u>		→	NVITE1(IAM)	
	SAM	→			02 Cassian Drawns with	
	007			<u> </u>	83 Session Progress without e	ncapsulated ACM
	СОТ	→		<u>→</u>	IPDATE	
	0.4.14			<u>+</u>	00 OK UPDATE	A.B.A
	SAM	→		<u>→</u>	NVITE2 (IAM with digits from S.	AIVI X + SAM Y)
				<u> </u>	84 Address Incomplete (1)	
	4014			<u>→</u>	CK	
	ACM	+		+	80 Ringing2(ACM)	
	ANM	+		<u>+</u>	00 OK INVITE(ANM)	
				→	.CK	
	551		Conversation		VE(DEL)	
	REL	→		<u>→</u>	YE(REL)	
	RLC	←		←	00 OK BYE(RLC)	

TP302017	SIP reference: RFC 3261	[4]	Q.19	ISUP reference: 912.5 [1], clause 7.2.1					
TSS reference	ISUP-SIP/Basic call/Receipt of SA	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent							
SIP selection criteria	PICS 3/2								
ISUP selection criteria	PICS 1/4								
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									
Comments	ISUP/BICC		SUT	SIP-I					
	IAM	→	→	INVITE(IAM)					
	SAM	→	→	INVITE(IAM)					
		T _{oiw}	₂ expired						
	SAM	→							
	- 1-11	←	+	180 Ringing(ACM)					
	ANM	(+	=======================================					
			→	ACK					
			versation						
	• • • • • • • • • • • • • • • • • • • •	→	→	-:=(::==)					
	RLC	(+	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 af	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	 The SUT in Idle state, on receipt of an IAM message. On receipt of a SAM from the BICC/ISUP the SUT shall: sends a INVITE message if the minimum number of digits for routing the call has been received in the IAM and the SAM; 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. 								
SIP parameter values									
ISUP parameter values									
Comments	ISUP/BICC	S	UT	SIP-I					
	IAM →								
	SAM →		→	INVITE(IAM)					
		T _{oiw2} 6	expired						
	SAM →								
	ACM ←		+	180 Ringing(ACM)					
	ANM ←		+	200 OK INVITE(ANM)					
			→	ACK					
		Conve	rsation						
	REL →		→	BYE(REL)					
	RLC +		←	200 OK BYE(RLC)					

5.2.2.3 Sending of the ACM message

TP303001	SIP reference: RFC 3261	[4]	Į;	SUP reference:			
				12.5 [1], clause 7.1,			
				[i.12], clause 2.1.4.8			
TSS reference	ISUP-SIP/Basic call/Sending of the	e ACM mess	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 sen						
	Sends the INVITE message to call			ssage with:			
	• the CPS indicator set to " no						
	 the Called party's category i 	ndicator se	t to "no indication	n(00)" or "ordinary subscriber			
	(01)" or "payphone (10)";						
	the interworking indicator set to " INT_IND_VAL";						
	the ISUP indicator set to "ISU						
	 the ISDN access indicator se 	et to "ISDN_	_ACC_IND_VAL".				
SIP parameter							
values							
ISUP parameter	IAM; Called party number: compl						
values	ACM, CPS indicator no indication						
	Called party's category indicator	r: no indicat	ion(00) or ordinar	y subscriber (01) or payphone			
	(10)	\	-				
	interworking indicator: INT_IND)				
	ISUP indicator: ISUP_IND_ID (PI		(D1)(IT)				
Cammanta	ISDN access indicator ISDN_AC			loip i			
Comments	ISUP/BICC		SUT	SIP-I			
		•	→	INVITE(IAM)			
				100 D: : (100 N)			
	\ 3/		-	180 Ringing(ACM)			
	ANM	-	<u> </u>	200 OK INVITE(ANM)			
			`	ACK			
	DEI.		ersation	DVE(DEL)			
	REL		→	BYE(REL)			
	RLC •		+	200 OK BYE(RLC)			

TP303002	SIP reference: RFC 326	[4]	Q.19	SUP reference: 12.5 [1], clause 7.1, [i.12], clause 2.1.4.8			
TSS reference	ISUP-SIP/Basic call/Sending of the	ne ACM me	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 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							
ISUP parameter values	ACM, Backward call indicator is sindication (00) Called party's category indicate (10) interworking indicator: INT_INI	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			
	IAM	→	→	INVITE(IAM)			
	ACM(no indication)	-					
	CPG(Alerting)	-	+	180 Ringing(ACM)			
	ANM	(+	200 OK INVITE(ANM)			
			→	ACK			
		Con	versation				
	REL	→	→	BYE(REL)			
	RLC	(+	200 OK BYE(RLC)			

TP303003	SIP reference: RFC 3261		Q.19 Q.764	SUP reference: 12.5 [1], clause 7.1, [i.12], clause 2.1.4.8		
TSS reference	ISUP-SIP/Basic call/Sending of the	e ACM mes	sage			
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 =		→	INVITE(IAM)		
	ACM(no indication)					
	CPG(Alerting)		+	180 Ringing(ACM)		
	ANM		+	200 OK INVITE(ANM)		
			→	ACK		
		Conv	ersation			
	REL =	>	→	BYE(REL)		
	RLC	-	+	200 OK BYE(RLC)		

TP303004	SIP reference: RFC 326	1 [4]			19	SUP reference:
11 000007	Cir reference. Nr 0 320	. [-]		Q.1912	-	clauses 7.1 1) d), 7.3.1, 7.4
TSS reference	ISUP-SIP/Basic call/Sending of t	he A0	CM messa		··· [·],	,,,
SIP selection	PICS 1/3			.3-		
criteria						
ISUP selection	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 where the end of address signalling is determined by the expiration timer T _{OIW1} after the receipt of the latest address message: • sends the INVITE message to called user; • sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "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	IAM; Called party number: com					
values	ACM, CPS indicator no indication					
	Called party's category indicat	or: no	o indicatio	n(00) or 0	ordina	ry subscriber (01) or payphone
	(10) interworking indicator: INT_IN	D \/A	I (DIVIT)			
	ISUP indicator: ISUP_IND_ID (
	ISDN access indicator ISDN_A			PIXIT)		
Comments	ISUP/BICC		SU			SIP-I
	IAM	→				
			T _{OIW1}	expiry		
	ACM(no indication)	+			→	INVITE(IAM)
	CPG(Alerting)	←			+	180 Ringing(ACM)
	ANM	←			+	200 OK INVITE(ANM)
					→	ACK
			Conve	sation		
	REL	→			→	BYE(REL)
	RLC	+			←	200 OK BYE(RLC)

TP303005	SIP reference: RFC 3261	[4]		ISUP reference:		
				5 [1], clauses 7.1, 7.3.1		
TSS reference	ISUP-SIP/Basic call/Sending of the	e ACM mes	sage			
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 minimum number of digits required for routing the call has been received (start timer TOIW2 and invoke the appropriate outgoing SIP signalling procedure): • sends an INVITE message to the called user and after the expiration of T _{OIW2} ; • sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "INT_IND_VAL", the ISUP indicator					
	set to "ISUP_IND_ID", the INTERWO					
SIP parameter	Set to 130F_IND_ID , the 13	DIN ACCESS	nuicator set to	ISDN_ACC_IND_VAL .		
values						
ISUP parameter	IAM; Called party number: comp	olete number	,			
values	ACM, CPS indicator no indication					
	Called party's category indicate	r: no indicat	ion(00) or ordina	ry subscriber (01) or payphone		
	(10)					
	interworking indicator: INT_IND		Γ)			
	ISUP indicator: ISUP_IND_ID (P					
_	ISDN access indicator ISDN_AC			T		
Comments	ISUP/BICC		SUT	SIP-I		
	11 11 11 11 11 11 11 11 11 11 11 11 11)				
	<u> </u>)				
	SAM	<u>→ </u>	<u></u>	INVITE(IAM)		
		T _{OIW}	₂ expiry			
	7 (011)(110 1114110411011)	(
	CPG(Alerting)	(+	180 Ringing(ACM)		
	ANM	(+	200 OK INVITE(ANM)		
			→	ACK		
			ersation			
	REL	→	→	BYE(REL)		
	RLC	←	+	200 OK BYE(RLC)		

TP303006	SIP reference: RFC 326	1 [4]		ISUP reference:				
				[1], clauses 7.1 1) a), 7.3.1				
TSS reference	ISUP-SIP/Basic call/Sending of the	he ACM mess	sage					
SIP selection	PICS 1/3							
criteria								
ISUP selection	NOT PICS 4/9							
criteria								
Test purpose	Ensure that the SUT in Idle state			ge containing the complete				
	called party number, on receipt	of a 180 Ring	ging message:					
	 Sends the ACM message wi 							
	 the CPS indicator set 							
				alue in the encapsulated ACM;				
	 the interworking indi- 							
	 the ISUP indicator se 							
	 the ISDN access indi 	cator set to the	ne value in the e	encapsulated ACM.				
SIP parameter								
values								
ISUP parameter	IAM; Called party number: com							
values	ACM, Backward call indicator is s							
Comments	ISUP/BICC		SUT	SIP-I				
	IAM	→	→	INVITE(IAM)				
	ACM	-	+	180 Ringing(ACM)				
	ANM	ANM ← 200 OK INVITE(ANM)						
			→	ACK				
		Conve	ersation					
	REL	→	→	BYE(REL)				
	RLC	←	+	200 OK BYE(RLC)				

TP303007	SIP reference: RFC 326	1 [4]		15	SUP reference:			
	0 10.0.0.001111 0 020		Q.191		1], clauses 7.1 1 a), 7.3.2			
TSS reference	ISUP-SIP/Basic call/Sending of t	he ACM mes			.1, 0.00000 0,,			
SIP selection criteria	PICS 3/1							
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 183 Session Progress with encapsulated ACM: sends the ACM message; the encapsulated ACM message is sent unchanged backward.							
SIP parameter values								
ISUP parameter values	IAM; Called party number: com	piete numbe	l					
Comments	ISUP/BICC		SUT		SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM(no indication)	+		+	183 Session Progress(ACM)			
	CPG(Alerting)	+		+	180 Ringing(CPG)			
	ANM	+		+	200 OK INVITE(ANM)			
				→	ACK			
		Conv	ersation					
	REL	→		→	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	PICS 4/2 AND 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 where the end of address signalling is determined by the expiration timer T _{OIW1} 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	30110 1001 _1110	<u>, UI</u>	uic iobit	4000	SS Indicator Set to TOBIN_NOO_IND_VALE.			
ISUP parameter	IAM; Called party nu	mber:	complete	num	her			
values	ACM,		complete					
	CPS indicator no ind	ication	(00)					
				o ind	ication(00) or ordinary subscriber (01) or payphone			
	(10)interworking ind	icator	: INT IND	VA	L (PIXIT)			
	ISUP indicator: ISUP							
	ISDN access indicate				VAL (PIXIT)			
Comments	ISUP/BICC		SUT		SIP-I			
	IAM	→			INVITE(IAM)			
					183 Session Progress without encapsulated ACM			
	COT	→			UPDATE			
					200 OK UPDATE			
		T	_{DIW1} expiry					
	ACM(no indication)	+						
			+					
		+		←	180 Ringing(ACM)			
	CPG(Alerting, BCi) ANM	+			180 Ringing(ACM) 200 OK INVITE(ANM)			
	CPG(Alerting, BCi)			→	200 OK INVITE(ANM)			
	CPG(Alerting, BCi)	+		←				
	CPG(Alerting, BCi)	+	onversation	<u></u> ←	200 OK INVITE(ANM)			

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/Sending of the INVITE message							
SIP selection	PICS 1/3 AND PICS 3/2 AND PICS 4/5 AND PICS 4/4 AND PICS 4/15							
criteria								
ISUP selection	PICS 4/2 AND NOT P	ICS 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 minimum number of digits required for routing the call 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 _{oiw2} : • sends the ACM message with the CPS indicator set to " no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "INT_IND_VAL", the ISUP indicator set							
CID noromotor	to "ISUP_IND_ID	, tne	ISDN acces	SS II	ndicator set to "ISDN_ACC_IND_VAL".			
SIP parameter values								
ISUP parameter	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	→	-		INVITE(IAM)			
					183 Session Progress without encapsulated ACM			
	COT	→			UPDATE			
				E	200 OK UPDATE			
		T	_{DIW2} expiry					
	ACM(no indication)	+						
	CPG(Alerting, BCi)	+		(180 Ringing(ACM)			
	ANM	+			200 OK INVITE(ANM)			
			-	}	ACK			
		Co	onversation					
	REL	→			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	DT PI	ICS 4/9				
criteria							
Test purpose					pt of an IAM message containing the complete		
					is performed (ISUP) or COT is expected (BICC)		
	indication receipt		0 0	ssag	ge:		
			nessage with:				
					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; 					
OID	the ISDN access indicator set to the value in the encapsulated ACM.						
SIP parameter							
values	IAM. Callad marts				sh a r		
ISUP parameter values	IAM; Called party						
Comments	ISUP/BICC	ali ir		tne	value in the encapsulated ACM SIP-I		
Comments		→	SUT	→			
	IAM	7		<u> </u>	INVITE(IAM)		
	COT	→		<u>▼</u>	183 Session Progress without encapsulated ACM UPDATE		
	COT	7		<u> </u>			
	ACM	+		-	200 OK UPDATE		
	ACM	-		-	180 Ringing(ACM)		
	ANM	_		<u>~</u>	200 OK INVITE(ANM)		
			Conversation	7	ACK		
	REL	→	Conversation	→	DVE/DEL\		
		7		<u>₹</u>	BYE(REL)		
	RLC	_		~	200 OK BYE(RLC)		

TP303014	SIP reference: RFC 3261 [4]			SUP reference:			
<i>(</i>	10115 015/5 : 11/0 !: (1)			1], clauses 7.1, 7.3.1, 7.4			
TSS reference	ISUP-SIP/Basic call/Sending of the IN	VIIE mes	sage				
SIP selection criteria	PICS 1/3 AND PICS 3/2 AND NOT PIC	JS 4/15					
ISUP selection	PICS 3/8 AND PICS 4/2 AND NOT PIC	CS 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 minimum number of digits required for routing the call has been received (start timer T _{OIW2} 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 _{Oiw2} :						
	 sends the ACM message with the CPS indicator set to " no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "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	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						
Comments	180 Ringing ISUP/BICC	SL	IT	SIP-I			
Comments	IAM →	SU	71	OIF-I			
	COT -		→	INVITE(IAM)			
	9	T _{OIW2}		JINVII E(IAWI)			
	ACM(no indication) ←						
	CPG(Alerting) ←		+	180 Ringing(ACM)			
	ANM ←		+	200 OK INVITE(ANM)			
			→	ACK			
		Conver	sation				
	REL →		→	BYE(REL)			
	RLC +		←	200 OK BYE(RLC)			

TP303015	SIP reference: RFC 3261 [4]		IS	SUP reference:				
		Q.19	12.5 [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 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 ISUP indicator set to the value in the encapsulated ACM;							
SIP parameter values								
ISUP parameter	IAM; Called party number: complete nu	umber						
values	ACM, Backward call indicator is set to th		capsul	ated ACM				
Comments	ISUP/BICC	SUT		SIP-I				
	IAM →							
	COT →		→	INVITE(IAM)				
	ACM ←		+	180 Ringing(ACM)				
	ANM ←		+	200 OK INVITE(ANM)				
			→	ACK				
		Conversation						
	REL →		→	BYE(REL)				
	RLC ←		+	200 OK BYE(RLC)				

5.2.2.4 Sending of the CPG message

TP304001	SIP reference: RFC 326	1 [4]	ISUP reference: Q.1912.5 [1], clauses 7.1, 7.3.1					
TSS reference	ISUP-SIP/Basic call/Sending of the CPG message							
SIP selection	PICS 3/1							
criteria								
ISUP selection	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 event indicator set to "Alerting".							
SIP parameter								
values								
ISUP parameter	ACM: BCi called party status ind	ACM: BCi called party status indicator = no indication						
values	CPG: Event Indicator = ALERTIN	NG, BCi as re	ceived from the e	ncapsulated ACM				
Comments	ISUP/BICC		SUT	SIP-I				
	IAM	→						
	SAM	→						
	SAM	→	→	INVITE(IAM)				
		T _{OIW}	₂ expiry					
	ACM(no indication)	+						
	CPG(Alerting BCi)	+	+	180 Ringing(ACM)				
	ANM	+	-	200 OK INVITE(ANM)				
			→	ACK				
		Conv	ersation					
	REL	→	→	BYE(REL)				
	RLC	+	+	200 OK BYE(RLC)				

TP304002	SIP reference: RFC	Q	-	SUP reference: 5 [1], clauses 7.1, 7.3.1					
TSS reference	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 event indicator set to "Alerting".								
SIP parameter values									
ISUP parameter values									
Comments	ISUP/BICC		SUT		SIP-I				
	IAM	→		→	INVITE(IAM)				
	ACM(no indication)	+		+	183 Session Progress(ACM)				
	CPG(Alerting)	+		+	180 Ringing(CPG)				
	ANM	+		+	200 OK INVITE(ANM)				
				→	ACK				
			Conversation	•					
	REL	→		→	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	Ensure that the SUT having sent the ACM message, on receipt of a 200 OK INVITE for this call, it shall stop timer TOIW2 (if running): send ANM as determined by BICC/ISUP procedures; stop any existing awaiting answer indication (e.g. ringing tone).								
SIP parameter values	200 OK INVITE;			,					
ISUP parameter values	ANM;								
Comments	ISUP/BICC	SUT		SIP-I					
	IAM →		→	INVITE(IAM)					
	ACM ←		+	180 Ringing(ACM)					
	ANM ←		+	200 OK INVITE(ANM)					
			→	ACK					
		Conversation	•						
	REL →		→	BYE(REL)					
	RLC ←		+	200 OK BYE(RLC)					

5.2.2.6 Sending of the CON message

TP306001	SIP reference: RFC 3261 [4]		-	SUP reference:						
				5 [1], clauses 7.5, 7.5.1						
TSS reference	ISUP-SIP/Basic call/Sending of the Co	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 t	the ACM me	essage, on r	eceipt of a 200 OK INVITE for						
	this call, it shall stop timer TOIW2 (if ru	ınning):								
	 send CON as determined by BICC 	C/ISUP prod	cedures.							
	Stop any existing awaiting answer indi	cation (e.g.	ringing tone	e) BCI encoded as received in						
	the encapsulated CON.	` ` `	0 0	'						
SIP parameter	200 OK INVITE;									
values										
ISUP parameter	CON; interworking indicator: INT_IN	D_VAL (PI	XIT)							
values	ISUP indicator: ISUP_IND_ID (PIXIT))								
	ISDN access indicator ISDN_ACC_II	ND_VAL (P	IXIT)							
	CPS indicator: no indication		•							
Comments	ISUP/BICC	SUT		SIP-I						
	IAM →		→	INVITE(IAM)						
	CON ←		+	200 OK INVITE(CON)						
			→	ACK						
		Conversat	on							
	REL →		→	BYE(REL)						
	RLC ←		+	200 OK BYE(RLC)						

5.2.2.7 Receipt of the Release message (REL)

TP307001	SIP reference: RFC 3	261 [4	•	ISUP reference: Q.1912.5 [1], clause 7.7.1, 1)				
TSS reference	ISUP-SIP/Basic call/Receipt c	of the F	Release message	(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		SUT	SIP-I				
	IAM	→						
	REL	→						
	RLC +							

TP307002	SIP reference: RFC 3	3261 [4]		ISUP reference: Q.1912.5 [1], clause 7.7.1 2)					
TSS reference	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 before 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		SUT		SIP-I				
	IAM	→		→	INVITE(IAM)				
	REL	→							
	RLC	+							
				+	200 OK INVITE(CON)				
				→	ACK				
				→	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	 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: 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. 								
SIP parameter									
values									
ISUP parameter									
values			_	Ta.= .					
Comments	ISUP/BICC	SL		SIP-I					
	IAM	→	→	INVITE(IAM)					
			+	100 Trying					
	REL	→							
	RLC	(→	CANCEL(REL)					
			+	200 OK INVITE(CON)					
			→	ACK					
			+	200 OK CANCEL					
			→	BYE(REL)					
			+	200 OK BYE(RLC)					

TP307004	SIP reference: RFC 32	61 [4]	I	SUP reference:					
			Q.1912.	5 [1], clause 7.7.1 2) 3)					
TSS reference	ISUP-SIP/Basic call/Receipt of	the Release m	essage (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 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	SL	IT	SIP-I					
	IAM	→	→	INVITE(IAM)					
	REL	→							
	RLC	+							
			+	100 Trying					
			→	CANCEL(REL)					
			+	200 OK CANCEL					
			←	487 Request terminated					
			→	ACK					

TP307005	SIP reference: RFC 3261 [4]			Q.19	ISUP reference: 12.5 [1], clause 7.7.1 4)				
TSS reference	ISUP-SIP/Basic call/Receipt of	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 after 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		SU	Γ	SIP-I				
	IAM	→		→	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
	REL	↑							
	RLC	+		→	ACK				
				→	BYE(REL)				
				+	200 OK BYE(RLC)				

TP307006	SIP reference: RFC 3261 [4]			-	SUP reference: 2.5 [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		SUT		SIP-I			
	IAM	→		→	INVITE(IAM)			
				←	SIP_MESSAGE_VA			
	REL	→						
	RLC	←						
	CASE A							
				→	CANCEL(REL)			
				←	200 OK CANCEL			
				←	487 Request terminated			
				→	ACK			
	CASE B							
				→	BYE(REL)			
				-	200 OK BYE			
				+	487 Request terminated			
				→	ACK			

Table 8

	Values for test purpose TP307106			
VA	SIP MESSAGE_VA			
VA_1	180 Ringing			
VA_2	181 Call Is Being Forwarded			
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]		Q.19	ISUP reference: 0.1912.5 [1], clause 7.7.2			
TSS reference	ISUP-SIP/Basic call/Sending of the Re	elease messa		• •			
SIP							
selection							
criteria							
ISUP							
selection							
criteria							
Test	Ensure that the SUT after receiving the	e IAM sends	out an INVIT	E message and on receipt of a			
purpose	BYE message in the confirmed dialogu	ue:					
-	 sends a REL message constructe 	d from the er	ncapsulated I	REL in the received BYE.			
SIP	-		•				
parameter							
values							
ISUP	REL; Cause value "Normal call clearin	g"					
parameter		•					
values							
Comments	ISUP/BICC	SU		SIP-I			
	IAM -		→	INVITE(IAM)			
	ACM ←		(180 Ringing(ACM)			
	ANM		+	200 OK INVITE(ANM)			
			→	ACK			
	Co	onversation	•				
	REL ←	i.	+	BYE(REL)			
	RLC ->		→	200 OK BYE(RLC)			

TP308002	SIP reference: RFC 3	261 [4	ı]			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	Τ		SIP-I			
	IAM	→			^	INVITE(IAM)			
					4	100 Trying			
	REL	+			4	SIP_Failure_VA(REL)			
	RLC	→			→	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 Rel	ease me			<u> </u>		
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			SIP-I		
	IAM	→		1	→	INVITE(IAM)		
					(100 Trying		
	REL	→		1	→	CANCEL(REL)		
	RLC	+			(200 OK CANCEL		
					,	107.5		
					<u>←</u> →	487 Request Terminated ACK		

TP308004	SIP reference: RFC 3261 [4]			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									
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								
Comments	ISUP/BICC		SU	Т		SIP-I			
	IAM → INVITE(IAM)								
	XXX	4			+	SIP MESSAGE_VA			
	REL	+			+	SIP_Failure_VA(REL)			
	RLC	→			→	ACK			

Table 10

	Values for test purpose TP308004				
VA	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 3	.]	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 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	Т		SIP-I		
	IAM → INVITE(IAM)							
	ACM ← 180 Ringing							
	REL	+			+	SIP_Failure_VA(REL)		
	RLC	→			→	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 3261 [4]		ISUP reference:					
	Q.1912.5 [1], clause 7.7.6							
TSS reference	ISUP-SIP/Basic call/Sending	of the Rele	ase m	essage (RI	EL)			
SIP selection	NOT PICS 4/21							
criteria								
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	REL; cause value: CV_ISUP							
values								
Comments	ISUP/BICC		SU	Т		SIP-I		
	IAM	→			→	INVITE(IAM)		
		← 100 Trying						
	REL	4			←	SIP_Response_VA		
	RLC	→			→	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 3	SIP reference: RFC 3261 [4] ISUP 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	T	SIP-I				
	IAM	→						
	RSC	→						
	RLC	+						

TP309002	SIP reference: RFC 3	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 the IAM with the complete called party number, sending a INVITE message on receipt RSC message before a SIP MESSAGE_VA response message has been received: the SUT shall hold the RSC message until a SIP response has been received; the SUT shall send a CANCEL or BYE request. The RSC is not encapsulated.							
SIP parameter values	CANCEL or BYE: A REL is er							
ISUP parameter values								
Comments	ISUP/BICC		SU	T	SIP-I			
	IAM	→		→	INVITE(IAM)			
	RSC	→						
	RLC	+						
				+	SIP_MESSAGE_VA			
	CASE A							
				→	CANCEL			
				+	200 OK CANCEL			
				+	487 Request terminated			
				→	ACK			
	CASE B							
				→	BYE(REL#31)			
				+	200 OK BYE(RLC)			
				+	487 Request terminated			
				→	ACK			

Table 15

	Values for test purpose TP309002					
VA	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:				
				, 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	n cause 31						
ISUP parameter values								
Comments	ISUP/BICC	SU	IT	SIP-I				
	IAM -	▶	→	INVITE(IAM)				
	RSC -	▶						
	RLC	-						
			+	200 OK INVITE(CON)				
			→	ACK				
			→	BYE(REL#31)				
			+	200 OK BYE(RLC)				

TP309005	SIP reference: RFC 3261 [4]			ı	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 block	ing mes	sage (CG	B) with the indi	cation hardware failure			
	oriented							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the SUT after re							
	sending a INVITE message							
		•		•	message has been received:			
	 the SUT shall send a BY 	YE requ	est The RS	SC is not encap	osulated.			
SIP parameter	BYE: A REL is encapsulated	l with ca	use 31					
values								
ISUP parameter								
values								
Comments	ISUP/BICC		SU	Т	SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		←	200 OK INVITE(ANM)			
				→	ACK			
	RSC	→		→	BYE(REL#31)			
	RLC	+		+	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					, Circuit group reset message			
	(GRS) or Circuit group blocking message (CGB) with the indication 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	ated with	cause 31				
ISUP parameter								
values								
Comments	ISUP/BICC		SU		SIP-I			
	IAM	→		→	INVITE(IAM)			
				←	SIP_MESSAGE_VA			
	RSC	→						
	RLC	←						
	CASE A							
				→	CANCEL			
				←	200 OK CANCEL			
				←	487 Request terminated			
				→	ACK			
	CASE B							
				→	BYE(REL#31)			
				+	200 OK BYE(RLC)			
				+	487 Request terminated			
				→	ACK			

Table 16

	Values for test purpose; TP309006						
VA	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 326	1 [4]	_	SUP reference: 5 [1], clauses 7.7.1, 1), 7.7.4, 7.7.5				
TSS reference		P-SIP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message S) or Circuit group blocking message (CGB) with the indication hardware failure nted						
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 SUT SIP-I							
	7 (1)	>						
	5	>						
	GRA							

TP309008	SIP reference: RFC 3	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	sending a INVITE message of response message has been the SUT shall hold the GF	and detailed and d					
SIP parameter values							
ISUP parameter values							
Comments	ISUP/BICC		SU	Т	SIP-I		
	IAM	→		→	INVITE(IAM)		
	GRS	→					
	GRA	←					
				+	SIP MESSAGE VA		
	CASE A						
				→	CANCEL		
				+	200 OK CANCEL		
				+	487 Request terminated		
				→	ACK		
	CASE B	1					
				→	BYE(REL#31)		
				+	200 OK BYE(RLC)		
				+	487 Request terminated		
				→	ACK		

Table 17

	Values for test purpose TP309008					
VA	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:		
			Q.1912.5 [1],	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	 Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt GRS message before a 200 OK response message has been received: the SUT shall hold the GRS message until a response has been received. A CANCEL is sent The GRS is not encapsulated; on subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent. 					
SIP parameter values						
ISUP parameter values						
Comments	ISUP/BICC	SL	IT	SIP-I		
	IAM	→	→	INVITE(IAM)		
			+	100 Trying		
	GRS	→				
	GRA	+	→	CANCEL		
			+	200 OK INVITE(CON)		
			→	ACK		
			+	200 OK CANCEL		
			→	BYE(REL#31)		
			+	200 OK BYE(RLC)		

TP309011	SIP reference: RFC 3	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 the IAM with the complete called party number, sending a INVITE message with the complete called party number, sending a INVITE message on receipt GRS message after a 200 OK response message has been received: • the SUT shall send a BYE request The GRS is not encapsulated.						
SIP parameter							
values							
ISUP parameter							
values							
Comments	ISUP/BICC		SU	T	SIP-I		
	IAM	→		→	INVITE(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
				→	ACK		
	GRS	→		→	BYE(REL#31)		
	GRA	←		+	200 OK BYE(RLC)		

TP309012	SIP reference: RFC	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 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	T	SIP-I	
	IAM	→		→	INVITE(IAM)	
				+	SIP_MESSAGE_VA	
	GRS	→				
	GRA	+				
	CASE A					
				→	CANCEL	
				+	200 OK CANCEL	
				←	487 Request terminated	
				→	ACK	
	CASE B					
				→	BYE(REL#31)	
				+	200 OK BYE(RLC)	
				+	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: 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 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 CSeg of INVITE1					
values	BYE2 contains the CS	Seg of INVITE2				
ISUP parameter values		•				
Comments	ISUP/BICC		SUT		SIP-I	
	IAM	→		→	INVITE1(IAM)	
	ACM	+		+	180 Ringing(ACM)	
	ANM	+		+	200 OK INVITE(ANM)	
				→	ACK	
	IAM	→		→	INVITE2(IAM)	
	ACM	+		+	180 Ringing(ACM)	
	ANM	+		+	200 OK INVITE(ANM)	
				→	ACK	
	GRS	→				
	GRA	+				
				→	BYE1(REL#31)	
				+	200 OK BYE(RLC)	
				→	BYE2(REL#31)	
				+	200 OK BYE(RLC)	

5.2.2.9.3 Receipt of Circuit group blocking message (CGB)

TP3090014	SIP reference: RFC 3261 [4] ISUP reference:							
TSS reference	Q.1912.5 [1], clauses 7.7.1, 1), 7.7.4 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	onened							
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	CGB -	SUT SIP-I						
	CGBA +	•						

TP309015	SIP reference: RFC	3261 [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 orien	ted)					
Comments	ISUP/BICC		SU	Т	SIP-I		
	IAM	→		→	INVITE(IAM)		
	CGB	→					
	CGBA	+					
				+	SIP_MESSAGE_VA		
	CASE A						
				→	CANCEL		
				+	200 OK CANCEL		
				+	487 Request terminated		
				→	ACK		
	CASE B						
				→	BYE(REL#31)		
				+	200 OK BYE(RLC)		
				+	487 Request terminated		
				→	ACK		

Table 19

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

TP3090016	SIP reference: RFC 3261	[4]	I -	SUP reference: 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 (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 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	SU	Т	SIP-I			
	IAM →		→	INVITE(IAM)			
			+	100 Trying			
	CGB →						
	CGBA ←		→	CANCEL			
			+	200 OK INVITE(CON)			
			→	ACK			
			-	200 OK CANCEL			
			→	BYE(REL#31)			
			+	200 OK BYE(RLC)			

TP309017	SIP reference: RFC 3	261 [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 (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 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	d)					
Comments	ISUP/BICC	S	UT	SIP-I			
	IAM	→	→	INVITE(IAM)			
	ACM	+	+	180 Ringing(ACM)			
	ANM ← 200 OK INVITE(ANM)						
	→ ACK						
	CGB	→	→	BYE(REL#31)			
	CGBA	←	+	200 OK BYE(RLC)			

TP309018	SIP reference: I	RFC 3261 [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" 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 CGB is not encapsulated.						
SIP parameter values							
ISUP parameter values	CGB(hardware failure oriented)						
Comments	ISUP/BICC		SUT		SIP-I		
	IAM	→		→	INVITE(IAM)		
				+	SIP_MESSAGE_VA		
	CGB	→					
	CGBA	←					
	CASE A	, , , , , , , , , , , , , , , , , , ,					
				→	CANCEL		
				+	200 OK CANCEL		
				+	487 Request terminated		
				→	ACK		
	CASE B						
				→	BYE(REL#31)		
				-	200 OK BYE(RLC)		
				-	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:				
], 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	CGB(hardware failure oriented)						
values	`	,					
Comments	ISUP/BICC	Sl	JT	SIP-I			
	IAM	→	→	INVITE1(IAM)			
	ACM	+	+	180 Ringing(ACM)			
	ANM	+	+	200 OK INVITE(ANM)			
			→	ACK			
	IAM	→	→	INVITE2(IAM)			
	ACM	+	+	180 Ringing(ACM)			
	ANM	+	+	200 OK INVITE(ANM)			
			→	ACK			
	CGB	→					
	CGBA	+					
			→	BYE1(REL#31)			
			+	200 OK BYE(RLC)			
			→	BYE2(REL#31)			
			+	200 OK BYE(RLC)			

5.2.2.10 Receipt of Confusion message

TP310001	SIP refer	ence: RFC 32	61 [4]		ISUP reference:					
TSS reference	Q.1912.5 [1], clause A.1.1.3 ISUP-SIP/ISUP Messages for special consideration / Confusion message									
SIP selection	1001 -011 /1001 Wessages for special consideration / Confusion message									
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	gress with end	apsulated CFN	1						
ISUP parameter values	CFN									
Comments	ISUP				SIP-I					
	IAM	→		→	INVITE(IAM with unknown parameter)					
	CFN	+		(183 Session Progress(CFN)					
	ACM	+		(180 Ringing(ACM)					
	ANM	+		(200 OK INVITE(ANM)					
				→	ACK					
		С	ommunication							
	REL	→		→	BYE(REL)					
	RLC	+		(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	 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: ensure that the ISUP message is transported through the SIP network encapsulated in the INFO message; ensure that the called subscriber can successfully clear back and reanswer the call. 							
SIP parameter				,				
values								
ISUP parameter								
values								
Comments	ISUP/BICC		SU	Γ	SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	←		←	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
				→	ACK			
		Conversation						
	SUS	+		(INFO(SUS)			
				→	200 OK INFO			
	RES	+		(INFO(RES)			
				→	200 OK INFO			
		(Convers	ation				
	REL	→		→	BYE(REL)			
	RLC	+		(200 OK BYE(RLC)			

5.2.2.12 Receipt of a Blocking message

TP312001	SIP reference: RFC 3261	[4]		ISUP reference:
			Q.191	2.5 [1], clause A.1.1.3.1
TSS reference	ISUP-SIP/ISUP Messages for spe	ecial conside	ration / Recei	pt of a Blocking message
SIP selection				
criteria				
ISUP selection				
criteria				
Test purpose	Ensure that the blocking/unblocking			ectly initiated. Ensure the BLO
	messages are not encapsulated v	vithin SIP me	essages	
SIP parameter				
values				
ISUP parameter				
values				
Comments	ISUP/BICC	SU	JT	SIP-I
	BLO =	•		
	BLA	-		
	UBL =	>		
	UBA •	-		

TP312002	SIP reference: RFC 326	1 [4]	ISUP reference: Q.1912.5 [1], clause A.1.1.3.1						
TSS reference	ISUP-SIP/ISUP Messages for sp	ecial conside	ration / Rece	ipt of a Blocking message					
SIP selection				·					
criteria									
ISUP selection									
criteria									
Test purpose	can be correctly initiated.	Ensure that the blocking from both ends; removal of blocking from one end can be correctly initiated. Ensure the BLO messages are not encapsulated within SIP messages.							
SIP parameter									
values									
ISUP parameter									
values									
Comments	ISUP/BICC	Sl	JT	SIP-I					
	BLO	→							
	BLA	←							
	BLO	BLO (+							
	BLA →								
	UBL	→							
	UBA	(

TP312003	SIP reference: RFC 3261 [4]		ISUP reference:							
	Q.1912.5 [1], clause A.1.1.									
TSS reference	ISUP-SIP/ISUP Messages for s	ISUP-SIP/ISUP Messages for special consideration / Receipt of a Blocking message								
SIP selection criteria										
ISUP selection criteria										
Test purpose	and on a Circuit group unblocki	CGB and CGU sent Ensure that the SUT is able to respond on a Circuit group blocking message with a CGBA and on a Circuit group unblocking message (both maintenance oriented) with a CGUA. Ensure the CGB/CGU messages are not encapsulated within SIP messages.								
SIP parameter values										
ISUP parameter values										
Comments	ISUP		SU	Т	SIP-I					
	CGB → CGBA ←									
	CGU	→								
	CGUA	+								

TP312004	SIP reference: RFC 3261 [4]	Q.	ISUP reference: Q.1912.5 [1], clause A.1.1.3.1				
TSS reference	of a Blocking message						
SIP selection criteria			•				
ISUP selection criteria							
Test purpose	Ensure that the SUT on receipt of a CGB, we messages, discards the ISUP information.	vhich is recei	ved en	ncapsulated 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 Message	es for special co	nsidera	ation / Receipt	t of a Blocking message	
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Ensure that a received IA	M will unblock a	a remo	tely blocked c	ircuit.	
SIP parameter						
values						
ISUP parameter						
values						
Comments	ISUP		SUT	Г	SIP-I	
	BLO	→				
	BLA	+				
	IAM	→		→	INVITE(IAM)	
	ACM	+		+	180 Ringing(ACM)	
	ANM	+		+	200 OK INVITE(ANM)	
				→	ACK	
	REL	→		→	BYE(REL)	
	RLC	+		←	200 OK BYE(RLC)	

5.2.2.13 Receipt of a user part test message

TP313001	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 Messages for special consideration / Receipt of a user part test message								
SIP selection									
criteria									
ISUP selection	PICS 4/22								
criteria									
Test purpose	Ensure that on receipt of a user part test mes	sage 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									
ISUP parameter									
values									
Comments	ISUP SU	JT SIP-I							
	UPT →								
	UPA ←								

TP313002	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 consideration / Receipt of a user part test message								
SIP selection criteria									
ISUP selection criteria	PICS 4/22	PICS 4/22							
Test purpose	Ensure that the SUT is able	to send a	user part test r	nessage.					
SIP parameter									
values ISUP parameter									
values									
Comments	ISUP SUT SIP-I								
	UPT	+							
	UPA	→							

TP313003	SIP reference: RFC 32	61 [4]	IS	UP reference:				
			Q.1912.5	[1], clause A.1.1.3.1				
TSS reference	ISUP-SIP/ISUP Messages for s	pecial conside	ration / 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	←						
	UPA	→						

5.2.2.14 Segmentation

TP314001	SIP reference: RFC	SIP reference: RFC 3261 [4]			ISUP reference:			
		Q.191						
TSS reference	ISUP-SIP/ISUP Messages f	or spec	cial conside	ration / Re	ceipt	t of a user part test message		
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that a call can be su	ccessf	ully complet	ted if segm	enta	tion applies in forward		
	direction.			•		• •		
SIP parameter	INVITE - encapsulated IAM:	Forwa	rd call indic	ator abser	nt or	set to "no additional		
values	information will be sent"							
	No action takes place on the	e SIP s	ide					
ISUP parameter	IAM: optional forward call in	dicator	: additional	informatio	n wil	I be sent in a segmentation		
values	message							
	SGM: optional parameters							
Comments	ISUP		SU	Г		SIP-I		
	IAM	→						
	SGM	→			→	INVITE(IAM)		
	ACM ← 180 Ringing(ACM) ANM ← 200 OK INVITE(AN)							
					→	ACK		
	Conversation							
	REL	→			→	BYE(REL)		
	RLC	+			-	200 OK BYE		

5.3 Test purposes for the Supplementary Services

5.3.1 Calling Line Identification Presentation (CLIP)

TP401001	SIP reference: RFC 3	261 [4]	c		SUP reference: 2.5 [1], clause 7.1.3		
TSS reference	ISUP-SIP-ISUP/SS/CLIP		<u> </u>					
SIP selection criteria								
ISUP selection criteria								
Test purpose	Calling Party number network provided, transferred in O-MGCF Ensure that the SUT can successfully transmit a call having a calling party number with the screening indicator set to "network provided" and the presentation restricted indicator set to "presentation allowed".							
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 presentation restricted indicator = presentation allowed, '00'B							
Comments	ISUP IAM	>	SUT		→	SIP-I INVITE(IAM)		
	,	(Conversa	tion	(180 Ringing(ACM) 200 OK INVITE(ANM)		
		→	00.110130		→	BYE(REL) 200 OK BYE(RLC)		

TP401002	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 netwo	rk prov	ided, Callin	g Subaddre	ess ti	ransferred in O-MGCF
	Ensure that the SUT can suc					
	the screening indicator set to			d" and an a	ссе	ss transport parameter
	containing the calling sub-a	iddres	s.			
SIP parameter						
values						
ISUP parameter	IAM:					
values	Calling party number para	motor				
	Address signals = PIXIT1	meter				
	Numbering plan indicator = '	001'B				
	Nature of address indicator :		0011'B			
	Screening indicator = '11'B	= 0000	ИПБ			
	presentation restricted indica	ator – r	recentation	n' bewelle	nη'R	
	Access transport parameter					tion
Comments	ISUP	inolaal	SU7		iiiia	SIP-I
Comments	IAM	→	501		>	INVITE(IAM)
	ACM	′			<u>,</u> +	180 Ringing(ACM)
	ANM	-				<u> </u>
	AINIVI		Convers			200 OK INVITE(ANM)
	DEL		Conversa		,	DVE(DEL)
	REL	<u>→</u>				BYE(REL)
	RLC	←		•	-	200 OK BYE(RLC)

TP401003	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 user provided transferred in O-MGCF Ensure that the SUT can successfully transmit a call having the calling party number with the screening indicator set to "user provided, verified and passed" and the presentation restricted indicator set to "presentation allowed".							
SIP parameter values								
ISUP parameter values	IAM; Calling party number para Address signals = PIXIT1 Numbering plan indicator = Nature of address indicator Screening indicator = '01'B presentation restricted indicator	'001'B = '0000	0011'B	n allowed	i, '00'E	3		
Comments	ISUP		SUT	-		SIP-I		
	IAM	→			→	INVITE(IAM)		
	ACM	+			+	180 Ringing(ACM)		
	ANM	+			+	200 OK INVITE(ANM)		
			Conversa	ation				
	REL	→			→	BYE(REL)		
	RLC	+			←	200 OK BYE(RLC)		

TP401004	SIP reference: RFC 32	261 [4]		-	SUP reference:			
T00 (LOUID OUD TOUR TOUR			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 provided and calling subaddress transferred in O-MGCF							
	Ensure that the SUT can succe							
	the screening indicator set to "u				sed" and an access			
	transport parameter containing	g the ca	lling sub-addi	ess.				
SIP parameter								
values								
ISUP parameter	IAM;							
values	Calling party number parame	eter						
	Address signals = PIXIT1							
	Numbering plan indicator = '00							
	Nature of address indicator = '0	0000011	'B					
	Screening indicator = '01'B							
	Presentation restricted indicate							
	Access transport parameter inc	cluding t		informa				
Comments	ISUP		SUT		SIP-I			
	IAM =	•		→	INVITE(IAM)			
	ACM	<u></u>		←	180 Ringing(ACM)			
	ANM	-		+	200 OK INVITE(ANM)			
		C	Conversation					
	REL =	>		→	BYE(REL)			
	RLC			+	200 OK BYE(RLC)			

TP401005	SIP reference: RFC 32	261 [4]		SUP reference:						
			Q.19 ²	12.5 [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 succe number with the screening ind containing the additional calling provided, not verified" and the allowed".	licator set to "no g party number	etwork provided with the screer	I" and a generic number ning indicator set to "user						
SIP parameter										
values										
ISUP parameter values	IAM;									
	Calling party number parame	eter								
	Address signals = PIXIT1									
	Numbering plan indicator = '00	1'B								
	Nature of address indicator = '0									
	Screening indicator = '11'B									
	Presentation restricted indicate	or = presentatio	n allowed, '00'E	3						
	Generic number parameter									
	Address signals = PIXIT2									
	Numbering plan indicator = '00									
	Nature of address indicator = '0	0000011'B								
	Screening indicator = '00'B									
	Presentation restricted indicate									
Comments	ISUP	SU	Т	SIP-I						
		→	→	INVITE(IAM)						
		-	+	180 Ringing(ACM)						
	ANM	-	+	200 OK INVITE(ANM)						
		Convers	ation							
	REL -	→	→	BYE(REL)						
	RLC	-	+	200 OK BYE(RLC)						

TP401006	SIP reference: RFC 3	261 [[4]	_	SUP reference: 2.5 [1], clause 7.1.3					
TSS reference	ISUP-SIP-ISUP/SS/CLIP			Q.10	2.0 [1], 0.0000 111.0					
SIP selection	1001 011 1001 700/0211									
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 se additional calling party numbe	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 param	eter								
	Address signals = PIXIT1 Numbering plan indicator = '00 Nature of address indicator = Screening indicator = '11'B		0011'B							
	Generic number parameter									
	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		SUT		SIP-I					
	IAM	→		→	INVITE(IAM)					
	ACM	(+	180 Ringing(ACM)					
	ANM	(+	200 OK INVITE(ANM)					
			Conversa	ation						
		→		→	BYE(REL)					
	RLC	(+	200 OK BYE(RLC)					

TP401007	SIP reference: RF	C 3261 [4	ij	ISUP reference: Q.1912.5 [1], clause 7.1.3					
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection criteria									
SUP selection criteria	PICS 6/8	PICS 6/8							
Test purpose	Ensure that the calling par	Calling party number discarded to due bilateral agreement in the I-MGCF. Ensure that the calling party number is discarded in case of bilateral agreements, if the address presentation restricted indicator is set to "presentation allowed" (see note).							
SIP parameter values	·								
SUP parameter values	IAM; No calling party number	paramete	er						
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
		Conversation							
	BYE(REL)	BYE(REL) → REL							
	200 OK BYE(RLC)								
	teral agreement prohibits the address presentation restrict		• • • • • • • • • • • • • • • • • • • •	•	-				

TP401008	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 6/7							
Test purpose	Additional Calling party number is discarded to due bilateral agreements in the I-MGCF Ensure that the additional calling party number in the generic number is discarded in case of bilateral agreements, if the address presentation restricted indicator is set to "presentation allowed".							
SIP parameter values								
ISUP parameter	IAM;							
values	No calling party number p	arame	er					
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
			Conversation	1				
	BYE(REL)	→	<u> </u>	→	REL			
	200 OK BYE(RLC)	←		←	RLC			
	eral agreement prohibits the t ess presentation restricted ind				nber in any case. The test with ed" is a CLIR test.			

TP401009	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	PICS 6/6								
Test purpose	Calling party number is omitted if the presentation restriction indicator is set to address not available in the I-MGCF Ensure that the calling party number is omitted, if the address presentation restricted indicator is set to "address not available".								
SIP parameter									
values									
ISUP parameter									
values									
Comments	SIP-I		SUT	-		ISUP			
	INVITE(IAM)	→			→	IAM			
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM)	←			+	ANM			
			Convers	ation					
	BYE(REL)	→			→	REL			
	200 OK BYE(RLC)	←			+	RLC			

TP401010	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										
Test purpose	Calling party number is se	ent as rec	eived		·					
SIP parameter values ISUP parameter values	number in the encapsulate		er in the Sent IAI	w is gener	ated from the calling party					
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	→	001	→	IAM					
	180 Ringing(ACM)	+			ACM					
	200 OK INVITE(ANM)	+		+	ANM					
		 	Conversation							
	BYE(REL)	→		→	REL					
	200 OK BYE(RLC)	+		+	RLC					

TP401011	SIP reference: RF	FC 3261	ISUP reference:				
				Q.19	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	ımber is s	sent as receive	d			
	Ensure that the additional additional calling party nu				M is generated from the		
SIP parameter values							
ISUP parameter							
values							
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		→	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
			Conversatio	n			
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP401012	SIP reference: RF	C 3261	[4]		SUP reference:
				Q.19	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 o	mitted in the	I-MGCF	
SIP parameter	Ensure that if the calling I number in a generic num INVITE: No calling party n	ber will b	e omitted.		additional calling party d IAM, additional calling party
values	number included.				, ,
ISUP parameter	IAM;				
values	No calling party number	parame	ter		
	No generic number para	•			
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Conversa	tion	
	BYE(REL)	→		→	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 convert the calling party number into an international number, setting the nature of address indicator to "international number" and can pass on the address presentation restricted indicator and the screening indicator transparently.									
SIP parameter										
values										
ISUP parameter	IAM;									
values	Calling party number para	meter								
	Address signals = PIXIT1									
	Numbering plan indicator = '									
	Nature of address indicator =	= '0000	100'B							
	Screening indicator = '11'B									
_	Presentation restricted indica	ator =p			'00'B	I				
Comments	SIP-I		SUT			ISUP				
	INVITE(IAM)	→			→	IAM				
	180 Ringing(ACM)	+			+	ACM				
	200 OK INVITE(ANM)	+			+	ANM				
			Convers	ation						
	BYE(REL)	→			→	REL				
	200 OK BYE(RLC)	←			+	RLC				

TP401014	SIP reference: RFC	3261 [4	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 con										
						ndicator is "ISDN Telephony",					
	setting the nature of address										
SIP parameter	address presentation restricted	ea mai	cator and tr	ne screer	iing in	dicator transparently.					
values											
ISUP parameter	IAM										
values	Calling party number paran	neter									
Valuoo	Address signals = PIXIT1										
	Numbering plan indicator = '0	001'B									
	Nature of address indicator =		100'B								
	Screening indicator = '11'B										
	Presentation restricted indica	ator =pr	esentation	allowed,	'00'B						
	Generic number parameter	•									
	Address signals = PIXIT2										
	Numbering plan indicator = '0										
	Nature of address indicator =	= '0000 <i>°</i>	100'B								
	Screening indicator = '00'B				10.015						
0	Presentation restricted indica	ator =pr			.00.R	lioup					
Comments	SIP-I		SUT			ISUP					
	INVITE(IAM)	→			<u>→</u>	IAM					
	180 Ringing(ACM)	(+	ACM					
	200 OK INVITE(ANM)	~	Converse	otion		ANM					
	BVE(DEL)	→	Conversa	au0H	→	REL					
	BYE(REL)	7			7	RLC					
	200 OK BYE(RLC)	~				KLU					

TP401015	SIP reference: RF	C 3261	[4]	1	SUP 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	PICS 1/7 AND NOT PICS 1/9								
Test purpose	Discarding an incomplete	calling p	arty number in	the I-MGCI	=					
	Ensure that the calling par number incomplete indicate				ived with the calling party					
SIP parameter values										
ISUP parameter values	IAM: No calling party number	parame	ter							
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	→		→	IAM					
	180 Ringing(ACM)	+		(ACM					
	200 OK INVITE(ANM)	+		(ANM					
		Conversation								
	BYE(REL)	→		→	REL					
	200 OK BYE(RLC)									
NOTE: This test	case is only applicable with	an ITU i	mplementation	١.						

TP401016	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 criteria	PICS 1/8									
Test purpose	Converting the calling party number to national format, if necessary in the O-MGCF Ensure that the country code in the address signals of the calling party 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 shall be transferred transparently.									
SIP parameter	INVITE: encapsulated IAM									
values	Calling party number Address signals = Planta Numbering plan indice Nature of address incomparts of Screening indicator = Presentation restricts	XIT1 cator = dicator = '11'B	'001'B = '0000011		allowe	d, '00'B				
ISUP parameter	IAM									
values	Calling party number param Address signals = PIXIT1 Numbering plan indicator = ' Nature of address indicator : Screening indicator = '11'B Presentation restricted indic	'001'B = '0000		n allowed	, '00'B					
Comments	SIP-I		SUT	•		ISUP				
	IAM	→			↑	INVITE(IAM)				
	ACM	←			+	180 Ringing(ACM)				
	ANM	+			+	200 OK INVITE(ANM)				
			Convers	ation						
	REL	→			→	BYE(REL)				
	RLC	←			+	200 OK BYE(RLC)				

TP401017	SIP reference: RFC	3261	[4]			SUP reference: 3.5/Q.731 [i.2]				
TSS reference	ISUP-SIP-ISUP/SS/CLIP									
SIP selection	1.55. 50. 155.144.50.									
criteria										
ISUP selection	PICS 1/8									
criteria										
Test purpose	Converting the additional calling party number to national format, if necessary in the O-MGCF									
	Ensure that the country code in the address signals of the generic number 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 transferred transparently.									
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 =					-l looib				
ISUP parameter	Presentation restricte	a inaic	ator = prese	entation all	owe	J, UUB				
values	Calling party number para	meter								
Values	Address signals = PIXIT1	ilicici								
	Numbering plan indicator = '	001'B								
	Nature of address indicator =	= '0000	0011'B							
	Screening indicator = '11'B									
	Presentation restricted indica		presentatior	allowed, '(00'B					
	Generic number paramete	r								
	Address signals = PIXIT2	00415								
	Numbering plan indicator = '		0044ID							
	Nature of address indicator = Screening indicator = '00'B	= 0000	DUTTB							
	Presentation restricted indica	ator =	oresentation	allowed '	nn'R					
Comments	SIP-I	ato: _	SUT		002	ISUP				
	IAM	→			→	INVITE(IAM)				
	ACM	+		١.	(180 Ringing(ACM)				
	ANM	+		٠	(200 OK INVITE(ANM)				
			Conversa	ation		` '				
	REL	→				BYE(REL)				
	RLC	+		•	←	200 OK BYE(RLC)				

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

TP401019	SIP reference: RF	C 3261	[4]		ISUP reference: 3.5/Q.731 [i.2]				
TSS reference	ISUP-SIP-ISUP/SS/CLIP				0.0, 4 0 . []				
SIP selection	1001 011 1001 700/0211								
criteria									
ISUP selection									
criteria									
Test purpose	Handling of address presentation restricted indicator set to "address not available" in the I-MGCF								
	Ensure that the screening presentation restricted ind available"(see note).								
SIP parameter	, , ,								
values									
ISUP parameter	IAM;								
values	Calling party number party	rameter							
	Address signals = PIXIT1								
	Numbering plan indicator =								
	Nature of address indicator	or = '*'B							
	Screening indicator = '11'E	3							
	Presentation restricted ind	licator =a	address not a	available, '10'	В				
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
			Conversa	ation					
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				
NOTE: The codir	ng "address not available" is	a natior	nal option (@	?).	•				

TP401020	SIP reference:	RFC 3261 [4]	-	SUP reference: I2.5 [1], clause 7.1.3				
TSS reference	ISUP-SIP-ISUP/SS/CL	IP							
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	O-MGCF: Calling party	number and	d Additional call	ing party n	umber not received				
SIP parameter values ISUP parameter values	parameter and the Ger Sends an INVITE mess	neric Numbe sage without onymous@a I Identity, Fro	r are not applica the "P-Asserted nonymous.inva om Header: ano	able. d-Identity h lid". No Pr nymous@					
Comments	ISUP		SUT		SIP-I				
	IAM	→		→	INVITE(IAM)				
	ACM	ACM ← 180 Ringing(ACM)							
	ANM ← 200 OK INVITE(ANM)								
			Conversation	1					
	REL	→		→	BYE(REL)				
	RLC	←		+	200 OK BYE(RLC)				

TP401021	SIP reference: RFC	3261	[4]		- 19	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	O-MGCF: Setting of From header									
SIP parameter values ISUP parameter values	parameter is not applicable presentation restriction para Address Indicator is set to N Sends an INVITE message field" where the user portion number and the country cod"+"CC+NDC+SN and no "Pr	e and the meter is loads. Note that without the is serivacy had been so that the interval of the is serivacy had been so that the interval of the is serivacy had been so the interval of the	ne Generic is set to "pre /ALUE. t the "P-Ass addr-spec i t to the cour Header field Privacy hea	Number is esentation a erted-Identi s set to valuitry where to be adder, From	appallow ity houe of the left hear hear hear hear hear hear hear hear	neader field", a "From header of the additional calling party MGCF is located in the format of the contains the value of the				
Comments	ISUP		SUT	-		SIP-I				
	IAM	→		-	>	INVITE(IAM)				
	ACM	+		•	-	180 Ringing(ACM)				
	ANM									
			Conversa	ation		` ,				
	REL	→		-	}	BYE(REL)				
	RLC	+		•	(200 OK BYE(RLC)				

TP401022	SIP reference: RFC	3261 [4]			SUP reference:						
	10115 015 10115 (00 (01 15			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-Asserted header 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 not 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; • a "From header field" where the "addr-spec" is set to 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;										
SIP parameter	 without "Privacy Header INVITE: P-Asserted-Identity 				mber Privacy-id From						
values	header derived from the add				11001, 1 11140y=14, 1 10111						
ISUP parameter values	IAM; Calling party number is				party number is present						
Comments	ISUP		SUT		SIP-I						
	IAM	→		→	INVITE(IAM)						
	ACM	+		+	180 Ringing(ACM)						
	ANM	+		+	200 OK INVITE(ANM)						
		Conversation									
	REL	→		→	BYE(REL)						
	RLC	+		+	200 OK BYE(RLC)						

TP401023	SIP reference: RFC	3261	[4]	Q.	ISUP reference: 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 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										
values	From header derived from th										
ISUP parameter values	IAM; Calling party number a	nd Add	litional callir	ng party num	·						
Comments	ISUP		SU	Γ	SIP-I						
	IAM	→		7							
	ACM	+		•	180 Ringing(ACM)						
	ANM	+		•							
			Convers	ation							
	REL	→		-	BYE(REL)						
	RLC	+		•	 200 OK BYE(RLC) 						

	Values for test purpose TP401021, TP	2401022, 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]	C		SUP reference: 2.5 [1], clause 7.1.3				
TSS reference	ISUP-SIP-ISUP/SS/CLIP									
SIP selection										
criteria										
ISUP selection	PICS 1/7									
criteria										
Test purpose	Calling party derived from the P-Asserted-Identity international number									
	Ensure when no calling party									
	party number in the in the en Privacy value "id" received.	ncapsu	lated IAM is	not identi	cal to	the P-Asserted-Identity, no				
	Send an IAM the calling part	y num	ber is derive	ed from SII	P P-A	Asserted-Identity. The				
	Address Presentation Restrict	cted In	dicator is se	et to Prese	entatio	on allowed.				
SIP parameter	INVITE: P-Asserted identity	user p	ortion is in t	he format	"+"C(C+NDC+SN, Privacy value				
values	"id" is not present									
ISUP parameter	IAM message with the Callir									
values	Address signals = nu				serte	d-Identity				
	Screening indicator =									
	Number Incomplete II									
	Numbering plan indic					Sava allassa d				
	Address Presentation NoAS: "international i			tor = Pres	entat	ion allowed				
Comments	SIP-I		SUT			ISUP				
	INVITE(IAM)	→			→	IAM				
	180 Ringing(ACM)	+			-	ACM				
	200 OK INVITE(ANM)	+			(ANM				
			Convers	ation						
	BYE(REL)	→			→	REL				
	200 OK BYE(RLC)	+			(RLC				

TP401025	SIP reference: RFC	3261 [4	1]	Q.19	ISUP reference: 912.5 [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 u	iser po	rtion is in th	ne format "+"	CC+NDC+SN, Privacy value					
values	"id" is not present	•			•					
ISUP parameter	IAM message with the Callin	g part	y number	parameter co	oded					
values	Address signals = nun Screening indicator =	networ	k provided		ted-Identity					
	Number Incomplete In Numbering plan indica Address Presentation NoAS: "national (signi	ator = I	SDN numb cted Indica		ation allowed					
Comments	SIP-I		SUT	•	ISUP					
	INVITE(IAM)	→		→	IAM					
	180 Ringing(ACM)	←		+	ACM					
	200 OK INVITE(ANM)	←		+	ANM					
	, ,		Conversa	ation						
	BYE(REL)	→		→	REL					
	200 OK BYE(RLC)	(+	RLC					

TP401026	SIP reference: RFC 32	61 [4]	_	SUP reference:						
T00 (101 15 015 101 15 (00 (01 15		Q.19	12.5 [1], clause 7.1.3						
TSS reference	ISUP-SIP-ISUP/SS/CLIP									
SIP selection										
criteria										
ISUP selection	PICS 1/7									
criteria										
Test purpose	Additional calling party number	derived from t	he From heade	r international number						
SIP parameter	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.									
values	INVITE: P-Asserted identity use	er portion is in	the format + C	C+NDC+SN, Privacy value						
	"id" is not present									
ISUP parameter values	IAM message with the Addition									
values	Address signals = numb			ider						
	Screening indicator = Us		ot verilled							
	Number Incomplete Indi									
	Numbering plan indicate Address Presentation R			tion allowed						
	NoAS: "international nur		aloi = Pieseila	lion allowed						
Comments	SIP-I	SU	-	ISUP						
Comments	INVITE(IAM)		<u>'</u>	IAM						
	\ /			11. 11.11						
	180 Ringing(ACM)		-	ACM						
	200 OK INVITE(ANM)		···	ANM						
		Convers								
	BYE(REL)		→	REL						
	200 OK BYE(RLC)	•	+	RLC						

TP401027	SIP reference: RFC	3261	[4]	Q.1		SUP reference: 2.5 [1], clause 7.1.3				
TSS reference	ISUP-SIP-ISUP/SS/CLIP									
SIP selection criteria										
ISUP selection criteria	NOT PICS 1/7									
Test purpose	Additional calling party number derived from the From header national (significant) number									
	Ensure when no additional of additional calling party numbers header field, no Privacy values and an IAM the additional Address Presentation Restr	oer in thue "id" in calling	ne in the en eceived. party numb	capsulated laterived	AM I froi	m From header field. The				
SIP parameter	INVITE: P-Asserted identity	user po	ortion is in t	he format "+	"CC	C+NDC+SN, Privacy value				
values	"id" is not present	•								
ISUP parameter	IAM message with the Addi									
values	Address signals = nu Screening indicator = Number Incomplete Numbering plan indic Address Presentation NoAS: "national (sign	= User Indicato cator = n Restr	provided, no or = PIXIT ISDN numb icted Indica onumber"	ot verified" pering plan utor = Presen	ntatio	on allowed				
Comments	SIP-I		SU	Γ		ISUP				
	INVITE(IAM)	^		→	•	IAM				
	180 Ringing(ACM)	+		+	-	ACM				
	200 OK INVITE(ANM)	+		+	-	ANM				
			Convers	ation						
	BYE(REL)	→		→	•	REL				
	200 OK BYE(RLC)	+		+		RLC				

5.3.2 Calling Line Identification Restriction (CLIR)

TP402001	SIP reference: RFC 3	3261	[4]	C	-	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	Calling party number network Ensure that the SUT can such	•	•			
	the screening indicator set to indicator set to "presentation"	"netw	ork provide			
SIP parameter values						
ISUP parameter	IAM;					
values	Calling party number paran Screening indicator = '11'B Address presentation restricte Generic number parameter Access transport paramete	ed pai	resent		ıddres	s information
Comments	ISUP		SU			SIP-I
	IAM	→			^	INVITE(IAM)
	ACM	←			4	180 Ringing(ACM)
	ANM	←			+	200 OK INVITE(ANM)
			Convers	ation		
	REL	→			→	BYE(REL)
	RLC	←			+	200 OK BYE(RLC)

TP402002	SIP reference: RFC 3	3261 [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 criteria						
Test purpose	Restricted calling party nun Ensure that the SUT can pass screening indicator set to "nei set to "presentation restricted sub-address.	s trans twork ¡	sparently a c provided", th	all havin ne addres	g a c a	alling party number with the esentation restricted indicator
SIP parameter values						
ISUP parameter values	IAM; Calling party number param Screening indicator = '11'B Address presentation restricte Generic number parameter Access transport paramete	ed para	esent		ormati	on
Comments	ISUP	i ii ioid	SUT	11033 11110	minati	SIP-I
	IAM	→			→	INVITE(IAM)
	ACM	′				180 Ringing(ACM)
	ANM	-			+	200 OK INVITE(ANM)
		ı	Conversa	tion		, ,
	REL	→			→	BYE(REL)
	RLC	←			+	200 OK BYE(RLC)

TP402003	SIP reference: RFC	2261	[4]		-	SUP reference:
17402003	SIF reference. KFC	3201	[4]			Q.1912.5 [1],
				0	731	[i.2], clause 4.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/CLIR			<u> </u>	731	[1.2], Clause 4.5.2.1.1
SIP selection	130F-31F-130F/33/CLIK					
criteria						
ISUP selection						
criteria						
Test purpose	Restricted calling party nu	ımber	(user provi	ded, verifi	ed a	and passed)
	Ensure that the SUT can pa	ss tran	sparently a	call having	the	calling party number with the
	screening indicator set to "u					
	presentation restricted indica	ator se	t to "present	ation restr	icted	d".
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')011'B			
	Screening indicator = '01'B					
	Address presentation restric	ted pa	rameter = '0	1'B		
Comments	ISUP		SUT	•		SIP-I
	IAM	→			→	INVITE(IAM)
	ACM	+			(180 Ringing(ACM)
	ANM	+			(200 OK INVITE(ANM)
			Conversa	ation		
	REL	→			→	BYE(REL)
	RLC	+			(200 OK BYE(RLC)

TP402004	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								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	sub-address Ensure that the SUT can pass screening indicator set to "us	Ensure that the SUT can pass transparently 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							
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 = '01'B								
	Address presentation restric								
Comments	Access transport parameter	er inci	laing subaddress SUT	<u>Informat</u>	SIP-I				
Comments	IAM	→	301	→					
		7		+	INVITE(IAM)				
	ACM ANM			-	180 Ringing(ACM) 200 OK INVITE(ANM)				
	AINIVI		Convergation	7	ZUU OK IINVITE(AINIVI)				
	DEL		Conversation		DVE(DEL)				
	REL	→		→	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP402005	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	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 paran	neter										
	Address signals = PIXIT1											
	Numbering plan indicator = '0											
	Nature of address indicator =	'0000	011'B									
	Screening indicator = '11'B											
	Address presentation restrict		ameter = 'C	11'B								
	Generic number parameter											
	Address signals = PIXIT2											
	Numbering plan indicator = '0											
	Nature of address indicator =	0000	011'B									
	Screening indicator = '00'B											
0	Address presentation restrict	ed par				loip i						
Comments	ISUP		SUT			SIP-I						
	IAM	→			→	INVITE(IAM)						
	ACM	(+	180 Ringing(ACM)						
	ANM	+			+	200 OK INVITE(ANM)						
			Convers	ation								
	REL	→			→	BYE(REL)						
	RLC	←			←	200 OK BYE(RLC)						

TP402006	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/CLIR								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	sub-address Ensure that the SUT can pass trans with the screening indicator set to additional calling party number with	stricted calling party number (user provided, not verified) with calling b-address sure that the SUT can pass transparently a call having a default calling party number in the screening indicator set to "network provided", a generic number containing the ditional calling party number with the screening indicator set to "user provided, not ified", both having the address presentation restricted indicator set to "presentation"							
SIP parameter	restricted and an access transpor	t parameter	containing the calling sub-address.						
values									
ISUP parameter	IAM;								
values	Calling party number parameter								
Comments	Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000 Screening indicator = '11'B Address presentation restricted par Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000 Screening indicator = '00'B Address presentation restricted par Access transport parameter inclu	ameter = '01 011'B ameter = '01	'B ress information						
Comments		SUI	SIP-I						
	IAM →		→ INVITE(IAM)						
	ACM ←		€ 180 Ringing(ACM)						
	ANM ←		€ 200 OK INVITE(ANM)						
	<u></u>	Conversat							
	REL →		→ BYE(REL)						
	RLC +		← 200 OK BYE(RLC)						

TP402007	SIP reference: RF	C 3261	[4]	_	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	PICS 6/4							
Test purpose	Discarding the calling pa Ensure that the calling pa address presentation resti	arty num	ber is discarded i	n case of	f bilateral agreements, if the			
SIP parameter values			•					
ISUP parameter	IAM;							
values	No Calling party number	r parame	eter					
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
		Conversation						
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP402008	SIP reference: RF	C 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/CLIR					
SIP selection						
criteria						
ISUP selection	PICS 6/4 AND PICS 6/5					
criteria						
Test purpose	Discarding the additiona					
						number is discarded in case
	of bilateral agreements, if	the addr	ess present	ation rest	tricted	indicator is set to
	"presentation restricted".					
SIP parameter						
values						
ISUP parameter	IAM;					
values	No Calling party number		eter			
	No Generic number para	ameter				
Comments	SIP-I		SU	Γ		ISUP
	INVITE(IAM)	→			→	IAM
	180 Ringing(ACM)	+			+	ACM
	200 OK INVITE(ANM)	+			+	ANM
			Convers	ation		
	BYE(REL)	→			→	REL
	200 OK BYE(RLC)	+			+	RLC

TP402009	SIP reference: RI	FC 3261	[4]	ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 4.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/CLIR		<u> </u>					
SIP selection criteria								
ISUP selection criteria								
Test purpose	I-MGCF: Calling party numbers that the calling party in the ISUP IAM.				t in the IAM sunchanged sent			
SIP parameter values								
ISUP parameter values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
			Conversati	on				
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		-	RLC			

TP402010	SIP reference: RF	C 3261	[4]	•	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	I-MGCF: Additional calling party number received in the INVITE is sent in the IAM Ensure that the additional 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)	→		→	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
			Conversat	ion					
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP402011	SIP reference: RFC 32	261 [4]	ı	-	SUP reference: I2.5 [1], clause 7.1.3		
TSS reference	ISUP-SIP-ISUP/SS/CLIR			Q.13	12.5 [1], clause 7.1.5		
SIP selection	ISST SIT ISST YES/SERV						
criteria							
ISUP selection							
criteria							
Test purpose	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 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 additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN; • 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; and • with "Privacy Header field" set to "id".						
SIP parameter values	INVITE: P-Asserted-Identity, F	rom he	eader field,	Privacy "id"			
ISUP parameter values	IAM: Calling party number. No	additio	onal calling	party numbe	r		
Comments	ISUP		SUT		SIP-I		
	IAM -	→		→	INVITE(IAM)		
	ACM	(+	180 Ringing(ACM)		
	ANM	f-		+	200 OK INVITE(ANM)		
			Conversati	on			
	REL	→		→	BYE(REL)		
	RLC	f-	•	+	200 OK BYE(RLC)		

TP402012	SIP reference: RFC	3261	[4]	-	SUP reference:			
T00 (IOUE OID IOUE/OO/OUE			Q.19°	12.5 [1], clause 7.1.3			
TSS reference	ISUP-SIP-ISUP/SS/CLIR							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	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. 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 • with "Privacy Header field" is set to "id".							
SIP parameter values	INVITE: P-Asserted-Identity,	From	header field	l, Privacy "id"				
ISUP parameter values	IAM: Calling party number. a	dditior	nal calling pa	arty number				
Comments	ISUP		SUT	-	SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	-		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
			Conversa	ation				
	REL	→		→	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)	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:				
				Q.191	12.5 [1], clause 7.1.3				
TSS reference	ISUP-SIP-ISUP/SS/CLIR								
SIP selection criteria									
ISUP selection	PICS 1/7	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				ed-Identity				
	Screening indicator =			l					
	Number Incomplete I								
	Numbering plan indic								
	Address Presentation			itor = Presenta	tion restricted				
0	NoAS: "international	numbe			loup				
Comments	SIP-I	_	SU		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		-	ACM				
	200 OK INVITE(ANM)	+		←	ANM				
			Convers						
	BYE(REL)	→		→	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	JP-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 portion is in the format "+"CC+NDC+SN, Privacy value									
values	"id" is present	•				•				
ISUP parameter	IAM message with the Callin	ng par	ty number	paramet	er cod	ded				
values	Address signals = nu Screening indicator = Number Incomplete I Numbering plan indic Address Presentation NoAS: "national (sign	netwondicate ator = Restr	ork provided or = PIXIT ISDN numb icted Indicat	ering pla	n	tion restricted				
Comments	SIP-I		SUT	•		ISUP				
	INVITE(IAM)	→			→	IAM				
	180 Ringing(ACM)	+			+	ACM				
	200 OK INVITE(ANM)	+			+	ANM				
			Conversa	ation						
	BYE(REL)	→			→	REL				
	200 OK BYE(RLC)	+			+	RLC				

TP402015	SIP reference: RFC	3261	[4]		-	SUP reference:			
TSS reference	Q.1912.5 [1], clause 7.1.3								
SIP selection	1301 -311 -1301 /33/02110	1001 -011 -1001 /00/0L111							
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								
values	Address signals = nu					der			
	Screening indicator =			ot verified	"				
	Number Incomplete I								
	Numbering plan indic								
	Address Presentation			itor = Pres	sentat	tion restricted			
	NoAS: "international	numbe	-			T			
Comments	SIP-I		SU	Γ		ISUP			
	INVITE(IAM)	→			→	IAM			
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
			Convers	ation					
	BYE(REL)	→			→	REL			
	200 OK BYE(RLC)	+			+	RLC			

TP402016	SIP reference: RFC	3261	[4]	Q.1		reference: 1], clause 7.1.3			
TSS reference	ISUP-SIP-ISUP/SS/CLIR	SUP-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 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					ter coded			
values	Address signals = nu				eader				
	Screening indicator =			ot verified"					
	Number Incomplete I								
	Numbering plan indic								
	Address Presentation			tor = Presen	tation re	estricted			
	NoAS: "national (sign	nificant							
Comments	SIP-I		SUT		ISUF	<u> </u>			
	INVITE(IAM)	→		→					
	180 Ringing(ACM)	+		+		1			
	200 OK INVITE(ANM)	+		+	- ANM	1			
			Convers	ation					
	BYE(REL)	→		7	REL	·			
	200 OK BYE(RLC)	(+	RLC				

5.3.3 Connected line identification presentation (COLP)

TP403001	SIP reference: RF	C 3261	[4]	Q.		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									
Test purpose	Initiate COLP request Ensure that the exchange can initiate successfully a call requesting the COLP service in the optional forward call indicators.								
SIP parameter values									
ISUP parameter	IAM;								
values	optional forward call ind	icators (Connected li	ne identity	requ	uest indicator = requested			
Comments	SIP-I		SUT			ISUP			
	INVITE(IAM)	→			→	IAM			
	180 Ringing(ACM)	+			←	ACM			
	200 OK INVITE(ANM)	+			←	ANM			
		Conversation							
	BYE(REL)	→			→	REL			
	200 OK BYE(RLC)	+			(RLC			

TP403002	SIP reference: RFC 3261 [4	1]	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	Connected number (user provided, verified and passed) with connected sub-address Ensure that the SUT passes transparently a default connected number with the screening indicator set to "verified and passed" and an access transport parameter								
			and an access transport parameter						
CID noromotor	containing the connected sub-addre	ess.							
SIP parameter values									
ISUP parameter	IAM;								
values	optional forward call indicators								
values	Connected line identity request indicators	cator: reque	sted						
	a)	bator. reque	sieu						
	ANM;								
	Connected number parameter								
		Address presentation restricted parameter = '00'B							
	Nature of address indicator = '00000		_						
	Numbering plan indicator = '001'B								
	Screening indicator = '01'B								
	Address signals = PIXIT								
	and an access transport paramete	r containing	the connected sub-address.						
	b)								
	CON;								
	Connected number parameter								
	Address presentation restricted para		'B						
	Nature of address indicator = '00000)11'B							
	Numbering plan indicator = '001'B								
	Screening indicator = '01'B								
	Address signals = PIXIT								
0	and an access transport paramete								
Comments	SIP-I	SUT	ISUP						
	INVITE(IAM) →		→ IAM						
	CASE A								
	180 Ringing(ACM)		← ACM						
	200 OK INVITE(ANM) ←		← ANM						
	CASE B								
	200 OK INVITE(CON) ←		← CON						
		Conversa							
	BYE(REL) →		→ REL						
	200 OK BYE(RLC)		← RLC						

TP403003	SIP reference: RFC 32	61 [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								
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	· · · · · · · · · · · · · · · · · · ·	ore						
	optional forward call indicators Connected line identity request indicator: requested a) ANM; Connected number parameter Address presentation restricted parameter = '00'B Nature of address indicator = '0000011'B Numbering plan indicator = '01'B Screening indicator = '11'B Address signals = PIXIT Additional connected number present Address presentation restricted parameter = '00'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '00'B Address signals = PIXIT b) CON; Connected number parameter Address presentation restricted parameter = '00'B							
	Nature of address indicator = '0 Numbering plan indicator = '00'							
	Screening indicator = '11'B							
	Address signals = PIXIT Additional connected numbe	r precont						
	Address presentation restricted		00'B					
	Nature of address indicator = 'C	0000011'B						
	Numbering plan indicator = '00'	1'B						
	Screening indicator = '00'B Address signals = PIXIT							
	Addiess signals = FIATI							
Comments	SIP-I	SL	it l	ISUP				
	INVITE(IAM)		→	IAM				
	CASE A							
	180 Ringing(ACM)		+	ACM				
	200 OK INVITE(ANM)	-	-	ANM				
	CASE B	<u>-</u>		CON				
	200 OK INVITE(CON) €	Conver	sation ←	CON				
	BYE(REL)		→	REL				
	200 OK BYE(RLC)		+	RLC				

TP403004	SIP reference: RFC 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	PICS 1/7								
criteria									
Test purpose	Converting the connected number to national format, if necessary								
						nnected number is removed			
	if it is the network's own cour								
	"national (significant) numbe				n rest	ricted indicator and the			
SIP parameter	screening indicator shall be t			arenuy.					
values	200 OK: encapsulated ANM Connected number								
values	Address presentation			ter - '00	'R				
	Nature of address ind								
	Numbering plan indica			_					
	Screening indicator =								
	Address signals = PI								
ISUP parameter	IAM;								
values	optional forward call indica	ators							
	Connected line identity reque		icator: requ	actad					
	a)	sst iiiu	icator. requ	esieu					
	ANM;								
	Connected number parame	eter							
	Address presentation restrict		ameter = 'C	0'B					
	Nature of address indicator =	= '0000	100'B						
	Numbering plan indicator = '0								
	Screening indicator = ISUP_								
	Address signals = CC+PIXIT	-							
	b)								
	CON;	.4							
	Connected number parameter								
	Address presentation restricted parameter = '00'B								
		Nature of address indicator = '000100'B							
	Numbering plan indicator = '001'B Screening indicator = ISUP_SI								
	Address signals = CC+PIXIT								
	Generic number parameter		resent						
Comments	SIP-I		SUT	•		ISUP			
	INVITE(IAM)	→			→	IAM			
	CASE A								
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
	CASE B								
	200 OK INVITE(CON)	+			+	CON			
			Convers	ation					
	BYE(REL)	→			→	REL			
	200 OK BYE(RLC)	+			+	RLC			

TP403005	SIP reference: RFC 326	1 [4]		SUP 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	PICS 1/7						
criteria	FIGS III						
Test purpose	Converting the additional connected number to national format, if necessary Ensure that the country code in the address signals of the generic number coded as an "additional connected number", if the numbering plan indicator is "ISDN Telephony" is removed if it is the network's owner, and the address addres						
	set to "national (significant) number", the address presentation restricted indicator a						
SIP parameter	screening indicator shall be transferred transparently. 200 OK: encapsulated ANM or CON						
values	additional connected number						
13.000	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 values	IAM;						
values	optional forward call indicators						
	Connected line identity request in	ndicator: requ	iested				
	a)						
	ANM;	nracent					
Connected number parameter present additional connected number							
	Address presentation restricted parameter = '00'B Nature of address indicator = '0000100'B Numbering plan indicator = '001'B Screening indicator = '01'B						
	b)						
	CON;						
	Connected number parameter present						
	additional connected number						
	Address presentation restricted parameter = '00'B						
	Nature of address indicator = '0000100'B						
	Numbering plan indicator = '001'B Screening indicator = '01'B						
	Address signals = CC+PIXIT						
Comments	SIP-I	SU [*]	т	ISUP			
	INVITE(IAM) →		→	IAM			
	CASE A	1	1	-			
	180 Ringing(ACM) ←		+	ACM			
	200 OK INVITE(ANM) ←		+	ANM			
	CASE B		,				
	200 OK INVITE(CON) ←		· ·	CON			
	D)(5(D51)	Convers		251			
	BYE(REL) →		→	REL			
	200 OK BYE(RLC) ←		←	RLC			

TP403006	SIP reference: RFC 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	PICS 1/8 AND PICS 7/5					
Test purpose	Adding a prefix to an international connected number Ensure that a prefix is added to the connected number and the nature of address indicator is set to "unknown" (see note).					
SIP parameter values	200 OK INVITE with encapsulated ANM or CON 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	Sl	JT	ISUP		
	IAM CASE A	→	→	INVITE(IAM)		
	ACM	(+	180 Ringing(ACM)		
	ANM	(+	200 OK INVITE(ANM)		
	CASE B					
	CON	(+	200 OK INVITE(CON)		
		Conver	sation	, , ,		
	REL	→	→	BYE(REL)		
	RLC	(-	200 OK BYE(RLC)		
NOTE: The codi	ng "unknown" is a national optio	n (@).				

TP403007	SIP reference: RFC 3261 [4	ij	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	PICS 1/8 AND PICS 7/3		
Test purpose		is discard	of bilateral agreements ded in case of bilateral agreements, if the t to "presentation allowed" (see note).
SIP parameter values	200 OK INVITE with encapsulated A Connected number parame Address presentation restrict Nature of address indicator = Numbering plan indicator = '0	eter ed parame : '0000011	eter = '00'B
	Screening indicator = '11'B Address signals = PIXIT	ЮТБ	
ISUP parameter	IAM		
values	optional forward call indicators Connected line identity request indical a) ANM No Connected number parameter b) CON; No Connected number parameter	·	
Comments	ISUP	SU	T SIP-I
	IAM →		→ INVITE(IAM)
	CASE A		
	ACM ←		← 180 Ringing(ACM)
	ANM ←		← 200 OK INVITE(ANM)
	CASE B		
	CON ←		← 200 OK INVITE(CON)
		Convers	ation
	REL →		→ BYE(REL)
	RLC ←		← 200 OK BYE(RLC)
	teral agreement prohibits the transferress presentation restricted indicator se		onnected number in any case. The test with

TP403008	SIP reference: RFC 32	261 [4]		ISUP reference: Q.1912.5 [1],				
			0.731	[i.2], clause 5.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/COLP							
SIP selection	1881 -811 -1881 78878821							
criteria								
ISUP selection	PICS 1/8 AND PICS 7/4							
criteria								
Test purpose	Discarding the additional co							
	Ensure that the additional con							
	of bilateral agreements, if the		tation restricted	I indicator is set to				
OID	"presentation allowed" (see no		211					
SIP parameter	200 OK INVITE with encapsula							
values	Additional Connected Address presentation re							
	Nature of address indic							
	Numbering plan indicat		1 0					
	Screening indicator = '0							
	Address signals = PIXI							
ISUP parameter	IAM;							
values	optional forward call indicators							
	Connected line identity reques		iested					
	a)	t indicator. requ	acsica					
	ANM;							
	ANWI,							
	No Connected number parame	stor						
	No Additional connected number							
	b)	ei pieseiii						
	CON;							
	No Connected number parame	eter						
	No Additional connected numb							
Comments	ISUP	SU	T	SIP-I				
		→	→	INVITE(IAM)				
	CASE A							
		-	+	180 Ringing(ACM)				
	7 (1 417)	-	+	200 OK INVITE(ANM)				
	CASE B	·	1 .	TOOO OK INDUITE (OON)				
	CON	Convers	←	200 OK INVITE(CON)				
	DEI	Convers		DVE(DEL)				
		>	→	BYE(REL) 200 OK BYE(RLC)				
NOTE: This bila	teral agreement prohibits the trai	-	_					
	in any case.	isiciiai ui tile a	dulional collie	cied namber in the generic				
Hamber	iii diiy odoo.							

TP403009	SIP reference: RFC	3261 I	[4]		ı	SUP reference:
						Q.1912.5 [1],
					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
OID	the address presentation res				reenir	ng indicator transparently.
SIP parameter	200 OK INVITE with encapsu			N		
values	Connected number				_	
	Address presentation				В	
	Nature of address ind			В		
	Numbering plan indica		001 B			
	Screening indicator = Address signals = CC		т			
ISUP parameter	IAM;	TI IAI	1			
values	optional forward call indica	ators				
Values	Connected line identity reque		icator: reque	ested		
	a))	iodior. roqui	otou		
	ANM					
	Connected number parame	eter				
	Address presentation restrict		rameter = '0	0'B		
	Nature of address indicator =	= '0000)100'B			
	Numbering plan indicator = '(001'B				
	Screening indicator = '11'B					
	Address signals = PIXIT					
	Presentation restricted indica					
	additional connected numb	per pre	esent			
	b)					
	CON;	4				
	Connected number parame			OID.		
	Address presentation restrict			0.B		
	Nature of address indicator = 'Numbering plan indicator = 'C		7100 B			
	Screening indicator = '11'B	логь				
	Address signals = PIXIT					
	Presentation restricted indica	ator – '	nn'B			
	additional connected number					
Comments	SIP-I	P'\	SUT	'		ISUP
	INVITE(IAM)	→			→	IAM
	CASE A		I		1	
	180 Ringing(ACM)	+			+	ACM
	200 OK INVITE(ANM)	+			+	ANM
	CASE B		1			Jan. 2007.
	200 OK INVITE(CON)	+			+	CON
			Conversa	ation		
	BYE(REL)	→		-	→	REL
	200 OK BYE(RLC)	+			+	RLC
L	· · - · - · - · /		L			

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 C Ensure that the call can be		sfully set up	if the SUT red	ceives an unsolicited COL.		
SIP parameter	200 OK INVITE with enca			V			
values	Connected number						
	Address presentati						
	Nature of address	indicator	= '0000011'	3			
	Numbering plan inc		'001'B				
	Screening indicator						
	Address signals = 1	PIXIT					
ISUP parameter	IAM;	_					
values	optional forward call ind						
	Connected line identity request indicator: not requested						
	a)						
	ANM;						
	Connected number para						
	Address presentation rest)'B			
	Nature of address indicate		J011'B				
	Numbering plan indicator						
	Screening indicator = '11'E	3					
	Address signals = PIXIT						
	additional connected nu	mber pr	esent				
	b) CON :						
	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	mhor or	ocont				
Comments	SIP-I	liibei pi	SUT		IISUP		
Comments	INVITE(IAM)	→	301	→	IAM		
	CASE A	7	L		IIVINI		
				L	IACNA		
	180 Ringing(ACM)	+	 	-	ACM		
	200 OK INVITE(ANM)			+	ANM		
	CASE B			1 #	loon		
	200 OK INVITE(CON)	+		· · ·	CON		
	D)(E(DEL)		Conversa		DEL		
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	←		←	RLC		

TP403012	SIP reference: RFC 3	3261 [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				
ISUP selection	PICS 1/7			
criteria	5 11 1 1 11 11 11 11 11 11 11 11 11 11 1	1 4 11	000 OK INN/ITE :	
Test purpose			a 200 OK INVITE is sent on the ISUP side	3
	connected sub address is inc		nchanged. The ATP contained the	
CID parameter	200 OK INVITE: encapsulate		naludad	
SIP parameter values	200 OK INVITE. encapsulate	d ANIVI OF CON I	nciuded	
ISUP parameter	a)			
values	ANM;			
values	Connected number parame	ter		
	Address presentation restricted		00'B	
	Nature of address indicator =			
	Numbering plan indicator = '0			
	Screening indicator = '11'B			
	Address signals = PIXIT			
		ameter containir	ng the connected sub-address.	
	b)			
	CON;			
	Connected number parame			
	Address presentation restricted		00'B	
	Nature of address indicator =			
	Numbering plan indicator = '0	01'B		
	Screening indicator = '11'B			
	Address signals = PIXIT			
			ng the connected sub-address.	
Comments	ISUP	SU		
	IAM	→	→ INVITE(IAM)	
	CASE A	- 1		
	ACM	(← 180 Ringing(ACM)	
	ANM	←	← 200 OK INVITE(ANM)	
	CASE B	- -		
	CON	(€ 200 OK INVITE(CON)	
		Convers		
	REL)	→ BYE(REL)	
	RLC	←	← 200 OK BYE(RLC)	

TP403013	SIP reference: RFC 3	261 [4]		ISUP reference: Q.1912.5 [1], [i.2], clause 5.5.2.1.1		
TSS reference	ISUP-SIP-ISUP/SS/CLIR		•				
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	O-MGCF: connected number	and a	ndditional co	nnected num	ber transferred transparently		
	Ensure that an ANM or CON e without changing. The connec connected sub address is incli	ted n	umber is un				
SIP parameter values	200 OK INVITE: encapsulated			cluded			
ISUP parameter	a)						
values	ANM;						
	Connected number paramet						
	Address presentation restricte)'B			
	Nature of address indicator =	'0000	011'B				
	Numbering plan indicator = '00	01'B					
	Screening indicator = '11'B						
	Address signals = PIXIT						
	Additional connected numb						
	Address presentation restricte)'B			
	Nature of address indicator =		011'B				
	Numbering plan indicator = '00	01'B					
	Screening indicator = '00'B						
	Address signals = PIXIT						
	and an access transport para	amete	er containing	the connecte	ed sub-address.		
	b)						
	CON;						
	Connected number paramet						
	Address presentation restricte)'B			
	Nature of address indicator =		011'B				
	Numbering plan indicator = '00	01'B					
	Screening indicator = '11'B						
	Address signals = PIXIT						
	Additional connected number present						
	Address presentation restricted parameter = '00'B						
	Nature of address indicator = '0000011'B						
	Numbering plan indicator = '00	บา.В					
	Screening indicator = '00'B						
	Address signals = PIXIT			. 41	- d b d - d		
0	and an access transport para	amete	•	the connecte			
Comments	ISUP		SUT		SIP-I		
		→		→	INVITE(IAM)		
	CASE A			1 -	1400 D: : (4.01.0		
		(-	180 Ringing(ACM)		
		←		←	200 OK INVITE(ANM)		
	CASE B						
	CON	←		←	200 OK INVITE(CON)		
			Conversa	tion			
	REL	→		→	BYE(REL)		
1	RLC	←		+	200 OK BYE(RLC)		

5.3.4 Connected Line Identification Restriction (COLR)

TP404001	SIP reference: RFC	3261	[4]		ISUP reference:				
				0.73	Q.1912.5 [1],				
TSS reference				Q.73	1 [i.2], clause 6.5.2.1.1				
SIP selection	ISUP-SIP-ISUP/SS/COLR	ISUP-SIP-ISUP/SS/COLK							
criteria ISUP selection	-								
criteria									
Test purpose	Passing on information rela	ating	to COLP						
rest purpose	Ensure that the SUT shall pa			all information	related to the COLR				
	supplementary service in the	addre	ess present	ation restricte	d indicator of the connected				
	number.	addic	oo prooona		a maidator or the commedica				
SIP parameter									
values									
ISUP parameter	IAM;								
values	optional forward call indica	ators							
	Connected line identity reque	est ind	icator: requ	ested					
	a)								
	ANM;								
	Connected number parame								
	Address presentation restrict)1' B					
	Nature of address indicator =		0011'B						
	Numbering plan indicator = 'C	J01'B							
	Screening indicator = '01'B Address signals = PIXIT								
	b)								
	CON;								
	Connected number parame	eter							
	Address presentation restrict		rameter = '()1' B					
	Nature of address indicator =			,, <u>J</u>					
	Numbering plan indicator = '0								
	Screening indicator = '01'B								
	Address signals = PIXIT								
Comments	SIP-I		SU	Γ	ISUP				
	INVITE(IAM)	→		→	IAM				
	CASE A								
	180 Ringing(ACM)	+		←					
	200 OK INVITE(ANM) ← ANM								
	CASE B								
	200 OK INVITE(CON)	+		+	CON				
			Convers	ation					
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP404002	SIP reference: RFC 3261	[4]		SUP reference: Q.1912.5 [1], [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 relating	to COLR					
	Ensure that the SUT shall pass tr						
	supplementary service in the add						
	number and the additional conne	ct number in	the generic n	umber.			
SIP parameter							
values							
ISUP parameter	IAM;						
values	optional forward call indicators						
	Connected line identity request in	aicator: requ	ested				
	a)						
	ANM;						
	Connected number parameter Address presentation restricted p	arametor – 'C	11' B				
	Nature of address indicator = '000		ט וי				
	Numbering plan indicator = '001'E						
	Screening indicator = '11'B	•					
	Address signals = PIXIT						
	Additional connected number	oresent					
	Address presentation restricted p)1' B				
	Nature of address indicator = '000						
	Numbering plan indicator = '001'E	3					
	Screening indicator = '00'B						
	Address signals = PIXIT						
	b)						
	CON;						
	Connected number parameter						
	Address presentation restricted p		01' B				
	Nature of address indicator = '000						
	Numbering plan indicator = '001'E	5					
	Screening indicator = '11'B Address signals = PIXIT						
	Additional connected number present						
	Address presentation restricted parameter = '01' B						
	Nature of address indicator = '0000011'B						
	Numbering plan indicator = '001'E						
	Screening indicator = '00'B						
	Address signals = PIXIT						
Comments	SIP-I	SUT		ISUP			
	INVITE(IAM) →		→	IAM			
	CASE A						
	180 Ringing(ACM) ←		+	ACM			
	200 OK INVITE(ANM) ←		+	ANM			
	CASE B						
	200 OK INVITE(CON) ←		+	CON			
		Convers	ation				
	BYE(REL) →		→	REL			
	200 OK BYE(RLC) ←		+	RLC			

TP404003	SIP reference: RFC 32	261 [4]		SUP reference: Q.1912.5 [1],		
			Q.731	[i.2], clause 6.5.2.1.1		
TSS reference	ISUP-SIP-ISUP/SS/COLR					
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Restricted connected number (user provided, verified and passed) with connect sub-address					
	Ensure that the SUT can pass indicator set to "user provided, restricted indicator set to "pres Additionally, an access transp	, verified and pa sentation restric	assed" and with ted", if the user	the address presentation provided COL is valid.		
	shall also be provided.					
SIP parameter						
values						
ISUP parameter	IAM;					
values	optional forward call indicate					
	Connected line identity reques	t indicator: requ	ıested			
	a)					
	ANM;					
	Connected number parameter					
	Address presentation restricted		01' B			
	Nature of address indicator = '					
	Numbering plan indicator = '00)1'B				
	Screening indicator = '01'B					
	Address signals = PIXIT			dus a s		
	access transport parameter co	ontaining the co	nnected sub-ad	aress		
	b) CON;					
	,	•				
	Connected number parameter Address presentation restricted		01' B			
	Nature of address indicator = '		UID			
	Numbering plan indicator = '00					
	Screening indicator = '01'B	71 0				
	Address signals = PIXIT					
	access transport parameter co	ntaining the co	nnected sub-ad	dress		
Comments	SIP-I	SU		ISUP		
		→	→	IAM		
	CASE A	<u>- 1</u>				
	1	(+	ACM		
200 OK INVITE(ANM)						
	CASE B	-		h 27.2141		
		(+	CON		
	255 51(1144112(0014)	Convers				
	BYE(REL)	→ Convers	• • • • • • • • • • • • • • • • • • •	REL		
		7	 	RLC		
	200 ON BTE(NLO)	•		INLO		

TP404004	SIP reference: RFC 3261	[4]	ISUP reference:			
			Q.1912.5 [1],			
			Q.731 [i.2], clause 6.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/COLR					
SIP selection						
criteria						
ISUP selection	PICS 7/1					
criteria						
Test purpose	Discarding the connected numb					
			ded in case of bilateral agreements, if the			
OID	address presentation restricted inc		t to "presentation restricted".			
SIP parameter	200 INVITE: encapsulated ANM of		.d. d			
values ISUP parameter	No Connected number para	ameter inclu	ided			
values	IAM; optional forward call indicators					
values	Connected line identity request inc	licator: requ	lested			
	a)	ilcator. requ	iesieu			
	ANM;					
	Connected number parameter					
	Address presentation restricted pa	rameter = '(01'B			
	Nature of address indicator = '000					
	Numbering plan indicator = '001'B					
	Screening indicator = '11'B					
	Address signals = PIXIT					
	b)					
	CON;					
	Connected number parameter		_			
	Address presentation restricted pa		01'B			
	Nature of address indicator = '0000	0011'B				
	Numbering plan indicator = '001'B					
	Screening indicator = '11'B					
Comments	Address signals = PIXIT SIP-I	SU ⁻	T ISUP			
Comments	INVITE(IAM) →	30	→ IAM			
	CASE A	l	IAIVI			
	180 Ringing(ACM) ←		← ACM			
	200 OK INVITE(ANM)		← ANM			
	CASE B					
	200 OK INVITE(CON)		← CON			
	200 01(11(11)2(0014)	Convers				
	BYE(REL) →	55117613	→ REL			
	200 OK BYE(RLC)		← RLC			
	200 OR BIL(NLO)	I	I INLO			

TP404005	SIP reference: RFC 3261	[4]		ISUP reference:					
				Q.1912.5 [1],					
			Q.731	[i.2], clause 6.5.2.1.1					
TSS reference	ISUP-SIP-ISUP/SS/COLR								
SIP selection	PICS 7/2	PICS 7/2							
criteria									
ISUP selection									
criteria									
Test purpose	Discarding the additional conne	cted number	er in the gene	ric number if the					
	presentation is restricted	ad number i	n tha manaria	number is discorded in some					
	Ensure that the additional connect of bilateral agreements, if the addr								
	"presentation restricted".	ess present	alion restricted	indicator is set to					
SIP parameter	200 INVITE: encapsulated ANM or	r CON							
values	No Additional Connected n		neter included						
ISUP parameter	IAM:	arribor parar	notor inoradoa						
values	optional forward call indicators								
	Connected line identity request inc	dicator: requ	ested						
	a)	•							
	ANM;								
	Connected number parameter pr								
	Additional Connected number p								
	Address presentation restricted pa		11'B						
	Nature of address indicator = '0000								
	Numbering plan indicator = '001'B								
	Screening indicator = '11'B								
	Address signals = PIXIT								
	b) CON;								
	Connected number parameter pr	esent							
	Additional Connected number p								
	Address presentation restricted pa		1'B						
	Nature of address indicator = '0000								
	Numbering plan indicator = '001'B								
	Screening indicator = '11'B								
	Address signals = PIXIT								
Comments	SIP-I	SUT		ISUP					
	INVITE(IAM) →		→	IAM					
	CASE A	.							
	180 Ringing(ACM) ←		-	ACM					
	200 OK INVITE(ANM) ←		←	ANM					
	CASE B	1	1 -	1					
	200 OK INVITE(CON) ←		· · ·	CON					
		Convers							
	BYE(REL)			REL					
	200 OK BYE(RLC) ←	ļ	←	RLC					

TP404007	SIP reference: RFC 3261 [4]		ISUP reference:			
		0.7	Q.1912.5 [1], 31 [i.2], clause 6.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/COLR	<u> </u>	o : [], o.aaoo o.o			
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	O-MGCF: Connected number, additional of	connected numb	er and connected subaddress			
	transferred					
	E (1 / ANIM OON)		V/ITE:			
	Ensure that an ANM or CON encapsulated					
	without changing. The connected number connected sub address is included.	is unchanged. I	ne ATP contained the			
SIP parameter	200 OK INVITE: encapsulated ANM or CC	M included				
values	200 OK INVITE. encapsulated AMM of OC	n included				
ISUP parameter	ANM;					
values	Connected number parameter					
	Address presentation restricted parameter	= '01'B				
	Nature of address indicator = '0000011'B					
	Numbering plan indicator = '001'B					
	Screening indicator = '11'B					
	Address signals = PIXIT					
	Additional connected number present	1041D				
	Address presentation restricted parameter Nature of address indicator = '0000011'B	= 016				
	Numbering plan indicator = '001'B					
	Screening indicator = '00'B					
	Address signals = PIXIT					
	and an access transport parameter conta	aining the conne	cted sub-address.			
	b)					
	CON;					
	Connected number parameter					
	Address presentation restricted parameter	' = '01'B				
	Nature of address indicator = '0000011'B					
	Numbering plan indicator = '001'B					
	Screening indicator = '11'B Address signals = PIXIT					
	Additional connected number present					
	Address presentation restricted parameter = '01'B					
	Nature of address indicator = '0000011'B					
	Numbering plan indicator = '001'B					
	Screening indicator = '00'B					
	Address signals = PIXIT					
Commonto	and an access transport parameter conta					
Comments	ISUP IAM →	SUT	SIP-I INVITE(IAM)			
	IAM →] 7	IIIAAIIE(IWIN)			
	ACM ←	•	180 Ringing(ACM)			
	ANM €		0 0\			
	CASE B		1200 OICHAVITE(ANNI)			
	CON ←	•	200 OK INVITE(CON)			
		versation				
	REL →	-	BYE(REL)			
	RLC +	•				

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/	ГР							
SIP selection criteria									
ISUP selection criteria									
Test purpose	requested by the cal	orms the called ling party upor	I party that a so receipt of use	uspend and er initiated S	a resume have been SUS and RES messages.				
SIP parameter values	INFO: Content-Type	INFO: Content-Type: application/ISUP; SUS and RES encapsulated in the MIME body							
ISUP parameter values									
Comments	ISUP		SUT		SIP-I				
	IAM	→		→	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
			Conversation	n					
	SUS	→		→	INFO(SUS)				
				+	200 OK INFO				
	RES	→		→	INFO(RES)				
				+	200 OK INFO				
	REL	→		→	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP405002	SIP reference	ce: RFC 3261 [4]		ISUP 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	requested by the ca	orms the calling lled party upon	party that a s receipt of user	uspend and initiated S	d a resume have been US and RES messages.
SIP parameter	INFO: Content-Type	e: application/IS	UP; SUS and	RES enca	osulated 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)
			Conversation	n	
	SUS	+		+	INFO(SUS)
				→	200 OK INFO
	RES	+		+	INFO(RES)
				→	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	+		/	200 OK BYE(RLC)

TP405003	SIP reference: RF	C 3261	[4]		ISUP reference: Q.1912.5 [1],
				Q.	733 [i.6], clause 4.5.2.1
TSS reference	ISUP-SIP-ISUP/SS/TP				
SIP selection					
criteria					
ISUP selection criteria					
Test purpose					o Resume after Suspend
					on timer expiry) by the SUT if
	timer T2 expires because t				
SIP parameter	INFO: Content-Type: appli				
values	BYE : Content-Type: applic	cation/IS	SUP ; REL e	ncapsulate	d in the MIME body
ISUP parameter					
values					
Comments	ISUP		SUT		SIP-I
	IAM	→			► INVITE(IAM)
	ACM	←		•	180 Ringing(ACM)
	ANM	←			200 OK INVITE(ANM)
			Convers	ation	
	SUS	→		-	► INFO(SUS)
				•	200 OK INFO
	REL	→			▶ BYE(REL)
	RLC	←		•	200 OK BYE(RLC)

TP405004	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.733 [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 Ensure that a suspended ca				user releases the call.			
SIP parameter values	INFO: Content-Type: applic BYE: Content-Type: applic	ation/IS	UP ; SUS er	ncapsulated in	n the MIME body			
ISUP parameter values					,			
Comments	ISUP		SUT		SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
			Conversa	tion				
	SUS → INFO(SUS)							
					200 OK INFO			
	REL	+		+	BYE(REL)			
	RLC	→		→	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/SUE	3							
SIP selection criteria									
ISUP selection criteria									
Test purpose	Ensure that the SUT caparameter in the encaps	n include th	ne called sub-addi	-					
SIP parameter values	INVITE: Content-Type:			osulated	in the MIME body				
ISUP parameter values									
Comments	ISUP		SUT		SIP-I				
	IAM	→		→	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
		Conversation							
	REL	→		→	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	Ensure that the SUT can in	Receiving the called sub-address in the access transport parameter Ensure that the SUT can include the called sub-address in the access transport								
OID	parameter in the ISUP IAM	1.								
SIP parameter values										
ISUP parameter										
values										
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	→		→	IAM					
	180 Ringing(ACM)	+		+	ACM					
	200 OK INVITE(ANM)	+		+	ANM					
			Conversat	ion						
	BYE(REL)	→		→	REL					
	200 OK BYE(RLC)	+		(RLC					

TP406003	SIP reference: RF	C 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 in parameter in the encapsul	nclude th	ne calling sub-a					
SIP parameter values	INVITE: Content-Type: ap	plication	/ISUP ; IAM en	capsulated	in the MIME body			
ISUP parameter values								
Comments	ISUP		SUT		SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM ← 200 OK INVITE(ANM)							
			Conversation	n				
	REL	→		→	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP406004	SIP reference: RI	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	Receiving the calling sub- Ensure that the SUT can parameter in the ISUP IA	include tl		•	•	
SIP parameter values						
ISUP parameter						
values						
Comments	SIP-I		SU	Γ		ISUP
	INVITE(IAM)	→			→	IAM
	180 Ringing(ACM)	+			←	ACM
	200 OK INVITE(ANM)	+			←	ANM
			Convers	ation		
	BYE(REL)	→			→	REL
	200 OK BYE(RLC)	+			←	RLC

5.3.7 Malicious Call Identification (MCID)

TP407001	SIP reference: RFC	SIP reference: RFC 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 can sue encapsulated IDR having than IRS with MCID respons number included. ISUP to see the second se	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.							
SIP parameter	183 Session Progress: Con	tent-Ty	pe: applicat	ion/ISUP; II	DR encapsulated in the MIME				
values	body INFO: Content-Type: applic	ation/IS	SUP; IRS er	ncapsulated	I in the MIME body				
ISUP parameter values				•					
Comments	ISUP		SUT		SIP-I				
	IAM	→		→	INVITE(IAM)				
	IDR	+		+	183 Session Progress(IDR)				
	IRS	→		→	INFO(IRS)				
				+	200 OK INFO				
	ACM	+		+	180 Ringing(ACM)				
	ANM	←		←	200 OK INVITE(ANM)				
			Conversa	tion					
	REL	→		→	BYE(REL)				
	RLC	←		←	200 OK BYE(RLC)				

TP407002	SIP reference: RFC 3261 [4]				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 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: Contented body INFO: Content-Type: application	,,	• •	•	·			
ISUP parameter values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	183 Session Progress(IDR)	+		+	IDR			
	INFO(IRS)	→		→	IRS			
	200 OK INFO	+						
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
			Conversatio					
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	←		←	RLC			

TP407003	SIP reference: RFC 326	61 [4]		Q.73		SUP reference: Q.1912.5 [1], [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 - after ACM								
	Ensure that the SUT will accept								
	been received. The SUT should								
	"MCID request" and pass on an					icator set to "MCID			
OID .	included" and the calling party					1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1			
SIP parameter	INFO: Content-Type: application								
values		INFO: Content-Type: application/ISUP; IRS encapsulated in the MIME body							
ISUP parameter values	IRS containing the calling party	numbe	r param	neter					
Comments	SIP-I		SI	UT		ISUP			
	INVITE(IAM)	→		-	→	IAM			
	CASE A								
	180 Ringing(ACM)	+		•	(ACM			
	183 Session Progress(IDR)	+		•	(IDR			
	INFO(IRS)	→		-	>	IRS			
	200 OK INFO	+							
	200 OK INVITE(ANM)	+		•	(ANM			
	CASE B								
	183 Session Progress(ACM)	+		•	(ACM(early)			
	183 Session Progress(IDR)	+		•	(IDR			
	INFO(IRS)	→		-	>	IRS			
	200 OK INFO	+							
	180 Ringing(CPG)	+		-	–	CPG(alerting)			
	200 OK INVITE(ANM)	+			(ANM			
	, ,		Conve	rsation					
	BYE(REL)	→		-	→	REL			
	200 OK BYE(RLC)	+		•	←	RLC			
NOTE: This situa	tion may occur e.g. if the call has	been f	orward	ed before re	eachi	ing the destination.			

TP407004	SIP reference: RFC	3261 [4	ij	ISUP refe Q.1912 Q.731.7 [i.3], cl	.5 [1],				
TSS reference	ISUP-SIP-ISUP/SS/MCID								
SIP selection criteria									
ISUP selection criteria									
Test purpose	Ensure that the SUT rejects	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	183 Session Progress: Cont				lated in the MIME				
values	body INFO: Content-Type: applica		• •	•					
ISUP parameter values				•					
Comments	ISUP		SUT	SIP-I					
	IAM	→		→ INVITE(IA	AM)				
	IDR	+		← 183 Sess	ion Progress(IDR)				
	IRS	→		→ INFO(IRS	S)				
				← 200 OK II	NFO				
	ACM ← 180 Ringing(ACM)								
	ANM	+		← 200 OK II	NVITE(ANM)				
			Conversa	on					
	REL	→		→ BYE(REL	.)				
	RLC	+		← 200 OK E	BYE(RLC)				

TP407005	SIP reference: RFC 32	261 [4]			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	Ensure that the SUT rejects a	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 values	183 Session Progress: Content body INFO: Content-Type: application	t-Type:	: application/	ISUP; IDR	encapsulated in the MIME					
ISUP parameter values			•		•					
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	→		→	IAM					
	183 Session Progress(IDR)	+		+	IDR					
	INFO(IRS)	→		→	IRS					
	200 OK INFO	+								
	180 Ringing(ACM)	+		+	ACM					
	200 OK INVITE(ANM)	←		←	ANM					
			Conversat	on						
	BYE(REL)	→		→	REL					
	200 OK BYE(RLC)	←		+	RLC					

TP407006	SIP reference: RFC 32	61 [4]		_	SUP reference: Q.1912.5 [1], ' [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	 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": the IDR request is transferred into the national network; the IRS is received from the national network having the calling party number coded as an "international number". Calling party sub-address in ATP. 									
SIP parameter values	183 Session Progress: Content body INFO: Content-Type: application	t-Type:	: application/ISUP	; IDR	encapsulated in the MIME					
ISUP parameter values					•					
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	→		→	IAM					
	183 Session Progress(IDR)	←		+	IDR					
	INFO(IRS)	→		→	IRS					
	200 OK INFO	+								
	180 Ringing(ACM) ← ← ACM									
	200 OK INVITE(ANM)	+		←	ANM					
	DVE(DEL)	\vdash	Conversation	_	DEL					
	BYE(REL) 200 OK BYE(RLC)	→		→	REL RLC					
	1200 ON DIL(INLO)			_	INLO					

TP407007	SIP reference: RFC	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 can su MCID request indicator se indicator set to "MCID included in the set of	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: Con body INFO: Content-Type: applic	,		•	DR encapsulated in the MIME					
ISUP parameter values					•					
Comments	ISUP		SUT		SIP-I					
	IAM	→		→	INVITE(IAM)					
	IDR	+		+	183 Session Progress(IDR)					
	IRS	→		→	INFO(IRS)					
				+	200 OK INFO					
	ACM	+		+	180 Ringing(ACM)					
	ANM	+		+	200 OK INVITE(ANM)					
			Conversa	tion						
	REL	→		→	BYE(REL)					
	RLC	+		+	200 OK BYE(RLC)					

TP407008	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" by send	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: Contented body INFO: Content-Type: application		• •			·				
ISUP parameter values			,	•		,				
Comments	SIP-I		Sl	JT		ISUP				
	INVITE(IAM)	→			→	IAM				
	183 Session Progress(IDR)	+			+	IDR				
	INFO(IRS)	→			→	IRS				
	200 OK INFO	+								
	180 Ringing(ACM)	+			+	ACM				
	200 OK INVITE(ANM)	+			+	ANM				
			Conver	sation						
	BYE(REL)	→			→	REL				
	200 OK BYE(RLC)	+		•	+	RLC				

TP407009	SIP reference: RFC	3261	[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	ISUP-SIP-ISUP/SS/MCID								
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	MCID timer (T39) expiry O									
						s is received within timer T39				
	expiry, after having sent the	IDR w	ith MCID re	quest	indica	tor set to "MCID requested".				
	ISUP to SIP-I interworking.									
SIP parameter	183 Session Progress: Con	tent-Ty	pe: applicat	ion/ISl	JP; IDF	R encapsulated in the MIME				
values	body									
ISUP parameter										
values										
Comments	ISUP		SUT			SIP-I				
	IAM	→			→	INVITE(IAM)				
	IDR	+			+	183 Session Progress(IDR)				
					T39 e	xpiry				
	ACM	←			+	180 Ringing(ACM)				
	ANM	-			+	200 OK INVITE(ANM)				
			Conversa	tion		, ,				
	REL	→			→	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	Ensure that call setup is contin	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 ISLIP interworking							
SIP parameter values	183 Session Progress: Contenbody INFO: Content-Type: application	,,	• •	·	•				
ISUP parameter values									
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	183 Session Progress(IDR)	←		←	IDR				
				T39 e	expiry				
	180 Ringing(ACM)	+		←	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
			Conversation						
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	←		←	RLC				

5.3.8 Call hold (HOLD)

TP408001	SIP reference: RF	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	0 "1 1 6				
Test purpose	Call hold after answer, requ	uested t	by the origin	nating user	
					d retrieved are sent with CPG
	messages having the even				
SIP parameter	INVITE: Content-Type: app	olication	ISUP; CPG	encapsulate	ed in the MIME body
values					
ISUP parameter values					
Comments	ISUP		SU	Т	SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	+		+	
			Convers	ation	
	CPG(progress, hold)	→		→	INVITE(CPG, sendonly)
				+	200 OK INVITE
				→	ACK
	CPG(progress, retrieve)	→		→	INVITE(CPG, sendrecv)
				(200 OK INVITE
				→	ACK
	REL	→		→	= : = (: :==)
	RLC	←		(200 OK BYE(RLC)

TP408002	SIP reference: RF	C 3261 [[4]	Q.733 [i.6], d	ISUP reference: Q.1912.5 [1], 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 Ensure that the notification messages having the even	s that a	call is place	ed on hold and	d retrieved are sent with CPG GCF interworking.
SIP parameter values	INVITE: Content-Type: app	olication/	ISUP; CPG	encapsulate	d in the MIME body
ISUP parameter values					
Comments	SIP-I		SUT	Г	ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Convers	ation	
	INVITE(CPG, sendonly)	→		→	CPG(progress, hold)
	200 OK INVITE	+			
	ACK	→			
	INVITE(CPG, sendrecv)	→		→	CPG(progress, retrieve)
	200 OK INVITE	+			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		+	RLC

TP408003	SIP reference: RF	C 3261	[4]	Q.733 [i.6]. c	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, req Ensure that the notification messages having the ever	ıs that a	call is place	ed on hold and	I retrieved are sent with CPG
SIP parameter values	INVITE: Content-Type: app				
ISUP parameter values					
Comments	ISUP		SU	Γ	SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
			Convers	ation	
	CPG(progress, hold)	+		+	INVITE(CPG, sendonly)
				→	200 OK INVITE
				+	ACK
	CPG(progress, retrieve)	+		+	INVITE(CPG, sendrecv)
				→	200 OK INVITE
				+	ACK
	REL	→		→	BYE(REL)
	RLC	←		+	200 OK BYE(RLC)

TP408004	SIP reference: RF0	C 3261	[4]	"	ISUP reference: Q.1912.5 [1],
				Q.733 [i.6], c	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	y the termi	nating user	
	Ensure that the notifications messages having the even	s that a t indica	call is place tor set to "	ed on hold and progress". I-M	I retrieved are sent with CPG GCF interworking.
SIP parameter values	INVITE: Content-Type: app				
ISUP parameter values					
Comments	SIP-I		SU	Γ	ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Convers	ation	
	INVITE(CPG, sendonly)	+		+	CPG(progress, hold)
	200 OK INVITE	→			
	ACK	+			
	INVITE(CPG, sendrecv)	+		+	CPG(progress, retrieve)
	200 OK INVITE	\			Cr O(progress, retrieve)
	ACK	+			
	7.010	†			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		+	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 (ing call	is placed on	hold and retri	
SIP parameter values	INVITE: Content-Type: app	olication	/ISUP; CPG	encapsulated	in the MIME body
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	CPG(progress, hold)	→		→	INVITE(CPG, sendonly) 200 OK INVITE
				→	ACK
	CPG(progress, retrieve)	→		→	INVITE(CPG, sendrecv)
				+	200 OK INVITE
				→	ACK
	ANM	+		+	200 OK INVITE(ANM)
			Conversa	ition	` '
	REL	→		→	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP408006	SIP reference: RF	C 3261	[4]	Q.733 [i.6	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 (ing call	is placed or	n hold and ret	rieved after alerting the rking.
SIP parameter values	INVITE: Content-Type: app				
ISUP parameter values					
Comments	SIP-I		SU	Т	ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		+	ACM
	INVITE(CPG, sendonly)	→		→	CPG(progress, hold)
	200 OK INVITE	+			
	ACK	→			
	INVITE(CPG, sendrecv)	→		→	CPG(progress, retrieve)
	200 OK INVITE	+			
	ACK	→			
	200 OK INVITE(ANM)	+		+	ANM
			Convers	ation	
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		+	RLC

TP408007	SIP reference: R	FC 3261	[4]	ISUP reference: Q.1912.5 [1], Q.764 [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, Ensure that a call in the h service. O-MGCF interwo	neld state			served user user who activated the Call hold					
SIP parameter		INVITE: Content-Type: application/ISUP; CPG encapsulated in the MIME body								
values				•	•					
ISUP parameter										
values										
Comments	ISUP		SUT	Γ	SIP-I					
	IAM	→		→	INVITE(IAM)					
	ACM	+		+	180 Ringing(ACM)					
	ANM	+		+	200 OK INVITE(ANM)					
			Convers	ation						
	CPG(progress, hold)	→		→	INVITE(CPG, sendonly)					
	, ,			+						
				→	ACK					
	REL	→		→	BYE(REL)					
	RLC	+		+	200 OK BYE(RLC)					

TP408008	SIP reference: RF	C 3261	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	encapsula	ated	in the MIME body	
ISUP parameter values							
Comments	SIP-I		SUT			ISUP	
	INVITE(IAM)	→			→	IAM	
	180 Ringing(ACM)	+			(ACM	
	200 OK INVITE(ANM)	+			(ANM	
			Convers	ation			
	INVITE(CPG, sendonly)	→			→	CPG(progress, hold)	
	200 OK INVITE	+					
	ACK	→					
	BYE(REL)	→			→	REL	
	200 OK BYE(RLC)	+			←	RLC	

TP408009	SIP reference: R	FC 3261 [4]	ISUP reference: Q.1912.5 [1], Q.764 [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,						
				by the us	er who did not activate the		
	Call hold service. O-MG0						
SIP parameter	INVITE: Content-Type: a	pplication/	ISUP; CPG enca	psulated	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)		
			Conversation				
	CPG(progress, hold)	+		+	INVITE(CPG, sendonly)		
			•	→	200 OK INVITE		
				+	ACK		
	REL	→		→	BYE(REL)		
	RLC	←		+	200 OK BYE(RLC)		

TP408010	SIP reference: RF	C 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	Ensure that a call in the he	Call hold after answer, release of the call by the terminating user Ensure that a call in the held state can be released by the user who did not activate the Call hold service. I-MGCF interworking.								
SIP parameter values	INVITE: Content-Type: app	olication	/ISUP; CPG	encapsulat	ted i	n the MIME body				
ISUP parameter values										
Comments	SIP-I		SUT			ISUP				
	INVITE(IAM)	→		-	→	IAM				
	180 Ringing(ACM)	+		•	H	ACM				
	200 OK INVITE(ANM)	+		•	H	ANM				
			Conversa	ation						
	INVITE(CPG, sendonly)	+		•	Т	CPG(progress, hold)				
	200 OK INVITE	→								
	ACK	+								
	BYE(REL)	→			>	REL				
	200 OK BYE(RLC)	+				RLC				

TP408011	SIP reference: R	FC 3261 [4	1	ISUP reference: Q.1912.5 [1], Q.764 [i.12], clause 2.3						
TSS reference	ISUP-SIP-ISUP/SS/HOLI)								
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		ser rated the Call hold service					
SIP parameter values	INVITE: Content-Type: a	INVITE: Content-Type: application/ISUP; CPG encapsulated in the MIME body								
ISUP parameter values										
Comments	ISUP		SUT		SIP-I					
	IAM	→		→	INVITE(IAM)					
	ACM	+		+	180 Ringing(ACM)					
			Ringing							
	CPG(progress, hold) → INVITE(CPG, sendonl									
				+	200 OK INVITE					
				→	ACK					
	REL	→		→	BYE(REL)					
	RLC	-		+	200 OK BYE(RLC)					

TP408012	SIP reference: RF	C 3261 [4	l]		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 alerting, re Ensure that a held call can without retrieving the call. I	be releas	sed by the	user who ac	user tivated the Call hold service
SIP parameter values	INVITE: Content-Type: app	olication/I	SUP; CPG	encapsulate	d in the MIME body
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		+	ACM
			Ringin	g	
	INVITE(CPG, sendonly)	→		→	CPG(progress, hold)
	200 OK INVITE	+			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		-	RLC

5.3.9 Call Waiting (CW)

TP409001	SIP reference: RI	FC 3261 [4	1	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 A Ensure that a call can be waiting call. O-MGCF inte	successfull	y establishe	d if the	ACM	indicates that it this call a			
SIP parameter values	180 Ringing: Content-Typ		on/ISUP; A	CM enca	apsula	ated in the MIME body			
ISUP parameter values	ACM: Generic notification	indicator "	Call is a wai	ting call'	II				
Comments	ISUP		SUT			SIP-I			
	IAM	→			→	INVITE(IAM)			
	ACM(waiting)	+			←	180 Ringing(ACM)			
	ANM ← 200 OK INVITE(ANM)								
			Conversati	on	•				
	REL	→			→	BYE(REL)			
	RLC	+			+	200 OK BYE(RLC)			

TP409002	SIP reference: RF	C 3261	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 A Ensure that a call can be s waiting call. I-MGCF interv	successf	ully establishe	d if the ACM	I indicates that this call is a				
SIP parameter values	180 Ringing: Content-Type		ation/ISUP; AC	CM encapsul	ated in the MIME body				
ISUP parameter values	ACM: Generic notification	indicator	"Call is a wait	ing call"					
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		+	ACM(waiting)				
	200 OK INVITE(ANM) ← ← ANM								
			Conversation	on					
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		←	RLC				

TP409003	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/C	W						
SIP selection criteria								
ISUP selection								
criteria								
Test purpose	Call waiting indication Ensure that a call can waiting call. O-MGCF	be successfu		d if the CF	PG indicates that this call is a			
SIP parameter values				G encaps	sulated in the MIME body			
ISUP parameter values	CPG: Generic notifica	ition indicator	"Call is a waiti	ng call"				
Comments	ISUP		SUT		SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	+		+	183 Session Progress(ACM)			
	CPG(waiting)	+		+	180 Ringing(CPG)			
	ANM ← 200 OK INVITE(ANM)							
			Conversation	1				
	REL	→		→	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	Ensure that a call can be succe waiting call. I-MGCF interworki 180 Ringing: Content-Type: ap CPG: Generic notification indic	ng. plicatio	n/ISUP; CPG end	capsul				
values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	183 Session Progress ACM)	←		+	ACM			
	180 Ringing(CPG)	+		+	CPG(waiting)			
	200 OK INVITE(ANM)	←		+	ANM			
			Conversation					
	BYE(REL)	→		→	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					
Test purpose	User rejects the waiting	g call			
	Ensure that the SUT pas	s on a	REL with cause	e #21 (d	call rejected) if a busy user rejects
	the waiting call. O-MGCI	F interv	vorking.		
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	otificat	ion indicator "Ca	all is a v	vaiting call"
values	REL: Cause #21 (call re	jected)			
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM(waiting)	+		+	180 Ringing(ACM)
	REL(#21)	+		+	480 Temporarily Unavailable(REL)
	RLC	→		→	ACK

TP409006	SIP reference: RFC 3261 [4]		ISUP reference:					
					1912.5 [1],			
			Q.733	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 with	h 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/	ISUI	; ACM or CPG	enc	apsulated in the MIME			
values	body							
	480 Temporarily unavailable: Content-T	уре:	application/ISU	P;R	EL encapsulated in the			
	MIME body							
ISUP parameter	ACM or CPG: Generic notification indica	ator '	Call is a waiting	call'	1			
values	REL: Cause #21 (call rejected)							
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	←		+	ACM(waiting)			
	,	←		+	REL(#21)			
		→		→	RLC			

TP409007	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	0.11							
Test purpose	Call waiting ignored (exp							
					ver from user, user alerted) if			
OID	a busy user does not answ							
SIP parameter values	180 Ringing: Content-Type	e: applic	ation/ISUP	, ACIVI OF CPG	encapsulated in the MilME			
values	body	Ja. Can		mmliantiam/ICLIF	O. DEL appearable to the			
	480 Temporarily unavailable: Content-Type: application/ISUP; REL encapsulated in the MIME body							
ISUP parameter	ACM or CPG: Generic notification indicator "Call is a waiting call"							
values	REL: Cause #19 (no answ	er from	user, user a	lerted)				
Comments	ISUP		SU	Γ	SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM(waiting)	+		+	180 Ringing(ACM)			
	T9 expiry							
	CASE A							
	REL(#19)	→		→	BYE(REL)			
	RLC ← 200 OK BYE(RLC)							
	← 487 Request 1							
	ACK							
	CASE B							
	REL(#19)	→		→	CANCEL			
	RLC	RLC ← 200 OK CANCEL						
				+	487 Request Terminated			
				→	ACK			

TP409008	SIP reference: RFC	3261	[4]		Į;	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 (expire						
						er from user, user alerted) if	
	a busy user does not answe						
SIP parameter	180 Ringing: Content-Type: application/ISUP; ACM or CPG encapsulated in the MIME						
values	body						
	480 Temporarily unavailable	e: Cont	ent-Type: a	pplication	/ISUP	; REL encapsulated in the	
	MIME body						
ISUP parameter	ACM or CPG: Generic notification indicator "Call is a waiting call"						
values	REL: Cause #19 (no answe	r from	user, user a	lerted)			
Comments	SIP-I		SUT	Γ		ISUP	
	INVITE(IAM)	→			→	IAM	
	180 Ringing(ACM) ← ACM(waiting)						
	T9 expiry						
	BYE(REL) → REL(#19)						
	200 OK BYE(RLC) ← RLC						
	487 Request Terminated	+					
	ACK	→					

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

		4]	ISUP reference: Q.1912.5 [1], Q.732 [i.4], clause 2.5			
ISUP-SIP-ISUP/SS/Call	Diversion					
"Call is diverting" indicati	on receive	d in 180 Ringin				
Verify that a call can be successfully established, if diversion occurs. The ACM contains the generic notification indicator set to "call is diverting", the call diversion information and the redirection number. The Redirection reason is set to CV_redirection_reason. CPG (alerting) is coded as if it has been mapped from the 180 Ringing (CPG). O-MCGF interworking. 183 Session Progress: Content-Type: application/ISUP; ACM encapsulated in the MIME body						
ACM: BCI Called party status indicator "No indication"						
Generic notification	n					
	Redirection number					
	erting	OUT	L loip i			
		501	SIP-I → INVITE(IAM)			
= \						
- 100 0000011 10gra						
	_		 ← 180 Ringing(CPG) ← 200 OK INVITE(ANM) 			
ZINIVI		Conversation	200 OK HAVITE(ANNI)			
RFI	→	Conversation	→ BYE(REL)			
RLC	+		€ 200 OK BYE(RLC)			
	"Call is diverting" indicati Verify that a call can be seen the generic notification and the redirection num. The Redirection reason in CPG (alerting) is coded at O-MCGF interworking. 183 Session Progress: Cobody 180 Ringing: Content-Tyle ACM: BCI Called party seen Call diversion information Redirection numb. CPG: Event indicator=ale ISUP IAM ACM(no indication) CPG(alerting) ANM REL	"Call is diverting" indication received Verify that a call can be successfull the generic notification indicator and the redirection number. The Redirection reason is set to CN CPG (alerting) is coded as if it has O-MCGF interworking. 183 Session Progress: Content-Type body 180 Ringing: Content-Type: applicated ACM: BCI Called party status indices Generic notification Call diversion information Redirection number CPG: Event indicator=alerting ISUP IAM → ACM(no indication) CPG(alerting) ANM ← REL	"Call is diverting" indication received in 180 Ringing Verify that a call can be successfully established, if the generic notification indicator set to "call is diverged in the redirection number. The Redirection reason is set to CV_redirection_r CPG (alerting) is coded as if it has been mapped fr O-MCGF interworking. 183 Session Progress: Content-Type: application/IS body 180 Ringing: Content-Type: application/ISUP; CPG ACM: BCI Called party status indicator "No indicated Generic notification Call diversion information Redirection number CPG: Event indicator=alerting ISUP IAM → ACM(no indication) CPG(alerting) ANM ← Conversation REL Conversation			

TP410002	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.732 [i.4], clause 2.5					
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion								
SIP selection criteria									
ISUP selection criteria									
Test purpose SIP parameter values	"Call is diverting" indication received in CPG Verify that a call can be successfully established, if diversion occurs. The ACM contains the generic notification indicator set to "call is diverting", the call diversion information and the redirection number. The Redirection reason is set to CV_redirection_reason. 180 Ringing (CPG (alerting)) is coded as if it has been mapped from the CPG. I-MCGF interworking. 183 Session Progress: Content-Type: application/ISUP; ACM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; CPG encapsulated in the MIME body								
ISUP parameter values	ACM: BCI Called party status indicator "No indication" Generic notification Call diversion information Redirection number CPG: Event indicator=alerting								
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	^		→	IAM				
	183 Session Progress(ACM)	+		+	ACM(no indication)				
	180 Ringing(CPG) ← CPG(alerting)								
	200 OK INVITE(ANM)	+		+	ANM				
	Conversation								
	BYE(REL)	→		→	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]	ISUP reference: Q.1912.5 [1], Q.732 [i.4], clause 2.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/Cal	I Diversion						
SIP selection criteria								
ISUP selection criteria								
Test purpose	"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.							
SIP parameter values	180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 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	→		→ INVITE(IAM)				
	ACM(free)	+		← 180 Ringing(ACM)				
	CPG ← 183 Session Progress(CPG)							
	CPG(alerting) ← 183 Session Progress(CPG)							
	ANM ← 200 OK INVITE(ANM)							
	Conversation							
	REL	→		→ BYE(REL)				
	RLC	+		← 200 OK BYE(RLC)				

TP410004	SIP reference: RFC 3261 [4]		SUP reference:					
			Q.1912.5 [1],					
		2 [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 AC		TI 4014					
	Verify that a call can be successfully establis							
	indicates that "call diversion may occur" in th							
	following CPG contains the generic notification diversion information and the redirection number 1.							
	The CPG (progress) contains CV_redirection							
	also Redirection number. The CPG (alerting)							
	with RnNbRes parameter (optional).	13 00000 43 11 10	nas been mapped nom New,					
	I-MCGF interworking.							
SIP parameter	180 Ringing: Content-Type: application/ISUF	; ACM encapsu	lated in the MIME body					
values	183 Session Progress: Content-Type: application							
	body							
ISUP parameter	ACM: BCI Called party status indicator "subscriber free", Optional backward call indicator:							
values	"Call diversion may occur"							
	CPG: Event information=progress, Call dive	rsion informatio	n; Generic notification;					
	Redirection number							
0	CPG: Event information=alerting							
Comments	_	SUT	ISUP					
	INVITE(IAM) →	→	IAM					
		180 Ringing(ACM) ← ACM(free)						
	183 Session Progress(CPG) ← ← CPG 183 Session Progress(CPG) ← ← CPG(alerting)							
	3 (/	←	CPG(alerting) ANM					
	200 OK INVITE(ANM) Conversation							
	BYE(REL) →	ersation -	REL					
	200 OK BYE(RLC)	- 1						
	200 ON DTE(NLO)		INLU					

CV_redirection_reason TP410003, TP410004					
VA_1 No reply					
VA_2	Deflection during alerting				

TP410005	SIP reference: RF	-	[4]	Q.	ISUP reference: Q.1912.5 [1], 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			10101011 11110	matien are received, as in				
	The CV_redirection_reas			ection reason					
	The Redirection number re								
	O-MCGF interworking.		•	•					
SIP parameter	183 Session Progress: Co	ntent-Typ	oe: applicat	ion/ISUP; AC	CM encapsulated in the MIME				
values	body								
		ntent-Typ	oe: applicat	ion/ISUP; CF	PG encapsulated in the MIME				
	body	body							
		180 Ringing: Content-Type: application/ISUP; CPG encapsulated in the MIME body							
ISUP parameter	ACM: BCI Called party status indicator "No indication"								
values	Generic notification				100				
	Call diversion inforr		edirection r	eason uncon	iditional				
	Redirection number								
	CPG1: Event information=		i						
	Generic notification Call diversion inforr		odiraction r	ooson CV ra	direction reason				
	Redirection number		edirection	eason CV_I	edirection_reason				
	Redirection number		on						
	CPG2: Event information=			number rest	triction				
Comments	ISUP	2.39,	SUT	1	SIP-I				
	IAM	→		→	INVITE(IAM)				
	ACM(no indication)	+		(183 Session Progress(ACM)				
	CPG1	+		(183 Session Progress(CPG)				
	CPG2(alerting)	+		+	180 Ringing(CPG)				
	ANM	+		+	200 OK INVITE(ANM)				
			Conversa	ition					
	REL	→		→	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP410006	SIP reference: RFC 3261	1 [4]			I	SUP reference:			
						Q.1912.5 [1],			
					Q.73	32 [i.4], clause 2.4.2			
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion	n							
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	ng the	call div	version i	nforn	nation are received, as if			
	multiple forwardings have occurre								
	The CV_redirection_reason is u								
	The Redirection number restriction	on pai	rameter	is passed	on.				
	I-MCGF interworking.								
SIP parameter	183 Session Progress: Content-T	Гуре:	applicat	ion/ISUP	; ACN	I encapsulated in the MIME			
values	body	_							
	183 Session Progress: Content-T	Гуре:	applicat	ion/ISUP	; CPG	encapsulated in the MIME			
	body	. ,.	"0115	000					
1011D	180 Ringing: Content-Type: appli				apsu	ated in the MIME body			
ISUP parameter values	ACM: BCI Called party status in	dicate	or "No in	dication"					
values	Generic notification Call diversion information	Dadie	ootion r	20000 110	oondi	tional			
	Redirection number	Keuii	ection	eason un	Condi	lionai			
	CPG: Event information=progres	ee							
	Generic notification	33							
	Call diversion information	Redir	ection r	eason CV	/ red	irection reason			
	Redirection number								
	Redirection number restric	ction							
	CPG: Event information=alerting	a, Red	direction	number	restric	ction			
Comments	SIP-I		Sl			ISUP			
	INVITE(IAM)	→			→	IAM			
	: 00 000:0::: : : : : : : : : : : : : :	←			+	ACM(no indication)			
	183 Session Progress(CPG)	(+	CPG1			
	3 3(/	(+	CPG2(alerting)			
	200 OK INVITE(ANM)	(+	ANM			
			Conver	sation					
	BYE(REL)	→			→	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	0		SUP reference: Q.1912.5 [1],							
TSS reference	ISUP-SIP-ISUP/SS/Call Div	oreion		Q.	132	[i.4], clause 2.5.2.2.1					
SIP selection	ISUT-SIT-ISUT/SS/Call DIVEISIUI										
criteria											
ISUP selection											
criteria											
Test purpose	Notification procedures for a	divert	ing call - afte	er the dive	rting	exchange					
	Verify that the IUT can successfully pass on in both directions (on the leg after the diversion) all the diversion information from the diverting exchange. It has to be checked that the following signalling information is passed on in the forward direction: redirecting number (see note); original called number (see note); redirection information. It has to be checked that the following signalling information is passed on in the backward direction: redirection number restriction parameter (in ACM/CPG/ANM/CON). O-MCGF interworking.										
SIP parameter values	INVITE: Content-Type: appl										
ISUP parameter	200 OK INVITE: Content-Ty IAM: Redirecting number, O	riginal	called numb	er Redire	oction	information					
values	ANM: Redirection address r			er, reduie	Cuoi	i illioittiation					
Comments	ISUP	-50.100	SUT			SIP-I					
	IAM	→	1		→	INVITE(IAM)					
	ACM	+			(180 Ringing(ACM)					
	ANM	+			-	200 OK INVITE(ANM)					
			Conversa	ition		, ,					
	REL	→			→	BYE(REL)					
	RLC ← 200 OK BYE(RLC)										
NOTE: Altered in	Gateways.			•		· ·					

TP410008	SIP reference: RFC	3261 [4]	_	SUP reference: Q.1912.5 [1], [i.4], clause 2.5.2.2.1						
TSS reference	ISUP-SIP-ISUP/SS/Call Div	ersion									
SIP selection											
criteria											
ISUP selection											
criteria											
Test purpose	Notification procedures for a Verify that the IUT can succ		·	·	, G						
	diversion) all the diversion in										
	It has to be checked that the										
	direction:	7 10110111	ng orginaling in	omation	is passed on in the forward						
	redirecting num	ber (se	e note):								
	original called n										
	redirection infor	mation	•								
	It has to be checked that the	e followi	ng signalling inf	ormation i	is passed on in the backward						
	direction:										
	redirection num	ber res	triction parame	eter (in AC	: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			Redirection	n information						
values	ANM: Redirection address r	estriction		1	_						
Comments	SIP-I		SUT		ISUP						
	INVITE(IAM)	→		→	IAM						
	180 Ringing(ACM)	+		←	ACM						
	200 OK INVITE(ANM)	+		←	ANM						
			Conversation								
	BYE(REL)	→		→	REL						
	200 OK BYE(RLC) ← RLC										
NOTE: Altered in	Gateways.		<u>-</u>								

TP410009	SIP reference: RF	FC 3261	[4]	•	SUP reference: Q.1912.5 [1], [i.2], clause 3.5.2.4.1			
TSS reference	ISUP-SIP-ISUP/SS/Call D	Diversion						
SIP selection								
criteria								
ISUP selection criteria	PICS 10/1 AND PICS 1/7							
Test purpose	Verify that the outgoing in number according to the	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.						
SIP parameter values	INVITE: Content-Type: ap encapsulated in the MIME		/ISUP; IAM con	taining an (Original called number			
ISUP parameter values	IAM: No original called nu	mber pre	sent					
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	, ,		Conversation	n .				
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP410010	SIP reference: RFC	3261 [4]	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 v									
SIP parameter				ntaining an (Original called number called					
values	number encapsulated in the									
ISUP parameter	IAM: Original called number	r "Internatio	nal numb	er"						
values										
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	→		→	IAM					
	180 Ringing(ACM)	+		+	ACM					
	200 OK INVITE(ANM) ← ← ANM									
		C	Conversati	on						
	BYE(REL)	→		→	REL					
	200 OK BYE(RLC)	+	•	+	RLC					

TP410011	SIP reference: RFC	3261	Γ 4 1		19	SUP reference:		
11 410011	On reference: At a	0201	L-1		•	Q.1912.5 [1],		
				0	731	[i.2], clause 4.5.2.1.1		
TSS reference	ISUP-SIP-ISUP/SS/Call Div	oroion		- 4	.731	[1.2], Clause 4.5.2.1.1		
	150P-51P-150P/55/Call Div	ersion						
SIP selection								
criteria								
ISUP selection	PICS 1/7							
criteria								
Test purpose	Original called number in	the ou	tgoing inte	rnational	gate	way		
	Verify that the outgoing inte	rnation	al gateway	checks an	id ma	nipulates the original called		
	number according to the pr	ocedur	es as define	ed for CLII	P:	_		
	Discarding the original	called	number, if t	the addres	ss is r	narked not available.		
	The PTC will send an IAM v							
SIP parameter	INVITE: Content-Type: appl	ication	/ISUP; IAM	containing	an C	Original called number called		
values	number encapsulated in the				•			
ISUP parameter	IAM: No original called num	ber pre	esent					
values		•						
Comments	SIP-I		SUT	Г		ISUP		
	INVITE(IAM)	1			→	IAM		
	180 Ringing(ACM)	4			+	ACM		
	200 OK INVITE(ANM) ← ANM							
			Convers	ation				
	BYE(REL)	^			→	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP410012	SIP reference: RF	C 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 Di	version			
SIP selection					
criteria					
ISUP selection	PICS 1/8				
criteria					
Test purpose	Original called number in				
					anipulates the original called
	number according to the p				
	Converting the original	l called	number to	national forma	t, if necessary (own country
	code).				
SIP parameter	INVITE: Content-Type: app	olication	/ISUP; IAM	containing an	Original called number called
values	number encapsulated in th	e MIME	body		-
ISUP parameter	IAM: Original called number	er "Natio	nal number	ı	
values					
Comments	SIP-I		SUT	-	ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
	, ,		Conversa	ation	
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		+	RLC

TP410013	SIP reference: RF0	[4]		I:	SUP reference: Q.1912.5 [1],	
					2.731	[i.2], clause 4.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/Call Div	version				
SIP selection criteria						
ISUP selection criteria	PICS 10/2 AND PICS 1/7					
Test purpose	Redirecting number in the					
	Verify that the outgoing inte					nipulates the redirecting
	number according to the p	rocedure	es as define	ed for CLI	P:	
	Discarding the redirect	ing nun	nber if case	of bilate	ral agr	reements.
SIP parameter	INVITE: Content-Type: app	lication/	ISUP; IAM	containin	g a Re	edirecting number
values	encapsulated in the MIME	body				
ISUP parameter values	IAM: No Redirecting number	er prese	nt			
Comments	SIP-I		SU			ISUP
	INVITE(IAM)	→			→	IAM
	180 Ringing(ACM)	+			+	ACM
	200 OK INVITE(ANM)	+			+	ANM
			Convers	ation		
	BYE(REL)	→			→	REL
	200 OK BYE(RLC)	+			+	RLC

TP410014	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	version						
SIP selection								
criteria								
ISUP selection	PICS 1/7							
criteria								
Test purpose	Redirecting number in the	e outgo	ing interna	tional ga	tewa	у		
	Verify that the outgoing inte	ernation	al gateway	checks an	d ma	nipulates the redirecting		
	number according to the pr	rocedur	es as define	ed for CLIF	⊃:	-		
	Discarding the redirect	ing nur	nber , if the	address is	mar	ked not available.		
	The PTC will send an IAM	with an	"address no	ot available	e" Rg	Nb.		
SIP parameter	INVITE: Content-Type: app	lication	/ISUP; IAM	containing	g a Re	edirecting number		
values	encapsulated in the MIME I	oody				-		
ISUP parameter	IAM: No Redirecting number	er prese	ent					
values		-						
Comments	SIP-I		SU	Г		ISUP		
	INVITE(IAM)	→			→	IAM		
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM) ← ANM							
	, , ,		Convers	ation				
	BYE(REL)	→			→	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP410015	SIP reference: RF	C 3261	[4]		.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 D	iversion							
SIP selection criteria									
ISUP selection criteria	PICS 1/7	PICS 1/7							
Test purpose	Verify that the outgoing int number according to the p Converting the redirect of address presentation	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				containing	a Re	edirecting number "National			
values	number" encapsulated in t								
ISUP parameter values	IAM: Redirecting number '	"Internati	onal numbe	er"					
Comments	SIP-I		SU	Γ		ISUP			
	INVITE(IAM)	→			→	IAM			
	180 Ringing(ACM)	+			←	ACM			
	200 OK INVITE(ANM)	←			←	ANM			
			Convers	ation					
	BYE(REL)	→			→	REL			
	200 OK BYE(RLC)	+			(RLC			

TP410016	SIP reference: RF	C 3261	[4]			SUP reference: Q.1912.5 [1],
				Q.		[i.4], clause 2.5.2.3,
						[i.2], clause 3.5.2.3
TSS reference	ISUP-SIP-ISUP/SS/Call D	iversion				
SIP selection						
criteria						
ISUP selection criteria	PICS 1/8					
Test purpose	Redirecting number in th					
	Verify that the incoming in					nipulates the redirecting
	number according to the p					
	Converting the redirec code).	ting nur	nber to nati	onal format	t, if n	ecessary (own country
	The PTC will send an IAM					
SIP parameter	INVITE: Content-Type: ap				a Re	directing number
values	"International number" end			/IE body		
ISUP parameter values	IAM: Redirecting number "	national	number"			
Comments	SIP-I		SUT	•		ISUP
	INVITE(IAM)	→			→	IAM
	180 Ringing(ACM)	+		•	(ACM
	200 OK INVITE(ANM)	+		•	(ANM
			Conversa	ation		
	BYE(REL)	→			→	REL
	200 OK BYE(RLC)	+		•	←	RLC

TP410017	SIP reference: RF	FC 3261	[4]	Q.732	SUP reference: Q.1912.5 [1], ! [i.4], clause 2.5.2.3, I [i.2], clause 3.5.2.3					
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion									
SIP selection criteria										
ISUP selection criteria	PICS 1/8 AND 10/4									
Test purpose	Redirecting number in the Verify that the incoming in number according to the Adding a prefix to an in The PTC will send an IAM	ternation procedur nternation	nal gateway of es as defined nal redirecti	checks and maded for CLIP:	ay anipulates the redirecting					
SIP parameter values	INVITE: Content-Type: ap encapsulated in the MIME		/ISUP; IAM c	ontaining a R	dedirecting number					
ISUP parameter values	IAM: Redirecting number	•								
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	→		→	IAM					
	180 Ringing(ACM)	←		←	ACM					
	200 OK INVITE(ANM)	200 OK INVITE(ANM) ← ANM								
		Conversation								
	BYE(REL)	→		→	REL					
	200 OK BYE(RLC)	+		+	RLC					

TP410018	SIP reference: F	RFC 3261 [4]		ISUP reference: Q.1912.5 [1], Q.732 [i.4], clause 2.5.2.4 Q.731 [i.2], clause 3.5.2.4					
TSS reference	ISUP-SIP-ISUP/SS/Call	Diversion							
SIP selection criteria									
ISUP selection criteria	PICS 10/5 AND PICS 1/5	8							
Test purpose	Redirection number in Verify that the incoming number according to the discarding the redirec removes the redirect	international e procedures ection numbe	gateway defined for er in case	checks and or COLP: of bilatera	d manipulates the redirection				
SIP parameter values	number encapsulated in	the MIME bo t-Type: applic	dy cation/ISU	JP; ANM c	ACM containing a Redirection ontaining a Redirection address				
ISUP parameter values	ACM: Called party statu Generic notification Call diversion info No Redirection no ANM: No Redirection no	on ormation Red umber	irection re		onditional				
Comments	ISUP		SUT		SIP-I				
	IAM	→		→	INVITE(IAM)				
	ACM(no indication)	+		+	183 Session Progress(ACM)				
	CPG	+		+	180 Ringing(CPG)				
	ANM	+							
		(Conversa	tion					
	REL	→		→	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP410019	SIP reference: RFC	3261	[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 1/7							
SIP parameter values ISUP parameter	number according to the pr Converting the redirecti code): 1. the PTC will provide the 2. ACM with CDInf, GenNo 183 Session Progress: Con number "International numb ACM: Called party status=	rnation ocedur on nur necess t = "ca tent-Ty er" end	al gateway of es defined for the national sary stimulur is diverting pe: applicate apsulated in the sary stimulur in the sary stimulur in the sary stimulur in the sary stimulur in the sary sary sary sary sary sary sary sary	checks or COL onal fo s; g" and ion/ISU	and m P: prmat, i an inte	ernational RnNb with own CC.		
values	Generic notification							
	Call diversion inform			eason	uncon	ditional		
•	Redirection number	"Nation		1		loip i		
Comments	ISUP		SUT			SIP-I		
	IAM	→			<u>→</u>	INVITE(IAM)		
	ACM(no indication)	-			<u>+</u>	183 Session Progress(ACM)		
	CPG	← 180 Ringing(CPG)						
	ANM	+	Camura :	4:	+	200 OK INVITE(ANM)		
	DEL		Conversa	tion		DVE(DEL)		
	REL	→			<u>→</u>	BYE(REL)		
	RLC	+			+	200 OK BYE(RLC)		

TP410020	SIP reference: RF	C 3261	[4]		ISUP	reference:				
					Q.19	912.5 [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 Di	version								
SIP selection criteria										
	PICS 1/8									
ISUP selection criteria	PICS 1/8									
Test purpose	Redirection number in th									
	Verify that the incoming int					lates the redirection				
	numbe r according to the p									
	Converting the redirec									
SIP parameter	183 Session Progress: Co					taining a Redirection				
values	number "National number"	encaps	ulated in the	MIME	body					
ISUP parameter	ACM: Called party status=	no indic	cation							
values	Generic notification									
	Call diversion inforr	nation R	dedirection r	eason ι	inconditional					
	Redirection number	r "Interna	ational num	oer"						
Comments	ISUP		SUT		SIP-I					
	IAM	→			→ INVIT	E(IAM)				
	ACM(no indication)	←			← 183 S	Session Progress(ACM)				
	CPG	+			← 180 R	Ringing(CPG)				
	ANM	+			← 200 C	OK INVITE(ANM)				
			Conversa	tion		·				
	REL	→			→ BYE(I	REL)				
	RLC	+			← 200 C	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 Div	ersion							
SIP selection criteria									
ISUP selection criteria	PICS 1/8 AND PICS 10/6								
Test purpose	Verify that the outgoing inte number according to the pr Adding a prefix to an inte	Redirection number in the outgoing international gateway /erify that the outgoing international gateway checks and manipulates the redirection number according to the procedures defined for COLP: Adding a prefix to an international redirection number. The PTC will provide the necessary stimulus.ACM with CDInf, GenNot = "call is diverting"							
SIP parameter values	183 Session Progress: Connumber "International numb				r; ACM containing a Redirection				
ISUP parameter values	ACM: Called party status=I Generic notification Call diversion inform Redirection number	no indication R	edirection re		•				
Comments	ISUP		SUT		SIP-I				
	IAM	→			→ INVITE(IAM)				
	ACM(no indication)	+			★ 183 Session Progress(ACM)				
	CPG								
	ANM	← 200 OK INVITE(ANM)							
			Conversat	ion	· · ·				
	REL	→			→ 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 SIP parameter values ISUP parameter	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 CPG: Generic notification: conference established								
values	CPG: Generic notification: other party a	added			_				
Comments	ISUP	<u> </u>	SUT	↓ _	SIP-I				
	IAM	→		→	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	+		←	200 OK INVITE(ANM)				
			Conversation						
	CPG(conference established)	→		→	INFO(CPG)				
				+	200 OK INFO				
	CPG(other party added) → INFO(CF								
	, - 1 - 2 /			+	200 OK INFO				
	REL	→		→	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP411002	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	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 values		olication	n/ISUP; CPG	encaps	sulated in the MIME body			
ISUP parameter values	CPG: Generic notification CPG: Generic notification			ished				
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	` '		Conversation	<u> </u>				
	INFO(CPG)	→		→	CPG(conference established)			
	200 OK INFO ←							
	INFO(CPG)	→		→	CPG(other party added)			
	200 OK INFO	+						
	BYE(REL)	→		→	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										
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 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.									
SIP parameter values	INFO: Content-Type: application/ISUP; CI	PG (encapsulated in	the I	MIME body					
ISUP parameter values	CPG: Generic notification: conference est CPG: Generic notification: isolated	abli	shed							
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					
	CPG(isolated)	>		→	INFO(CPG)					
				+	200 OK INFO					
		>		→	BYE(REL)					
	RLC	(←	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.							
SIP parameter	I-MGCF interworking.		n/ISLIP: CPG	encan	sulated in the MIME body			
values	in o. contont Typo. app	modilo	1,11001 , 01 0	опоарс	salated in the Minit Body			
ISUP parameter values	CPG: Generic notification CPG: Generic notification			shed				
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	←		+	ANM			
			Conversation)				
	INFO(CPG)	→		→	CPG(conference established)			
	200 OK INFO	+						
	INFO(CPG) → CPG(isolated)							
	200 OK INFO	+						
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP411005	SIP reference: RFC 3261 [Q.7	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 "confer	ence esta	blished", "isola	ated" a	nd "reattached"					
	CPG message:1. Assist a call set up from ISUP to2. Check that the notification "conformer conferee at SIP-I;	 Assist a call set up from ISUP to SIP-I; Check that the notification "conference established" is received in the CPG from conferee at SIP-I; Check the notification "reattached" in the CPG. 								
SIP parameter	INFO: Content-Type: application/IS	UP; CPG	encapsulated	in the	MIME body					
values										
ISUP parameter values	CPG: Generic notification: conferen CPG: Generic notification: isolated CPG: Generic notification: reattache		shed							
Comments	ISUP		SUT		SIP-I					
	IAM	→		→	INVITE(IAM)					
	ACM	+		+	\ /					
	ANM	+		+	200 OK INVITE(ANM)					
	7 11 1111		Conversation	<u> </u>	200 31(111112(711111)					
	CPG(conference established)	→		<u>`</u>	INFO(CPG)					
	or e(cornerence established)			+	200 OK INFO					
	CPG(isolated)	→		→	INFO(CPG)					
				+	200 OK INFO					
	CPG(reattached)	→		→	INFO(CPG)					
	, ,			+	200 OK INFO					
	REL	→		→	BYE(REL)					
	RLC	+	_	+	200 OK BYE(RLC)					

TP411006	SIP reference: F	RFC 32	261 [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", "isolated" and "reattached"								
	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 "reattached" in the CPG. I-MGCF interworking.								
SIP parameter			on/ISUP; CPG e	encap	sulated in the MIME body				
values ISUP parameter	CPG: Generic notificatio	n: oon	forance catablic	hod					
values	CPG: Generic notificatio	n: isola	ated	neu					
	CPG: Generic notificatio	n: reat							
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
			Conversation						
	INFO(CPG)	→		→	CPG(conference established)				
	200 OK INFO	+							
	INFO(CPG)	→		→	CPG(isolated)				
	200 OK INFO								
	INFO(CPG)	→		→	CPG(reattached)				
	200 OK INFO	+							
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP411007	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 "conferenc disconnected"	e esta	blished", "other	party	added" and "other party
	To verify that the IUT can successfully to CPG message: 1. assist a call set up from ISUP to SIF 2. check the notification "other party di O-MGCF interworking.	P-I;			d notifications in/from the
SIP parameter	INFO: Content-Type: application/ISUP;	CPG	encapsulated in	the I	MIME body
values					
ISUP parameter	CPG: Generic notification: conference		shed		
values	CPG: Generic notification: other party a CPG: Generic notification: other party of		nected		
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
	CPG(other party added)	→		→	INFO(CPG)
	, - 1 - 2 - 2 - 2 - 2			+	200 OK INFO
	CPG(other party disconnected)	→		→	INFO(CPG)
				+	200 OK INFO
	DEL	\downarrow		_	DVE(DEL)
	REL	→		→	BYE(REL)
	RLC	←		+	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	ISUP-SIP-ISUP/SS/CONF							
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	disconnected"								
	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 the notification "other party disconnected" in the CPG. I-MGCF interworking.								
SIP parameter values	INFO: Content-Type: ap	plication	n/ISUP; CPG	encaps	sulated in the MIME body				
ISUP parameter	CPG: Generic notificatio	n: confe	erence establi	shed					
values	CPG: Generic notificatio CPG: Generic notificatio	n: other	party added						
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	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	+							
	INFO(CPG)	→		→	CPG(other party disconnected)				
	200 OK INFO	+							
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP411009	SIP reference: RFC 3261 [4]	Q.	Preference: 1912.5 [1], 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 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.						
SIP parameter values	INFO: Content-Type: application/ISUP; CPG	encapsulated in the	MIME body				
ISUP parameter values	CPG: Generic notification: conference establ	ished					
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	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 trans	fer "co	onference estab	olished	d", and disconnect the conference		
SIP parameter	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						
values	in O. Content-Type. app	meane	, 01 0 0	ысара	sulated in the MiME body		
ISUP parameter values	CPG: Generic notification	n: conf	erence establis	shed			
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		→	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
			Conversation	1			
	INFO(CPG)	→		→	CPG(conference established)		
	200 OK INFO	+					
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	+		+	RLC		

5.3.12 ECT

TP412001	SIP reference: RFC 3261 [4	<u>.]</u>			SUP 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 transf							
	Verify that the IUT is able to send th							
	the service activation parameter "ca			call tra	ansfer number, received in the			
	ISUP FAC, in an INFO request for the	ne active	user.					
	O-MGCF interworking.							
SIP parameter	INFO: Content-Type: application/ISU	JP; FAC	encapsulat	ted in	the MIME body			
values								
ISUP parameter values	FAC: Generic notification=call transf	er active	e, Call trans	fer nu	umber (PIXIT)			
Comments	ISUP		SUT		SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
	Conversation							
	FAC(call transfer active, CTNb)	→		→	INFO(FAC)			
				+	200 OK INFO			
	REL	→		→	BYE(REL)			
	RLC	←		←	200 OK BYE(RLC)			

TP412002	SIP reference: RF	nce: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15		
TSS reference	ISUP-SIP-ISUP/SS/ECT		J		Q.104 [1.1], clause 1.0.10		
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 send the Generic notification parameter "Call transfer active", the service activation parameter "call transfer" and the call transfer number, received in the INFO request containing the encapsulated FAC, in a ISUP FAC for the active user. I-MGCF interworking.						
SIP parameter values	INFO: Content-Type: appli	cation/IS	SUP; FAC er	ncapsu	lated in the MIME body		
ISUP parameter values	FAC: Generic notification=	call trans	sfer active, (Call tra	nsfer number (PIXIT)		
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		→	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	Conversation						
	INFO(FAC)	→		→	FAC(call transfer active, CTNb)		
	200 OK INFO	+					
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP412005	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	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; CPG	encapsulate	ed in	the MIME body			
values	INFO: Content-Type: application/ISU							
ISUP parameter	CPG: Event indicator=progress, Gen	eric not	ification=hol	d				
values	FAC: Generic notification=call transfe	er active	·	er nu				
Comments	ISUP		SUT		SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	←		+	180 Ringing(ACM)			
	ANM	←		+	200 OK INVITE(ANM)			
			conversation					
	CPG(hold)	→		→	INVITE(CPG, sendonly)			
				-	200 OK INVITE(recvonly)			
				→	ACK			
	FAC(call transfer active, CTNb)	→		→	INFO(FAC)			
				+	200 OK INFO			
				→	INVITE(sendrecv)			
				←	200 OK INVITE(sendrecv)			
				→	ACK			
	REL	→		→	BYE(REL)			
	RLC	←		+	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 er	ncapsu	lated in the MIME body		
values	INFO: Content-Type: applica						
ISUP parameter	CPG: Event indicator=progre	ess, G	eneric notific	ation=	hold		
values	FAC: Generic notification=ca						
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		→	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
			Conversatio	n			
	INVITE(CPG, sendonly)	→		→	CPG(hold)		
	200 OK INVITE(recvonly)	+			<u> </u>		
	ACK	→					
	INFO(FAC)	→		→	FAC(call transfer active, CTNb)		
	200 OK INFO	+					
	INVITE(sendrecv)	→					
	200 OK INVITE(sendrecv)	+					
	ACK	→					
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP412009	SIP reference: RFC 3261 [4]	SIP reference: RFC 3261 [4]								
			0.72	Q.1912.5 [1], Q.732.7 [i.5], clause 7.5.2.1.1.1 a)						
TSS reference	ISUP-SIP-ISUP/SS/ECT		Q.73)Z./ [1.5], Clause 7.5.2.1.1.1 a)					
SIP selection	130F-31F-130F/33/EC1									
criteria										
ISUP selection										
criteria										
Test purpose	Loop prevention procedure - initiation									
	Verify that the SUT is able to transfer	r the lo	op request re	eceiv	ed in an ISUP LOP in an					
	INFO request containing the LOP me									
	received in an ISUP LOP in an SIP II	NFO re	quest contai	ining	the ISUP LOP message.					
	O-MGCF interworking.									
SIP parameter	INFO: Content-Type: application/ISU	P; CPC	encapsula	ted ir	the MIME body					
values	INFO: Content-Type: application/ISU									
IOLID	INFO: Content-Type: application/ISU				the MIME body					
ISUP parameter	CPG: Event indicator=progress, Gen		tification=ho	ıa						
values	LOP: request: Call transfer reference									
	LOP: response: Call transfer reference FAC: Generic notification=call transfer active, Call transfer number (PIXIT)									
Comments	ISUP	or activ	SUT		SIP-I					
Comments	IAM	→	301	→	INVITE(IAM)					
	ACM	/		′	180 Ringing(ACM)					
	ANM	+		È	200 OK INVITE(ANM)					
	Conversation									
	CPG(hold)	→	Jonversation	<u> </u>	INVITE(CPG, sendonly)					
	Ci C(ilola)			+	200 OK INVITE(recvonly)					
				→	ACK					
				Ť	7.6.1					
	LOP(request)	→		→	INFO(LOP)					
				+	200 OK INFO					
				Ť						
	LOP(response)	+		+	INFO(LOP)					
	- (→	200 OK INFO					
				1						
	FAC(call transfer active, CTNb)	→		→	INFO(FAC)					
	,			+	200 OK INFO					
				→	INVITE(sendrecv)					
				+	200 OK INVITE(sendrecv)					
				→	ACK					
	REL	→		→	BYE(REL)					
	RLC(RLC)	+		+	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	CPG: Event indicator=progre			cation=l	nold			
values	LOP: request: Call transfer r							
	LOP: response: Call transfe			.	() (D)((T)			
0	FAC: Generic notification=ca	all tran		Call tran				
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		←	ACM			
	200 OK INVITE(ANM)	+	<u> </u>	←	ANM			
			Conversation					
	INVITE(CPG, sendonly)	→		→	CPG(hold)			
	200 OK INVITE(recvonly)	+						
	ACK	→						
	INFO(LOP)	→		→	LOP(request)			
	200 OK INFO	+						
	INFO(LOP)	+		←	LOP(response)			
	200 OK INFO	→						
	INFO(FAC)	→		→	FAC(call transfer active, CTNb)			
	200 OK INFO	+						
	INVITE(sendrecv)	→						
	200 OK INVITE(sendrecv)	/						
	ACK	→						
	7.01	+*						
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+	1	-	RLC			
	ZUU UN BTE(NLU)	_	L		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	ic no	tification=hol	d				
values	LOP: request: Call transfer reference							
Comments	ISUP		SUT		SIP-I			
	IAM	→		1	INVITE(IAM)			
	ACM	←		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
		(Conversation					
	CPG(hold)	→		^	INVITE(CPG, sendonly)			
				+	200 OK INVITE(recvonly)			
				^	ACK			
	LOP(request)	→		→	INFO(LOP)			
				+	200 OK INFO			
	REL	→		^	BYE(REL)			
	RLC	←		+	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							
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	To verify that SUT is able to containing the LOP messag							
SIP parameter	INFO: Content-Type: applica	ation/IS	SUP; CPG enc	apsu	lated in the MIME body			
values	INFO: Content-Type: applica							
ISUP parameter	CPG: Event indicator=progr							
values	LOP: request: Call transfer i	referer	ice					
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	←		←	ANM			
			Conversation					
	INVITE(CPG, sendonly)	→		→	CPG(hold)			
	200 OK INVITE(recvonly)	+						
	ACK	→						
	INFO(LOP)	→		→	LOP(request)			
	200 OK INFO	+						
	DVE(DEL)	+		_	DEL			
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	←		←	RLC			

TP412013	SIP reference: RFC 3261 [4]	l	Q.732.7	ISUP 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	Verify that the SUT is able to transfel INFO request containing the LOP me loop detection procedure is unsuccestor-MGCF interworking.	r the looessage.	op request rece	ived in an ISUP LOP in an
SIP parameter	INFO: Content-Type: application/ISU	P: CPC	encapsulated	in the MIME body
values	INFO: Content-Type: application/ISU			
	INFO: Content-Type: application/ISU			in the MIME body
ISUP parameter	CPG: Event indicator=progress, Gen		ification=hold	
values	LOP: request: Call transfer reference			
	FAC: Generic notification=call transfe	er activ		
Comments	ISUP		SUT	SIP-I
	IAM	→	-	
	ACM	←	€	100 11119119(710111)
	ANM	←	€	200 OK INVITE(ANM)
			Conversation	
	CPG(hold)	→	-	
			€	 200 OK INVITE(recvonly)
			-	ACK
	LOP(request)	→		
			•	200 OK INFO
	510(II () () () () () () ()	+		11150(540)
	FAC(call transfer active, CTNb)	→	-	
			•	- 200 OK INFO
			-	INVITE(sendrecv)
		+		
		+	-	\ '
		+	7	ACK
	REL	+_	<u> </u>	BYE(REL)
	RLC	→	-	
	KLC	~	•	- 200 OK BYE(RLC)

TP412014	SIP reference: RFC				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	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. I-MGCF interworking.								
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								
ISUP parameter	CPG: Event indicator=progre								
values	LOP: request: Call transfer r								
	FAC: Generic notification=call transfer active, Call transfer number (PIXIT)								
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
	Conversation								
	INVITE(CPG, sendonly)	→		→	CPG(hold)				
	200 OK INVITE(recvonly)	+		<u> </u>	OT C(Hold)				
	ACK	→							
	AOR	+							
	INFO(LOP)	→		→	LOP(request)				
	200 OK INFO	/			LOI (lequest)				
	200 OK INFO								
	INFO(FAC)	→		→	FAC(call transfer active, CTNb)				
	200 OK INFO	+			i AC(call transfer active, CTNb)				
	200 OK INI O	+							
	INVITE(sendrecv)	→							
	200 OK INVITE(sendrecv)	-							
	ACK	→							
	ACR	+-							
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	-		/	RLC				
	1200 ON DIL(NLO)		<u> </u>	_	INEO				

TP412015	SIP reference: RFC 3261 [4] ISUP reference: Q.1912.5 [1],						
			0.732.7	7 [i.5], clause 7.5.2.1.1.1 a)			
TSS reference	ISUP-SIP-ISUP/SS/ECT			[], e.aaco :			
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Facility message with generic noti	ficatior	sent to the r	emote user			
	Verify that the SUT is able to transfer						
	transfer, alerting" and the service ac	tivation	n parameter se	et to "call transfer" received in an			
	ISUP FAC in a SIP INFO request cor	ntaining	the ISUP FAC	C. O-MGCF interworking.			
SIP parameter	INFO: Content-Type: application/ISU						
values	INFO: Content-Type: application/ISU			in the MIME body			
ISUP parameter	CPG: Event indicator=progress, Generic notification=hold						
values	FAC: Generic notification=call transfe	er active	e, Call transfer				
Comments	ISUP		SUT	SIP-I			
	IAM	→	1				
	ACM	+	•	180 Ringing(ACM)			
	ANM	+	•	200 OK INVITE(ANM)			
		С	onversation				
	CPG(hold)	→		INVITE(CPG, sendonly)			
			•	200 OK INVITE(recvonly)			
			-	▶ ACK			
	FAC(call transfer active, CTNb)	→	-	INFO(FAC)			
			•	200 OK INFO			
			=	NVITE(sendrecv)			
			•	200 OK INVITE(sendrecv)			
			=	▶ ACK			
	REL	→	=	▶ BYE(REL)			
	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 values	CPG: Event indicator=progre FAC: Generic notification=ca	ess, G	eneric notific	ation=	hold		
Comments	SIP-I INVITE(IAM) 180 Ringing(ACM) 200 OK INVITE(ANM)	→ ← ←	SUT	→ + +	ISUP IAM ACM ANM		
	INVITE(CPG, sendonly) 200 OK INVITE(recvonly) ACK	→ ← →	Conversatio	n →	CPG(hold)		
	INFO(FAC) 200 OK INFO	→		→	FAC(call transfer active, CTNb)		
	INVITE(sendrecv) 200 OK INVITE(sendrecv) ACK	→ ← →					
	BYE(REL) 200 OK BYE(RLC)	→		→	REL RLC		

TP412017	SIP reference: RFC 3261 [4	1	Q.73	-	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 gene Verify that the transfer the CPG with and the service activation paramet containing the ISUP CPG message. O-MGCF interworking.	the ge ler set to	neric notific	ation	n set to "call transfer, active"
SIP parameter values	INFO: Content-Type: application/ISU	JP; CPC	encapsula	ted in	the MIME body
ISUP parameter values	CPG: Generic notification=call trans	fer activ	e, Call trans	fer n	umber (PIXIT)
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	CPG(call transfer active, CTNb)	→		→	INFO(CPG)
				+	200 OK INFO
	ANM	+		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	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							
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 values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body							
ISUP parameter	CPG: Event indicator=progre	ess, Ge	eneric notific	cation=	hold			
values	CPG: Generic notification=c							
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	,		Conversation	on				
	INVITE(CPG, sendonly)	→		→	CPG(hold)			
	200 OK INVITE(recvonly)	+						
	ACK	→						
	INFO(CPG) 200 OK INFO	→		→	CPG(call transfer active, CTNb)			
	INVITE(sendrecv) 200 OK INVITE(sendrecv)	→						
	ACK	→						
	BYE(REL)	→		→	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 Verify that the exchange removes the call transfer number in the SIP-I INFO request containing a FAC or CPG before sending it to the next exchange, if its indicator is set to "presentation restricted" and there is no bilateral agreement to transfer the number.							
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC(CTNb=restricted) encapsulated in the MIME body							
ISUP parameter values	CPG: Event indicator=progre FAC: Generic notification=ca							
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
		Conversation						
	INVITE(CPG, sendonly)	→		· -	CPG(hold)			
	200 OK INVITE(recvonly)	+			0. 0()			
	ACK	→						
	INFO(FAC) 200 OK INFO	→		→	FAC(call transfer active)			
	INVITE(sendrecv)	→						
	200 OK INVITE(sendrecv) ACK	←						
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		+	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	ISUP-SIP-ISUP/SS/ECT							
SIP selection criteria									
ISUP selection									
criteria									
Test purpose	Call transfer number - conversion to international number Verify that the IUT converts the call transfer number 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/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		<u>all tran</u>		all trai	nsfer number=international (PIXIT)				
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		←	ANM				
	Conversation								
	INVITE(CPG, sendonly)	→		→	CPG(hold)				
	200 OK INVITE(recvonly)	+							
	ACK	→							
	INFO(FAC)	→		→	FAC(call transfer active, CTNb)				
	200 OK INFO	+							
	INVITE(sendrecv)	→							
	200 OK INVITE(sendrecv) ACK	←							
	DVE(DEL)	1			DEL				
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	←		←	RLC				

TP412021	SIP reference: RFC 3261 [4	4]	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 values	CPG: Event indicator=progress, Ge FAC: Generic notification=call trans			number-national (PIXIT)			
Comments	ISUP	lei active	SUT	SIP-I			
Comments	IAM	→		► INVITE(IAM)			
	ACM	-		180 Ringing(ACM)			
	ANM	+		200 OK INVITE(ANM)			
	ANIVI		Conversation	200 OK INVITE(ANIVI)			
	CPG(hold)	→		► INVITE(CPG, sendonly)			
	Of O(floid)			200 OK INVITE(recvonly)			
				ACK			
	FAC(call transfer active, CTNb)	→	-	► INFO(FAC)			
			•				
			-	=(00::0:00)			
			•				
			-	→ ACK			
	DEL		—	DVE(DEL)			
	REL	→	-	()			
	RLC	←	•	200 OK BYE(RLC)			

TP412022	SIP reference: RFC 3261 [4	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	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/ISUP; FAC encapsulated in the MIME body						
values	71 11	•	•					
ISUP parameter	FAC: Generic notification=call trans	fer active	e, Call transfe	r nu	ımber (PIXIT)			
values	FAC: ATP contained the connected	sub add	ress					
Comments	ISUP		SUT		SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	+		←	180 Ringing(ACM)			
	ANM	+		←	200 OK INVITE(ANM)			
		С	onversation					
	FAC(call transfer active, CTNb)	→		→	INFO(FAC)			
				←	200 OK INFO			
	FAC(ATP=SUB)	+		+	INFO(FAC)			
				→	200 OK INFO			
	FAC(ATP=SUB)	→		→	INFO(FAC)			
				(200 OK INFO			
	REL	→		→	BYE(REL)			
	RLC	+		(200 OK BYE(RLC)			

5.3.13 3PTY

TP413001	SIP reference: RFC 3261 [4]			SUP reference: Q.1912.5 [1],			
		Q.73	34.2	[i.8], clauses 2.4; 2.2.1			
TSS reference	ISUP-SIP-ISUP/SS/3PTY						
SIP selection							
criteria							
ISUP selection							
criteria	O I ODTV						
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; CP	G encapsulate	d in	the MIME body			
values	, , , , , , , , , , , , , , , , , , , ,						
ISUP parameter	CPG: Event indicator=progress, Generic no	otification=holo	1				
values	CPG: Event indicator=progress, Generic no	otification=con	ferer	nce established			
Comments	ISUP	SUT		SIP-I			
	IAM →		→	INVITE(IAM)			
	ACM ←		+	180 Ringing(ACM)			
	ANM ←		+	200 OK INVITE(ANM)			
		Conversation					
	CPG(hold) →		→	INVITE(CPG, sendonly)			
			+	200 OK INVITE(recvonly)			
			→	ACK			
	CPG(conference established) →		→	INVITE(CPG, sendrecv)			
			←	200 OK INVITE(sendrecv)			
			→	ACK			
		Conversation					
	REL →		→	BYE(REL)			
	RLC +		←	200 OK BYE(RLC)			

TP413002	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.734.2 [i.8], clauses 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/ISU	P; CPG en	capsu	lated in the MIME body			
values				•	·			
ISUP parameter	CPG: Event indicator=progre							
values	CPG: Event indicator=progre	ss, Gen		tion=				
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	←		←	ANM			
			onversation	_				
	INVITE(CPG, sendonly)	→		→	CPG(hold)			
	200 OK INVITE(recvonly)	(
	ACK	→						
	INVITE(CPG, sendrecv)	→		→	CPG(conference established)			
	200 OK INVITE(sendrecv)	+						
	ACK	→						
		C	onversation					
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP413003	SIP reference: RFC 3261 [4]		ISUP reference:
			Q.1912.5 [1],
		Q.734.2	[i.8], clauses 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 ca (remote active user) to a three-way conversa accordingly. The IUT should send a CPG message with th "conference established" to both implied part be set to "progress": 1. setup a call to user B; 2. establish a conference. O-MGCF interworking.	ition, and notify the generic notif	he implied remote party ication indicator set to
SIP parameter values	INFO: Content-Type: application/ISUP; CPG	encapsulated in	the MIME body
ISUP parameter values	CPG: Event indicator=progress, Generic noti	fication=confere	nce established
Comments	ISUP	SUT	SIP-I
	IAM →	→	INVITE(IAM)
	ACM ←	+	180 Ringing(ACM)
	ANM ←	+	200 OK INVITE(ANM)
	C	onversation	
	CPG(conference established) →	→	INFO(CPG)
		+	200 OK INFO
	C	onversation	
	REL →	→	BYE(REL)
	RLC ←	+	200 OK BYE(RLC)

TP413004	SIP reference: RF	C 3261 [4]	(ISUP reference: Q.1912.5 [1], Q.734.2 [i.8], clauses 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 enc	apsı	ulated in the MIME body		
ISUP parameter values	CPG: Event indicator=prog	gress, Ge	eneric notificat	ion=	conference established		
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		→	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
		(Conversation				
	INFO(CPG)	→		→	CPG(conference established)		
	200 OK INFO	+					
	Conversation						
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	+		←	RLC		

TP413005	SIP reference: RFC 3261 [4]	ISUP reference:							
		Q.1912.5 [1],							
			Q.73	4.2	[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 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/ISUI	P; CPG	encapsulate	ed in	the MIME body				
ISUP parameter	CPG 1, 4: Event indicator=progress, (
values	CPG 5: Event indicator=progress, Ge								
		CPG 2: Event indicator=progress, Generic notification=conference established							
_	CPG 3: Event indicator=progress, Ge	<u>neric n</u>		onfe					
Comments	ISUP		SUT		SIP-I				
	IAM	→		→	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	← 200 OK INVITE(ANM)							
	Conversation								
	CPG 1(hold)	→		^	INVITE(CPG, sendonly)				
				+	200 OK INVITE(recvonly)				
				→	ACK				
	CPG 2(conference established)	→		→	INVITE(CPG, sendrecv)				
				←	200 OK INVITE(sendrecv)				
				→	ACK				
	CPG 3(conference disconnected)	→		→	INFO(CPG)				
	CFG 3(contenence disconnected)	+		+	200 OK INFO				
					200 OK INFO				
	CPG 4(hold)	→		→	INVITE(CPG, sendonly)				
	CFG 4(floid)	+		/	200 OK INVITE(recvonly)				
				1	ACK				
		+			/ COR				
	CPG 5(retrieve)	→		→	INVITE(CPG, sendrecv)				
	5. 5 5(16th 16v6)	+*		′	200 OK INVITE(sendrecv)				
				<u>`</u>	ACK				
	Conversation								
	CPG 6(conference established)	→	2.110.000.011	→	INFO(CPG)				
		†-		+	200 OK INFO				
		1							
		(Conversation						
	REL	→	- CATTO TO GALLOTT	→	BYE(REL)				
	RLC	/		+	200 OK BYE(RLC)				
	INCO			_	200 ON DIL(NLO)				

TP413006	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 a 3PTY call can successfully create private communication with the active-held user. The appropriate notification received in a INVITE/INFO (CPG) and is sent in CPG messages to the ISUP. I-MGCF interworking.							
SIP parameter	INFO: Content-Type: applica	tion/l	SUP: CPG e	encansi	lated in the MIME body			
values	S. Sement Type. applied		23. , 3. 3 0	Подрос	Jody			
ISUP parameter	CPG: Event indicator=progre	ess. C	Seneric notifi	cation=	hold			
values	CPG: Event indicator=progre							
	CPG: Event indicator=progress, Generic notification=conference established							
	CPG: Event indicator=progress, Generic notification=conference disconnected							
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	Conversation							
	INVITE(CPG, sendonly)	→		→	CPG(hold)			
	200 OK INVITE(recvonly)	+						
	ACK	→						
	INVITE(CPG, sendrecv)	→		→	CPG(conference established)			
	200 OK INVITE(sendrecv)	+						
	ACK	→						
	INFO(CPG)	→		→	CPG(conference disconnected)			
	200 OK INFO	+						
	INVITE(CPG, sendonly)	→		→	CPG(hold)			
	200 OK INVITE(recvonly)	+						
	ACK	→						
		Ť						
	INVITE(CPG, sendrecv)	→		→	CPG(retrieve)			
	200 OK INVITE(sendrecv)	+						
	ACK	→						
	-	1	Conversation	on				
	INFO(CPG)	→	2	<u>→</u>	CPG(conference established)			
	200 OK INFO	+		<u> </u>				
		Ť	Conversation	on				
	BYE(REL)	→	Jonversan		REL			
	200 OK BYE(RLC)	+		-	RLC			

TP413007	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 con Verify that the IUT (controlling the co- private communication with the activ- messages to the user. O-MGCF interworking.	onferenc	e) on a 3PTY o	all can successfully create				
SIP parameter values	INFO: Content-Type: application/ISU	JP; CPG	encapsulated	in the MIME body				
ISUP parameter	CPG: Event indicator=progress, Generic notification=conference established							
values	CPG: Event indicator=progress, Ger	neric not	ification=confer	ence disconnected				
Comments	ISUP		SUT	SIP-I				
	IAM	→	-					
	ACM	←	•	10011119119(71011)				
	ANM	←	€	200 OK INVITE(ANM)				
			Conversation					
	CPG(conference established)	→	-					
			•	200 OK INFO				
	CPG(conference disconnected)	→	-	5 (5. 5)				
			€	200 OK INFO				
			Conversation					
	CPG(conference established)	→	7					
			€	200 OK INFO				
			Conversation					
	REL	→	-	- · - \· · · = -/				
	RLC	←	€	200 OK BYE(RLC)				

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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 values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body						
ISUP parameter values	CPG: Event indicator=prog						
	CPG: Event indicator=prog	jiess, G	1	cation=			
Comments	SIP-I	→	SUT		ISUP		
	INVITE(IAM)	7	1	→	IAM		
	180 Ringing(ACM)	-		-	ACM		
	200 OK INVITE(ANM)	-		←	ANM		
	W. (2.7.0)		Conversation				
	INFO(CPG)	→		→	CPG(conference established)		
	200 OK INFO	+					
	INFO(CPG)	→		→	CPG(conference disconnected)		
	200 OK INFO	+					
	INFO(CPG)	→		→	CPG(conference established)		
	200 OK INFO	←					
			Conversation	on			
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	←		←	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	encapsulate	d in	the MIME body		
values	урагарриания,			-			
ISUP parameter	CPG: Event indicator=progress, Generic	c not	fication=con	fere	nce established		
values	CPG: Event indicator=progress, Generic						
Comments	ISUP		SUT		SIP-I		
	IAM	→		→	INVITE(IAM)		
	ACM	+		←	180 Ringing(ACM)		
	ANM	+		←	200 OK INVITE(ANM)		
		C	onversation				
	CPG(conference established)			→	INFO(CPG)		
				←	200 OK INFO		
	CPG(conference disconnected)	→		→	INFO(CPG)		
				+	200 OK INFO		
		C	onversation				
	REL	←		→	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

TP413010	SIP reference: RFC	3261	[4]	Q	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: applic	ation/I	SUP; CPG end	capsu	lated in the MIME body		
ISUP parameter	CPG: Event indicator=progr	ess, G	eneric notifica	tion=	conference established		
values	CPG: Event indicator=progr	ess, G	eneric notifica	tion=	conference disconnected		
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		→	IAM		
	180 Ringing(ACM)	←		←	ACM		
	200 OK INVITE(ANM)	←		←	ANM		
			Conversation				
	INFO(CPG)	→		→	CPG(conference established)		
	200 OK INFO	+					
	INFO(CPG)	→		→	CPG(conference disconnected)		
	200 OK INFO	+					
			Conversation				
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP413011	SIP reference: RFC 3261 [4]			IS	SUP reference: Q.1912.5 [1],		
			0.734	2 [i	.8], clause 2.5.2.1.1.3 b		
TSS reference	ISUP-SIP-ISUP/SS/3PTY		Q., 04.	<u> [.</u>	.0], 014400 210.211110 5		
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-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". O-MGCF interworking.						
SIP parameter values	INFO: Content-Type: application/ISUF	; CPG	encapsulated	l in t	the MIME body		
ISUP parameter	CPG: Event indicator=progress, Gene						
values	CPG: Event indicator=progress, Gene	ric not	ification=confe				
	CPG: Event indicator=progress, Gene	ric not					
Comments	ISUP		SUT		SIP-I		
	IAM	→			INVITE(IAM)		
	ACM	←			180 Ringing(ACM)		
	ANM	+		(200 OK INVITE(ANM)		
	0004 10		Conversation		W N ((TT) (ODO)		
	CPG(hold)	→			INVITE(CPG, sendonly)		
					200 OK INVITE(recvonly)		
			-	}	ACK		
		→		•	INIVITE (CDC condrag)		
	CPG(conference established)	7			INVITE(CPG, sendrecv) 200 OK INVITE(sendrecv)		
					ACK		
		1	-	7	ACK		
	CPG(conference disconnected)	→	 	>	INFO(CPG)		
	Of O(contenence discontracted)	+*			200 OK INFO		
				_	200 0.1111 0		
	CPG(hold)	→	-	}	INVITE(CPG, sendonly)		
					200 OK INVITE(recvonly)		
			-		ACK		
	REL	→	-	→	BYE(REL)		
	RLC	+	•		200 OK BYE(RLC)		

TP413012	SIP reference: RFC	3261	[4]	ISUP reference:				
11 410012					Q.1912.5 [1],			
				Q.	.734.2 [i.8], clause 2.5.2.1.1.3 b			
TSS reference	ISUP-SIP-ISUP/SS/3PTY				<u> </u>			
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-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: applica	tion/I	SLID: CDC (ncancu	lated in the MIME body			
values	in O. Content-Type, applica	111011/14	301 , CF G 6	ricapsu	nated in the Milvic body			
ISUP parameter	CPG: Event indicator=progre	288 G	eneric notifi	cation-l	hold			
values	CPG: Event indicator=progre							
Value	CPG: Event indicator=progre							
Comments	SIP-I	1	SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	Conversation							
	INVITE(CPG, sendonly)	→	1	→	CPG(hold)			
	200 OK INVITE(recvonly)	+						
	ACK	→						
	7.0.1							
	INVITE(CPG, sendrecv)	→		→	CPG(conference established)			
	200 OK INVITE(sendrecv)	+						
	ACK	→						
	_							
	INFO(CPG)	→		→	CPG(conference disconnected)			
	200 OK INFO	+			,			
	INVITE(CPG, sendonly)	→		→	CPG(hold)			
	200 OK INVITE(recvonly)	+			, ,			
	ACK	→						
			•	•				
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		+	RLC			

5.3.14 User-to-user service

5.3.14.1 User-to-user service 1

TP414001	SIP reference: RFC 32	61 [4]		ISUP reference: 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 implicit request: User Ensure that the SUT can succe the encapsulated IAM. O-MGC	essfully transfer		
SIP parameter values	INVITE: Content-Type: applicate parameter encapsulated in the		containing the	user-to-user information
ISUP parameter values	IAM: User-to-user information p			
Comments	ISUP	SUT	Г	SIP-I
	IAM -	•	→	INVITE(IAM)
	ACM •		+	180 Ringing(ACM)
	ANM		+	200 OK INVITE(ANM)
		Convers	ation	
	REL -	•	→	BYE(REL)
	RLC •		+	200 OK BYE(RLC)

TP414002	SIP reference: RF	C 3261	[4]	=	SUP reference:], clauses A.1.1, 1.1.5.2.3 and 4.
					Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS1		Į.		
SIP selection criteria					
ISUP selection criteria					
Test purpose	Service 1 implicit request: Ensure that the SUT can sithe encapsulated IAM. I-Mo	uccessf	ully transfer th		//TE er service 1 implicit request in
SIP parameter	INVITE: Content-Type: app	olication	/ISUP; IAM co	ntaining the	user-to-user information
values	parameter encapsulated in	the MIN	ЛE body		
ISUP parameter values	IAM: User-to-user informat	ion para	ımeter		
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Conversati	on	
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		+	RLC

TP414003	SIP reference: RFC	3261			SUP reference: I], 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		Ensure that the SUT can successfully transfer the User-to-user service 1 explicit request not essential in the encapsulated IAM. O-MGCF interworking.						
SIP parameter values	INVITE: Content-Type: appl parameter encapsulated in			taining the	user-to-user indicator			
ISUP parameter	IAM: User-to-user information	on para	meter, User-to-	user indica	tor = service 1 explicit			
values	request	-			·			
Comments	ISUP		SUT		SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
			Conversation	1				
	REL	→		→	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP414004	SIP reference: RFC	3261	[4]	Q.191	-	SUP reference:], clauses A.1.1, 1.1.5.2.3 and 4, Q.737 [i.10]				
TSS reference	ISUP-SIP-ISUP/SS/UUS1	ISUP-SIP-ISUP/SS/UUS1								
SIP selection criteria										
ISUP selection criteria										
Test purpose	Service 1 explicit request: User-to-user indicator in the INVITE Ensure that the SUT can successfully transfer the User-to-user service 1 explicit request essential received in the IAM. O-MGCF interworking.									
SIP parameter values	INVITE: Content-Type: appl parameter encapsulated in 180 Ringing: Content-Type: parameter encapsulated in	the MIN applica	/IE body ation/ISUP;		_					
ISUP parameter	IAM: User-to-user information			r-to-user	indica	tor				
values	ACM: User-to-user indicator									
Comments	ISUP		SU	Γ		SIP-I				
	IAM	→			→	INVITE(IAM)				
	ACM	+			+	180 Ringing(ACM)				
	ANM	←			+	200 OK INVITE(ANM)				
			Convers	ation						
	REL	→			→	BYE(REL)				
	RLC	+			+	200 OK BYE(RLC)				

TP414005	SIP reference: RF	C 3261	[4]	-	SUP reference: I], 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: Ensure that the SUT can sessential received in the er	uccessf	ully transfer th	e User-to-us	er service 1 explicit request
SIP parameter values	INVITE: Content-Type: app parameter encapsulated in	the MIN	/ISUP; IAM co /IE body ation/ISUP; A(ntaining the	
ISUP parameter values	IAM: User-to-user informat ACM: User-to-user indicato				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Conversati	on	
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP414006	SIP reference: RFC	3261	[4]	Q.191		SUP reference: I], clauses A.1.1, 1.1.5.2.3 and 4, Q.737 [i.10]				
TSS reference	ISUP-SIP-ISUP/SS/UUS1	ISUP-SIP-ISUP/SS/UUS1								
SIP selection criteria										
ISUP selection criteria										
Test purpose	Ensure that the SUT can su in the encapsulated ACM. O-MGCF interworking.									
SIP parameter values	Service 1 implicit response: INVITE: Content-Type: appl parameter encapsulated in t 180 Ringing: Content-Type: parameter encapsulated in t	ication/ the MIN applica	ISUP; IAM IE body ation/ISUP;	containin	g the					
ISUP parameter	IAM: User-to-user information		•							
values	ACM: User-to-user informat	ion par	ameter							
Comments	ISUP		SUT			SIP-I				
	IAM	→			→	INVITE(IAM)				
	ACM	←			+	180 Ringing(ACM)				
	ANM	←			←	200 OK INVITE(ANM)				
			Conversa	ation	•					
	REL	→			→	BYE(REL)				
	RLC	←			←	200 OK BYE(RLC)				

TP414007	SIP reference: RF	C 3261	[4]		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				Q.737 [1.10]
SIP selection	1301 -311 -1301 /33/0031				
criteria					
ISUP selection criteria					
Test purpose	Service 1 implicit response	: User-to	o-user inform	ation in the A	ACM
	Ensure that the SUT can s	uccessfu	ully transfer th	ne User-to-us	ser service 1 implicit response
	in the encapsulated ACM.		,		ээ ээг гээр г нирион гээр эггээ
	I-MGCF interworking.				
SIP parameter	INVITE: Content-Type: app	olication	/ISUP; IAM c	ontaining the	user-to-user information
values	parameter encapsulated in				
				CM containir	ng the user-to-user information
	parameter encapsulated in				
ISUP parameter	IAM: User-to-user informat				
values	ACM: User-to-user informa	tion par			
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		←	ACM
	200 OK INVITE(ANM)	+		←	ANM
			Conversa	tion	
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		+	RLC

TP414008	SIP reference: RFC	3261	[4]	Q.191	-	SUP reference:], clauses A.1.1, 1.1.5.2.3 and 4, Q.737 [i.10]				
TSS reference	ISUP-SIP-ISUP/SS/UUS1	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 180 Ringing: Content-Type: parameter encapsulated in	the MIN	/IE body ation/ISUP;							
ISUP parameter	IAM: User-to-user information			r-to-user	indica	tor set to service 1 request				
values	ACM: User-to-user indicator	r set to	service 1 n	ot suppor	ted re	sponse				
Comments	ISUP		SU			SIP-I				
	IAM	→			→	INVITE(IAM)				
	ACM	+			+	180 Ringing(ACM)				
	ANM	+			+	200 OK INVITE(ANM)				
			Convers	ation						
	REL	→			→	BYE(REL)				
	RLC	+			+	200 OK BYE(RLC)				

TP414009	SIP reference: RFC	3261	[4]	Q.191	-	SUP reference:], 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 response service 1 not supported in the ACM Ensure that the SUT can successfully transfer the User-to-user service 1 explicit response not supported in the ACM. I-MGCF interworking.						
SIP parameter	INVITE: Content-Type: appl	ication	ISUP; IAM	containin	g the	user-to-user information	
values	parameter encapsulated in the 180 Ringing: Content-Type: parameter encapsulated in the 180 Ringing:	applica	ation/ISUP;	ACM cor	ntainin	g the user-to-user indicator	
ISUP parameter	IAM: User-to-user information			r-to-user	indica	tor set to service 1 request	
values	ACM: User-to-user indicator	set to	service 1 n	ot suppor	ted re	sponse	
Comments	SIP-I		SU	Γ		ISUP	
	INVITE(IAM)	1			→	IAM	
	180 Ringing(ACM)	4			+	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
			Convers	ation			
	BYE(REL)	↑			→	REL	
	200 OK BYE(RLC)	+		· ·	+	RLC	

TP414010	SIP reference: RFC	3261	[4]	Q.191		SUP reference:], 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	Ensure that the SUT can sunetwork in the encapsulated O-MGCF interworking.		ully transfer	the User	-to-use	er service 1 discarded by the
SIP parameter values	INVITE: Content-Type: appl parameter encapsulated in 180 Ringing: Content-Type: parameter encapsulated in	the MIN applica	ME body ation/ISUP;		•	
ISUP parameter	IAM: User-to-user information					
values	ACM: User-to-user indicator	r set to	discarded b	y the net	work r	esponse
Comments	ISUP		SUT			SIP-I
	IAM	→			→	INVITE(IAM)
	ACM	+			4	180 Ringing(ACM)
	ANM	+			+	200 OK INVITE(ANM)
			Conversa	ation		
	REL	→			↑	BYE(REL)
	RLC	+			+	200 OK BYE(RLC)

TP414011	SIP reference: RFC	3261	[4]	Q.191	-	SUP reference:], 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	Ensure that the SUT can su network in the encapsulated I-MGCF interworking.		ully transfer	the User-	-to-us	er service 1 discarded by the
SIP parameter values	INVITE: Content-Type: appl parameter encapsulated in 180 Ringing: Content-Type: parameter encapsulated in	the MIN applica	ME body ation/ISUP;		-	
ISUP parameter	IAM: User-to-user information	on para	meter			
values	ACM: User-to-user indicator	set to	discarded b	y the net	work r	response
Comments	SIP-I		SUT			ISUP
	INVITE(IAM)	→			→	IAM
	180 Ringing(ACM)	+			+	ACM
	200 OK INVITE(ANM)	+			+	ANM
			Convers	ation		
	BYE(REL)	→			→	REL
	200 OK BYE(RLC)	+			+	RLC

5.3.14.2 User-to-user service 2

TP414101	SIP reference: RFC	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				Q.737 [1.10]		
SIP selection	130F-31F-130F/33/0032						
criteria							
ISUP selection							
criteria							
Test purpose	Service 2 request not esser	ntial tran	sferred in th	ne INVITE			
	Ensure that the SUT can su	ıccessfi	ılly transfer t	the User-to-us	ser service 2 explicit request		
	and User-to user informatio						
	information is sent in a USF						
	O-MGCF interworking.		igo onoapoo	iatod iii dir ii v	. o requees.		
SIP parameter	INVITE: Content-Type: app	lication/	ISUP; IAM o	containing the	user-to-user indicator and		
values	User-to-user information en						
	INFO: Content-Type: applic	ation/IS	UP; USR co	ontaining the	Jser-to-user information		
	parameter encapsulated in	the MIN	1E body	_			
ISUP parameter	IAM: User-to-user information	on para	meter, User	to-user indication	ator		
values	USR: User-to-user informat	ion					
Comments	ISUP		SUT		SIP-I		
	IAM	→		→	INVITE(IAM)		
	ACM	←		+	180 Ringing(ACM)		
	ANM	←		+	200 OK INVITE(ANM)		
			Conversa	ıtion			
	USR	→		→	INFO(USR)		
				+	200 OK INFO		
	USR ← INFO(USR)						
				<u> </u>	200 OK INFO		
					200 010 1141 0		
	REL	→		→	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

TP414102	SIP reference: RI	FC 3261 [4	4]		ISUP reference: [1], clauses A.1.1, 1.2.5.2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS2	<u>)</u>			
SIP selection criteria					
ISUP selection criteria					
Test purpose	the encapsulated IAM. An encapsulated in an INFO O-MGCF interworking.	successfu additiona request.	lly transfer all User-to-us	the User-to-u ser informatio	ser service 2 explicit request in n is sent in a USR message
SIP parameter values	INVITE: Content-Type: apencapsulated in the MIME		ISUP; IAM o	containing the	user-to-user indicator
, and a	INFO: Content-Type: appl parameter encapsulated i	lication/IS		ontaining the	User-to-user information
ISUP parameter values	IAM: User-to-user informatus: User-to-user information		meter, User	-to-user indic	ator
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Conversa	ation	
	INFO(USR)	→		→	USR
	200 OK INFO	+			
	INFO(USR)	+		+	USR
	200 OK INFO	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		+	RLC

TP414103	SIP reference: RFC	3261	[4]	Q.191	-	SUP reference:], 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	Service 2 response not provided transferred in the INVITE Ensure that the SUT can successfully transfer the User-to-user service 2 explicit response not supported in the encapsulated ACM. I-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;		_		
ISUP parameter	IAM: User-to-user information			r-to-user	indica	tor set to service 2 request	
values	ACM: User-to-user indicator						
Comments	ISUP		SU	Γ		SIP-I	
	IAM	→			→	INVITE(IAM)	
	ACM	+			+	180 Ringing(ACM)	
	ANM	+			+	200 OK INVITE(ANM)	
			Convers	ation			
	REL	→			→	BYE(REL)	
	RLC	+			+	200 OK BYE(RLC)	

TP414104	SIP reference: RF	C 3261	[4]	=	SUP reference: I], 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 so not provided in the encaps	successfu	ully transfer	the User-to-us	er service 2 explicit response
SIP parameter	INVITE: Content-Type: ap	plication/	ISUP; IAM	containing the	user-to-user indicator
values	parameter encapsulated in			•	
ISUP parameter values	IAM: User-to-user informa	tion para	meter, User	-to-user indica	itor
Comments	SIP-I		SUT	•	ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Conversa	ation	
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		+	RLC

5.3.14.3 User-to-user service 3

TP414201	SIP reference: RFC	3261	[4]		SUP reference: 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		tional l			ser service 3 explicit request in sent in several USR message
SIP parameter	INVITE: Content-Type: appl	ication	/ISUP; IAM	containing the	user-to-user indicator
values	encapsulated in the MIME b			-	
	INFO: Content-Type: application			ontaining the l	Jser-to-user information
	parameter encapsulated in t				
ISUP parameter	IAM: User-to-user information		meter, Use	r-to-user indica	ator
values	USR: User-to-user informati	ion	1		_
Comments	ISUP		SU		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	+		←	180 Ringing(ACM)
	ANM	+		←	200 OK INVITE(ANM)
			Convers		
	USR	→		→	INFO(USR)
				←	200 OK INFO
	USR	+		+	INFO(USR)
				→	200 OK INFO
	USR	+		+	INFO(USR)
				→	200 OK INFO
	USR	→		→	INFO(USR)
				+	200 OK INFO
			_		
	REL	→		→	BYE(REL)
	RLC	+	_	+	200 OK BYE(RLC)

TP414202	SIP reference: RI	FC 3261 [4]	Q.1912.	ISUP reference: 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		Iditional U			user service 3 explicit request is sent in several USR messa			
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	IAM: User-to-user informa		neter, Use	r-to-user ind	icator			
values	USR: User-to-user inform	ation						
Comments	SIP-I		SU		ISUP			
	INVITE(IAM)	→		-	11. 11.11			
	180 Ringing(ACM)	+		•				
	200 OK INVITE(ANM)	+		- €	- ANM			
			Convers					
	INFO(USR)	→		-	USR			
	200 OK INFO	+						
	INFO(USR)	+		•	- USR			
	200 OK INFO	→						
	INFO(USR)	+		•	- USR			
	200 OK INFO	→						
	INFO(USR)	→		-	USR			
	200 OK INFO	+						
	BYE(REL)	→		-				
	200 OK BYE(RLC)	+		•	- RLC			

TP414203	SIP reference	: RFC 3261	[4]		ISUP reference: 1], clauses A.1.1, 1.3.5.2.3 and 4, Q.737 [i.10]			
TSS reference	ISUP-SIP-ISUP/SS/U	US3						
SIP selection criteria								
ISUP selection criteria								
Test purpose	confirmed state can s several encapsulated O-MGCF interworking	uccessful pro USR messaç ı.	ceeded. The ges.	e User-to-use	n an INFO request during the rinformation is passed on in			
SIP parameter values	encapsulated in the M INFO: Content-Type: encapsulated in the M INFO: Content-Type: parameter encapsulat	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 ind FAA: User-to-user ind USR: User-to-user inf	licator service	e 3 request i					
Comments	ISUP IAM ACM ANM	→ ← ←	SUT	→ ← ←	SIP-I INVITE(IAM) 180 Ringing(ACM) 200 OK INVITE(ANM)			
	FAR	→	Conversa	ation → ←	INFO(FAR) 200 OK INFO			
	FAA	+		←	INFO(FAA) 200 OK INFO			
	USR	→		→ ←	INFO(USR) 200 OK INFO			
	USR	+		←	INFO(USR) 200 OK INFO			
	USR	+		←	INFO(USR) 200 OK INFO			
	USR	→		→ ←	INFO(USR) 200 OK INFO			
	REL RLC	→		→	BYE(REL) 200 OK BYE(RLC)			

TP414204	SIP reference: RI	FC 3261	[4]	Q.1912.5	ISUP reference: [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	confirmed state can succe several encapsulated USI I-MGCF interworking.	essful pro R messaç	ceeded. Th ges.	e User-to-use	n an INFO request during the r information is passed on in			
SIP parameter values	encapsulated in the MIME INFO: Content-Type: applencapsulated 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 indicat	FAR: User-to-user indicator service 3 request not essential FAA: User-to-user indicator service 3 response provided						
Comments	SIP-I INVITE(IAM) 180 Ringing(ACM) 200 OK INVITE(ANM)	→ ← ←	SU	+ + +	ISUP IAM ACM ANM			
	INFO(FAR) 200 OK INFO	→	Convers	ation ->	FAR			
	INFO(FAA) 200 OK INFO	←		+	FAA			
	INFO(USR) 200 OK INFO	→		→	USR			
	INFO(USR) 200 OK INFO	←		+	USR			
	INFO(USR) 200 OK INFO	←		+	USR			
	INFO(USR) 200 OK INFO	→		→	USR			
	BYE(REL) 200 OK BYE(RLC)	→		→	REL RLC			

TP414205	SIP reference: RFC	3261	[4]	-	SUP reference: 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-to-us	er service 3 explicit response
SIP parameter values	INVITE: Content-Type: appl parameter encapsulated in t 200 OK INVITE: Content-Ty indicator parameter encapsi	the MIN pe: ap	/IE body plication/ISU	P; ANM conta	
ISUP parameter	IAM: User-to-user indicator				
values	ANM: User-to-user indicator	set to	service 3 pro	vided respon	se
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	4		+	200 OK INVITE(ANM)
			Conversa	tion	
	REL	^		→	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP414206	SIP reference: RFC	3261	[4]	-	SUP reference: 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	the User-to-us	ser service 3 explicit response
SIP parameter values	INVITE: Content-Type: appl parameter encapsulated in t 200 OK INVITE: Content-Ty indicator parameter encapsi	the MIN pe: ap	/IE body plication/ISU	IP; ANM conta	
ISUP parameter	IAM: User-to-user indicator	set to s	service 3 rec	uest	
values	ANM: User-to-user indicator	set to	service 3 pr	ovided respon	se
Comments	SIP-I		SUT	•	ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Conversa	ation	
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		+	RLC

TP414207	SIP reference: RF	FC 3261	[4]		ISUP reference: 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 a User-to-use confirmed state can succe				n an INFO request during the request is rejected.		
SIP parameter values	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; FRJ containing the user-to-user indicator					
ISUP parameter	FAR: User-to-user indicate		e 3 request r	not essential			
values	FRJ: User-to-user indicate	or service	3 response	not provided			
Comments	ISUP		SUT	•	SIP-I		
	IAM	→		→	INVITE(IAM)		
	ACM	(+	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
			Conversa	ation			
	FAR	→		→	INFO(FAR)		
				+	200 OK INFO		
	FRJ	+		+	INFO(FRJ)		
				→	200 OK INFO		
	REL	→		→	BYE(REL)		
	RLC	←		←	200 OK BYE(RLC)		

TP414208	SIP reference: RI	FC 3261	[4]	•	SUP reference: 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	confirmed state can succe	essful pro	ceeded. The	e user to user	
SIP parameter	INFO: Content-Type: app		SUP; FAR co	ontaining the u	ser-to-user indicator
values	encapsulated in the MIME				
	INFO: Content-Type: app		SUP; FRJ co	ntaining the u	ser-to-user indicator
	encapsulated in the MIME				
ISUP parameter	FAR: User-to-user indicat				
values	FRJ: User-to-user indicate	or service			liaiin
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Conversa	ation	
	INFO(FAR)	→		→	FAR
	200 OK INFO	+			
	INFO(FRJ)	+		+	FRJ
	200 OK INFO	→			
	BYE(REL)	→		→	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/Sendi	ng_of th	e_SETUF	_messag	ge		
SIP selection criteria	NOT PICS 4/4 AND NOT PIC	CS 4/5					
ISDN selection criteria	NOT 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 μ-law (PCMU) 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 values	SIP INVITE: Audio RTP/AV 200 OK: Audio RTP/AV						
ISDN parameter values	SETUP BC: A-law						
Comments	SIP-I		SU	Т		ISDN	
	INVITE(IAM)	→			→	SETUP	
	180 Ringing(ACM)	+			+	ALERTING	
	200 OK INVITE(ANM)	+			←	CONNECT	
	ACK	→					
	Conversation						
	BYE(REL)	→			→	DISCONNECT	
	200 OK BYE(RLC)	+			+	RELEASE	
					→	RELEASE COMPLETE	

TP501002	SIP reference: RFC 3	3261 [4	1			ISDN reference:			
		-	-	Q.	1912	2.5 [1], clause 6.1.2 (i,2b)			
TSS reference	SIP-I-ISDN/Basic_call/Sending_of the_SETUP_message								
SIP selection	PICS 4/4 AND PICS 4/5								
criteria									
ISDN selection	NOT PICS 1/6								
criteria									
Test purpose	 offer and 100rel extensions a including a SDP for A-law (Petalon the SUT shall delete μ-las send back in the SDP and the SETUP shall be defer constructed from the ISU. The bearer path shall be consare satisfied: the I-IWU determines (Beautine proceed (if applicable). In addition, if BICC is perform Outgoing bearer set-up proceed and the I-IWU determines (the preconditions have been metalon the I-IWU determines (the IIWU determines) 	and preceded and p	conditions and µ-law MU), if pretil all precor USI. in both direction the Control of the Control	extension (PCMU): sent, from the conditions where the conditions whenever sent the conditions whenever sent the conditions whenever the conditions where the conditions	ons in the shave when ed in for set-up pe is the ined	e media description that it will we been met. The BC is a both of the following conditions RFC 3312 [5]) that sufficient ession establishment to			
SIP parameter	SIP INVITE: Audio RTP/AVI								
values ISDN parameter	200 OK: Audio RTP/AVI SETUP BC: A-law	P 8							
values	SETUP BC. A-law								
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	→		•					
	183 session Progress	+							
	PRACK	→							
	200 OK PRACK	+							
	UPDATE	→			→	SETUP			
	200 OK UPDATE	+			<u> </u>				
	180 Ringing(ACM)	+			+	ALERTING			
	200 OK INVITE(ANM)	+			+	CONNECT			
	ACK	→			Ť				
			Convers	sation	1				
	BYE(REL)	→	23111311		→	DISCONNECT			
	200 OK BYE(RLC)	+			-	RELEASE			
		<u> </u>			<u>`</u>	RELEASE COMPLETE			
	. 1		<u> </u>		-				

TP501003	SIP reference: RFC 32	261 [4]			ISDN reference:		
	Q.1912.5 [1], clause 6.1.2 (i,1)							
TSS reference	SIP-I-ISDN/Basic_call/Sending	_of th	e_SETUF	_messa	ge			
SIP selection criteria	NOT PICS 4/4 AND NOT PICS	8 4/5						
ISDN selection criteria	NOT 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 μ-law (PCMU) 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 values	SETUP BC: A-law							
Comments	SIP-I		SU	ΙΤ		ISDN		
	INVITE(IAM)	→			→	SETUP		
	180 Ringing(ACM)	+			+	ALERTING		
	200 OK INVITE(ANM)	+			+	CONNECT		
	ACK	→						
	Conversation							
	BYE(REL)	→			→	DISCONNECT		
	200 OK BYE(RLC)	+			+	RELEASE		
				•	→	RELEASE COMPLETE		

TP501004	SIP reference: RFC	3261 [4]				ISDN reference:			
11 301004	on reference. Ki e	0201 [4]		Q.	1912	2.5 [1], clause 6.1.2 (i,2b)			
TSS reference	SIP-I-ISDN/Basic_call/Sending_of the_SETUP_message								
SIP selection	PICS 4/4 AND PICS 4/5								
criteria									
ISDN selection	NOT PICS 1/6								
criteria									
Test purpose	Ensure that if the SUT upon	receipt o	f the first	INVITE	with	sufficient digits, with an SDP			
	offer and 100rel extensions					n the SIP Require header			
	including a SDP for A-law (PCMA) ar	nd µ-law	(PCMU):					
			U), if pre	sent, froi	m the	e media description that it will			
	send back in the SDP a								
	 the SETUP shall be def 	ferred unt	il all prec	onditions	s hav	e been met. The BC is			
	constructed from the IS		or USI.						
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	JT		ISDN			
	INVITE(IAM)	→							
	183 session Progress	+							
	PRACK	→							
	200 OK PRACK	+							
	UPDATE	→			→	SETUP			
	200 OK UPDATE	+							
	180 Ringing(ACM)	+			←	ALERTING			
	200 OK INVITE(ANM)	←			←	CONNECT			
	ACK	→							
			Conver	sation					
	BYE(REL)	→			→	DISCONNECT			
	200 OK BYE(RLC)	+			←	RELEASE			
					→	RELEASE COMPLETE			

TP501005	SIP reference: RFC 32	261 [4]	ISDN reference:				
						2.5 [1], clause 6.1.2 (i,1)		
TSS reference	SIP-I-ISDN/Basic_call/Sending	_of th	e_SETUF	_messa	ge			
SIP selection	NOT PICS 4/4 AND NOT PICS	3 4/5						
criteria								
ISDN selection	PICS 1/6							
criteria								
Test purpose	Ensure that if the SUT upon re offer including a SDP for A-law the SUT shall delete A-law	v (PC	MA) and I	u-law (Po	CMU			
	the SDP answer;	•	,			C is constructed from the ISUP		
SIP parameter	SIP INVITE: Audio RTP/AVP	Λ <u>8</u>						
values	200 OK: Audio RTP/AVP							
ISDN parameter	SETUP BC: A-law	0						
values	02.01 Bo.71 law							
Comments	SIP-I		SL	JT		ISDN		
	INVITE(IAM)	→			→	SETUP		
	180 Ringing(ACM)	+			+	ALERTING		
	200 OK INVITE(ANM)	+			+	CONNECT		
	ACK	→						
	Conversation							
	BYE(REL)	→			→	DISCONNECT		
	200 OK BYE(RLC)	+			+	RELEASE		
					→	RELEASE COMPLETE		

TP501006	SIP reference: RFC 3261 [4]			ISDN reference: Q.1912.5 [1], clause 6.1.2 (i,2b)				
TSS reference	SIP-I-ISDN/Basic_call/Sending	_of the	SETUP	_message				
SIP selection	PICS 4/4 AND PICS 4/5							
criteria								
ISDN selection	PICS 1/6							
criteria				IND 0775 101	// L L L L L L L L L L L L L L L L L L			
Test purpose	 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): the SUT shall delete A-law (PCMA), if present, from the media description that it will send back in the SDP answer; the SETUP shall be deferred until all preconditions have been met. The BC is constructed from the ISUP TMR or USI. 							
SIP parameter	SIP INVITE: Audio RTP/AVP		0. 00					
values	200 OK: Audio RTP/AVP							
ISDN parameter	SETUP BC: A-law							
values								
Comments	SIP-I		SU	T	ISDN			
	INVITE(IAM)	→						
	183 session Progress	+						
	PRACK	→						
	200 OK PRACK	+						
	UPDATE	→		→	SETUP			
	200 OK UPDATE	+						
	180 Ringing(ACM)	+		(ALERTING			
	200 OK INVITE(ANM)	+		(CONNECT			
	ACK	→						
			Convers	sation				
	BYE(REL)	→		→	DISCONNECT			
	200 OK BYE(RLC)	+		(RELEASE			
				→	RELEASE COMPLETE			

TP501007	SIP reference: RFC 3261 [4]					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	NOT PICS 4/4 AND NOT PICS	4/5						
criteria								
ISDN selection	PICS 1/6							
criteria								
SIP parameter values ISDN parameter	 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 INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 8 SETUP BC: A-law 							
values	OID I		0.1	-	1	IODNI		
Comments	SIP-I	<u> </u>	SU	1	-	ISDN		
	INVITE(IAM)	→			→	SETUP		
	180 Ringing(ACM)	+				ALERTING		
	200 OK INVITE(ANM)	+			+	CONNECT		
	ACK	→						
			Convers	sation				
	BYE(REL)	→			→	DISCONNECT		
	200 OK BYE(RLC)	+			←	RELEASE		
			_	•	→	RELEASE COMPLETE		

TP501008	SIP reference: RFC 3261 [4]			ISDN reference: Q.1912.5 [1], clause 6.1.2 (i,2b)						
TSS reference	SIP-I-ISDN/Basic_call/Sending	SIP-I-ISDN/Basic_call/Sending_of the_SETUP_message								
SIP selection	PICS 4/4 AND PICS 4/5									
criteria										
ISDN selection	PICS 1/6	PICS 1/6								
criteria										
Test purpose	 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): the SUT shall delete A-law (PCMA), if present, from the media description that it will send back in the SDP answer; the SETUP shall be deferred until all preconditions have been met. The BC is constructed from the ISUP TMR or USI. 									
SIP parameter	INVITE: Audio RTP/AVP 0 8									
values	200 OK: Audio RTP/AVP 8									
ISDN parameter	SETUP: BC: A-law									
values										
Comments	SIP-I		SU	IT	ISDN					
	INVITE(IAM)	→								
	183 session Progress	+								
	PRACK	→								
	200 OK PRACK	+								
	UPDATE	→		→	SETUP					
	200 OK UPDATE	+								
	180 Ringing(ACM)	+		←	ALERTING					
	200 OK INVITE(ANM)	+		+	CONNECT					
	ACK	→								
			Conver	sation						
	BYE(REL)	→		→	DISCONNECT					
	200 OK BYE(RLC)	+		+	RELEASE					
				→	RELEASE COMPLETE					

TP501009	SIP reference: RFC 32	61 [4]		ISDN reference:				
			E	N 383	001 [2], clause 6.1.3.5.2.2			
TSS reference	SIP-I-ISDN/Basic_call/Sending_of the_SETUP_message							
SIP selection	NOT PICS 4/4 AND NOT PICS	4/5 AND I	PICS 1/9					
criteria								
ISDN selection								
criteria								
Test purpose	Ensure that the SUT on receipt							
	A-Law, then independent from	n the rece	ived order	of pre	ference.			
	Sends a SETUP message:							
	 the G.711 a-law codec shall 	be return	ed in the SE	P ans	wer as preferred codec.			
SIP parameter	Offer: m=audio 4711 RTP/A	4VP 0 8						
values	Answer: m=audio 4712 RTP/A	4VP 8 0						
ISDN parameter								
values								
Comments	SIP-I		SUT		ISDN			
	INVITE(IAM)	→		→	SETUP			
	180 Ringing(ACM)	+		+	ALERTING			
	200 OK INVITE(ANM)	+		+	CONNECT			
	ACK	→						
	Conversation							
	BYE(REL)	→		→	DISCONNECT			
	200 OK BYE(RLC)	+		+	RELEASE			
				→	RELEASE COMPLETE			

TP501010	SIP reference: RFC 3261 [4]			EN 383	ISDN reference: 001 [2], clause 6.1.3.5.2.2				
TSS reference	SIP-I-ISDN/Basic_call/Sending	g_of th	e_SETUF						
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								
					in the SIP Supported header,				
	then independent from the r								
	 the SETUP shall be defer 								
	 the G.711 a-law codec sh 	all be	eturned i	n the SDP ar	nswer as preferred codec.				
SIP parameter	Offer: m=audio 4711 RTP	,							
values	Answer: m=audio 4712 RTP	P/AVP 8	3 0						
ISDN parameter									
values		1			_				
Comments	SIP-I		SU	JT	ISDN				
	INVITE(IAM)	→							
	183 session Progress	←							
	PRACK	→							
	200 OK PRACK	←							
	UPDATE	→		→	SETUP				
	200 OK UPDATE	←							
	180 Ringing(ACM)	←		+	ALERTING				
	200 OK INVITE(ANM)	←		←	CONNECT				
	ACK	→							
			Conver	sation					
	BYE(REL)	→	_	→	DISCONNECT				
	200 OK BYE(RLC)	+		+	RELEASE				
				→	RELEASE COMPLETE				

TP501011	SIP reference: RFC 3261 [4]		=11.00	ISDN reference:					
					3 001 [2], clause 6.1.3.5.2.2				
TSS reference	SIP-I-ISDN/Basic_call/Send			P_message					
SIP selection	PICS 4/4 AND PICS 4/5 AN	ND PICS 1	/9						
criteria									
ISDN selection									
criteria									
Test purpose					h a SDP offer for μ-Law and				
					s in the SIP Require header,				
	then independent from th								
	 the SETUP shall be de 								
	 the G.711 a-law codec 	shall be r	eturned i	n the SDP a	nswer as preferred codec.				
SIP parameter	Offer: m=audio 4711 R	TP/AVP (8 (
values	Answer: m=audio 4712 R	TP/AVP 8	3 0						
ISDN parameter									
values									
Comments	SIP-I		SL	JT	ISDN				
	INVITE(IAM)	→							
	183 session Progress	(
	PRACK	←							
	200 OK PRACK	+							
	UPDATE	→		→	SETUP				
	200 OK UPDATE	+							
	180 Ringing(ACM)	+		+	ALERTING				
	200 OK INVITE(ANM)	+		+	CONNECT				
	ACK	→							
			Conver	sation					
	BYE(REL)	→		→	DISCONNECT				
	200 OK BYE(RLC)	+		+					
	,			→	RELEASE COMPLETE				

TP501012	SIP reference: RFC 32	261 [4]				ISDN reference:				
				Q.1	912	2.5 [1], clause 6.1.2 (i,1)				
TSS reference	SIP-I-ISDN/Basic_call/Sending_of the_SETUP_message									
SIP selection	NOT PICS 4/4 AND NOT PICS 4/5 AND PICS 1/9 AND NOT 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	3			_				
values	m= video 4712 RTP	P/AVP	31							
	Answer: m=audio 4711 RTP/ m=video 0 RTP/AVI		3							
ISDN parameter values										
Comments	SIP-I		SU	Т		ISDN				
	INVITE(IAM)	→			>	SETUP				
	180 Ringing(ACM)	+		•	(ALERTING				
	200 OK INVITE(ANM)	+		•	(CONNECT				
	ACK →									
	Conversation									
	BYE(REL)	→			→	DISCONNECT				
	200 OK BYE(RLC)	+		•	(RELEASE				
					→	RELEASE COMPLETE				

TP501013	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [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 NOT PICS 4/19								
criteria									
ISDN selection criteria									
Test purpose	one media streams, 100rel exheader and based on operato 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 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							
SIP parameter	Offer: m=audio 4711 RTF	P/AVP 8	3						
values	m= video 4712 RTI	P/AVP	31						
	Answer: m=audio 4711 RTF m=video 0 RTP/AV		3						
ISDN parameter									
values				_	I				
Comments	SIP-I		SU	T	ISDN				
	INVITE(IAM)	→							
	183 session Progress	+							
	PRACK	→							
	200 OK PRACK	-							
	UPDATE	→		-	SETUP				
	200 OK UPDATE	+							
	180 Ringing(ACM)	+		€					
	200 OK INVITE(ANM) ← CONNECT								
	ACK	→							
			Convers						
	BYE(REL)	→		-					
	200 OK BYE(RLC)	+		•					
				-	RELEASE COMPLETE				

TP501014	SIP reference: RFC 3	3261 [4]	0	ISDN reference: Q.1912.5 [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					9			
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 Require 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	3						
values	m= video 4712 RTI	P/AVP	31						
ISDN parameter	Answer: m=audio 4711 RTF m=video 0 RTP/AV		3						
values									
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	→							
	183 session Progress	+							
	PRACK	→							
	200 OK PRACK	+							
	UPDATE	→			→	SETUP			
	200 OK UPDATE	+							
	180 Ringing(ACM)	+			+	ALERTING			
	200 OK INVITE(ANM) ← CONNECT								
	ACK →								
			Convers	sation					
	BYE(REL)	→			→	DISCONNECT			
	200 OK BYE(RLC)	←			+	RELEASE			
					→	RELEASE COMPLETE			

TP501015	SIP reference: RFC 3261 [4]	[4] ISDN reference:									
			Q.1912.5 [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	NOT PICS 4/4 AND NOT PICS 4/5 AND			CS 4/1	9						
criteria											
ISDN selection criteria											
Test purpose	Ensure that the SUT on receipt of an IN one media streams and based on ope the call is refused with a 415 Unsupport	erator	policy then:		P offer with more than						
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)	E(IAM) →									
	415 Unsupported media type	•									
	ACK -	,									

TP501016	SIP reference: RFC 3261 [4	1]		ISDN reference:					
			Q.	Q.1912.5 [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 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 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 →								

TP501017	SIP reference: RFC 3261 [4] ISDN reference:								
			Q.1912.5 [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	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 police	cy thei	n:	•					
	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 →								

TP501018	SIP reference: RFC 32		ISDN reference:						
		Q	Q.1912.5 [1], clause 6.1.3						
TSS reference	SIP-I-ISDN/Basic_call/Sending_of the_SETUP_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: CCITT standardized coding								
values	Information transfer capabilit	y : Co	nstructed	from the U	SI or fron	n the TMR			
	transfer mode: circuit mode								
	information transfer rate: 64 kbits/s								
Comments	SIP-I		SU		ISDN				
	INVITE(IAM)	→		-	00				
	180 Ringing(ACM)	+		€					
	200 OK INVITE(ANM)	+		€	CONN	IECT			
	ACK	→							
	Conversation								
	BYE(REL)	→		-		DNNECT			
	200 OK BYE(RLC)	+		€	• RELE	ASE			
	→ RELEASE COMPLETE								

				Values for test purp	oses TP501018			
a_b_m_LINE_VALUE								
	m= line		b= line	a= line	BC_VALUE			
test purpose s	<media></media>	<transport></transport>	<fmt-list></fmt-list>	<modifier>:<bandwidth-value></bandwidth-value></modifier>	rtpmap: <dynamic-pt> <encoding name="">/<clock rate="">[/encoding parameters></clock></encoding></dynamic-pt>	Information Transport Capability	User Information Layer 1 Protocol Indicator	
				See note				
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		
NOTE:	<bandwid< td=""><td>th value> for <m< td=""><td>nodifier> of AS</td><td>is evaluated to be B kbit/s.</td><td></td><td></td><td></td></m<></td></bandwid<>	th value> for <m< td=""><td>nodifier> of AS</td><td>is evaluated to be B kbit/s.</td><td></td><td></td><td></td></m<>	nodifier> of AS	is evaluated to be B kbit/s.				

TP501019	SIP reference: RFC 3261 [4]			ISDN reference: Q.1912.5 [1], clause 6.1.3.5					
TSS reference	SIP-I-ISDN/Rasic call/Send	12.5 [1], clause 0.1.5.5							
SIP selection	SIP-I-ISDN/Basic_call/Sending_of the_SETUP_message								
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 HLC information element constructed from the encapsulated ATP (HLC).								
SIP parameter values									
ISDN parameter values									
Comments	SIP-I		SU	Τ		ISDN			
	INVITE(IAM)	→			→	SETUP			
	180 Ringing(ACM)	+			+	ALERTING			
	200 OK INVITE(ANM)	+			+	CONNECT			
	ACK	→							
	Conversation								
	BYE(REL) → → 200 OK BYE(RLC) ← ←					DISCONNECT			
						RELEASE			
						RELEASE COMPLETE			

TP501020	SIP reference: RFC 326	1 [4]		ISDN reference: EN 300 403-1 [i.16]					
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 number								
Comments	SIP-I		SUT		ISDN				
	INVITE(IAM)	→		→	SETUP				
	180 Ringing(ACM)	←		←	ALERTING				
	=00 011 11111 = (/ 11111)	←		←	CONNECT				
	ACK	→							
	Conversation								
	D 1 = (1 (= =)	→		→	DISCONNECT				
	200 OK BYE(RLC)	←		←	RELEASE				
				→	RELEASE COMPLETE				

TP501021	SIP reference: RFC	3261 [4]			ISDN reference:		
T00 (- 05711			EN 300 403-1 [i.16]		
TSS reference	SIP-I-ISDN/Basic_call/Sending_of the_SETUP_message							
SIP selection								
criteria								
ISDN selection								
criteria								
Test purpose	NDC SN where CC is the co component of the Request-L	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.						
					NDC	C" and use the remaining digits		
	to fill the Address signals co							
	Numbering plan Indicator	ISDN (T	elephony) numbe	ring	plan.		
SIP parameter values								
ISDN parameter values	SETUP : Called party number	er						
Comments	SIP-I		SL	JT		ISDN		
	INVITE(IAM)	→			→	SETUP		
	180 Ringing(ACM)	+			+	ALERTING		
	200 OK INVITE(ANM)	+			+	CONNECT		
	ACK	→						
			Conver	sation				
	BYE(REL)	→			→	DISCONNECT		
	200 OK BYE(RLC)	+			←	RELEASE		
					→	RELEASE COMPLETE		

TP501022	SIP reference: RFC	ı	ISDN reference: EN 300 403-1 [i.16]						
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: "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 number	er							
Comments	SIP-I		SUT		ISDN				
	INVITE(IAM)	→		→	SETUP				
	180 Ringing(ACM)	←		+	ALERTING				
	200 OK INVITE(ANM)	(+	CONNECT				
	ACK	→							
			Conversat	ion					
	BYE(REL)	→		→	DISCONNECT				
	200 OK BYE(RLC)	+		+	RELEASE				
				→	RELEASE COMPLETE				

A.1.1.1.1 Void

A.1.1.1.2 Sending of the INFO

TP502001	SIP reference: RFC		ı	ISDN reference: EN 300 403-1 [i.16]					
TSS reference	SIP-I-ISDN/Basic_call/Sending_of INFO_message								
SIP selection criteria	PICS 3/4	PICS 3/4							
ISDN selection criteria	PICS 3/8								
Test purpose	Ensure that the SUT receives an INVITE with the INFOe Call-ID and From tag as a previous INVITE which whereby the number of digits in the Request-URI is greater than the number of digits already accumulated for the call: • sends a INFO and pass it to outgoing ISDN procedures; • the INFO 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									
ISDN parameter values									
Comments	SIP-I		SU	IT		ISDN			
	INVITE(IAM)	→			→	SETUP			
	INVITE(SAM)	→			→	INFO			
	INVITE(SAM)	→			→	INFO			
	180 Ringing(ACM)	+			(ALERTING			
	200 OK INVITE(ANM)	+			(CONNECT			
	ACK	→							
			Convers	sation					
	BYE(REL)	→	•		→	DISCONNECT			
	200 OK BYE(RLC)	+	•		(RELEASE			
					→	RELEASE COMPLETE			

TP502002	SIP reference: RFC 326	ISDN reference: EN 300 403-1 [i.16]							
TSS reference	SIP-I-ISDN/Basic_call/Sending_of INFO_message								
SIP selection criteria	PICS 3/4								
ISDN selection criteria	PICS 3/8								
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 fewer than the number of digits already accumulated for the call: • then the SUT shall immediately send a 484 Address Incomplete 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(SAM) →								
	INVITE(SAM)								
	10 17 14441 000 11100111 11010	(
	ACK	→							

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	PICS 3/8 AND PICS 1/6							
criteria								
Test purpose	the 180 Ringing SIP	 Ensure that the SUT in call state N25, on receipt the ALERTING message: the 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	JT		ISDN		
	INVITE(IAM)	→			→	SETUP		
					←	SETUP ACK		
	180 Ringing(ACM)	+			←	ALERTING		
	Inl	band Info						
	BYE(REL)	→			→	DISCONNECT		
	200 OK BYE(RLC)	+			←	RELEASE		
					→	RELEASE COMPLETE		

TP503002	SIP reference: RFC 32	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								
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		SL	IT		ISDN		
	INVITE(IAM)	→			→	SETUP		
	180 Ringing(ACM)	+			(ALERTING		
	Inband	Info						
	BYE(REL)	→			→	DISCONNECT		
	200 OK BYE(RLC)	+			(RELEASE		
	·				→	RELEASE COMPLETE		

TP503003	SIP reference: RFC 32	61 [4]				ISDN reference:		
				G	2.19	12.5 [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		a recommendation to contact and an early and an early						
SIP parameter values								
ISDN parameter values								
Comments	SIP-I		SL	IT		ISDN		
	INVITE(IAM)	→		•	→	SETUP		
	183 Session Progress(ACM)	+		•	(CALL PROCEEDING		
	180 Ringing(CPG)	+		•	(ALERTING		
	Inband	Info						
	BYE(REL)	→		•	→	DISCONNECT		
	200 OK BYE(RLC)	+		•	(RELEASE		
				•	<u>→</u>	RELEASE COMPLETE		

TP503004	SIP reference: RFC 3261 [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	Ensure that the SUT in call state N25, on receipt of the CALL PROCEEDING message:						
	 a 183 Session Progress w 	 a 183 Session Progress with an encapsulated ACM is sent to the previous entity. 						
SIP parameter	183 Session Progress encapsu	183 Session Progress encapsulated ACM: BCi Called party status = no indication						
values								
ISDN parameter	CALL PROCEEDING							
values								
Comments	SIP-I		SL	ΙΤ		ISDN		
	INVITE(IAM)	→			→	SETUP		
	183 Session Progress(ACM)	+			+	CALL PROCEEDING		
	CANCEL(REL)	→			→	RELEASE		
	200 OK CANCEL(RLC)	←			←	RELEASE COMPLETE		

TP503005	SIP reference: RFC 32	61 [4]]		O 10	ISDN reference: 12.5 [1], clause 6.5 2)		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message							
SIP selection		_0.7.12	<u></u>	_0, (;				
criteria								
ISDN selection criteria								
Test purpose	Ensure that the SUT in call state N6, on receipt of the CALL PROCEEDING message containing a progress indicator set to PI_VALUE: • a 183 Session Progress with an encapsulated ACM is sent to the previous entity.							
SIP parameter values	183 Session Progress encapsu with Progress indicator							
ISDN parameter values	CALL PROCEEDING							
Comments	SIP-I		SU	IT		ISDN		
	INVITE(IAM)	→			→	SETUP		
	183 Session Progress(ACM)	+			+	CALL PROCEEDING(PI)		
	CANCEL(REL)	1			→	RELEASE		
	200 OK CANCEL(RLC)	+			+	RELEASE COMPLETE		

TP503006	SIP reference: RFC 32	61 [4]	ISDN reference:							
TSS reference	Q.1912.5 [1], clause 6.5 2) SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message										
SIP selection	CII 1 I IODIN/ Basic_call/ Neccipt_	_01 / (L	LIXTINO_	_0/\LL	100	NOCKEOO_message					
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 progress indicator set to PI_VALUE: • a 183 Session Progress with an encapsulated ACM is sent to the previous entity.									
SIP parameter values	183 Session Progress with end	183 Session Progress with encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with									
ISDN parameter values	CALL PROCEEDING										
Comments	SIP-I		SU	ΙΤ		ISDN					
	INVITE(IAM)	→			→	SETUP					
	183 Session Progress(ACM)	+			(CALL PROCEEDING					
	183 Session Progress(CPG)	+			+	PROGRESS(PI)					
				•							
	CANCEL(REL)	→		•	→	RELEASE					
	200 OK CANCEL(RLC)	+		•	+	RELEASE COMPLETE					

TP503007	SIP reference: RFC 3	3261 [4]		Q	.19	ISDN reference: 12.5 [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 PI_	Ensure that the SUT in call state N9, on receipt of the ALERTING message containing a progress indicator set to PI_VALUE: the 180 Ringing SIP response is sent.						
SIP parameter	180 Ringing encapsulated AC	M: BCi	called pa	rty status=	sul=	oscriber free, ATP with		
values	Progress indicator		•	•				
ISDN parameter values	ALERTING(PI)							
Comments	SIP-I		SU	Т		ISDN		
	INVITE(IAM)	→			→	SETUP		
	180 Ringing(ACM)	+		•	(ALERTING(PI)		
		•	•					
	BYE(REL)	→	•		>	DISCONNECT		
	200 OK BYE(RLC)	+	•	•	(RELEASE		
					→	RELEASE COMPLETE		

TP503008	SIP reference: RFC 326	1 [4]	0.1	ISDN reference:				
TCC reference	CID LICON/Design cell/Designs	4 ALEDTING		912.5 [1], clause 6.5 2)				
TSS reference	SIP-I-ISDN/Basic_call/Receipt_c	TALERTING	_CALL-PROC	_PROGRESS_message				
SIP selection criteria								
ISDN selection criteria	PICS 1/6	PICS 1/6						
Test purpose	progress indicator set to PI_V/	Insure 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	180 Ringing encapsulated ACM:	180 Ringing encapsulated ACM: BCi called party status=subscriber free, ATP with						
values	Progress indicator	·	•	•				
ISDN parameter	ALERTING(PI)							
values	,							
Comments	SIP-I	SI	JT	ISDN				
	INVITE(IAM)	→	→	SETUP				
	INVITE(SAM)	→	→	INFO				
	180 Ringing(ACM)	(+	ALERTING(PI)				
	BYE(REL)	→	→	DISCONNECT				
	200 OK BYE(RLC)	(+	RELEASE				
	, ,		→	RELEASE COMPLETE				

TP503009	SIP reference: RFC 3261 [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 a sufficient number of digits has been received 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)	→			→	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROCEEDING			
		•		•	,				
	CANCEL(REL)	→			→	RELEASE			
	200 OK CANCEL(RLC)	+			(RELEASE COMPLETE			

TP503010	SIP reference: RFC 3261 [4]			ISDN reference: Q.1912.5 [1], clause 6.5 2)				
TSS reference	SIP-I-ISDN/Basic_call/Receipt	_of ALERTI	NG_CA	LL-PROC	_PROGRESS_message			
SIP selection criteria					-			
ISDN selection criteria								
Test purpose	progress indicator:		-		GRESS message containing no sent to the previous entity.			
SIP parameter values	183 Session Progress encaps	ulated ACM	BCi Ca	alled party	status = no indication			
ISDN parameter values	CALL PROCEEDING							
Comments	SIP-I		SUT		ISDN			
	INVITE(IAM)	→		→	SETUP			
	183 Session Progress(ACM)	+		+	CALL PROCEEDING			
				+	PROGRESS			
			•					
	CANCEL(REL)	→		→	RELEASE			
	200 OK CANCEL(RLC)	+		+	RELEASE COMPLETE			

TP503011	SIP reference: RFC 3	261 [4]			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	message, receives an ALERT	Ensure that the SUT in the Idle state on receipt of a INVITE message sends out a SETUP nessage, receives an ALERTING message, having sent a 180 Ringing message, on eceipt of a PROGRESS message:							
SIP parameter values									
ISDN parameter values									
Comments	SIP-I		SUT		ISDN				
	INVITE(IAM)	→		→	SETUP				
	180 Ringing(ACM)	←		+	ALERTING				
				+	PROGRESS				
	BYE(REL)	→		→	DISCONNECT				
	200 OK BYE(RLC)	+		+	RELEASE				
			•	→	RELEASE COMPLETE				

TP503012	SIP reference: RFC 3261 [4]			ISDN reference: Q.1912.5 [1], clause 6.6						
TSS reference	SIP-I-ISDN/Basic_call/Receipt	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message								
SIP selection	·					<u> </u>				
criteria										
ISDN selection	PICS 1/6									
criteria										
Test purpose	message, receives a CALL PR message: no message is sent.									
SIP parameter values	183 Session Progress: Encaps	sulated	d ACM, ca	alled part	y sta	tus indicator=no indication				
ISDN parameter values										
Comments	SIP-I		SL	JT		ISDN				
	INVITE(IAM)	→			→	SETUP				
	183 Session Progress(ACM)	+			+	CALL PROCEEDING				
		← PROGRESS								
	CANCEL(REL)	→			→	RELEASE				
	200 OK CANCEL(RLC)	+			+	RELEASE COMPLETE				

TP503013	SIP reference: RFC 32	61 [4]				ISDN reference:	
						912.5 [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 criteria	PICS 1/6						
Test purpose	message, receives a CALL PR sends a 180 Ringing with	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 an ALERTING: • sends a 180 Ringing with encapsulated CPG Alerting.					
SIP parameter	183 Session Progress with end	apsul	ated ACN	1: called p	arty	status indicator=no indication	
values	180 Ringing encapsulated CPC	3: Eve	nt indicat	or=Alertin	ng		
ISDN parameter							
values							
Comments	SIP-I		SL	JT		ISDN	
	INVITE(IAM)	←			→	SETUP	
	183 Session Progress(ACM)	+			4	CALL PROCEEDING	
	180 Ringing(CPG)	+			+	ALERTING	
	Inband	Info					
	BYE(REL)	→			→	DISCONNECT	
	200 OK BYE(RLC)	+	•		+	RELEASE	
					^	RELEASE COMPLETE	

TP503014	SIP reference: RFC 32	261 [4]			ISDN reference:				
					2.1912.5 [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	Ensure that the SUT in the Idle message, receives a CALL PR				E message sends out a SETUP				
					n value is set to PI_VALUE, on				
	receipt of a ALERTING Messa		p 9						
	 sent a 180 Ringing messa 								
SIP parameter			ated ACN	1: called pa	ty status indicator=no indication				
values	183 Session Progress with end	apsul	ated CPG	event indic	cator=Progress, ATP with				
	Progress indicator								
	180 Ringing encapsulated CP0	3: Eve	nt indicat	or=Alerting					
ISDN parameter values									
Comments	SIP-I		SL	JT	ISDN				
	INVITE(IAM)	→		-	SETUP				
	183 Session Progress(ACM)	+		•	- CALL PROCEEDING				
	183 Session Progress(CPG)			•	PROGRESS(PI)				
	180 Ringing(CPG) ←								
	Inband	Info							
	BYE(REL)	→		-	DISCONNECT				
	200 OK BYE(RLC)	+		•	11227102				
					RELEASE COMPLETE				

TP503015	SIP reference: RFC 32	261 [4]			0.1	ISDN reference:				
T00 (EDTINIO	0411 00		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	message, on receipt of a ALER progress indicator where the	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 progress indicator 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	Л: BCi	called pa	arty status:	=su	bscriber free				
values	183 Session Progress with end									
	Progress indicator	•				3 ,				
ISDN parameter values										
Comments	SIP-I		SL	JT		ISDN				
	INVITE(IAM)	→			→	SETUP				
	180 Ringing(ACM)	+			(ALERTING				
	183 Session Progress(CPG) ← PROGRESS(PI)									
	Inband	Inband Info								
	BYE(REL)	→			→	DISCONNECT				
	200 OK BYE(RLC)	+			(RELEASE				
			_		→	RELEASE COMPLETE				

TP503016	SIP reference: RFC 32	261 [4]		Q.	ISDN reference: 1912.5 [1], clause 6.6					
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	PICS 1/6									
Test purpose	message, receives a CALL PR	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:								
SIP parameter values	183 Session Progress with end 183 Session Progress with end Progress indicator				status indicator=no indication tor=Progress, ATP with					
ISDN parameter values										
Comments	SIP-I		SUT		ISDN					
	INVITE(IAM)	→		→	SETUP					
	183 Session Progress(ACM)	+		+	CALL PROCEEDING					
	183 Session Progress(CPG)			+	PROGRESS(PI)					
	Inband	l Info								
	BYE(REL)	→		→	DISCONNECT					
	200 OK BYE(RLC)	+		+	RELEASE					
				→	RELEASE COMPLETE					

A.1.1.1.4 Receipt of the CONNECT Message

TP504001	SIP reference: RFC 3261	[4]	Q.	ISDN reference: 1912.5 [1], clause 6.7				
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of	CONNECT	_message	<u> </u>				
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 [i.14]) 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	SU	JT	ISDN				
	INVITE(IAM) →		→	SETUP				
	180 Ringing(ACM) ←		+	ALERTING				
	200 OK INVITE(ANM) ← CONNECT							
	ACK →							
		Conver	sation					
	BYE(REL) →		→	DISCONNECT				
	200 OK BYE(RLC) ←		+	RELEASE				
			→	RELEASE COMPLETE				

TP504002	SIP reference: RFC 3261 [4]			Q	ISDN reference: .1912.5 [1], clause 6.7
TSS reference	SIP-I-ISDN/Basic_call/Rece	eipt_of CC	NNECT_		[-],
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the message, receives a ALER sends a 200 OK INVITE to	TING mes	ssage, or	receipt of a	E message sends out a SETUP CONNECT message
SIP parameter values			_		
ISDN parameter values					
Comments	SIP-I		SL	JT	ISDN
	INVITE(IAM)	→			
	183 session Progress	←			
	PRACK	→			
	200 OK PRACK	+			
	UPDATE	→		→	SETUP
	200 OK UPDATE	+			
	180 Ringing(ACM)	+		+	ALERTING
	200 OK INVITE(ANM)	+		+	CONNECT
	ACK	→			
			Conver	sation	
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	+		+	RELEASE
				→	RELEASE COMPLETE

TP504003	SIP reference: RFC 3261 [4]			ISDN reference: Q.1912.5 [1], clause 6.7					
TSS reference	SIP-I-ISDN/Basic_call/Receipt_o	SIP-I-ISDN/Basic call/Receipt of CONNECT message							
SIP selection criteria		· =							
ISDN selection									
criteria									
Test purpose	ALERTING message, on receipt • sends a 200 OK INVITE to the	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		SUT			ISDN			
	INVITE(IAM)	→			→	SETUP			
	180 Ringing(ACM)	+			-	ALERTING			
	200 OK INVITE(ANM; SDP1)	+			-	CONNECT			
	ACK(SDP2)	→							
	· ,		Conversa	ation					
	BYE(REL)	→			→	DISCONNECT			
	200 OK BYE(RLC)	←			←	RELEASE			
					→	RELEASE COMPLETE			

TP504004	SIP reference: RFC 3261 [4]			0	101	ISDN reference: 2.5 [1], clauses 6.4, 6.7			
TSS reference	SIP-I-ISDN/Basic call/Receip	SIP-I-ISDN/Basic_call/Receipt_of CONNECT_message							
SIP selection									
criteria									
ISDN selection									
criteria									
Test purpose	message:	L COO OK BUITE & d							
SIP parameter values	200 OK INVITE: encapsulated	d CON		•					
ISDN parameter values									
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	→			→	SETUP			
	200 OK INVITE(CON)	+			+	CONNECT			
	ACK	→							
			Convers	sation					
	BYE(REL)	→			→	DISCONNECT			
	200 OK BYE(RLC)	+			+	RELEASE			
					→	RELEASE COMPLETE			

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

TP505001	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_REL	EASE				
SIP selection	NOT PICS 4/10							
criteria								
ISDN selection								
criteria								
Test purpose SIP parameter values	CV_ISDN, location LOC_ISDN:the SUT immediately red	an RELE quests the appropria	EASE CO e disconi te SIP st	OMPLETE	TE message, sends out a message with the Cause value fithe internal bearer path; ned as SIP_FAILURE_VA.			
ISDN parameter values	REL_COMP: cause value: CV_	_ISDN (P	IXIT)					
Comments	SIP-I	SIP-I SUT ISDN						
	INVITE(IAM)	→		→	SETUP			
	SIP FAILURE VA(REL)	+		+	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_o	f_DI	SC_or_RELE		- - - - - - - - - -		
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 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 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	_					
ISDN parameter values	RELEASE; cause value: CV_ISI	ON (I	PIXIT)				
Comments	SIP-I		SUT		ISDN		
	INVITE(IAM)	→		→	SETUP		
	SIP_FAILURE_VA(REL)	(+	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_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, on receipt of an 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 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_FAILUR	VA (FI)	XII)						
ISDN parameter values	DISC; cause value: CV_ISDN (PIXIT)							
Comments	SIP-I		SUT		ISDN				
	INVITE(IAM)	→		→	SETUP				
	SIP_FAILURE_VA(REL)	+	<u> </u>	←	DISCONNECT				
	ACK	→		→	RELEASE				
				+	RELEASE COMPLETE				

TP505004	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 SETUP message, receives a S RELEASE COMPLETE messa the SUT immediately requirate the SUT shall send the applications SIP Statue-Code: SIP_FAILUR	ETUP Age with ests the propriat	ACKNOV the Cau disconn e SIP sta	VLEDGE se value ection of	me CV f the	ssage, and on receipt of a '_ISDN, location LOC_ISDN: internal bearer path;			
ISDN parameter values	REL_COMP: cause value: CV	_ISDN	(PIXIT)						
Comments	SIP-I		SU	Γ		ISDN			
	INVITE(IAM)	→			1	SETUP			
	← SETUP ACK								
	SIP_FAILURE_VA(REL)	+			₩	RELEASE COMPLETE			
	ACK	→							

TP505005	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 SETUP message, receives a SI RELEASE message with the Ca the SUT immediately reque bearer channel is available side; the SUT shall send the app SIP Statue-Code: SIP_FAILURI	ETUP ause vests the for re	ACKNOV /alue CV_ e disconn- selection te SIP sta	VLEDGE _ISDN, I d lection of l, a REL_	me cat the CO	ssage, and on receipt of a ion LOC_ISDN: internal bearer path. When the MP is returned to the ISDN				
values										
ISDN parameter values	RELEASE; cause value: CV_IS	SUN (I	PIXII)							
Comments	SIP-I		SU	Τ		ISDN				
	INVITE(IAM)	→			→	SETUP				
			•		+	SETUP ACK				
	SIP_FAILURE_VA(REL)	+	•		+	RELEASE				
	ACK	→	·		→	RELEASE COMPLETE				

TP505006	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	C_or_R	ELEASE						
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 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 Statue-Code: SIP_FAILURE_VA (PIXIT									
ISDN parameter	DISC: cause value: CV ISDN	(PIXIT)								
values	Dies. sauss value. sv_lobit	(1 17(11)								
Comments	SIP-I		SU	Т		ISDN				
	INVITE(IAM)	→			1	SETUP				
	← SETUP ACK									
	SIP_FAILURE_VA(REL) ← DISCONNECT									
	ACK	→			→	RELEASE				
					+	RELEASE COMPLETE				

TP505007	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_D	ISC_or_R			. .		
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 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 parameter values	SIP Statue-Code: SIP_FAILUR	E_VA	(PIXIT)					
ISDN parameter values	REL_COMP: cause value: CV_	_ISDI	N (PIXIT)					
Comments	SIP-I		SU	Τ		ISDN		
	INVITE(IAM)	→			→	SETUP		
				·	←	SETUP ACK		
	INVITE(IAM)	→			→	INFO		
	SIP_FAILURE_VA(REL)	+		, and the second second	←	RELEASE COMPLETE		
	ACK	→						

TP505008	SIP reference: RFC 326	1 [4]			2 40	ISDN reference:				
	Q.1912.5 [1], clause 6.11.2									
TSS reference	SIP-I-ISDN/Basic_call/Receipt_c	ot_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 SETUP ACKNOWLEDGE message, on receipt of a re-INVITE message sends an INFORMATION message and on receipt of a REL EASE 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	SIP Statue-Code: SIP_FAILURE	_VA	(PIXIT)							
values										
ISDN parameter	RELEASE; cause value: CV_IS	DN ((PIXIT)							
values										
Comments	SIP-I		SU	T		ISDN				
	INVITE(IAM)	→			→	SETUP				
	← SETUP ACK									
	INVITE(IAM)	→			→	INFO				
	SIP_FAILURE_VA(REL)	-			+	RELEASE				
	ACK	→			→	RELEASE COMPLETE				

TP505009	SIP reference: RFC 32	61 [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 criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose SIP parameter	message with the Cause value the SUT immediately reque	ETUP ACKNONFORMATION CV_ISDN, Ico ests the disco efor re-selection propriate SIP	DWLEDGE N message ocation LO nnection of on, an ISD	me e and C_I the N R	ssage, on receipt of a d on receipt of a DISCONNECT SDN: internal bearer path. When the ELEASE message is returned
values	Sil Statue-Code. Sil _l AlEON	L_VA (1 1X11			
ISDN parameter values	DISC: cause value: CV_ISDN	(PIXIT)			
Comments	SIP-I	S	UT		ISDN
	INVITE(IAM)	→		→	SETUP
				←	SETUP ACK
	INVITE(IAM)	→		→	INFO
	SIP_FAILURE_VA(REL)	+		←	DISCONNECT
	ACK	→		→	RELEASE
				←	RELEASE COMPLETE

	Values for test purp	poses TP108001 - TP108009
	←SIP Message SIP_FAILURE_VA	← REL 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)
\/A 00	else 480 Temporarily unavailable	Course Value in the Class 040 (resource unousileble
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 to 79) (79 is class default)
VA_27	500 Server Internal Error	Cause Value No. 87 ("user not member of CUG")
VA_27	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)

TP505010	SIP reference: RFC 32	P reference: RFC 3261 [4] ISDN reference:							
	Q.1912.5 [1], clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_	of_DIS	C_or_REL	_EASE					
SIP selection	NOT PICS 4/10								
criteria									
ISDN selection criteria									
SIP parameter values	Ensure that the SUT in the Idle message, and receives a CALL COMPLETE message with the the SUT immediately reque the SUT shall send the app 183 Session Progress encapsus SIP Statue-Code: SIP_FAILUR	PROC Cause ests the propriat lated A E_VA (EEDING r value CV disconne e SIP statu CM: BCi C PIXIT)	message _ISDN, I ction of t us define	e, o l oc the ed a	n receipt of a RELEASE ation LOC_ISDN: internal bearer path; as SIP_FAILURE_VA.			
ISDN parameter values	REL_COMP: cause value: CV_	_ISDN	(PIXII)						
Comments	SIP-I		SUT			ISDN			
	INVITE(IAM)	→		-	>	SETUP			
	183 Session Progress(ACM)	+		•	-	CALL PROC			
	SIP_FAILURE_VA(REL)	+		•	F	RELEASE COMPLETE			
	ACK	→							

TP505011	SIP reference: RFC 32	reference: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic call/Receipt	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	NOT PICS 4/10	<u> </u>								
ISDN selection criteria										
Test purpose	message, and receives a CALL message with the Cause value the SUT immediately reque bearer channel is available is returned to the ISDN side the SUT shall send the app	bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side;								
SIP parameter values	183 Session Progress encapsu SIP Statue-Code: SIP_FAILUR			i Called p	oarty	status = no indication				
ISDN parameter values	RELEASE; cause value: CV_IS	SDN	(PIXIT)							
Comments	SIP-I		SU	Т		ISDN				
	INVITE(IAM)	→			→	SETUP				
	183 Session Progress(ACM)	+			+	CALL PROC				
	SIP_FAILURE_VA(REL)	+			←	RELEASE				
	ACK → 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 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, and receives a CALL PROCEEDING message, 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	183 Session Progress encapsu SIP Statue-Code: SIP_FAILUR			i Called p	party	status = no indication				
ISDN parameter values	DISC; cause value: CV_ISDN									
Comments	SIP-I		SU	Т		ISDN				
	INVITE(IAM)	→			→	SETUP				
	183 Session Progress(ACM)	+			←	CALL PROC				
	SIP_FAILURE_VA(REL)	+			+	DISCONNECT				
	ACK	→			→	RELEASE				
					+	RELEASE COMPLETE				

TP505013	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 SIP Statue-Code: SIP_FAILUR	•		_,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,	u. 0 u.	g			
ISDN parameter values	REL_COMP: cause value: CV								
Comments	SIP-I		SU	T		ISDN			
	INVITE(IAM)	^			^	SETUP			
	183 Session Progress(ACM)	+		•	+	CALL PROC			
	183 Session Progress(CPG)	+			+	PROGRESS(PI)			
	SIP_FAILURE_VA(REL)	+			+	RELEASE COMPLETE			
	ACK	→							

TP505014	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	NOT PICS 4/10								
criteria									
ISDN selection									
criteria									
Test purpose	Ensure that the SUT in the Idle								
	message, receives a CALL PRO								
	with a progress indicator PI_\			eipt of a F	KELE	ASE message with the Cause			
	value CV_ISDN, location LOC	_							
						internal bearer path. When the			
			e-selectio	n, an ISD	NK	ELEASE COMPLETE message			
	is returned to the ISDN side	,	-4- OID -4	- 4 I - 6 '		OID EAULIDE \/A			
CID noromotor	the SUT shall send the app 402 Consider Browners are appropriately								
SIP parameter values	183 Session Progress encapsu								
values	183 Session Progress with ence Progress indicator	apsui	aled CPG	e veni in	uica	ioi= Progress, ATP with			
	SIP Statue-Code: SIP_FAILUR	□ \/∧	(DIVIT)						
ISDN parameter	RELEASE; cause value: CV_IS								
values	NELEAGE, cause value. CV_N	ווטכ	(1 1/11)						
Comments	SIP-I		SU	JT		ISDN			
	INVITE(IAM)	→			→	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROC			
	183 Session Progress(CPG)	+			←	PROGRESS(PI)			
	SIP_FAILURE_VA(REL)	+			←	RELEASE			
	ACK	→			→	RELEASE COMPLETE			

TP505015	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	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.								
values	183 Session Progress encapsu 183 Session Progress with enc Progress indicator SIP Statue-Code: SIP_FAILUR	apsul	ated CPG						
ISDN parameter values	DISC; cause value: CV_ISDN								
Comments	SIP-I		SL	JT		ISDN			
	INVITE(IAM)	→			1	SETUP			
	183 Session Progress(ACM)	←			←	CALL PROC			
	183 Session Progress(CPG) ← PROGRESS(PI)								
	SIP_FAILURE_VA(REL)	+			+	DISCONNECT			
	ACK	→				RELEASE			
					+	RELEASE COMPLETE			

Table 21

	Values for test	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_FAILUR								
ISDN parameter values	REL_COMP: cause value: CV_	_ISDI	N (PIXIT)						
Comments	SIP-I		SU	Τ		ISDN			
	INVITE(IAM)	→			→	SETUP			
	← SETUP ACK								
	180 Ringing(ACM) ← ALERTING								
	SIP_FAILURE_VA(REL)	+			+	RELEASE COMPLETE			
	ACK	→							

TP505017	SIP reference: RFC 32		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									
SIP parameter values	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 values	RELEASE; cause value: CV_	ISDN	(PIXIT)						
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	→			→	SETUP			
					(SETUP ACK			
	180 Ringing(ACM) ← ALERTING								
	SIP_FAILURE_VA(REL)	SIP_FAĬLURE_VÁ(REL) ← RELEASE							
	ACK	→			→	RELEASE COMPLETE			

TP505018	SIP reference: RFC 3	261 [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	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_ISDN	N (PIXI	Τ)							
Comments	SIP-I		SU	T		ISDN				
	INVITE(IAM)	→			→	SETUP				
					+	SETUP ACK				
	180 Ringing(ACM) ← ALERTING									
	SIP_FAILURE_VA(REL)	+			+	DISCONNECT				
	ACK	→			^	RELEASE				
					4	RELEASE COMPLETE				

TP505019	SIP reference: RFC 32	61 [4	ı	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	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_FAILUR			i Called	party	status = no indication				
ISDN parameter values	REL_COMP: cause value: CV	_ISDI	N (PIXIŤ)							
Comments	SIP-I		SU	Т		ISDN				
	INVITE(IAM)	→			→	SETUP				
	183 Session Progress(ACM)	4			+	CALL PROC				
	180 Ringing(CPG)	+			+	ALERTING				
	SIP_FAILURE_VA(REL) ← RELEASE COMPLETE									
	ACK	→								

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 encapsu SIP Statue-Code: SIP_FAILUF			i Called p	arty	status = no indication			
ISDN parameter values	RELEASE; cause value: CV_I	SDN	(PIXIT)						
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	→			→	SETUP			
	183 Session Progress(ACM) ← CALL PROC								
	180 Ringing(CPG) ← ALERTING								
	SIP_FAILURE_VA(REL)	+			+	RELEASE			
	ACK	→			→	RELEASE COMPLETE			

TP505021	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 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	lated	ACM: BC	i Called p	arty	status = no indication				
values	SIP Statue-Code: SIP_FAILUR	E_VA	(PIXIT)							
ISDN parameter values	DISC; cause value: CV_ISDN	(PIXI	T)							
Comments	SIP-I		SU	T		ISDN				
	INVITE(IAM)	→			→	SETUP				
	183 Session Progress(ACM)	+			←	CALL PROC				
	180 Ringing(CPG) ← ALERTING									
	SIP_FAILURE_VA(REL)	+			+	DISCONNECT				
	ACK	^			1	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 progress indicator 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 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								
ICDN naramatar	SIP Statue-Code: SIP_FAILUR								
ISDN parameter values	REL_COMP: cause value: CV	וטטו_	N (PIAII)						
Comments	SIP-I		SL	ΙΤ		ISDN			
	INVITE(IAM)	→			→	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROC			
	183 Session Progress(CPG)	+			+	PROGRESS(PI)			
	180 Ringing(CPG)	+		-	←	ALERTING			
	SIP_FAILURE_VA(REL)	+			+	RELEASE COMPLETE			
	ACK	→		_					

TP505023	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	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 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 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 enca Progress indicator SIP Statue-Code: SIP_FAILUR	apsul	ated CPG					
ISDN parameter values	RELEASE; cause value: CV_IS	SDN ((PIXIT)					
Comments	SIP-I		SL	JT		ISDN		
	INVITE(IAM)	1			→	SETUP		
	183 Session Progress(ACM)	+			+	CALL PROC		
	183 Session Progress(CPG)	+			←	PROGRESS(PI)		
	180 Ringing(CPG)	+			←	ALERTING		
	SIP_FAILURE_VA(REL)	+			←	RELEASE		
	ACK	→			→	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_DISC_or_RELEASE								
SIP selection	NOT PICS 4/10								
criteria									
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, receives a PROGRESS message with a progress indicator 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 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. 183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with								
	Progress indicator	•				3			
	SIP Statue-Code: SIP_FAILUR								
ISDN parameter values	DISC; cause value: CV_ISDN	(PIXII)						
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	→			→	SETUP			
	183 Session Progress(ACM)	4			4	CALL PROC			
	183 Session Progress(CPG)	+			+	PROGRESS(PI)			
	180 Ringing(CPG)	+			+	ALERTING			
	SIP_FAILURE_VA(REL)	+			+	DISCONNECT			
	ACK	→			→	RELEASE			
					+	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 progress indicator where the progress description value is set to PI_VALUE, 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. 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								
ISDN parameter values	REL_COMP: cause value: CV	_ISDI	N (PIXIT)						
Comments	SIP-I		SU	T		ISDN			
	INVITE(IAM)	^			1	SETUP			
	183 Session Progress(ACM)	+	-	-	+	CALL PROC			
	180 Ringing(CPG)	+			4	ALERTING			
	183 Session Progress(CPG)	+			+	PROGRESS(PI)			
	SIP_FAILURE_VA(REL)	+			4	RELEASE COMPLETE			
	ACK	→			•				

TP505026	SIP reference: RFC 32	61 [4]	l		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	NOT PICS 4/10									
criteria										
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 progress indicator where the progress description value is set to PI_VALUE, on receipt of an 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. 183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with									
	SIP Statue-Code: SIP_FAILUR	E_VA	(PIXIT)							
ISDN parameter values	RELEASE; cause value: CV_IS	SDN ((PIXIT)							
Comments	SIP-I		SL	IT		ISDN				
	INVITE(IAM)	→			→	SETUP				
	183 Session Progress(ACM)	+			+	CALL PROC				
	180 Ringing(CPG)	+			+	ALERTING				
	183 Session Progress(CPG)	+			+	PROGRESS(PI)				
	SIP_FAILURE_VA(REL)	+			+	RELEASE				
	ACK	↑			→	RELEASE COMPLETE				

TP505027	SIP reference: RFC 32	61 [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									
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 progress indicator where the progress description value is set to PI_VALUE, on receipt of an 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. 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	SIP Statue-Code: SIP_FAILUR DISC; cause value: CV_ISDN								
Comments	SIP-I		SL	JT		ISDN			
	INVITE(IAM)	→			→	SETUP			
	183 Session Progress(ACM)	+			←	CALL PROC			
	180 Ringing(CPG)	+			←	ALERTING			
	183 Session Progress(CPG)	+		_	←	PROGRESS(PI)			
	SIP_FAILURE_VA(REL)	←	-		←	DISCONNECT			
	ACK	→			→	RELEASE			
					←	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								
ISDN selection criteria									
Test purpose	SETUP message, receives a C message, a 200 OK message i with the Cause value CV_ISDI	and do i initinduately requeste the disconnection of the internal boards pain,							
SIP parameter values	183 Session Progress encapsu			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)	→			→	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROC			
	200 OK INVITE(ANM)	200 OK INVITE(ANM) ← CONNECT							
			Commun	ication					
	BYE(REL)	+			+	RELEASE COMPLETE			
	200 OK BYE(RLC)	→							

TP505029	SIP reference: RFC 32	261 [4]				ISDN reference:			
						912.5 [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 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	ISDN (PIXIT)						
values						1			
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	→			→	SETUP			
	183 Session Progress(ACM)	←			←	CALL PROC			
	200 OK INVITE(ANM)	+			←	CONNECT			
			Commun	ication					
	BYE(REL)	←			+	RELEASE			
	200 OK BYE(RLC)	→			→	RELEASE COMPLETE			

TP505030	SIP reference: RFC 32	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 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	arty	status = no indication			
ISDN parameter values	DISC; cause value: CV_ISDN	(PIXI	T)						
Comments	SIP-I		SU	Τ		ISDN			
	INVITE(IAM)	→			1	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROC			
	200 OK INVITE(ANM)	+			+	CONNECT			
			Commun	ication					
	BYE(REL)	+			+	DISCONNECT			
	200 OK BYE(RLC)	→			→	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		NOO_OI_INEEI	LAGE					
criteria	110111034/10								
ISDN selection									
criteria									
Test purpose SIP parameter values ISDN parameter	SETUP message, receives a 0 of a RELEASE COMPLETE m LOC_ISDN: the SUT immediately requ	the SUT immediately requests the disconnection of the internal bearer path;							
values	_								
Comments	SIP-I		SUT		ISDN				
	INVITE(IAM)	→		→	SETUP				
	200 OK INVITE(ANM)	+		+	CONNECT				
			Communica	tion					
	BYE(REL)	+		+	RELEASE COMPLETE				
	200 OK BYE(RLC)	→							

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	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 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.								
ISDN parameter	RELEASE; cause value: CV_ISD	N (PIXIT)							
values		, ,							
Comments	SIP-I	SL	JT	ISDN					
	INVITE(IAM)	→	→	SETUP					
	200 OK INVITE(ANM)	-	+	CONNECT					
		Commur	nication						
	BYE(REL) ◀	-	+	RELEASE					
	200 OK BYE(RLC)	>	→	RELEASE COMPLETE					

TP505033	SIP reference: RFC 320	61 [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	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 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								
ISDN parameter	DISC; cause value: CV_ISDN ((PIXI	Γ)					
values								
Comments	SIP-I		SU	Τ		ISDN		
	INVITE(IAM)	→			→	SETUP		
	200 OK INVITE(ANM)	←			4	CONNECT		
			Commun	ication				
	BYE(REL)	+			+	DISCONNECT		
	200 OK BYE(RLC)	→		·	^	RELEASE		
					+	RELEASE COMPLETE		

TP505034	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	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, via a broadcast data link, after time-out of T303 : • the SUT shall send a 480 Temporarily unavailable final response.									
SIP parameter values	480 Temporarily unavailable: Encapsula	ated	REL with cause 1	8						
ISDN parameter values										
Comments	SIP-I		SUT		ISDN					
	INVITE(IAM)	NVITE(IAM) → SETUP								
	→ SETUP									
		T303 expiry								
	480 Temporarily unavailable(REL)	←								
	ACK	→		·						

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	reference: RFC 3261 [4] ISDN reference Q.1912.5 [1], clause							
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 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 parameter values	SIP Statue-Code: SIP_FAILURE_VA (PIXIT), Reason header value: CV_SIP (PIXIT)								
ISDN parameter values	REL_COMP: cause value: CV_ISD	N (PIXIT)							
Comments	SIP-I	SU	Т	ISDN					
	INVITE(IAM) →		→	SETUP					
	SIP_FAILURE_VA(REL) ←		+	RELEASE COMPLETE					
	ACK →								

TP505036	SIP reference: RFC 3261 [SIP reference: RFC 3261 [4] ISDN Q.1912.5 [
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, 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	RELEASE; cause value: CV_ISDN	(PIXIT)								
values										
Comments	SIP-I	SL	JT T	ISDN						
	INVITE(IAM) →		→	SETUP						
	SIP_FAILURE_VA(REL) ←		+	RELEASE						
	ACK →		→	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")

Values for test purposes 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 326	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 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; • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.								
SIP parameter values	183 Session Progress encapsul								
ISDN parameter values	SIP Statue-Code: SIP_FAILURE_VA (PIXIT), Reason header value: CV_SIP (PIXIT) REL_COMP: cause value: CV_ISDN (PIXIT)								
Comments	SIP-I		SL	JΤ		ISDN			
	INVITE(IAM)	\			→	SETUP			
	183 Session Progress(ACM)	1			+	CALL PROC			
	SIP_FAILURE_VA(REL)	+	-		+	RELEASE COMPLETE			
	ACK	→							

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 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 encapsul SIP Statue-Code: SIP_FAILURE								
ISDN parameter	RELEASE; cause value: CV_IS			iveasoli	iieat	Jei Value. CV_SIF (FIXII)			
values			()						
Comments	SIP-I		SU	IT		ISDN			
	INVITE(IAM)	→			→	SETUP			
	183 Session Progress(ACM)	+			←	CALL PROC			
	SIP_FAILURE_VA(REL)	(+	RELEASE			
	ACK	→			→	RELEASE COMPLETE			

TP505039	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	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 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 SIP Statue-Code: SIP_FAILUR									
ISDN parameter values	DISC; cause value: CV_ISDN					_ ,				
Comments	SIP-I		SU	Τ		ISDN				
	INVITE(IAM)	→			→	SETUP				
	183 Session Progress(ACM)	+			(CALL PROC				
	SIP_FAILURE_VA(REL)	+			+	DISCONNECT				
	ACK	→			→	RELEASE				
					(RELEASE COMPLETE				

TP505040	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 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; • 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	REL_COMP: cause value: CV_ISDN (PIXIT)								
Comments	SIP-I		SL	JT		ISDN			
	INVITE(IAM)	↑			→	SETUP			
	183 Session Progress(ACM)	+			←	CALL PROC			
	183 Session Progress(CPG)	+			←	PROGRESS(PI)			
	SIP_FAILURE_VA(REL)	+			←	RELEASE COMPLETE			
	ACK	→							

TP505041	SIP reference: RFC 32	SIP reference: RFC 3261 [4]							
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		SL	JT		ISDN			
	INVITE(IAM)	→			1	SETUP			
	183 Session Progress(ACM)	←			←	CALL PROC			
	183 Session Progress(CPG)	+			+	PROGRESS(PI)			
	SIP_FAILURE_VA(REL)	←			+	RELEASE			
	ACK	→			→	RELEASE COMPLETE			

TP505042	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	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 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	DISC; cause value: CV_ISDN					· ,				
Comments	SIP-I		SU	IT		ISDN				
	INVITE(IAM)	→			→	SETUP				
	183 Session Progress(ACM)	+			←	CALL PROC				
	183 Session Progress(CPG)	+			+	PROGRESS(PI)				
	SIP_FAILURE_VA(REL)	+			←	DISCONNECT				
	ACK	→			→	RELEASE				
					←	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	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_FAILUR	E_VA	(PIXIT),	Reason I	head	der value: CV_SIP (PIXIT)			
ISDN parameter values	REL_COMP: cause value: CV_	ISDN	(PIXIT)						
Comments	SIP-I		SU	Τ		ISDN			
	INVITE(IAM)	→			1	SETUP			
					+	SETUP ACK			
	180 Ringing(ACM)	+			+	ALERTING			
	SIP_FAILURE_VA(REL)	+			+	RELEASE COMPLETE			
	ACK	→							

TP505044	SIP reference: RFC 32	61 [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 criteria	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 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 Statue-Code: SIP_FAILURE_VA (PIXIT), Reason header value: CV_SIP (PIXIT)							
values			· (1 17 (1 1),	rrouson i	···ouc	101 Value: 0 V_011 (1 1741)		
ISDN parameter values	RELEASE; cause value: CV_IS	SDN	(PIXIT)					
Comments	SIP-I		SL	JT		ISDN		
	INVITE(IAM)	^			→	SETUP		
					←	SETUP ACK		
	180 Ringing(ACM)	+			←	ALERTING		
	SIP_FAILURE_VA(REL)	+			←	RELEASE		
	ACK	→			→	RELEASE COMPLETE		

TP505045	SIP reference: RFC 32	261 [4]			- 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_FAILUF	RE_VA	(PIXIT),	Reason	head	der value: CV_SIP (PIXIT)		
values								
ISDN parameter values	DISC: cause value: CV_ISDN	(PIXI	Γ)					
Comments	SIP-I		SU	T		ISDN		
	INVITE(IAM)	←			→	SETUP		
					+	SETUP ACK		
	180 Ringing(ACM)	+			←	ALERTING		
	SIP_FAILURE_VA(REL)	+			+	DISCONNECT		
	ACK	↑			→	RELEASE		
					←	RELEASE COMPLETE		

Table 26

	Values for test purpo	ses 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 32	261 [4]	C	Ղ.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	PICS 4/10	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 values	183 Session Progress encapsu SIP Statue-Code: SIP_FAILUR								
ISDN parameter values	REL_COMP: cause value: CV			iteu30ii i	icac	variation ov_on (FIXIT)			
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	→			→	SETUP			
	183 Session Progress(ACM)	+			(CALL PROC			
	200 OK INVITE(ANM)	+			←	CONNECT			
			Commun	ication					
	BYE(REL)	+			(RELEASE COMPLETE			
	200 OK BYE(RLC)	→							

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 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 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 SIP Statue-Code: SIP_FAILUR								
ISDN parameter values	RELEASE: cause value: CV_I		, ,						
Comments	SIP-I		SL	JT		ISDN			
	INVITE(IAM)	→			→	SETUP			
	183 Session Progress(ACM)	+			←	CALL PROC			
	200 OK INVITE(ANM)	+			←	CONNECT			
			Commun	nication					
	BYE(REL)	+			←	RELEASE			
	200 OK BYE(RLC)	→			→	RELEASE COMPLETE			

TP505048	SIP reference: RFC 32	261 [4]		0 40	ISDN reference:			
T00(OID LIODNI/DiII/Di-	- (D	100 0			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	5 4 44 007				<i></i> —				
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	183 Session Progress encapsu								
values	SIP Statue-Code: SIP_FAILUR			Reason	nead	der value: CV_SIP (PIXIT)			
ISDN parameter values	DISC: cause value: CV_ISDN	(PIXI	1)						
Comments	SIP-I		SU	IT.	1	ISDN			
Comments	INVITE(IAM)	→	30	'	→	SETUP			
	/	-			7				
	183 Session Progress(ACM)					CALL PROC			
	200 OK INVITE(ANM)	+			←	CONNECT			
		ļ	Commun	ication	-	21000111100			
	BYE(REL)	(+	DISCONNECT			
	200 OK BYE(RLC)	→			→	RELEASE			
					←	RELEASE COMPLETE			

TP505049	SIP reference: RFC 326	ce: 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 SIP parameter	Ensure that the SUT in the Idle message, receives a CONNECTRELEASE COMPLETE message the SUT immediately requeshall send a BYE message the ISDN Cause Value field header field. SIP Statue-Code: SIP_FAILURI	T me ge wit ests th ; d in th	ssage, a 2 th the Cau ne disconi	200 OK me ise value (nection of EL messa	ess CV_ the age	age is sent, on receipt of a LISDN, location LOC_ISDN: internal bearer path. the SUT is mapped to the Reason		
values	SIP Statue-Code. SIP_PAILORI	=_v <i>F</i>	((F I X I I), I	Reason n	eau	der value. CV_SIP (PIXII)		
ISDN parameter values	REL_COMP: cause value: CV_	_ISDI	N (PIXIT)					
Comments	SIP-I		SU	T		ISDN		
	INVITE(IAM)	→			→	SETUP		
	200 OK INVITE(ANM)	+		4	(CONNECT		
			Commun	ication				
	BYE(REL)	+		•	(RELEASE COMPLETE		
	200 OK BYE(RLC)	→						

TP505050	SIP reference: RFC 3261	1 [4]			. 40	ISDN reference:				
T00 (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 SIP parameter	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.									
values	SIP Statue-Code: SIP_FAILURE	_ v	(F1X11), IX	eason n	ica	der value. CV_SII (FIXII)				
ISDN parameter values	RELEASE: cause value: CV_ISE	ON (PIXIT)							
Comments	SIP-I		SUT	-		ISDN				
	INVITE(IAM)	→			→	SETUP				
	200 OK INVITE(ANM)	←			(CONNECT				
			Communic	cation						
	BYE(REL)	((RELEASE				
	200 OK BYE(RLC)	→			→	RELEASE COMPLETE				

TP505051	SIP reference: RFC 3261	[4]	Q.1	ISDN reference: 912.5 [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 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	SIP Statue-Code: SIP_FAILURE_	VA (PIXIT),	Reason hea	der value: CV_SIP (PIXIT)				
ISDN parameter values	DISC: cause value: CV_ISDN (PI	XIT)						
Comments	SIP-I	SU	JT	ISDN				
	INVITE(IAM)	>	→	SETUP				
	200 OK INVITE(ANM) ◀	-	+	CONNECT				
		Commu	nication					
	BYE(REL)	-	+	DISCONNECT				
	200 OK BYE(RLC))	→	RELEASE				
	, ,		+	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	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 BYE , 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 mapp	ed from	the encaps	ulated REL in the received BYE					
Comments	SIP-I		SU	Т	ISDN					
	INVITE(IAM)	→		-	SETUP					
	180 Ringing(ACM)	+		+	- ALERTING					
	200 OK INVITE(ANM)	+		+	- CONNECT					
	ACK →									
	Conversation									
	BYE(REL)	→		-	DISCONNECT					
	200 OK BYE(RLC)	+		+	RELEASE					
				-	RELEASE COMPLETE					

TP506002	SIP reference: RFC 32	261 [4]	I	Q.19	ISDN reference: 12.5 [1], clause 6.11.1				
TSS reference	SIP-I-ISDN/Basic_call/Receipt	of_B`	YE_or_CANC	EL					
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 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									
ISDN parameter	DISC: Cause value and locatio	n map	ped from the	encapsul	ated REL in the received				
values	CANCEL								
Comments	SIP-I		SUT		ISDN				
	INVITE(IAM)	→		→	SETUP				
	180 Ringing(ACM)	+		+	ALERTING				
	CANCEL(REL)	CANCEL(REL) → DISCONNECT							
	200 OK CANCEL ← RELEASE								
	487 Request Terminated	+		→	RELEASE COMPLETE				
	ACK	→							

TP506003	SIP reference: RFC 3	G	Q.19	ISDN reference: 112.5 [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	SETUP message, the SUT or	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 CANCEL , 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 locat CANCEL	ion map	ped from	the enca	psu	lated REL in the received				
Comments	SIP-I		SU	Т		ISDN				
	INVITE(IAM)	→								
	100 Trying	+			→	SETUP				
	CANCEL(REL)	CANCEL(REL) → DISCONNECT								
	200 OK CANCEL	+			(RELEASE				
	487 Request Terminated	+			→	RELEASE COMPLETE				
	ACK	→								

TP506004	SIP reference: RFC 3	3261 [4]		Q.19	ISDN reference: 12.5 [1], clause 6.11.1				
TSS reference	SIP-I-ISDN/Basic_call/Receip	t_of_B`	YE_or_CA	NCEL					
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 CANCEL , 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	tion ma	apped fron	n the end	capsi	ulated REL in the received			
Comments	SIP-I		SU	T		ISDN			
	INVITE(IAM)	→			→	SETUP			
					+	SETUP ACK			
	CANCEL(REL) → DISCONNECT								
	200 OK CANCEL	+			(RELEASE			
	487 Request Terminated	+			→	RELEASE COMPLETE			
	ACK	→							

TP506005	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	SETUP message, receives a INFO message on receipt of S	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 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							
SIP parameter values									
ISDN parameter	DISC: Cause value and locati	on mappe	d from	the encapsu	lated REL in the received				
values	CANCEL								
Comments	SIP-I		SU	Т	ISDN				
	INVITE(IAM)	→		→	SETUP				
				+	SETUP ACK				
	INVITE(IAM)	→		→	INFO				
	CANCEL(REL)	→		→	DISCONNECT				
	200 OK CANCEL	+		+	RELEASE				
	487 Request Terminated	+		→	RELEASE COMPLETE				
	ACK	→							

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_B	YE_or_C/	ANCEL				
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	ılated	ACM: BC	i Called	party	status = no indication		
ISDN parameter	DISC: Cause value and locatio	n map	ped from	the enca	apsul	ated REL in the received		
values	CANCEL	-			-			
Comments	SIP-I		SU	Т		ISDN		
	INVITE(IAM)	→			→	SETUP		
	183 Session Progress(ACM)	+			+	CALL PROC		
	CANCEL(REL)	→			→	DISCONNECT		
	200 OK CANCEL ← RELEASE							
	487 Request Terminated	+			→	RELEASE COMPLETE		
	ACK	→		-				

TP506007	SIP reference: RFC 326	61 [4]		0	10	ISDN reference:		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_c	of BY	F or CA		1.19	12.5 [1], clause 6.11.1		
SIP selection criteria	On Tresty Buolo_out//Tooolpt_	<u>01_D1</u>	<u></u>	WOLL				
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 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 encapsul 183 Session Progress with enca Progress indicator							
ISDN parameter values	DISC: Cause value and location CANCEL	map	oed from	the encap	osul	ated REL in the received		
Comments	SIP-I		SU	Т		ISDN		
	INVITE(IAM)	→		1	→	SETUP		
	183 Session Progress(ACM)	←			(CALL PROC		
	183 Session Progress(CPG)	←			(PROGRESS(PI)		
	CANCEL(REL)	→		1	→	DISCONNECT		
	200 OK CANCEL	←			(RELEASE		
	487 Request Terminated	+			→	RELEASE COMPLETE		
	ACK	→	•					

TP506008	SIP reference: RFC	3261 [4]		Q.		ISDN reference: 12.5 [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	SETUP message, receives a ALERTING message, on rece	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 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										
ISDN parameter values	DISC: Cause value and locat CANCEL	ion map	ped from	the encaps	sula	ated REL in the received				
Comments	SIP-I		SU [*]	Γ		ISDN				
	INVITE(IAM)	→		-	•	SETUP				
				•	.1.	SETUP ACK				
	180 Ringing(ACM)	+		•	Т	ALERTING				
	CANCEL(REL)	→		-	•	DISCONNECT				
	200 OK CANCEL	+		€		RELEASE				
	487 Request Terminated	+		7	•	RELEASE COMPLETE				
	ACK	→				·				

TP506009	SIP reference: RFC 32	61 [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 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	ılated A	CM: BCi	Called par	ty status = no indication				
	DIOC. O			·	ulated DEL in the maniford				
ISDN parameter	DISC: Cause value and location	n mapp	ea trom	tne encaps	ulated REL in the received				
values	CANCEL		<u> </u>		T				
Comments	SIP-I		SU		ISDN				
	INVITE(IAM)	→		→	SETUP				
	183 Session Progress(ACM)	←		€	CALL PROC				
	180 Ringing(CPG)	+		+	ALERTING				
	CANCEL(REL)								
	200 OK CANCEL ← RELEASE								
	487 Request Terminated	487 Request Terminated ← → RELEASE COMPLETE							
	ACK	→							

TP506010	SIP reference: RFC 32	(Q.19	ISDN reference: 12.5 [1], clause 6.11.1					
TSS reference	SIP-I-ISDN/Basic_call/Receipt_	of_B	YE_or_C/	ANCEL					
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 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 183 Session Progress with enc Progress indicator								
ISDN parameter values	DISC: Cause value and locatio CANCEL	n map	ped from	the enca	ıpsul	ated REL in the received			
Comments	SIP-I		SU	JT		ISDN			
	INVITE(IAM)	→			→	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROC			
	180 Ringing(CPG)	+			(ALERTING			
	183 Session Progress(CPG)	+			+	PROGRESS(PI)			
	CANCEL(REL) → DISCONNECT								
	200 OK CANCEL	+			←	RELEASE			
	487 Request Terminated	+			→	RELEASE COMPLETE			
	ACK	→							

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 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										
ISDN parameter values	DISC: Cause value and location	on map	ped from	the enca	psul	ated REL in the received BYE				
Comments	SIP-I		SU	Т		ISDN				
	INVITE(IAM)	→			→	SETUP				
	180 Ringing(ACM)	+			←	ALERTING				
	BYE(REL)	→			→	DISCONNECT				
	200 OK BYE(RLC)	+			←	RELEASE				
	487 Request Terminated	+			→	RELEASE COMPLETE				
	ACK	→								

TP506012	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_B	YE_or_C/	NCEL					
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: BC	i Called	party	status = no indication			
ISDN parameter values	DISC: Cause value and locatio	n map	ped from	the enc	apsul	ated REL in the received BYE			
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	→			→	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROC			
	BYE(REL) → DISCONNECT								
	200 OK BYE(RLC)	200 OK BYE(RLC) ← RELEASE							
	487 Request Terminated	Request Terminated ← RELEASE COMPLETE							
	ACK	→							

TP506013	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_CANCEL										
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 followed by a PROGRESS message, on receipt of SIP BYE, the I-IWU shall send an ISDN DISC with the cause and ocation 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	he encapsu	lated REL in the received BYE						
Comments	SIP-I		SU	Γ	ISDN						
	INVITE(IAM)	→		→	SETUP						
	183 Session Progress(ACM)	+		+	CALL PROC						
	183 Session Progress(CPG)	+		+	PROGRESS(PI)						
	BYE(REL)	→		→	DISCONNECT						
	200 OK BYE(RLC)	+		+	RELEASE						
	487 Request Terminated	+		→	RELEASE COMPLETE						
	ACK	→									

TP506014	SIP reference: RFC 3	261 [4]		ISDN reference:						
					<u> </u>	12.5 [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 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										
	D100 0 1 11 11			d						
ISDN parameter values	DISC: Cause value and locati	on mapp	ea trom	tne enca	psui	ated REL in the received BYE				
Comments	SIP-I		SU	ΙΤ		ISDN				
	INVITE(IAM)	→			→	SETUP				
					←	SETUP ACK				
	180 Ringing(ACM)	+			+	ALERTING				
	BYE(REL)	→			→	DISCONNECT				
	200 OK BYE(RLC)	+		•	+	RELEASE				
	487 Request Terminated	+		•	→	RELEASE COMPLETE				
	ACK	→								

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_CANCEL										
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 ocation 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										
ISDN parameter values	DISC: Cause value and location	n mapp	ed from	the enca	psul	ated REL in the received BYE					
Comments	SIP-I		SU	Т		ISDN					
	INVITE(IAM)	→			→	SETUP					
	183 Session Progress(ACM)	+			(CALL PROC					
	180 Ringing(CPG)	+			(ALERTING					
	BYE(REL)	→			→	DISCONNECT					
	200 OK BYE(RLC)	+			(RELEASE					
	487 Request Terminated	+			→	RELEASE COMPLETE					
	ACK	→									

TP506016	SIP reference: RFC 32	61 [4]		Q.19	ISDN reference: 12.5 [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, a ALERTING Message, a PROGRESS 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 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 location	n map	ped from	the enca	psul	ated REL in the received BYE				
Comments	SIP-I		SU	Τ		ISDN				
	INVITE(IAM)	→			→	SETUP				
	183 Session Progress(ACM)	+			+	CALL PROC				
	180 Ringing(CPG)	+			←	ALERTING				
	183 Session Progress(CPG)	+			+	PROGRESS(PI)				
	BYE(REL)	→			→	DISCONNECT				
	200 OK BYE(RLC)	+			+	RELEASE				
	487 Request Terminated	+			→	RELEASE COMPLETE				
	ACK	→								

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	Q.1912.5 [1], clause 7.1 1 a)				
TSS reference	ISDN-SIP/Basic call/Sending of the INVITE message								
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 and the sending complete indication: • sends the INVITE message.								
SIP parameter values	<u> </u>								
ISDN parameter values	SETUP; Called party nur	nber:	with send co	mplete indi	cation				
Comments	ISDN		SUT		SIP-I				
	SETUP	→		→	INVITE(IAM)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONNECT	+		←	200 OK INVITE(ANM)				
				→	ACK				
			Conversat	tion					
	DISCONNECT	→	•	→	BYE(REL)				
	RELEASE	+		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	→							

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

TP601003	SIP reference: RF	C 3261	[4]	ISDN/ISDN reference: Q.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	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 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		-							
ISDN parameter values	SETUP; Called party nur	mber: s	ufficient nui	mber of dig	its to route to the called party				
Comments	ISDN		SUT		SIP-I				
	SETUP	→		→	INVITE(IAM)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONNECT	+		+	200 OK INVITE(ANM)				
				→	ACK				
	Conversation								
	DISCONNECT	→		→	BYE(REL)				
	RELEASE	+		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	→							

TP601004	SIP reference: RF0	C 3261	[4]			ISDN/ISDN reference:				
			• •		Q.	.1912.5 [1], clause 7.1 1 d)				
TSS reference	ISDN-SIP/Basic call/Sending of the INVITE message									
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	→			→	INVITE(IAM)				
	ALERTING	←			←	180 Ringing(ACM)				
	CONNECT	←			←	200 OK INVITE(ANM)				
					→	ACK				
			Conversat	tion						
	DISCONNECT	CONNECT → BYE(REL)								
	RELEASE	+			+	200 OK BYE(RLC)				
	RELEASE COMPLETE	→								

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 en	capsulated i	in a N	IIME-body					
Comments	ISDN		SUT		SIP-I					
	SETUP	→		→	INVITE(IAM)					
	ALERTING	←		+	180 Ringing(ACM)					
	CONNECT	+		+	200 OK INVITE(ANM)					
				→	ACK					
		Con	versation							
	DISCONNECT	→		→	BYE(REL)					
	RELEASE	←		+	200 OK BYE(RLC)					
	RELEASE COMPLETE	→								

Table 28

				Values for t	test purpose TP301005						
VA		IS	DN		SDP - a_b_m_LINE_VALUE						
		BC para	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></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)		
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: RFC 3261 [4]			ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2					
TSS reference									
SIP selection	ISDN-SIP/Basic call/Sending of the INVITE message								
criteria									
· · · · · · · · · ·									
ISDN selection									
criteria 									
Test purpose				y addres	ss information contained in the				
	Called Party Number para								
					which shall include the "user=phone"				
	URI parameter if the	To header f	ield contair	ns a sip:	URI.				
SIP parameter	INVITE: To: sip:; user:	=phone							
values									
ISDN parameter									
values									
Comments	ISDN		SUT		SIP-I				
	SETUP	→		→	INVITE(IAM)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONNECT	+		+	200 OK INVITE(ANM)				
				→	ACK				
		Co	-						
	DISCONNECT	→		→	BYE(REL)				
	RELEASE	+		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	→	· · · · · · · · · · · · · · · · · · ·						

TP601007	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	Called Party Number para	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter of the SETUP and the and the followed INFO: to the addr-spec component of the To header field.							
SIP parameter values									
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	→							
	INFO	→							
	INFO	→		1	INVITE(IAM)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONNECT	+		+	200 OK INVITE(ANM)				
				7	ACK				
			Conversa	tion					
	DISCONNECT	→		→	BYE(REL)				
	RELEASE	+		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	→							

TP601008	SIP reference: RF	C 3261	[4]	ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2				
TSS reference	ISDN-SIP/Basic call/Send	ling of th	ne INVITE r	nessage				
SIP selection criteria								
ISDN selection criteria								
Test purpose	Called Party address informationto the addr-spec community	Ensure that the SUT is mapping in the Called Party Number parameter contained in the Called Party address information of the SETUP and followed INFO: to the addr-spec component of the To header field which shall include the "user=phone" URI parameter if the To header field contains a sip: URI.						
SIP parameter values	INVITE: To: sip:; user=				·			
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	→						
	INFO	→						
	INFO	→		→	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONNECT	+		+	200 OK INVITE(ANM)			
				→	ACK			
			Conversat	tion				
	DISCONNECT	→		→	BYE(REL)			
	RELEASE	+		+	200 OK BYE(RLC)			
	RELEASE COMPLETE	→						

TP601009	SIP reference: RF	C 3261	[4]		ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2	
TSS reference	ISDN-SIP/Basic call/Send	ling of th	ne INVITE r	nessage		
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 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						
ISDN parameter values						
Comments	ISDN		SUT		SIP-I	
	SETUP	→		→	INVITE(IAM)	
	ALERTING	+		+	180 Ringing(ACM)	
	CONNECT	+		+	200 OK INVITE(ANM)	
				→	ACK	
			Conversat	tion		
	DISCONNECT	→	_	→	BYE(REL)	
	RELEASE	+		+	200 OK BYE(RLC)	
	RELEASE COMPLETE	→	_			

TP601010	SIP reference: RF	C 3261 [4	i]		ISDN/ISDN reference:			
TSS reference	Q.1912.5 [1], clause 7.1.2 ISDN-SIP/Basic call/Sending of the INVITE message							
SIP selection	TODIV-OII /Basic call/Geria	ing or the		icssage				
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	·							
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	→		→	INVITE(IAM)			
	ALERTING	←		←	180 Ringing(ACM)			
	CONNECT	+		←	200 OK INVITE(ANM)			
				→	ACK			
		(Conversation	on				
	DISCONNECT	→		→	BYE(REL)			
	RELEASE	+		←	200 OK BYE(RLC)			
	RELEASE COMPLETE	→						

TP601011	SIP reference: RF	C 3261 [4]		(ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2	
TSS reference	ISDN-SIP/Basic call/Send	ing of the IN	IVITE mess	age		
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						
ISDN parameter values						
Comments	ISDN		SUT		SIP-I	
	SETUP	→		→	INVITE(IAM)	
	ALERTING	←		+	180 Ringing(ACM)	
	CONNECT	←		+	200 OK INVITE(ANM)	
				→	ACK	
		Cor	nversation			
	DISCONNECT	→		→	BYE(REL)	
	RELEASE	+		+	200 OK BYE(RLC)	
	RELEASE COMPLETE	→				

TP601012	SIP reference: RF	C 3261 [4]		ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2			
TSS reference	ISDN-SIP/Basic call/Send	ling 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							
ISDN parameter							
values							
Comments	ISDN	SUT	-	SIP-I			
	SETUP	→					
	INFO	→					
	INFO	→	→	INVITE(IAM)			
	ALERTING	+	+	180 Ringing(ACM)			
	CONNECT	+	+	200 OK INVITE(ANM)			
			→	ACK			
		Convers	ation				
	DISCONNECT	→	→	BYE(REL)			
	RELEASE	+	+	200 OK BYE(RLC)			
	RELEASE COMPLETE	→					

TP601013	SIP reference: RF	C 3261 [4]		ISDN/ISDN reference:				
TSS reference	Q.1912.5 [1], clause 7.1.2 ISDN-SIP/Basic call/Sending of the INVITE message							
SIP selection		ing or the invitic	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								
values								
ISDN parameter values								
Comments	ISDN	SUT		SIP-I				
	SETUP	→						
	INFO	→						
	INFO	→	→	INVITE(IAM)				
	ALERTING	←	←	180 Ringing(ACM)				
	CONNECT	+	+	200 OK INVITE(ANM)				
			→	ACK				
		Conversa	ation					
	DISCONNECT	→	→	BYE(REL)				
	RELEASE	+	+	200 OK BYE(RLC)				
	RELEASE COMPLETE	→						

TP601014	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 = "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 values									
ISDN parameter values									
Comments	ISDN		SUT			SIP-I			
	SETUP	→							
	INFO	→							
	INFO	→		-	>	INVITE(IAM)			
	ALERTING	+		•	(180 Ringing(ACM)			
	CONNECT	+		•	(200 OK INVITE(ANM)			
				-	>	ACK			
			Conversa	tion					
	DISCONNECT	→		-	>	BYE(REL)			
	RELEASE	+		•	(200 OK BYE(RLC)			
	RELEASE COMPLETE	→							

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	nding					
SIP selection criteria	PICS 3/1							
ISDN selection criteria								
Test purpose		Ensure if the SUT is supporting en bloc addressing towards the SIP network, subsequent NFOs received after the SUT has sent the INVITE are ignored.						
SIP parameter values		J. J						
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	→		→	INVITE(IAM)			
	INFO	→						
	ALERTING	+		+	180 Ringing(ACM)			
	CONNECT	+		+	200 OK INVITE(ANM)			
		→ ACK						
			Conversat	tion				
	DISCONNECT	→		→	BYE(REL)			
	RELEASE	+		+	200 OK BYE(RLC)			
	RELEASE COMPLETE	→						

TP602002	SIP reference: RF	C 326	1 [4]		ISDN/ISDN reference: Q.1912.5 [1], clause 7.2.1		
TSS reference	ISDN-SIP/Basic call/Over	lap se	nding				
SIP selection criteria	PICS 3/2		-				
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 resource parameters in que	INFO is rur ted are and the this continued the life sent. In the life sent sent life sent. In the life sent life sent. In the life sent life	from the ISDN nning); nd the SUT shate To header find. NFOe Call-ID and ontain a new Stady been reset to be reflected with the ISDN NFOE Call-ID and the ISDN not shate the ISDN not sha	I access all invoke eld of the and From DP offer. rved for t vithin the			
SIP parameter values ISDN parameter							
values							
Comments	ISDN		SUT		SIP-I		
	SETUP	→		→	INVITE(IAM)		
	INFO	→		→	INVITE(IAM)		
	INFO	→		→	INVITE(IAM)		
	ALERTING	+		+	180 Ringing(ACM)		
	CONNECT	+		-	200 OK INVITE(ANM)		
	→ ACK						
			Conversation	on	-		
	DISCONNECT	→		<u>→</u>	BYE(REL)		
	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	nding			
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	→		→	INVITE(IAM)	
	INFO	→		→	INVITE(IAM)	
			T _{oiw2} expi	red		
	INFO	→				
	ALERTING	+		+	180 Ringing(ACM)	
	CONNECT	+		+	200 OK INVITE(ANM)	
				→	ACK	
			Conversat	ion		
	DISCONNECT	→		→	BYE(REL)	
	RELEASE	+		+	200 OK BYE(RLC)	
	RELEASE COMPLETE	→				

TP602004	SIP reference: R	EC 2264 [41		ISDN/ISDN reference:		
17002004	SIF reference. K	FC 3201 [4]		Q.1912.5 [1], clause 7.2.1		
TSS reference	ISDN-SIP/Basic call/Ove	erlan send	ina	`	Q. 1312.0 [1], 010030 1.2.1		
SIP selection	PICS 3/1	map coma	9				
criteria							
ISDN selection							
criteria							
Test purpose	 The SUT in Idle state, on receipt of a SETUP message. On receipt of a INFO from the BICC/ISDN the SUT shall: 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; ensure that if timer TOIW2 has expired, subsequent INFOs received after the SUT has sent the INVITE are ignored. 						
SIP parameter values	30.11.11.12.41.0	.g					
ISDN parameter							
values							
Comments	ISDN		SUT		SIP-I		
	SETUP	→					
	INFO	→		→	INVITE(IAM)		
			T _{oiw2} expire	ed			
	INFO	→					
	ALERTING	←		+	180 Ringing(ACM)		
	CONNECT	+		+	200 OK INVITE(ANM)		
				→	ACK		
	DIGGONNEGT		Conversation		D)(E(DEL)		
	DISCONNECT	→		→	BYE(REL)		
	RELEASE	-		+	200 OK BYE(RLC)		
	RELEASE COMPLETE	→					

TP602005	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.2.1	
TSS reference	ISDN-SIP/Basic call/Over	lap sei	nding			
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	→		→	INVITE(IAM)	
				+	404/484	
				→	ACK	
	INFO	→		→	INVITE(IAM)	
				+	404/484	
				→	ACK	
	INFO	→		→	INVITE(IAM)	
	111.0	+ -		+	404/484	
				→	ACK	
	INFO	→		→	INVITE(IAM)	
	ALERTING	←		+	180 Ringing(ACM)	
	CONNECT	+		+	200 OK INVITE(ANM)	
				→	ACK	
			Conversat	tion		
	DISCONNECT	→		→	BYE(REL)	
	RELEASE	+		+	200 OK BYE(RLC)	
	RELEASE COMPLETE	→			. ,	

TP602006	SIP reference: RFC 32	261 [4]			DN/ISDN reference: 912.5 [1], clause 7.7.6					
TSS reference	ISDN-SIP/Basic call/Overlap	ISDN-SIP/Basic call/Overlap sending								
SIP selection criteria	NOT PICS 3/2									
ISDN selection criteria										
Test purpose	Ensure that the SUT after received that the SUT before having remessage (4xx, 5xx, 6xx) defines sends a DISCONNECT of	eceived ned as	d an backward SIP_Failure_V	messa 'A:	•					
SIP parameter values	SIP_Failure_VA: ISUP REL 6	encaps	ulated in the M	IME bo	dy					
ISDN parameter values	DISCONNECT/RELEASE: C	ause v	alue constructe	ed from	the encapsulated REL					
Comments	ISDN		SUT		SIP-I					
	SETUP	→		→	INVITE(IAM)					
				+	484 Address Incomplete					
	CASE A			→	ACK					
	RELEASE	+								
	RELEASE COMPLETE	→								
	CASE B									
	DISCONNECT	+								
	RELEASE	→								
	RELEASE COMPLETE	+								

A.1.1.2.3 Sending of the CALL PROCEEDING / ALERTING message

TP603001	SIP reference: RF	C 326	1 [4]	Q.191	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> </u>								
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	→		→	INVITE(IAM)					
	CALL PROCEEDING	+		+	183 Session Progress(ACM)					
	PROGRESS(PI)	+		+	183 Session Progress(CPG)					
	ALERTING	+		+	180 Ringing(ACM					
	CONNECT	+		+	200 OK INVITE(ANM)					
				→	ACK					
		1	Conversa							
	DISCONNECT	→		→	BYE(REL)					
	RELEASE	+		←	200 OK BYE(RLC)					
	RELEASE COMPLETE	→								

TP603002	SIP reference: RF0	C 326	1 [4]	Q.1	912	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 criteria	PICS 3/1	<u> </u>				-		
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 values	183 Session Progress with with PI	n enca	apsulated AC	M: BCi C	alle	ed party status = no indication, ATP		
ISDN parameter values	CALL PROCEEDING(PI v	alue=	PI_VAL)					
Comments	ISDN		SUT			SIP-I		
	SETUP	→		-3	>	INVITE(IAM)		
	CALL PROCEEDING	+		•	Т.	183 Session Progress(ACM)		
	ALERTING	+		•		180 Ringing(ACM)		
	CONNECT	+		•	-	200 OK INVITE(ANM)		
				-	•	ACK		
			Conversat	tion				
	DISCONNECT	→		-3	•	BYE(REL)		
	RELEASE	+		•		200 OK BYE(RLC)		
	RELEASE COMPLETE	→						

TP603003	SIP reference: RFC 3261 [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 _{OIW2} sends the CALL PROCEEDING message.							
SIP parameter								
values								
ISDN parameter	CALL PROCEEDING							
values								
Comments	ISDN		SUT		SIP-I			
	SETUP	→						
	INFO	→		→	INVITE(IAM)			
			T _{oiw2} expired	d				
	CALL PROCEEDING	+						
	ALERTING	+		-	180 Ringing(ACM)			
	CONNECT	+		+	200 OK INVITE(ANM)			
				→	ACK			
			Conversat	ion				
	DISCONNECT	→		→	BYE(REL)			
	RELEASE	+		+	200 OK BYE(RLC)			
	RELEASE COMPLETE	→						

TP603004	SIP reference: RF	C 3261 [4	4]	0 101	ISDN/ISDN reference:			
TSS reference	ISDN-SIP/Basic call/Send	ling of C	ALL DDO		2.5 [1], clauses 7.1 1) a), 7.3.1			
SIP selection criteria	PICS 3/1	iiiig_oi_c	ALL FRO	CEDING_A	LENTING			
ISDN selection criteria								
Test purpose	called party number and message:	sends the ALERTING message with the with the with progress description value						
SIP parameter values	180 Ringing encapsulated Progress indicator PI_VAI		Ci Called	party status	s = subscriber free, ATP with			
ISDN parameter values	ALERTING: Progress indi	icator valu	ue PI_VAI	_ included				
Comments	ISDN		SUT		SIP-I			
	SETUP	→		→	INVITE(IAM)			
	CALL PROCEEDING	+						
	ALERTING	+		(180 Ringing(ACM(PI))			
	CONNECT	+		←	200 OK INVITE(ANM)			
				→	ACK			
		(Conversat	ion				
	DISCONNECT	→		→	BYE(REL)			
	RELEASE	+	-	+	200 OK BYE(RLC)			
	RELEASE COMPLETE	→						

TP603005	SIP reference: RF	C 3261 [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 _{OIW2} after the receipt of the latest address message on receipt of a 183 Session Progress with encapsulated ACM: • a PROGRESS is sent backward.							
SIP parameter	183 Session Progress en	capsulated	ACM: BCi Ca	alled p	arty status = no indication, ATP with			
values	Progress indicator			г				
ISDN parameter	PROGRESS							
values								
Comments	ISDN		SUT		SIP-I			
	SETUP	→						
	INFO	→		→	INVITE(IAM)			
		T _c	_{oiw2} expired					
	CALL PROCEEDING	+						
	PROGRESS	+		+	183 Session Progress(ACM(PI))			
	ALERTING	+		+	180 Ringing(CPG)			
	CONNECT	+		+	200 OK INVITE(ANM)			
			·	→	ACK			
		Co	onversation	,				
	DISCONNECT	→	·	→	BYE(REL)			
	RELEASE	+		+	200 OK BYE(RLC)			
	RELEASE COMPLETE	→						

TP603006	SIP reference: RF	C 3261	[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 _{OIW2} 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	capsula	ted ACM: B	Ci Called ր	party status = no indication, without				
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	→							
	INFO	→		→	INVITE(IAM)				
			T _{oiw2} expi	red					
				+	183 Session Progress(ACM)				
	ALERTING	+		+	180 Ringing(ACM				
	CONNECT	+		+	200 OK INVITE(ANM)				
				→	ACK				
			Conversat	ion					
	DISCONNECT	→		→	BYE(REL)				
	RELEASE	+		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	→							

TP603007	SIP reference: RF	C 326	61 [4]		ISDN/ISDN reference:				
				Q	.1912.5 [1], clause 7.1 1 a)				
TSS reference	ISDN-SIP/Basic call/Send	ISDN-SIP/Basic call/Sending_of_CALL PROCEDING_ALERTING							
SIP selection	PICS 3/2								
criteria									
ISDN selection									
criteria									
Test purpose	called party number who timer T _{OIW2} after the rece Progress:	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 _{OIW2} after the receipt of the latest address message on receipt of a 183 Session Progress:							
	no information is sent								
SIP parameter	183 Session Progress wit	hout e	encapsulated	ISUP mess	sage				
values									
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	→							
	INFO	→		→	INVITE(IAM)				
			T _{oiw2} expi	red					
				←	183 Session Progress				
	ALERTING	+		←	180 Ringing(ACM				
	CONNECT	+		+	200 OK INVITE(ANM)				
				→	ACK				
			Conversat	tion					
	DISCONNECT	→		→	BYE(REL)				
	RELEASE	+		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	→							

A.1.1.2.4 Sending of the CONNECT message

TP604001	SIP reference: RF	C 3261 [4]		ISDN/ISDN reference: Q.1912.5 [1], clause 7.5
TSS reference	ISDN-SIP/Basic call/Send	ling_of_CONNE	CT	
SIP selection criteria				
ISDN selection criteria				
Test purpose	Ensure that the SUT havi for this call, it shall stop til send CONNECT as control stop any existing away.	mer T _{OIW2} (if ru determined by I	nning): SDN procedure	
SIP parameter values	200 OK INVITE: encapsu	lated ANM in th	e MIME body	
ISDN parameter values	CONNECT			
Comments	ISDN	SI	JT	SIP-I
	SETUP	→	→	INVITE(IAM)
	ALERTING	+	+	180 Ringing(ACM
	CONNECT	+	+	200 OK INVITE(ANM)
			→	ACK
		Conve	rsation	
	DISCONNECT	→	→	BYE(REL)
	RELEASE	+	+	200 OK BYE(RLC)
	RELEASE COMPLETE	→		

TP604002	SIP reference: RF	C 326	1 [4]			ISDN/ISDN reference: Q.1912.5 [1], clause 7.5		
TSS reference	ISDN-SIP/Basic call/Send	ing of	CONNECT			[1], 0.0000		
SIP selection criteria		<u> </u>						
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 _{OIW2} (if running): send CON as determined by ISDN procedures; Stop any existing awaiting answer indication (e.g. ringing tone).							
SIP parameter	200 OK INVITE: encapsul					,		
values								
ISDN parameter	CONNECT							
values								
Comments	ISDN		SUT			SIP-I		
	SETUP	→			→	INVITE(IAM)		
	CONNECT	+			←	200 OK INVITE(CON)		
					→	ACK		
			Conversat	ion				
	DISCONNECT	→			→	BYE(REL)		
	RELEASE	(←	200 OK BYE(RLC)		
	RELEASE COMPLETE	→						

A.1.1.2.5 Receipt of the RELEASE or DISCONNECT

TP605001	SIP reference: RF	C 3261	1 [4]		ISDN/ISDN reference Q.1912.5 [1], clause 7.7.				
TSS reference	ISDN-SIP/Basic call/Receipt_of_RELEASE_or_DISCONNECT								
SIP selection									
criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT after receipt of a RELEASE CC no action is required of in progress.	MPLE	TĔ messag	e:	efore an INVITE has beer o terminate local procedur				
SIP parameter values									
ISDN parameter values	RELEASE COMPLETE; cause value: (PIXIT), location (PIXIT)								
Comments	ISDN		SUT		SIP-I	•			
	SETUP	→	•						
	RELEASE COMPLETE	→							

TP605002	SIP reference: RF	C 3261	[4]		ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1, 1)				
TSS reference	ISDN-SIP/Basic call/Rece	call/Receipt_of_RELEASE_or_DISCONNECT							
SIP selection criteria									
ISDN selection criteria									
Test purpose	receipt of a RELEASE me	essage:	Ü		efore an INVITE has been sent. On o terminate local procedures if any are				
SIP parameter values									
ISDN parameter values	RELEASE; cause value:	(PIXIT)	location (PIXIT)					
Comments	ISDN	SUT SIP-I							
	SETUP	→							
	RELEASE	→							
	RELEASE COMPLETE	+							

TP605003	SIP reference: RF0	C 326	1 [4]		Q.	ISDN/ISDN reference: 1912.5 [1], clause 7.7.1, 1)		
TSS reference	ISDN-SIP/Basic call/Recei	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: 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	DISCONNECT; cause value: (PIXIT), location (PIXIT)							
Comments	ISDN		SUT			SIP-I		
	SETUP →							
	DISCONNECT	DISCONNECT →						
	RELEASE	+						
	RELEASE COMPLETE →							

TP605004	SIP reference: RF	C 3261 [4]	0	ISDN/ISDN reference: .1912.5 [1], clause 7.7.1 2)			
TSS reference	\(\text{\lambda} \) \(\text{\lambda} \)						
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 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.						
SIP parameter values							
ISDN parameter values	RELEASE COMPLETE; cause value: (PIXIT), location (PIXIT)						
Comments	ISDN	SUT		SIP-I			
	SETUP → INVITE(IAM)						
	RELEASE COMPLETE →						
	← 200 OK INVITE(ANM)						
			→	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	<u>ipt_of_RELEASE_</u>	or_DISCON	INECT			
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 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	RELEASE; cause value: (PIXIT), location (PIXIT)						
Comments	ISDN	SUT		SIP-I			
	SETUP	→	→	INVITE(IAM)			
	RELEASE	→					
	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/Rece	ipt_of_RELEASE_	or_DISCON	INECT		
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 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 va	lue: (PIXIT), locat	ion (PIXIT)			
Comments	ISDN	SUT		SIP-I		
	SETUP	→	→	INVITE(IAM)		
	DISCONNECT	→				
	RELEASE					
	RELEASE COMPLETE → 200 OK INVITE(ANM)					
	→ ACK					
			→	BYE(REL)		
			←	200 OK BYE(RLC)		

TP605007	SIP reference: RF	C 3261 [4]	ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 2) 3)			
TSS reference	ISDN-SIP/Basic call/Rece	eipt_of_RELEASE_	or_DISCON	NECT		
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 ISDN message before a 200 OK SIP 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. 					
SIP parameter values						
ISDN parameter values	RELEASE COMPLETE; c	ause value: (PIXI	T), location	ı (PIXIT)		
Comments	ISDN	SUT		SIP-I		
	SETUP	→	→	INVITE(IAM)		
			+	100 Trying		
	RELEASE COMPLETE	→				
			→	CANCEL(REL)		
	 ← 200 OK INVITE(ANM) → ACK ← 200 OK CANCEL → BYE(REL) 					
			+	200 OK BYE(RLC)		

TP605008	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	NOT PICS 4/10					
criteria						
ISDN selection						
criteria						
Test purpose	sending an INVITE messa SIP response message ha	age. On as been	receipt of a received:	RELEASE	e complete called party number, E message ISDN before a 200 OK	
	 the SUT shall hold the was been received; 	e releas	e procedur	e. A CANC	EL is sent when any SIP response	
					les, 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	→		→	INVITE(IAM)	
				+	100 Trying	
	RELEASE	→				
	RELEASE COMPLETE ← → CANCEL(REL)					
	← 200 OK INVITE(ANM)					
				→	ACK	
				+	200 OK CANCEL	
				→	BYE(REL)	
		→		+	200 OK BYE(RLC)	

TP605009	SIP reference: RF	C 3261 [4]		ISDN/ISDN reference:		
	Q.1912.5 [1], clause 7.7.1 2) 3)					
TSS reference	ISDN-SIP/Basic call/Rece	ipt_of_RELEAS	_or_DISCO	NNECT		
SIP selection	NOT PICS 4/10					
criteria						
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 DISCONNECT message ISDN before a 200 OK SIP 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. 					
values						
ISDN parameter values	DISCONNECT; cause va	lue: (PIXIT), loc	ation (PIXIT)			
Comments	ISDN	SU	Τ	SIP-I		
	SETUP	→	→	INVITE(IAM)		
			+	100 Trying		
	DISCONNECT	→				
	RELEASE ← CANCEL(REL)					
	RELEASE COMPLETE →					
	→ ACK					
			+	200 OK CANCEL		
			→	BYE(REL)		
			+	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/Rece	eipt_of_	_RELEASE_	or_DISCON	INECT
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 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 COMPLETE; cause value: (PIXIT), location (PIXIT)				
Comments	ISDN		SUT		SIP-I
	SETUP	→		→	INVITE(IAM)
				+	100 Trying
	RELEASE COMPLETE	→			
				←	SIP_MESSAGE_VA
				CAS	E A
				→	CANCEL(REL)
				←	200 OK CANCEL
				←	487 Request Terminated
				→	ACK
	CASE B				
				→	BYE(REL)
				+	200 OK BYE(RLC)
				+	487 Request Terminated
				→	ACK

TP605011	SIP reference: RF	C 3261 [4]	ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 2) 3)					
TSS reference	ISDN-SIP/Basic call/Rece	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	→	→	INVITE(IAM)				
			+	100 Trying				
	RELEASE	→						
	RELEASE COMPLETE	←						
			+	SIP_MESSAGE_VA				
			CAS	SE A				
			→	CANCEL(REL)				
			←	200 OK CANCEL				
			←	487 Request Terminated				
			→	ACK				
			CAS	_ =				
			→	BYE(REL)				
			←	200 OK BYE(RLC)				
			←	487 Request Terminated				
			→	ACK				

TP605012	SIP reference: RFC 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	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: (P	PIXIT), locati	ion (PIXIT)			
Comments	ISDN		SUT		SIP-I		
	SETUP	→		→	INVITE(IAM)		
				+	100 Trying		
	DISCONNECT	→					
	RELEASE	+					
	RELEASE COMPLETE	→					
				+	SIP MESSAGE VA		
	CASE A						
				→	CANCEL(REL)		
				+	200 OK CANCEL		
				+	487 Request Terminated		
				→	ACK		
	CASE B						
				→	BYE(REL)		
				+	200 OK BYE(RLC)		
				+	487 Request Terminated		
				→	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_of_l	RELEASE_	or_DISC	ON	INECT		
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 v	alue: (PIXI ⁻	Γ), locat	ion	(PIXIT)		
Comments	ISDN		SUT			SIP-I		
	SETUP	→		•	→	INVITE(IAM)		
	ALERTING	+		•	(180 Ringing(ACM		
	CONNECT	+		•	(200 OK INVITE(ANM)		
	RELEASE COMPLETE	→						
					→	ACK		
					→	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/Rece	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 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	→	→	INVITE(IAM)					
	ALERTING	-	←	180 Ringing(ACM					
	CONNECT	-	+	200 OK INVITE(ANM)					
	RELEASE	RELEASE →							
	RELEASE COMPLETE	RELEASE COMPLETE ← → ACK							
			→	BYE(REL)					
			←	200 OK BYE(RLC)					

TP605015	SIP reference: RF	C 3261 [4]	Q	ISDN/ISDN reference: .1912.5 [1], clause 7.7.1 4)					
TSS reference	ISDN-SIP/Basic call/Rece	ipt_of_RELEASE_	or_DISCON	NECT					
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 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	DISCONNECT; cause va	lue: (PIXIT), locat	tion (PIXIT)						
Comments	ISDN	SUT		SIP-I					
	SETUP	→	→	INVITE(IAM)					
	ALERTING	+	+	180 Ringing(ACM					
	CONNECT	←	←	200 OK INVITE(ANM)					
	DISCONNECT	→							
	RELEASE	RELEASE ← → ACK							
	RELEASE COMPLETE	→	→	BYE(REL)					
			+	200 OK BYE(RLC)					

TP605016	SIP reference: RF	C 3261	1 [4]	Q	ISDN/ISDN reference: 1.1912.5 [1], clause 7.7.1 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 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 COMPLETE; c	ause	value: (PIXI	T), location	n (PIXIT)			
Comments	ISDN		SUT		SIP-I			
	SETUP	→		→	INVITE(IAM)			
				+	SIP_MESSAGE_VA			
	RELEASE COMPLETE	→						
	CASE A							
				→	CANCEL(REL)			
				←	200 OK CANCEL			
				+	487 Request Terminated			
				→	ACK			
	CASE B							
				→	BYE(REL)			
				+	200 OK BYE(RLC)			
				+	487 Request Terminated			
				→	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)	, location (PIXIT)			
Comments	ISDN		SUT			SIP-I	
	SETUP	→		-	>	INVITE(IAM)	
				€	_	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)	
				€		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/Rece	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: (Pl	XIT), locati	ion (PIXIT)			
Comments	ISDN		SUT		SIP-I		
	SETUP	→		→	INVITE(IAM)		
				+	SIP_MÈSSÁGE_VA		
	DISCONNECT	→					
	RELEASE	+					
	RELEASE COMPLETE	→					
	CASE A						
				→	CANCEL(REL)		
				+	200 OK CANCEL		
				+	487 Request Terminated		
				→	ACK		
	CASE B						
				→	BYE(REL)		
				+	200 OK BYE(RLC)		
				+	487 Request Terminated		
				→	ACK		

TP605019	SIP reference: RF	C 3261	[4]	ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 2) 4)			
TSS reference	ISDN-SIP/Basic call/Rece	ipt_of_R	RELEASE_	or_DISC	ON	INECT	
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 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	e CV_SI	P, encapsu	ılated RI	EL (constructed from the ISDN	
ISDN parameter values	RELEASE COMPLETE; c	ause va	alue: (PIXI	Γ), locat	ion	(PIXIT)	
Comments	ISDN		SUT			SIP-I	
	SETUP	→			→	INVITE(IAM)	
	RELEASE COMPLETE	→		İ			
				•	(200 OK INVITE(ANM)	
					→	ACK	
					→	BYE(REL)	
					(200 OK BYE(RLC)	

TP605020	SIP reference: RF0	C 3261 [4]	Q.	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 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 values	BYE:Reason header value RELEASE	e CV_SIP, enca	sulated REL	constructed from the ISDN				
ISDN parameter values	RELEASE; cause value:	(PIXIT), location	ı (PIXIT)					
Comments	ISDN	SU	Γ	SIP-I				
	SETUP	→	→	INVITE(IAM)				
	RELEASE	→						
	RELEASE COMPLETE	+						
			+	200 OK INVITE(ANM)				
			→	ACK				
			→	BYE(REL)				
			+	200 OK BYE(RLC)				

TP605021	SIP reference: RF0	C 3261	[4]	0	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	PICS 4/10	ipt_oi_i	KELE/KOE_	01_01000	INICOT			
criteria	1.00 1,10							
ISDN selection								
criteria								
SIP parameter values ISDN parameter	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message on receipt DISCONNECT 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. BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN DISCONNECT DISCONNECT; cause value: (PIXIT), location (PIXIT)							
values					1			
Comments	ISDN		SUT		SIP-I			
	SETUP	→		→	INVITE(IAM)			
	ALERTING	+		-	180 Ringing(ACM			
	CONNECT	+		+	200 OK INVITE(ANM)			
				→	ACK			
		Conversation						
	DISCONNECT	→		→	BYE(REL)			
	RELEASE	+		←	200 OK BYE(RLC)			
	RELEASE COMPLETE	→						

TP605022	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	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		e CV_SIP	, encapsı	ulated REI	constructed from the ISDN				
values	RELEASE COMPLETE		(51) (1		(DI) (IT)				
ISDN parameter values	RELEASE COMPLETE; c	ause valu	ue: (PIXI	I), locatio	n (PIXII)				
Comments	ISDN		SUT		SIP-I				
	SETUP	→		→					
				←	100 Trying				
	RELEASE COMPLETE	→							
				→					
				←	` '				
				→					
				←					
				→	-:=(::==)				
				←	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), location	(PIXIT)				
Comments	ISDN	SUT		SIP-I			
	SETUP	→	→	INVITE(IAM)			
			+	100 Trying			
	RELEASE	→					
	RELEASE COMPLETE	←	→	CANCEL(REL)			
			+	200 OK INVITE(ANM)			
			→	ACK			
			+	200 OK CANCEL			
			→	BYE(REL)			
		→	(200 OK BYE(RLC)			

TP605024	SIP reference: RF	C 3261	1 [4]		Q .1	ISDN/ISDN reference: 1912.5 [1], clause 7.7.1 2) 3
TSS reference	ISDN-SIP/Basic call/Rece	ipt_of_	_RELEASE_	or_DISC	ON	INECT
SIP selection criteria	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 DISCONNECT message with Cause value CV_ISDN, location LOC_ISDN 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. BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN					
values	DISCONNECT	# C v	Sir, encapsi	ulateu KE	<u> </u>	constructed from the ISDN
ISDN parameter values	DISCONNECT; cause va	lue: (F	PIXIT), locat i	ion (PIXI	T)	
Comments	ISDN		SUT			SIP-I
	SETUP	→			→	INVITE(IAM)
				•		100 Trying
	DISCONNECT	→				
	RELEASE	+			→	CANCEL(REL)
	RELEASE COMPLETE	→				200 OK INVITE(ANM)
					>	ACK
						200 OK CANCEL
					>	BYE(REL)
				•	<u> </u>	200 OK BYE(RLC)

TP605025	SIP reference: RF	C 3261	l [4]	Q	ISDN/ISDN reference: .1912.5 [1], clause 7.7.1 3)	
TSS reference	ISDN-SIP/Basic call/Rece	ipt_of_	RELEASE_	or_DISCON	NECT	
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	BYE:Reason header valu	e CV_S	SIP, encaps	ulated REL	constructed from the ISDN	
values	RELEASE COMPLETE					
ISDN parameter values	RELEASE COMPLETE; c	ause	value: (PIXI	T), location	ı (PIXIT)	
Comments	ISDN		SUT		SIP-I	
	SETUP	→		→	INVITE(IAM)	
				+	100 Trying	
	RELEASE COMPLETE	→				
				+	SIP_MESSAGE_VA	
	CASE A					
				→	CANCEL(REL)	
				+	200 OK CANCEL	
				+	487 Request Terminated	
				→	ACK	
	CASE B	1 1				
				→	BYE(REL)	
		1 1		+	200 OK BYE(RLC)	
		1 1		+	487 Request Terminated	
				→	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/Rece	eipt_of	_RELEASE_	or_DISCON	NECT		
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 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		e CV_	SIP, encaps	ulated REL	constructed from the ISDN		
ISDN parameter values	RELEASE; cause value:	(PIXI7	「), location ((PIXIT)			
Comments	ISDN		SUT		SIP-I		
	SETUP	→		→	INVITE(IAM)		
				-	100 Trying		
	RELEASE	→					
	RELEASE COMPLETE	+					
				-	SIP_MESSAGE_VA		
	CASE A						
				→	CANCEL(REL)		
				+	200 OK CANCEL		
				←	487 Request Terminated		
				→	ACK		
	CASE B						
				→	BYE(REL)		
				+	200 OK BYE(RLC)		
				+	487 Request Terminated		
				→	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/Rece	eipt_of_	_RELEASE_	or_DISCO	DNNECT	
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 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	BYE:Reason header valu	e CV_	SIP, encaps	ulated RE	L constructed from the ISDN	
values	DISCONNECT					
ISDN parameter	DISCONNECT; cause va	ilue: (F	PIXIT), locat	ion (PIXIT	-)	
values Comments	IODN		OUT		OID I	
Comments	ISDN	→	SUT	-	SIP-I	
	SETUP	7		- 		
	DIOCOMMENT	+			100 Trying	
	DISCONNECT	→				
	RELEASE	+				
	RELEASE COMPLETE	→			OID MEGGAGE VA	
	0.405.4			•	SIP_MESSAGE_VA	
	CASE A					
				→	0, (0 = 1)	
				•		
				+	101 110 401001 101111111010	
					ACK	
	CASE B					
				→	()	
				+		
				+		
				→	ACK	

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	ipt_of_RELEAS	E_or_DISC	ON	NECT	
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 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 RELEASE COMPLETE	e CV_SIP, enca	psulated RE	EL c	constructed from the ISDN	
ISDN parameter values	RELEASE COMPLETE; C	ause value: (P	XIT), locati	ion	(PIXIT)	
Comments	ISDN	SI	Т		SIP-I	
	SETUP	→	-	}	INVITE(IAM)	
	ALERTING	+	•	(180 Ringing(ACM	
	CONNECT	-	•	(200 OK INVITE(ANM)	
	RELEASE COMPLETE	→				
			•	→	ACK	
			-		BYE(REL)	
			•	(200 OK BYE(RLC)	

TP605029	SIP reference: RF	C 3261 [4]	Q	ISDN/ISDN reference: .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 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 RELEASE	e CV_SIP, encaps	sulated REL	constructed from the ISDN			
ISDN parameter values	RELEASE; cause value:	(PIXIT), location	(PIXIT)				
Comments	ISDN	SUT		SIP-I			
	SETUP	→	→	INVITE(IAM)			
	ALERTING	+	+	180 Ringing(ACM			
	CONNECT	←	+	200 OK INVITE(ANM)			
	RELEASE →						
	RELEASE COMPLETE	-	→	ACK			
			→	BYE(REL)			
			←	200 OK BYE(RLC)			

TP605030	SIP reference: RF0	326	1 [4]		Q.	ISDN/ISDN reference: 1912.5 [1], clause 7.7.1 3)
TSS reference	ISDN-SIP/Basic call/Recei	pt_of	_RELEASE_	or_DIS	CON	INECT
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	BYE:Reason header value	CV_	SIP, encapsu	ulated F	REL (constructed from the ISDN
values	DISCONNECT					
ISDN parameter	DISCONNECT; cause val	ue: (l	PIXIT), locat i	ion (PI)	XIT)	
values						
Comments	ISDN		SUT			SIP-I
	SETUP	1			→	INVITE(IAM)
	ALERTING	+			←	180 Ringing(ACM
	CONNECT	+			+	200 OK INVITE(ANM)
	DISCONNECT	→				
	RELEASE	+			→	ACK
	RELEASE COMPLETE	→			→	BYE(REL)
					+	200 OK BYE(RLC)

TP605031	SIP reference: RF	C 326	61 [4]		ISDN/ISDN reference: 0.1912.5 [1], clause 7.7.1 3)	
TSS reference	ISDN-SIP/Basic call/Rece	int of	DELEVEE			
SIP selection	PICS 4/10	ipt_oi	_NELEASE_	01_013001	MEGI	
criteria	FICS 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 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					constructed from the ISDN	
values	RELEASE COMPLETE	C O V_	_On , encaps	ulated INEL	constructed from the fobit	
ISDN parameter	RELEASE COMPLETE; c	ause	value: (PIXI	T). location	(PIXIT)	
values	,		(,,	. ()	
Comments	ISDN		SUT		SIP-I	
	SETUP	→		→	INVITE(IAM)	
				(SIP_MESSAGE_VA	
	RELEASE COMPLETE	→				
	CASE A					
				→	CANCEL(REL)	
				(200 OK CANCEL	
				+	487 Request Terminated	
				→	ACK	
	CASE B					
				→	BYE(REL)	
				+	200 OK BYE(RLC)	
				+	487 Request Terminated	
				→	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/Rece	eipt_of_RELEASE_	or_DISCON	INECT		
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				constructed from the ISDN		
values	RELEASE	c ov_on , choaps	diated IVEE			
ISDN parameter values	RELEASE; cause value:	(PIXIT), location ((PIXIT)			
Comments	ISDN	SUT		SIP-I		
	SETUP	→	→	INVITE(IAM)		
			+	SIP_MESSÁGE_VA		
	RELEASE	→				
	RELEASE COMPLETE	+				
	CASE A					
			→	CANCEL(REL)		
			-	200 OK CANCEL		
			+	487 Request Terminated		
			→	ACK		
	CASE B					
			→	BYE(REL)		
			←	200 OK BYE(RLC)		
			←	487 Request Terminated		
			→	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/Rece	eipt_of_RELEASE	_or_DISCON	NECT		
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 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 valu DISCONNECT	e CV_SIP, encap	sulated REL	constructed from the ISDN		
ISDN parameter values	DISCONNECT; cause va	ilue: (PIXIT), loca	tion (PIXIT)			
Comments	ISDN	SUT	•	SIP-I		
	SETUP	→	→	INVITE(IAM)		
			+	SIP_MESSAGE_VA		
	DISCONNECT	→				
	RELEASE	-				
	RELEASE COMPLETE	→				
	CASE A					
			→	CANCEL(REL)		
			+	200 OK CANCEL		
			+	487 Request Terminated		
			→	ACK		
	CASE B					
			→	BYE(REL)		
			+	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
-	-	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]			DN/ISDN reference: 912.5 [1], clause 7.7.2		
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 and on receipt of a BYE message where a Reason header field with Q.850 Cause Value is not included: • sends a DISCONNECT message constructed from the encapsulated REL in the received BYE MIME body.						
SIP parameter values	BYE: ISUP REL encapsula	BYE: ISUP REL encapsulated in the MIME body					
ISDN parameter values	DISCONNECT: Cause value	ue constr	ructed from	the encaps	sulated REL		
Comments	ISDN		SUT		SIP-I		
	SETUP	→		→	INVITE(IAM)		
	ALERTING	+		+	180 Ringing(ACM		
	CONNECT	+		+	200 OK INVITE(ANM)		
				→	ACK		
			Conversati	on			
	DISCONNECT	+		+	BYE(REL)		
	RELEASE	→		→	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/Recei	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 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 encapsula	ated in t	he MIME	body, Re	eason	header value = (PIXIT)		
ISDN parameter values	DISCONNECT: Cause val	ue cons	structed f	rom the e	ncaps	sulated REL		
Comments	ISDN		SI	JT		SIP-I		
	SETUP	→			→	INVITE(IAM)		
	ALERTING	+			+	180 Ringing(ACM		
	CONNECT	+			+	200 OK INVITE(ANM)		
					→	ACK		
			Conve	rsation				
	DISCONNECT	+			+	BYE(REL)		
	RELEASE	→			→	200 OK BYE(RLC)		
	RELEASE COMPLETE	+						

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

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

TP606003	SIP reference: RFC 32	261 [4]			DN/ISDN reference: 912.5 [1], clause 7.7.6			
TSS reference	ISDN-SIP/Basic call/Receipt	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 encapsulated in the MIME body						
ISDN parameter values	DISCONNECT/RELEASE: c	ause va	lue: mapped	from th	e encapsulated REL			
Comments	ISDN		SUT		SIP-I			
	SETUP	→		→	INVITE(IAM)			
	SETUP ACK	+						
	CASE A			+	SIP_Failure_VA(REL)			
	RELEASE	+		→	ACK			
	RELEASE COMPLETE	→						
	CASE B							
	DISCONNECT	+	-					
	RELEASE	→	-					
	RELEASE COMPLETE	+	·					

TP606004	SIP reference: RFC 3261 [4]				DN/ISDN reference: 912.5 [1], clause 7.7.6			
TSS reference	ISDN-SIP/Basic call/Receipt_	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 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 REL.							
SIP parameter values	SIP_Failure_VA: ISUP REL 6	encaps	ulated in the N	MIME bo	ody			
ISDN parameter values	DISCONNECT/RELEASE: ca	ause v	alue: mapped	I from th	e encapsulated REL			
Comments	ISDN		SUT		SIP-I			
	SETUP	→		→	INVITE(IAM)			
	CALL PROCEEDING	+						
	CASE A			+	SIP_Failure_VA(REL)			
	RELEASE	+		→	ACK			
	RELEASE COMPLETE	→						
	CASE B							
	DISCONNECT	+						
	RELEASE	→						
	RELEASE COMPLETE	+						

TP606005	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 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 parameter values	SIP_Failure_VA: no ISUP	REL encap	sulated in th	ne MIME	body		
ISDN parameter values	DISCONNECT/RELEASE:	cause va	lue: mapped	I from th	e encapsulated REL		
Comments	ISDN		SUT		SIP-I		
	SETUP	→		→	INVITE(IAM)		
	CALL PROCEEDING	(
	CASE A			←	SIP_Failure_VA(REL)		
	DISCONNECT	+		→	ACK		
	RELEASE	→					
	RELEASE COMPLETE	+					
	CASE B						
	RELEASE	+	•				
	RELEASE COMPLETE	→					

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 320	61 [4]		ı		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 4/12							
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 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 o CV_ ISDN. 	r REL	EASE r	nessage w	ith t	he Cause value set to		
SIP parameter								
values								
ISDN parameter values	DISCONNECT/RELEASE; ca	use v	alue: C	V_ISDN (F	IXI	Γ)		
Comments	ISDN		S	UT		SIP-I		
	SETUP	→		·	→	INVITE(IAM)		
	SETUP ACK	+						
	CASE A				(SIP_Failure_VA		
	RELEASE	+		ı)	ACK		
	RELEASE COMPLETE	→						
	CASE B							
	DISCONNECT	←						
	RELEASE	+						
	RELEASE COMPLETE	+						

TP606007	SIP reference: RFC 320	61 [4]				DN/ISDN reference: 912.5 [1], clause 7.7.6
TSS reference	ISDN-SIP/Basic call/Receipt_o	of_bac	ckward_f	inal_resp	ons	e_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 not 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		SU	Т		SIP-I
	SETUP	→			→	INVITE(IAM)
	CALL PROCEEDING	+				
	CASE A				←	SIP_Failure_VA
	RELEASE	←			→	ACK
	RELEASE COMPLETE	→				
	CASE B					
	DISCONNECT	←				
	RELEASE	→				
	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: RFC 3261 [4]			ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6			
TSS reference	ISDN-SIP/Basic call/Rec	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	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 not 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	→			→	INVITE(IAM)	
					+	SIP_Failure_VA	
		,		•	→	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/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, on receipt of a Failure message 487 Request terminated where a Reason header field with Q.850 Cause Value is not 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	→			→	INVITE(IAM)	
					+	100 Trying	
	RELEASE	→			→	CANCEL(REL)	
	RELEASE COMPLETE	+			+	200 OK CANCEL	
				•	+	487 Request terminated	
					→	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	Ensure that the SUT after receiving the SETUP sends out an INVITE message, on receipt of a Failure message 491 Request Pending where a Reason header field with Q.850 Cause Value is not included: • no action is taken on the ISDN.					
SIP parameter values						
ISDN parameter values						
Comments	ISDN		SUT			SIP-I
	SETUP	→			→	INVITE(IAM)
					+	100 Trying
					+	491 Request Pending
					→	ACK

TP606011	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	NOT PICS 4/11					
criteria						
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 not included: • sends a RELEASE or DISCONNECT message with the Cause value set to CV_ISDN.					
SIP parameter values				<u> </u>		
ISDN parameter values	RELEASE/DISCONNECT; ca	ause v	alue : C	V_ISDN	١	
Comments	ISDN		S	UT		SIP-I
	SETUP	→			→	INVITE(IAM)
					+	SIP MESSAGE_VA
	CASE A				+	SIP_Failure_VA
	RELEASE	+			→	ACK
	RELEASE COMPLETE	→				
	CASE B					
	DISCONNECT	+				
	RELEASE	→				
	RELEASE COMPLETE	+				

Table 36

	Values for test purpose TP606011				
VA	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/Receipt_	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 values	SIP_Failure_VA						
ISDN parameter values	RELEASE/DISCONNECT; ca	RELEASE/DISCONNECT; cause value: CV_ISDN					
Comments	ISDN		SUT			SIP-I	
	SETUP	→		-	>	INVITE(IAM)	
	ALERTING	+		€	-	180 Ringing(ACM)	
	DISCONNECT	+		€	-	SIP_Failure_VA	
	RELEASE	→		-	>	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 3	3261 [4]			SDN/ISDN reference: 912.5 [1], clause 7.7.6	
TSS reference	ISDN-SIP/Basic call/Receip	t_of_bac	kward_final_r	espons	e_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 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	ader CV	_ SIP (PIXIT)			
ISDN parameter values	DISCONNECT/RELEASE of	ause val	ue CV_ ISDN	l (PIXIT		
Comments	ISDN		SUT		SIP-I	
	SETUP	→		→	INVITE(IAM)	
	SETUP ACK	+				
				+	SIP MESSAGE_VA	
	CASE A			+	SIP_Failure_VA	
	RELEASE	+		→	ACK	
	RELEASE COMPLETE	→				
	CASE B					
	DISCONNECT	+				
	RELEASE	→				
	RELEASE COMPLETE	+				

TP606014	SIP reference: RFC 32	61 [4]			DN/ISDN reference: 912.5 [1], clause 7.7.6			
TSS reference	ISDN-SIP/Basic call/Receipt_	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 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 head	der CV_ S	IP (PIXIT)					
ISDN parameter values	DISCONNECT/RELEASE car	use value	CV_ ISDN	(PIXIT				
Comments	ISDN		SUT		SIP-I			
	SETUP	→		→	INVITE(IAM)			
	CALL PROCEEDING	+						
				+	SIP MESSAGE_VA			
	CASE A			+	SIP_Failure_VA			
	RELEASE	+		→	ACK			
	RELEASE COMPLETE	→						
	CASE B							
	DISCONNECT	+						
	RELEASE	→						
	RELEASE COMPLETE	←						

TP606015	SIP reference: RFC 326	1 [4]		ISDN/ISDN reference: .1912.5 [1], clause 7.7.6		
TSS reference	ISDN-SIP/Basic call/Receipt_c	f_backwa	rd_final_respor	nse_or_BYE		
SIP selection	PICS 4/11					
criteria						
ISDN selection criteria						
Test purpose	 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, 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; 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. 					
SIP parameter values	SIP_Failure_VA Reason head	er CV_ SI	P (PIXIT)			
ISDN parameter values	DISCONNECT/RELEASE cau	se value C	V_ ISDN (PIXI	T)		
Comments	ISDN		SUT	SIP-I		
	SETUP	→	-	INVITE(IAM)		
	CALL PROCEEDING	←				
			+	SIP MESSAGE_VA		
	CASE A		(SIP_Failure_VA		
	RELEASE	+)	ACK		
	RELEASE COMPLETE	→				
	CASE B					
	DISCONNECT	+				
	RELEASE	→				
	RELEASE COMPLETE	+				

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) 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_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]		SDN/ISDN reference: 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	→	→	INVITE(IAM)					
	SETUP ACK	+							
	CASE A		+	SIP_Response_VA					
	RELEASE	+	→	ACK					
	RELEASE COMPLETE	→							
	CASE B								
	DISCONNECT	+							
	RELEASE	→							
	RELEASE COMPLETE	+							

TP606017	SIP reference: RFC 32	:61 [4]			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	that the SUT in state N3, beforesponse message (3xx) defi	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_ISDN	I							
Comments	ISDN		SU	IT	SIP-I				
	SETUP	→		7	► INVITE(IAM)				
	CALL PROCEEDING	←							
	CASE A			•	SIP_Response_VA				
	RELEASE	+		-	▶ ACK				
	CASE B								
	DISCONNECT	+							
	RELEASE	→							
	RELEASE COMPLETE	←							

TP606018	SIP reference: RFC 320	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	NOT PICS 4/17							
criteria								
ISDN selection								
criteria								
Test purpose	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 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	DISC; cause value: CV_ISDN	1						
Comments	ISDN		SUT		SIP-I			
	SETUP	→		→	INVITE(IAM)			
	CALL PROCEEDING	+						
	CASE A			+	SIP_Response_VA			
	RELEASE	←		→	ACK			
	RELEASE COMPLETE	→						
	CASE B							
	DISCONNECT	←						
	RELEASE	→						
	RELEASE COMPLETE	+						

TP606019	SIP reference: RFC	3261 [4	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	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	→			→	INVITE(IAM)			
	SETUP ACK	+							
					+	SIP_Response_VA			
					→	ACK			
					→	INVITE(IAM)			
	ALERTING	+				180 Ringing(ACM			
	CONNECT	+			+	200 OK INVITE(ANM)			
	CONNECT				→ -	ACK			
			Conve	rsation		, non			
	DISCONNECT	+			←	BYE(REL)			
	RELEASE	→			→	200 OK BYE(RLC)			
	RELEASE COMPLETE	+							

TP606020	SIP reference: RF0	C 3261 [4]		DN/ISDN reference: 912.5 [1], clause 7.7.6			
TSS reference	ISDN-SIP/Basic call/Rece	eipt_of_ba	ackward_fir	nal_response	e_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 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 values		SIP_Response_VA Contact: URI of new destination INVITE: Request URI of new destination						
ISDN parameter values	IIIVITE. Request SIXI of I	iow dooi!	lation					
Comments	ISDN		SUT		SIP-I			
	SETUP	→		→	INVITE(IAM)			
	CALL PROCEEDING	+			, ,			
				+	SIP_Response_VA			
				→	ACK			
				→	INVITE(IAM)			
	ALERTING	←		+	180 Ringing(ACM			
	CONNECT	+		+	200 OK INVITE(ANM)			
				→	ACK			
			Conversa	ition				
	DISCONNECT	+		+	BYE(REL)			
	RELEASE	→		→	200 OK BYE(RLC)			
	RELEASE COMPLETE	+						

TP606021	SIP reference: RF0	C 3261 [4	1]			DN/ISDN reference: 912.5 [1], clause 7.7.6		
TSS reference	ISDN-SIP/Basic call/Rece	eipt_of_b	ackward	_final_re				
SIP selection criteria	PICS 4/17							
ISDN selection criteria								
Test purpose	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 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 values	SIP_Response_VA Contact: URI of new destination INVITE: Request URI of new destination							
ISDN parameter values	·							
Comments	ISDN		Sl	JT		SIP-I		
	SETUP	→			→	INVITE(IAM)		
	CALL PROCEEDING	+						
					+	SIP_Response_VA		
					→	ACK		
						INDUCTE (IABA)		
	AL EDTING				→	INVITE(IAM)		
	ALERTING	(-	180 Ringing(ACM		
	CONNECT	+			+	200 OK INVITE(ANM)		
			0		→	ACK		
	DISCONNECT	-	Conve	rsation	Z	DVE(DEL)		
	DISCONNECT	→			←	BYE(REL)		
	RELEASE COMPLETE	7			7	200 OK BYE(RLC)		
	RELEASE COMPLETE	~						

Table 40

Values for test purpose TP606016, TP606017, TP606018 TP606019, TP606020, TP606021						
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	261 [4]		Q		N/ISDN reference: 2.5 [1], clause 7.7.6.1			
TSS reference	ISDN-SIP/Basic call/Autonon	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 _{OIW3} . The RELEASE or DISCONNECT message shall only be sent if T _{OIW3} expires.								
SIP parameter values									
ISDN parameter values									
Comments	ISDN		SU	Τ		SIP-I			
	SETUP	→			→	INVITE(IAM)			
			Start time	r T _{OIW3}	+	484 Address incomplete			
					→	ACK			
			Timeout	T _{OIW3}					
	CASE A								
	RELEASE	←							
	RELEASE COMPLETE	→							
	CASE B								
	DISCONNECT	+							
	RELEASE	→							
	RELEASE COMPLETE	←							

TP607002	SIP reference: RFC 3	261 [4]			N/ISDN reference:				
	Q.1912.5 [1], clause 7.7.6.1								
TSS reference	ISDN-SIP/Basic call/Autonom	ous_rele	ease						
SIP selection	NOT PICS 3/4								
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 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	→		→	INVITE(IAM)				
				+	484 Address incomplete				
	CASE A			→	ACK				
	RELEASE	+							
	RELEASE COMPLETE	→							
	CASE B								
	DISCONNECT	+	<u> </u>						
	RELEASE	→							
	RELEASE COMPLETE	+							

TP607003	SIP reference: RFC 3261 [4]			(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										
	Ensure that the SUT when	. 41 :			tion is a second and					
Test purpose	preconditions used, the S		nternai resot	irce reserva	tion is unsuccessiul and					
	'		a tha CID nat	worle.						
	sends a CANCEL or I									
CID managed an	 sends a RELEASE to 	the i	SDN terminal							
SIP parameter values										
ISDN parameter values										
Comments	ISDN		SUT		SIP-I					
	SETUP	→		→	INVITE(IAM)					
					183 Session Progress					
					PRACK					
					200 OK PRACK					
			Internal reso	ource						
		res	servation uns	uccessful						
	CASE A									
	RELEASE	+		→	CANCEL(REL)					
	RELEASE COMPLETE	→		+	200 OK CANCEL					
				+	487 Request Terminated					
				→	ACK					
	CASE B									
	RELEASE	←		→	BYE(REL)					
	RELEASE COMPLETE	→		+	200 OK BYE(RLC)					
				+	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:	RFC 3261 [4]		I	SUP reference: Q.1912.5 [1],			
				Q.731	[i.2], clause 3.5.2.1.1			
TSS reference	ISDN-(ISUP)-SIP/SS/C	LIP						
SIP selection criteria								
ISUP selection criteria								
Test purpose	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 presentation restricted indicator set to "presentation allowed".							
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	→		→	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				→	ACK			
	Communication							
	DISC	→		→	BYE(REL)			
	REL	+		+	200 OK BYE(RLC)			
	REL_COM	→	•					

TP701002	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	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	INVITE: Content-Type: applica						
values	180 Ringing: Content-Type: ap	plication/ISUP;	ACM encapsu	lated in the MIME body			
ISDN parameter values							
Comments	ISDN	Sl	JT	SIP-I			
	SETUP	→	→	INVITE(IAM)			
	ALERTING	+	+	180 Ringing(ACM)			
	CONN	+	+	200 OK INVITE(ANM)			
			→	ACK			
	Communication						
	DISC	→	→	BYE(REL)			
	REL	+	+	200 OK BYE(RLC)			
	REL_COM	→					

TP701003	SIP reference: RF0	3261 [4]		-	SUP reference: Q.1912.5 [1], [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 succeed the screening indicator set	cessfully c	riginate a call h	aving the	calling party number with
SIP parameter	INVITE: Content-Type: app	lication/IS	UP; IAM encap	sulated in	n the MIME body
values	180 Ringing: Content-Type	: application	on/ISUP; ACM	encapsul	ated in the MIME body
ISDN parameter values					
Comments	ISDN		SUT		SIP-I
	SETUP	→		→	INVITE(IAM)
	ALERTING	+		+	180 Ringing(ACM)
	CONN	+		+	200 OK INVITE(ANM)
				→	ACK
			Communication	1	
	DISC	→		→	BYE(REL)
	REL	+		+	200 OK BYE(RLC)
	REL_COM	→			

TP701004	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	Calling party number (user Verify that the IUT can succes the screening indicator set to transport parameter containi	ssfully c "user p	originate a call havi rovided, verified ar	ing a o	calling party number with	
SIP parameter	INVITE: Content-Type: applic	ation/IS	SUP; IAM encapsul	ated i	n the MIME body	
values	180 Ringing: Content-Type: a	pplicati	on/ISUP; ACM end	capsul	ated in the MIME body	
ISDN parameter values						
Comments	ISDN		SUT		SIP-I	
	SETUP	→		→	INVITE(IAM)	
	ALERTING	+		+	180 Ringing(ACM)	
	CONN	+		+	200 OK INVITE(ANM)	
				→	ACK	
	Communication					
	DISC	→		1	BYE(REL)	
	REL	+		+	200 OK BYE(RLC)	
	REL_COM	→				

TP701005	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/C	LIP				
SIP selection criteria						
ISUP selection criteria						
Test purpose	with the screening indic	successfully of ator set to "ne	riginate a c twork prov	all having a ided" and a	default calling party number generic number containing tor set to "user provided, not	
SIP parameter values	INVITE: Content-Type: 180 Ringing: Content-T				in the MIME body lated in the MIME body	
ISDN parameter values					•	
Comments	ISDN		SUT		SIP-I	
	SETUP	→		→	INVITE(IAM)	
	ALERTING	+		+	180 Ringing(ACM)	
	CONN	+		+	200 OK INVITE(ANM)	
				→	ACK	
	Communication					
	DISC	→		→	BYE(REL)	
	REL	+		+	200 OK BYE(RLC)	
	REL_COM	→				

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/CL	.IP					
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: a	application/IS	SUP; IAM	encapsu	lated in	n the MIME body	
values	180 Ringing: Content-Ty	pe: applicati	ion/ISUP;	ACM en	capsul	ated in the MIME body	
ISDN parameter values							
Comments	ISDN		SU	Т		SIP-I	
	SETUP	→			→	INVITE(IAM)	
	ALERTING	←			←	180 Ringing(ACM)	
	CONN	←			←	200 OK INVITE(ANM)	
					→	ACK	
	Communication						
	DISC	→			→	BYE(REL)	
	REL	+			+	200 OK BYE(RLC)	
	REL_COM	→					

A.1.1.2.2 Calling Line Identification Restriction (CLIR)

TP702001	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 (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: applic	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	→		→	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				→	ACK			
		Communication						
	DISC	→		→	BYE(REL)			
	REL	+		+	200 OK BYE(RLC)			
	REL_COM	→						

TP702002	SIP reference: RFC	3261 [4]	9.7	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) 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: appl	lication/ISUP; IAI	/ encapsulat	ed in the MIME body			
values	180 Ringing: Content-Type:	application/ISUI	; ACM encap	osulated in the MIME body			
ISDN parameter values							
Comments	ISDN	5	SUT	SIP-I			
	SETUP	→	1	➤ INVITE(IAM)			
	ALERTING	←	•	► 180 Ringing(ACM)			
	CONN	(► 200 OK INVITE(ANM)			
				→ ACK			
		Comm	unication				
	DISC	→		▶ BYE(REL)			
	REL	←	•	200 OK BYE(RLC)			
	REL_COM	→					

TP702003	SIP reference: RF	C 3261 [4]	Q.	-	SUP reference: Q.1912.5 [1], [i.2], clause 4.5.2.1.1		
ISDN parameter values	ISDN-(ISUP)-SIP/SS/CLIF	₹						
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/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body						
ISDN parameter values			-			·		
Comments	ISDN SETUP ALERTING CONN DISC REL	+ + + +	Communic	eation	→ ← ← → →	SIP-I INVITE(IAM) 180 Ringing(ACM) 200 OK INVITE(ANM) ACK BYE(REL) 200 OK BYE(RLC)		
	REL_COM	→						

TP702004	SIP reference: RFC	3261 [4]		ISUP 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: applic			capsulated	in the MIME body		
values	180 Ringing: Content-Type: a						
ISDN parameter values							
Comments	ISDN		SUT		SIP-I		
	SETUP	→		→	INVITE(IAM)		
	ALERTING	+		+	180 Ringing(ACM)		
	CONN	+		+	200 OK INVITE(ANM)		
				→	ACK		
			Communica	ation			
	DISC	→		→	BYE(REL)		
	REL	+		+	200 OK BYE(RLC)		
	REL_COM	→					

TP702005	SIP reference: RFC 3	261 [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: applica	ation/ISUP; IAM	encapsulated i	n the MIME body			
values	180 Ringing: Content-Type: a	oplication/ISUP	; ACM encapsu	ated in the MIME body			
ISDN parameter values							
Comments	ISDN	SI	JT	SIP-I			
	SETUP	→	→	INVITE(IAM)			
	ALERTING	←	+	180 Ringing(ACM)			
	CONN	+	+	200 OK INVITE(ANM)			
			→	ACK			
	Communication						
	DISC	→	→	BYE(REL)			
	REL	+	+	200 OK BYE(RLC)			
	REL_COM	→					

TP702006	SIP reference: RFC 3	3261 [4]			ISUP 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	pplication	on/ISUP; A	CM encapsu	lated in the MIME body				
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	→		→	INVITE(IAM)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONN	+		+	200 OK INVITE(ANM)				
				→	ACK				
		Communication							
	DISC	→	· · · · · · · · · · · · · · · · · · ·	→	BYE(REL)				
	REL	+		+	200 OK BYE(RLC)				
	REL_COM	→							

TP702007	SIP reference: RFC 3	3261 [4]	_	SUP reference: Q.1912.5 [1],			
			Q.73	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 - 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: applic 180 Ringing: Content-Type: a						
ISDN parameter values				•			
Comments	ISDN	SI	JT	SIP-I			
	SETUP	→	→	INVITE(IAM)			
	ALERTING	+	+	180 Ringing(ACM)			
	CONN	+	+	200 OK INVITE(ANM)			
			→	ACK			
		Commu	nication				
	DISC	→	→	BYE(REL)			
	REL	+	+	200 OK BYE(RLC)			
	REL_COM	→					

TP702008	SIP reference: RFC 3	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: applic	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	→		→	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				→	ACK			
		Co	mmunication					
	DISC	→		→	BYE(REL)			
	REL	+		+	200 OK BYE(RLC)			
	REL_COM	→						

TP702009	SIP reference: RF	C 3261 [4	1	-	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	→		→	INVITE(IAM)		
	ALERTING	+		+	180 Ringing(ACM)		
	CONN	+		+	200 OK INVITE(ANM)		
				→	ACK		
			Communication				
	DISC	→		→	BYE(REL)		
	REL	+		+	200 OK BYE(RLC)		
	REL_COM	→					

TP7020010	SIP reference: R	FC 3261 [4]	_	SUP reference: Q.1912.5 [1], :1 [i.2], clause 4.2.1			
TSS reference	ISDN-(ISUP)-SIP/SS/CL	IR						
SIP selection								
criteria								
ISUP selection	DLE							
criteria								
Test purpose	Verify that the calling pa	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: a	pplication/IS	SUP; IAM encapsi	ulated i	n the MIME body			
values	180 Ringing: Content-Ty	pe: applicati	on/ISUP; ACM er	ncapsul	ated in the MIME body			
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	→		→	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				→	ACK			
		Communication						
	DISC	→		→	BYE(REL)			
	REL	+		+	200 OK BYE(RLC)			
	REL_COM	→						

A.1.1.2.3 Connected Line Identification Presentation (COLP)

TP703001	SIP reference:	RFC 3261 [4]		_	SUP reference: Q.1912.5 [1], i.2], clause 5.5.2.5.1 i)
TSS reference	ISDN-(ISUP)-SIP/SS/C	COLP			
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Initiate COLP request	t		•	
	Verify that the exchang optional forward call		uccessfully a c	all reque	sting the COLP service in the
SIP parameter	INVITE: Content-Type:	: application/IS	UP; IAM encap	sulated i	n the MIME body
values	180 Ringing: Content-	Type: application	on/ISUP; ACM	encapsul	ated in the MIME body
ISDN parameter values					
Comments	ISDN		SUT		SIP-I
	SETUP	→		→	INVITE(IAM)
	ALERTING	+		+	180 Ringing(ACM)
	CONN	+		+	200 OK INVITE(ANM)
				→	ACK
			Communication)	
	DISC	→		→	BYE(REL)
	REL	+		+	200 OK BYE(RLC)
	REL_COM	→			

TP703002	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	1						
SIP selection criteria								
ISUP selection criteria								
Test purpose	Verify that the IUT can prov	Connected number (user provided, verified and passed) /erify 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 values	180 Ringing: Content-Type:	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	→		→	INVITE(IAM)			
	CASE A							
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				→	ACK			
	CASE B							
	CONN	←		+	200 OK INVITE(CON)			
				→	ACK			
			Communication	n				
	DISC	→		→	BYE(REL)			
	REL	+		←	200 OK BYE(RLC)			
	REL_COM	→						

TP703003	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	address Verify that the IUT car "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	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	→		→	INVITE(IAM)			
	CASE A							
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				→	ACK			
	CASE B							
	CONN	+		+	200 OK INVITE(CON)			
				→	ACK			
		(Communication	n				
	DISC	→		→	BYE(REL)			
	REL	+		+	200 OK BYE(RLC)			
	REL_COM	→						

TP703004	SIP reference: RFC 3	3261 [4	l		ISUP 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 provid	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 values	180 Ringing: Content-Type: a	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	ΙΤ	SIP-I			
	SETUP	→		→	INVITE(IAM)			
	CASE A							
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				→	ACK			
	CASE B							
	CONN	+		+	200 OK INVITE(CON)			
				→	ACK			
			Commun	ication				
	DISC	→		→	BYE(REL)			
	REL	+		+	200 OK BYE(RLC)			
	REL_COM	→						

TP703005	SIP reference: RFC	3261 [4	.]	ISUP reference:			
				0.724	Q.1912.5 [1],		
T00 (IODAL (IOLID) OLD/OO/OOLD			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 (netwo	ork prov	ided) with co	nnected s	ub-address		
					with the screening indicator		
					d and an access transport		
	parameter containing the co	nnected	sub-address.		-		
SIP parameter	INVITE: Content-Type: appl	ication/I	SUP; IAM enc	apsulated i	n the MIME body		
values	180 Ringing: Content-Type:						
	200 OK INVITE: Content-Ty	pe: appl	ication/ISUP;	ANM enca	psulated in the MIME body		
ISDN parameter							
values							
Comments	ISDN		SUT		SIP-I		
	SETUP	→		→	INVITE(IAM)		
	CASE A						
	ALERTING	+		+	180 Ringing(ACM)		
	CONN	+		+	200 OK INVITE(ANM)		
				→	ACK		
	CASE B						
	CONN	+		+	200 OK INVITE(CON)		
				→	ACK		
			Communicat	ion			
	DISC	→		→	BYE(REL)		
	REL	+		+	200 OK BYE(RLC)		
	REL_COM	→					

TP703006	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/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: appli							
values	180 Ringing: Content-Type: 200 OK INVITE: Content-Ty							
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	→		→	INVITE(IAM)			
	CASE A							
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				→	ACK			
	CASE B							
	CONN	+		+	200 OK INVITE(CON)			
				→	ACK			
		С	ommunication					
	DISC	→		→	BYE(REL)			
	REL	+		+	200 OK BYE(RLC)			
	REL_COM	→						

TP703007	SIP reference: RF	FC 3261 [4]			ISUP reference: Q.1912.5 [1], [i.2], clause 5.5.2.5.1 i)				
TSS reference	ISDN-(ISUP)-SIP/SS/COL	_P	-						
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Verify that the IUT can proset to "network provided", with the screening indicate	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: ap	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body							
					psulated in the MIME body				
ISDN parameter values		7			,				
Comments	ISDN		SUT		SIP-I				
	SETUP	→		→	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	→		→	BYE(REL)				
	REL	+		+	200 OK BYE(RLC)				
	REL_COM	→							

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 number or full ISDN number as the connected number on call answer.						
SIP parameter values	180 Ringing: Content-Type:	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	→		→	INVITE(IAM)			
	CASE A							
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				→	ACK			
	CASE B							
	CONN	+		+	200 OK INVITE(CON)			
				→	ACK			
			Communicat	ion				
	DISC	→		→	BYE(REL)			
	REL	+		+	200 OK BYE(RLC)			
	REL_COM	→						

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/0	COLR						
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	Verify that a local exch system when a conne presentation restricted	Presentation of restricted COL Verify that a local exchange will not pass the information on to the access signalling system when a connected number 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 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	→		→	INVITE(IAM)			
	CASE A							
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				→	ACK			
	CASE B							
	CONN	(+	200 OK INVITE(CON)			
				→	ACK			
		C	ommunication	1				
	DISC	→		→	BYE(REL)			
	REL	(+	200 OK BYE(RLC)			
	REL_COM	→						

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/COL	R							
SIP selection criteria									
ISUP selection criteria									
Test purpose	Verify that the received co number in the generic nur	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	→		→	INVITE(IAM)				
	CASE A								
	ALERTING	+		+	180 Ringing(ACM)				
	CONN	+		+	200 OK INVITE(ANM)				
				→	ACK				
	CASE B								
	CONN	+		+	200 OK INVITE(CON)				
			-	→	ACK				
			Communicat	ion					
	DISC	→		→	BYE(REL)				
	REL	+		+	200 OK BYE(RLC)				
	REL_COM	→							

TP704003	SIP reference: RFC	3261 [4]			SUP 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	DLE				
criteria					
Test purpose	Restricted connected number				
	Verify that the IUT can provid				
					esentation restricted indicator
OID	set to "presentation restricted				
SIP parameter	INVITE: Content-Type: applic				
values	180 Ringing: Content-Type: a				
ICDM managements	200 OK INVITE: Content-Typ	e: appii	cation/iSUP;	Anivi enca	psulated in the MilME body
ISDN parameter values					
Comments	ISDN		SUT		SIP-I
Comments	SETUP	→	301	→	INVITE(IAM)
	CASE A				IIIVII E(IAW)
	ALERTING	+		+	180 Ringing(ACM)
	CONN	+		-	200 OK INVITE(ANM)
	COMM			→ -	ACK
	CASE B			7	ACK
		+			
	CONN	-		←	200 OK INVITE(CON)
			0 : 1		ACK
	DIGG	-	Communicat		DVE(DEL)
	DISC	→		→	BYE(REL)
	REL	+		+	200 OK BYE(RLC)
	REL_COM	→			

TP704004	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1],			
				Q.73	1 [i.2], clause 6.5.2.1		
TSS reference	ISDN-(ISUP)-SIP/SS/COL	_R					
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	→		→	INVITE(IAM)		
	CASE A						
	ALERTING	+		+	180 Ringing(ACM)		
	CONN	+		+	200 OK INVITE(ANM)		
				→	ACK		
	CASE B						
	CONN	←		+	200 OK INVITE(CON)		
				→	ACK		
			ommunicatio				
	DISC	→		→	BYE(REL)		
	REL	+		+	200 OK BYE(RLC)		
	REL_COM	→					

TP704005	SIP reference:	RFC 3261 [4]			ISUP reference: Q.1912.5 [1], 1 [i.2], clause 6.5.2.5			
TSS reference	ISDN-(ISUP)-SIP/SS/0	COLR						
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	Postrioted connector	l number (net	work provide	1/				
	Verify that the IUT can set to "network provide "presentation restricted	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							
values					lated in the MIME body psulated in the MIME body			
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	←		→	INVITE(IAM)			
	CASE A							
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	1		+	200 OK INVITE(ANM)			
				→	ACK			
	CASE B							
	CONN	+		+	200 OK INVITE(CON)			
				→	ACK			
			Communicatio	n				
	DISC	→		→	BYE(REL)			
	REL	+		+	200 OK BYE(RLC)			
	REL_COM	→						

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/COLR	2						
SIP selection criteria								
ISUP selection criteria								
Test purpose	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 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	→		→	INVITE(IAM)			
	CASE A							
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				→	ACK			
	CASE B							
	CONN	+		+	200 OK INVITE(CON)			
				→	ACK			
		(Communication	on				
	DISC	→		→	BYE(REL)			
	REL	+		+	200 OK BYE(RLC)			
	REL_COM	→						

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/COLR	₹					
SIP selection criteria							
ISUP selection criteria							
Test purpose	Restricted connected number (user provided, not verified) with connected sub-address 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: 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 SETUP	→	SU	T →	SIP-I INVITE(IAM)		
	CASE A ALERTING CONN	+		+	200 OK INVITE(ANM)		
	CASE B			→	ACK		
	CONN	+		←	=======================================		
			Commun				
	DISC REL	→		→	(/		
	REL_COM	→			200 OR BIL(RLO)		

A.1.1.2.5 Terminal Portability (TP)

TP705001	SIP reference	e: RFC 3261 [4]	Q.191	ISUP reference: 2.5 [1], clause 4.5.2.1.1 a)/ Q.733 [i.6]			
TSS reference	ISDN-(ISUP)-SIP/SS/	TP						
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	To verify that the calli	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								
values								
ISDN parameter								
values								
Comments	ISDN		SU		SIP			
	SETUP	→			/ ITTTITE (I7 1111)			
	ALERTING	←		•	100 11119119(110111)			
	CONN	←		•	200 OK INVITE(ANM)			
			Commun	ication				
	SUSPEND	→		7	INFO(SUS)			
				•	200 OK INFO			
	RESUME	→		7	INFO(RES)			
				•	200 OK INFO			
			Commun	ication				
	DISC	→		7	BYE(REL)			
	REL	+		•	200 OK BYE			
	REL_COM	→						

TP705002	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 4.5.2.1.1 b)/ Q.733 [i.6]				
TSS reference	ISDN-(ISUP)-SIP/SS/	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)			
			Communica	tion				
	SUSPEND	+		+	INFO(SUS)			
				→	200 OK INFO			
	RESUME	+		+	INFO(RES)			
				→	200 OK INFO			
			Communica	tion				
	DISC	→		→	BYE(REL)			
	REL	-		+	200 OK BYE			
	REL_COM	→						

TP705003	SIP reference: RFC 3	261 [4]		ISUP reference: 2.5 [1], clause 4.5.2.1.2/ Q.733 [i.6]
TSS reference	ISDN-(ISUP)-SIP/SS/TP			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Terminal portability, reques To verify that the call is releas timer T2 expires because the	ed with cause	#102 (recovery	on timer expiry) by the IUT if
SIP parameter values	·			
ISDN parameter values				
Comments	ISDN	S	SUT	SIP
	SETUP	→	→	INVITE(IAM)
	ALERTING	+	+	180 Ringing(ACM)
	CONN	+	+	200 OK INVITE(ANM)
		Commi	unication	
	SUSPEND	→	→	INFO(SUS)
			+	200 OK INFO
		T2 e	expiry	
			→	BYE(REL)
			+	200 OK BYE

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

TP705005	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 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	Terminal portability, red To verify that the IUT information requested by the calling	orms the cal	led party t	hat suspend	and resume have been SUS and RES messages.		
SIP parameter		-					
values							
ISDN parameter							
values							
Comments	ISDN		SU	Т	SIP		
	SETUP	+		+	INVITE(IAM)		
	ALERTING	→		→	180 Ringing(ACM)		
	CONN	→		→	200 OK INVITE(ANM)		
			Commun	ication			
	NOTIFY(suspend)	+		+	INFO(SUS)		
				→	200 OK INFO		
	NOTIFY(resume)	+		+	INFO(RES)		
				→	200 OK INFO		
	Communication						
	DISC	+		+	BYE(REL)		
	REL	→		→			
	REL_COM	+					

TP705006	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 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	To verify that the called p	Terminal portability, requested by the called party To verify that the called party can suspend and resume an incoming call and that user initiated SUS and RES messages are sent to the preceding exchange.						
SIP parameter								
values								
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	+		+	INVITE(IAM)			
	ALERTING	→		→	180 Ringing(ACM)			
	CONN	→		→	200 OK INVITE(ANM)			
			Communica	tion				
	NOTIFY(suspend)	→		→	INFO(SUS)			
				←	200 OK INFO			
	NOTIFY(resume)	→		→	INFO(RES)			
				+	200 OK INFO			
			Communica	tion				
	DISC	+		+	BYE(REL)			
	REL	→		→	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:	RFC 3261 [4]			ISUP reference: 5 [1], clauses 1.1.5.2.1.1.1; 5.2.1.1.3; 1.1.5.2.2-4.1/ Q.737 [i.10]
TSS reference	ISDN-(ISUP)-SIP/SS/L	JUS1			Q.707 [1.10]
SIP selection criteria					
ISUP selection criteria					
Test purpose		an successfully in a			with an UUS 1 implicit request, , without the user-to-user
SIP parameter values					
ISDN parameter values	SETUP: User-to-user i	nformation			
Comments	ISDN		SUT		SIP
	SETUP(UUInf)	→		→	INVITE(IAM UUInf)
	ALERTING	+		+	180 Ringing(ACM)
	CONN(UUInf)	+		+	200 OK INVITE(ANM)
				→	ACK
		Co	mmunicat	ion	
	DISC	+		+	BYE(REL)
	REL	→		→	200 OK BYE
	REL_COM	←			

TP706002	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	To verify that the IUT can, after successfully request, continue normal call set up if the firs user-to-user indicators set to "user-to-user note).	initiating/transiting a call with an UUS1 implicit the backward message is received with the
SIP parameter values		
ISDN parameter values		
Comments	ISDN SL	T SIP
	SETUP(UUInf) →	→ INVITE(IAM UUInf)
	ALERTING ←	← 180 Ringing(ACM UUInd)
	CONN ←	← 200 OK INVITE(ANM)
		→ ACK
	Commu	nication
	DISC ←	★ BYE(REL)
	REL →	→ 200 OK BYE
	REL_COM ←	
NOTE: The user-	to-user information is discarded because the f	ollowing network does not support it.

TP706003	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/UU	S1						
SIP selection criteria								
ISUP selection criteria								
Test purpose	To verify that the IUT car	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)	→		→	INVITE(IAM UUInf)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				→	ACK			
		С	ommunicatior	1				
	DISC	+		+	BYE(REL)			
	REL	→		→	200 OK BYE			
	REL_COM	+	<u> </u>					
	to-user information is disc		se:					

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

TP706004	SIP reference:	SIP reference: RFC 3261 [4]		ISUP reference: Q.1912.5 [1], clauses 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	UUS1 implicit - acceptance To verify that the IUT can successfully transit/accept a call with an UUS1 implicit request, and transfer/include the user-to-user information parameter in the ACM, CPG, ANM or CON as implicit acceptance (no user-to-user indicators).							
SIP parameter values								
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP(UUInf)	→		→	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)			
				→	ACK			
	CASE D							
	ALERTING	+		+	180 Ringing(ACM)			
	CONN(UUInf)	+		+	200 OK INVITE(ANM UUInf)			
				→	ACK			
		C	ommunicati	on				
	DISC	+		+	BYE(REL)			
	REL	→		→	200 OK BYE			
	REL_COM	+						

TP706005	SIP reference: RFC	3261 [4	1]	Q.19		SUP reference: [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 with indication generated To verify that the IUT can successfully transit/accept a call with an UUS1 implicit request and set the user-to-user indicators to "user-to-user information discarded by the network" in the first backward message, if the network is unable to support it (see note).						
SIP parameter values			J - ,				
ISDN parameter values							
Comments	SIP		SU	T		ISDN	
	INVITE(IAM UUInf)	→			→	SETUP	
	180 Ringing(ACM UUInd)	+			+	ALERTING	
	200 OK INVITE(ANM)	+			+	CONN	
	ACK	→					
			Commun	ication			
	BYE(REL)	+			+	DISC	
	200 OK BYE	→			→	REL	
					+	REL_COM	
NOTE: The user-	to-user information is discarde	ed beca	use the ne	twork do	es no	t support it.	

TP706006	SIP reference: RFC 32	261 [4]	Q.19		SUP reference: 1], clauses 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	UUS1 explicit non-essential - request To verify that the IUT can successfully initiate/transit a call with an UUS1 explicit non-essential request, by including/transferring the user-to-user information parameter and the user-to-user indicators in the IAM set to "request, not essential".						
SIP parameter values							
ISDN parameter values							
Comments	ISDN		SUT		SIP		
	SETUP(FAC uus1reqness)				INVITE(IAM UUInd, UUInf)		
	ALERTING(FAC uus1rr)				180 Ringing(ACM)		
	CONN				200 OK INVITE(ANM)		
					ACK		
			Communication				
	BYE(REL)	+		+	DISC		
	200 OK BYE	→		→	REL		
				+	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	UUS1 explicit non-essential - 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 the UUS1 service is explicitly rejected (the user-to-user indicators parameter is received as "service not provided" in the ACM or CPG or ANM or CON) (see note).							
SIP parameter								
values								
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP(FAC uus1regness)	→		→	INVITE(IAM UUInd, UUInf)			
	CASE A							
	ALERTING(FAC uus1rr)	+		+	180 Ringing(ACM UUInd s1 prov)			
	CONN	+		+	200 OK INVITE			
	CASE B							
				+	183 Session Progress(ACM)			
	ALERTING(FAC uus1rr)	+		+	180 Ringing(CPG UUInd s1 prov)			
	CONN	+		+	200 OK INVITE			
	CASE C							
	ALERTING	+		+	180 Ringing(ACM)			
	CONN(FAC uus1rr)	+		+	200 OK INVITE(ANM UUInd s1 prov)			
	CASE D							
	CONN(FAC uus1rr)	+		+	200 OK INVITE(ANM UUInd s1 prov)			
	,	Con	municati	on				
	BYE(REL)	+		+	DISC			
	200 OK BYE	→		→	REL			
				+	REL_COM			
1) the ne	-to-user information is discarde etwork is unable to pass the ex emote user may not be able to i	plicit se	ervice 1 ir		message;			

TP706008	SIP reference: RFC 3	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)	→	→	INVITE(IAM UUInd, UUInf)		
	ALERTING(FAC uus1rr)	+	(180 Ringing(ACM)		
	CONN(FAC uus1reterr)	+	(200 OK INVITE(ANM)		
			→	ACK		
		Comm	unication			
	BYE(REL)	+	+	DISC		
	200 OK BYE	→	→	REL		
			+	REL_COM		
NOTE: The user-to-user information is discarded because: 1) the network is unable to pass the explicit service 1 in any message; 2) the remote user may not be able to interpret incoming UUS information.						

TP706009	SIP reference: RFC 3261 [4	Q.191		ISUP reference: [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 - acceptance To verify that the IUT can successfully transit/accept a call with an UUS1 explicit non-essential request, by transferring/including 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)	→		→	SETUP(FAC uus1reqness)		
					CASE A		
	180 Ringing(ACM UUInd s1 prov)	+		←	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	+		←	CONN		
					CASE C		
	180 Ringing(ACM)	←		←	ALERTING		
	200 OK INVITE(ANM UUInd s1 prov)	←		←	CONN(FAC uus1rr)		
					CASE D		
	200 OK INVITE(ANM UUInd s1 prov)		←	CONN(FAC uus1rr)			
	,	Co	mmunicatio	n			
	DISC	+		+	BYE(REL)		
	REL	→		→	200 OK BYE		
	REL_COM			+			

TP706010	SIP reference: RFC 3	3261 [4]	ISUP reference: Q.1912.5 [1], clauses 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	To verify that the IUT can trar request, and reject the service	UUS1 explicit non-essential - implicit (no explicit) rejection sent To verify that the IUT can transfer/accept a call with an UUS1 explicit non-essential request, and reject the service by not providing any user-to-user indicators parameter in the ACM, CPG, ANM or CON (see note).						
SIP parameter values		·						
ISDN parameter values								
Comments	SIP		SUT		ISDN			
	INVITE(IAM UUInd, UUInf)	→		→	SETUP(FAC uus1reqness)			
	180 Ringing(ACM)	+		+	ALERTING			
	200 OK INVITE(ANM)	+		+	CONN			
	ACK	→						
			Communic	ation				
	DISC	+		+	BYE(REL)			
	REL	→		→	200 OK BYE			
	REL_COM			+				
NOTE: The netwo	ork or the user cannot support	UUS1.						

TP706011	SIP reference: RFC 32	61 [4]	Q.19	-	SUP reference: [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 - 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	→		→	REL	
				+	REL_COM	

TP706012	SIP reference: RFC	3261 [4	-	-	SUP reference: [1], clauses 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)	→		→	INVITE(IAM UUInd, UUInf)		
	ALERTING(FAC uus1rr)	+		+	180 Ringing(ACM)		
	CONN(FAC uus1reterr)	+		+	200 OK INVITE(ANM)		
				→	ACK		
			Communication				
	BYE(REL)	+		+	DISC		
	200 OK BYE	→		→	REL		
				+	REL_COM		
NOTE: The netwo	ork does not understand the se	ervice 1	request. In this of	case the	call should be released.		

TP706013	SIP reference: RFC 3261 [4	Q.191		SUP reference: [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)	→		→	SETUP(FAC uus1reqess)		
	CASE A						
	180 Ringing(ACM UUInd s1 prov)	+		←	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	←		←	CONN		
	CASE C						
	180 Ringing(ACM)	←		←	ALERTING		
	200 OK INVITE(ANM UUInd s1 prov)	+		←	CONN(FAC uus1rr)		
	CASE D						
	200 OK INVITE(ANM UUInd s1 prov)	+		-	CONN(FAC uus1rr)		
	E/	Co	mmunication	1			
	DISC	+		′	BYE(REL)		
	REL	→		→	200 OK BYE		
	REL_COM			+			

TP706014	SIP reference: RFC 3	261 [4]		Q.	ISUP reference: 1912.5 [1], clauses 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)	→		→	INVITE(IAM UUInd, UUInf)			
	RELEASE	+		+	500 Server Internal Error(REL#29)			
	RELEASE COLMPLETE	→		→	ACK			
NOTE: The netwo	ork or the called user cannot su	pport t	he service.					

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

TP706101	SIP reference: RF	C 3261 [4]		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/UUS	32							
SIP selection criteria									
ISUP selection criteria									
Test purpose	To verify that the IUT can essential request, having the essential. To verify that the	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 set to "service provided"							
SIP parameter values									
ISDN parameter values									
Comments	ISDN		SUT		SIP				
	SETUP	→		→	INVITE(IAM UU2 not ess)				
	ALERTING	+		+	180 Ringing(ACM)				
	USER INFO	→		→	INFO(USR)				
				+	200 OK INFO				
	USER INFO	+		+	INFO(USR)				
				→	200 OK INFO				
	CONN	+		+	200 OK INVITE(ANM)				
		С	ommunication	on	, ,				
	DISC	+		+	BYE(REL)				
	RELEASE	→		→	200 OK BYE				
	REL_COM	+							

TP706102	SIP reference: RFC 3	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	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 uus2regness)	→		→	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)	+		+	DISC					
	200 OK BYE(RLC)	→		→	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	To verify that the IUT can suc	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 uus1regness)	→		→	INVITE(IAM UU2 ness)		
	ALERTING(FAC uus1rr)	+		+	180 Ringing(ACM)		
	CONN(FAC uus1reterr)	+		+	200 OK INVITE(ANM)		
				→	ACK		
		•	Communication				
	BYE(REL)	+		+	DISC		
	200 OK BYE	→		→	REL		
				+	REL COM		

TP706104	SIP reference	: RFC 3261 [4]	Q		ISUP reference: [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	essential request, hav	can successfully ori	r indicato	rs set to	having an UUS2 explicit "request, essential" and the cators in the IAM set to "ISUP
SIP parameter	·				
values					
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP	→		→	INVITE(IAM UU2 ess)
	ALERTING	+		+	180 Ringing(ACM)
	USER INFO	→		→	INFO(USR)
				+	200 OK INFO
	USER INFO	+		+	INFO(USR)
				→	200 OK INFO
	CONN	+		+	200 OK INVITE(ANM)
		Con	nmunication	1	` , ,
	DISC	+		+	BYE(REL)
	RELEASE	→		→	200 OK BYE
	REL_COM	+			

TP706105	SIP reference:	SIP reference: RFC 3261 [4]			ISUP reference: 12.5 [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/U	JUS2						
SIP selection criteria								
ISUP selection criteria								
Test purpose	To verify that the IUT c	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	→		→	180 Ringing(ACM)			
	USER INFO	+		+	INFO(USR)			
				→	200 OK INFO			
	USER INFO	→		→	183 Session Progress(USR)			
	CONN	→		→	200 OK INVITE(ANM)			
		Co	mmunicat	ion				
	DISC	+		+	BYE(REL)			
	RELEASE	→		→	200 OK BYE			
	REL_COM	+						

TP706106	SIP referen	nce: RI	FC 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	S2				
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	↑		→	500 Server Internal error(REL#29, 69, 88)		
	REL_COM	+		+	ACK		

TP706107	SIP refer	ence: RI	FC 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)-SIF	P/SS/UUS	32		
SIP selection criteria					
ISUP selection criteria					
Test purpose		service	can be reje	cted if	on no indication is received (no user-to-user d message (implicit rejection of service 2)
SIP parameter values	180 Ringing: the	encapsu	lated ACM	does	not contain an user-to-user response indicator
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP	→		→	INVITE(IAM UU2 ess)
				+	180 Ringing(ACM)
	RELEASE	+		→	CANCEL(REL)
	REL_COMP	→		+	200 OK CANCEL
				+	487 Request Terminated
				→	ACK
NOTE: The remo	ote network does r	ot under	stand the s	ervice	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], clauses 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)	→		→	INVITE(IAM UU3 not ess)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONN(UU3 ret res)	+		+	200 OK INVITE(ANM UU3 prov)				
		C	ommunication	on					
	USER INFO	→		→	INFO(USR)				
		_		+	200 OK INFO				
	USER INFO	←		+	INFO(USR)				
				→	200 OK INFO				
	DISC	→		→	BYE(REL)				
	RELEASE	-		+	200 OK BYE				
	REL_COM	→							

TP706202	SIP reference: RF	C 3261 [4]	Q.	ISUP reference: 1912.5 [1], clauses 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 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	parameter est to est tree		-				
values							
ISDN parameter							
values							
Comments	ISDN		SUT		SIP		
	SETUP(UU3 req ess)	→		→	INVITE(IAM UU3 ess)		
	ALERTING	+		+	180 Ringing(ACM)		
	CONN(UU3 ret res)	+		+	200 OK INVITE(ANM UU3 prov)		
			ommunicat	tion			
	USER INFO	→		→	INFO(USR)		
				←	200 OK INFO		
	USER INFO	←		←	INFO(USR)		
				→	200 OK INFO		
	USER INFO	→		→	INFO(USR)		
				←	200 OK INFO		
	USER INFO	+		+	INFO(USR)		
				→	200 OK INFO		
		C	ommunicat	tion			
	DISC	→		→	BYE(REL)		
	RELEASE	+		+	200 OK BYE		
	REL_COM	→					

TP706203	SIP reference:	e: RFC 3261 [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/U	JUS3						
SIP selection criteria								
ISUP selection criteria								
Test purpose	UUS3 explicit essential - explicit rejection To verify that the service can be rejected with a REL having the Cause value #29 "facility rejected", #69 "requested facility not implemented", either with diagnostics (user-to-user indicators name) (see note).							
SIP parameter values								
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	+		+	INVITE(IAM UU2 ess)			
	RELEASE	→		→	500 Server Internal error(REL#29, 69)			
	REL_COM	+		+	ACK			
NOTE: The network or the called user cannot support the service.								

TP706204	SIP reference: RF	C 3261 [4		Q.19	ISUP reference: 012.5 [1], clauses 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	call	-		•	ce during the active phase of the			
	To verify that the IUT can successfully generate/transit an UUS3 explicit non-essential request, with a FAR having the facility indicator parameter set to "user-to-user service" and the Service 3 field in the user-to-user indicators set to "request, not essential". To verify that the IUT can successfully reply to an UUS3 explicit non-essential request with a FAA having the facility indicator parameter set to "user-to-user service" and the Service 3 field in the user-to-user indicators parameter set to "service provided".							
SIP parameter			•		•			
values								
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	→		→	INVITE(IAM)			
	ALERTING	+		+				
	CONN	+		+	200 OK INVITE			
		C	ommunication					
	FAC(UU3 req not ess)	→		→	INFO(FAR reg UU3 not ess)			
	(+				
	FAC(UU3 ret res)	+		+				
	USER INFO	→		→	INFO(USR)			
	002111111			+	200 OK INFO			
	USER INFO	+		+	INFO(USR)			
				→	200 OK INFO			
	USER INFO	→		→	INFO(USR)			
	552771115			+	200 OK INFO			
	USER INFO	+		+	INFO(USR)			
	552	+ +		→	200 OK INFO			
		C	ommunication					
	DISC	→ Ĭ	5a	→	BYE(REL)			
	RELEASE	-		′	200 OK BYE			
	REL_COM	+		Ť	200 011212			
	KEL_COM	7						

TP706205	SIP reference: RFC 3261 [4]			Q.191	ISUP reference: 2.5 [1], clause 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 phase of the call and the	UUS3 explicit non-essential - explicit rejection during call (service not provided - FRJ) To verify that the UUS3 explicit non-essential service can be rejected during the active phase of the call and the Service 3 field in the user-to-user indicators in the FRJ are services.							
	to "service 3 not provided	".							
SIP parameter values									
ISDN parameter values									
Comments	ISDN		SUT		SIP				
	SETUP	→		→	INVITE(IAM)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONN	+		+	200 OK INVITE				
			Communicat	tion					
	FAC(UU3 req not ess)	→		→	INFO(FAR req UU3 not ess)				
				+	200 OK INFO				
	FAC(UU3 ret err)	+		+	INFO(FRJ resp UU3 not prov)				
				→	200 OK INFO				
			Communicat	tion					
	DISC	→		→	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	S3							
SIP selection criteria									
ISUP selection criteria	PICS 11/3								
Test purpose	FAA or FRJ)	To verify that the IUT can successfully complete a call with an UUS3 request in the FAR, if							
SIP parameter values									
ISDN parameter values									
Comments	ISDN		SUT		SIP				
	SETUP	→		→	INVITE(IAM)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONN	+		+	200 OK INVITE				
		(Communication	n					
	FAC(UU3 req not ess)	→		→	INFO(FAR req UU3 not ess)				
				+	200 OK INFO				
		(Communication	n					
	DISC	→		→	BYE(REL)				
	RELEASE	+		+	200 OK BYE				
	REL_COM	→							

A.1.1.2.7 Closed User Group (CUG)

TP707001	SIP reference: RF0	C 3261 [4]	I	Q.19	-	SUP reference: [1], clause 1.5.2.1.1 i) a)/ Q.735 [i.9]				
TSS reference	ISDN-(ISUP)-SIP/SS/CUG									
SIP selection criteria										
ISUP selection criteria										
Test purpose	To verify that the IUT can sinterlock code together with optional forward call it	CUG without outgoing access in IAM To verify that the IUT can successfully establish a CUG call by including the CUG interlock code together with an indication of "CUG call, outgoing access not allowed" in the optional forward call indicators in the IAM. The ISUP preference indicator of the forward call indicators in the IAM should be set to "ISUP required all the way".								
SIP parameter values										
ISDN parameter values										
Comments	ISDN		SU			SIP				
	SETUP	→			^	INVITE(IAM, CUG -OA)				
	ALERTING	←			+	180 Ringing				
	CONN	←			+	200 OK INVITE				
					→	ACK				
			Communi	cation						
	DISC	→			→	BYE				
	REL	+			+	200 OK BYE				
	REL_COM	→								

TP707002	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/S	SS/CUG							
SIP selection criteria									
ISUP selection criteria									
Test purpose	activated	CUG call without outgoing access; class of called user CUG without IA, 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	→		→	180 Ringing(ACM)				
	CONN	→		→	200 OK INVITE(ANM)				
			Communication	n					
	DISC	+		+	BYE(REL)				
	REL	→		→	200 OK BYE				
	REL_COM	-							

TP707002	SIP reference: F	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/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)			
				1	500 Server Internal Error(REL#55)			
				+	ACK			

TP707003	SIP referer	nce: 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	SS/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	→		→	180 Ringing(ACM)				
	CONN	→		→	200 OK INVITE(ANM)				
			Communication	n	. ,				
	DISC	+		+	BYE(REL)				
	REL	→		→	200 OK BYE				
	REL_COM	+							

TP707004	SIP reference	e: RFC 320	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/SS/	CUG						
SIP selection criteria								
ISUP selection criteria								
Test purpose	CUG call with outgoing access; class of called user CUG with IA and no ICB activated To verify that the IUT can successfully establish a CUG call with outgoing access.							
SIP parameter values					<u> </u>			
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	+		+	INVITE(IAM, CUG +OA)			
	ALERTING	→		→	180 Ringing(ACM)			
	CONN	→		→	200 OK INVITE(ANM)			
			Communicati	on				
	DISC	+		+	BYE(REL)			
	REL	→		→	200 OK BYE			
	REL_COM	+						

TP707005	SIP reference	e: RFC	3261 [4]	Q	ISUP reference: .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 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)
				→	500 Server Internal Error(REL#55)
				+	ACK

TP707006	SIP reference: F	RFC 32	61 [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	To verify that the IUT rej the REL cause #87 "Use	ects the	e CUG call with	n a 50	d user non-CUG O Server Internal Error encapsulated
SIP parameter values					
ISDN parameter values					
Comments	ISDN		SUT		SIP
				+	INVITE(IAM CUG -OA)
				→	500 Server Internal Error(REL#87)
				+	ACK

TP707007	SIP reference	: RFC 3261 [4]	Q.191	-	SUP reference: , 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 outgoi				
SIP parameter values		-			
ISDN parameter values					
Comments	ISDN		SUT		SIP-I
	SETUP	+		+	INVITE(IAM CUG, +OA)
	ALERTING	→		→	180 Ringing(ACM)
	CONN	→		→	200 OK INVITE(ANM)
				+	ACK
		Con	nmunication	•	
	DISC	+		+	BYE(REL)
	REL	→		→	200 OK BYE(RLC)
	REL_COM	+			

TP707008	SIP reference: I	RFC 3261 [4]	Q.19	ISUP reference: 912.5 [1], clause 1.5.2.5.1; table 1-2/ Q.735 [i.9]			
TSS reference	ISDN-(ISUP)-SIP/SS/CI	JG					
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)			
			→	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)		SU	Т		SIP-I	
	SETUP	+			+	INVITE(IAM)	
	ALERTING	→			→	180 Ringing(ACM)	
	CONN	→			→	200 OK INVITE(ANM)	
					+	ACK	
		С	Commun	ication			
	DISC	+	•		+	BYE(REL)	
	REL	→			→	200 OK BYE(RLC)	
	REL_COM	+	•				

TP707010	SIP reference:	RFC 326	61 [4]	Q.19	ISUP reference: 012.5 [1], clause 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		jects the	CUG call with	h a 500	d user other CUG without IA) Server Internal Error encapsulated
SIP parameter					
values					
ISDN parameter					
values					
Comments	ISDN (CUG -IA)		SUT		SIP-I
				+	INVITE(IAM CUG -OA)
				→	500 Server Internal Error(REL#87)
				+	ACK

TP707011	SIP reference: I	RFC 3261 [4]	Q.19	ISUP reference: 912.5 [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		ects the CUG call wit	h a 500	ser other CUG without IA) Server Internal Error encapsulated
SIP parameter values				
ISDN parameter values				
Comments	ISDN (CUG -IA)	SUT		SIP-I
			+	INVITE(IAM CUG +OA)
			→	500 Server Internal Error(REL#87)
			←	ACK

TP707012	SIP reference:	RFC 3261 [4]	Q.19	ISUP reference: 912.5 [1], clause 1.5.2.5.1; table 1-2/ Q.735 [i.9]
TSS reference	ISDN-(ISUP)-SIP/SS/CI	JG		
SIP selection criteria				
ISUP selection criteria				
Test purpose	CUG call without outgoi To verify that the IUT re the REL cause #87 "Us	jects the CUG call wit	th a 500	ser: other CUG with IA O Server Internal Error encapsulated
SIP parameter values				
ISDN parameter values				
Comments	ISDN (CUG A +IA)	SUT		SIP-I
			←	INVITE(IAM CUG B -OA)
			→	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/0	CUG							
SIP selection criteria									
ISUP selection criteria									
Test purpose		CUG call with outgoing access; class of called user other CUG with IA To verify that the IUT can successfully establish a non-CUG call.							
SIP parameter			-						
values									
ISDN parameter values									
Comments	ISDN (CUG A +IA)		SUT		SIP-I				
	SETUP	+		+	INVITE(IAM, CUG B +OA)				
	ALERTING	→		→	180 Ringing(ACM)				
	CONN	→		→	200 OK INVITE(ANM)				
			Communicatio	n					
	DISC	+		+	BYE(REL)				
	REL	→		→	200 OK BYE				
	REL_COM	+							

A.1.1.2.8 SUB-addressing (SUB)

TP708001	SIP reference: RFC	3261 [4]	Q.1912	ISUP reference: 2.5 [1], clause 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 the MIME body ATP with called sub-address included							
ISDN parameter values	SETUP: called sub-address i	ncluded						
Comments	ISDN	5	SUT	SIP-I				
	SETUP(SUB)	→	→	INVITE(IAM, ATP(SUB))				
	ALERTING	+	+	180 Ringing				
	CONN	+	+	200 OK INVITE				
			→	ACK				
	Communication							
	DISC	→	→	BYE				
	REL	+	+	200 OK BYE				
	REL_COM	→						

TP708002	SIP reference: RF0	C 3261 [4	•	-	SUP reference: .5 [1], clause 8.5.2.5.1/ Q.731 [i.2]		
TSS reference	ISDN-(ISUP)-SIP/SS/SUB						
SIP selection criteria							
ISUP selection criteria							
Test purpose		successi	ully established if t	he IAN	t parameter M contains the sub-address in ress is passed on to the user		
SIP parameter values	INVITE: IAM encapsulated	in the MI	ME body ATP with	called	sub-address included		
ISDN parameter values	SETUP: called sub-address	s included	d				
Comments	ISDN		SUT		SIP-I		
	SETUP(SUB)	+		+	INVITE(IAM, ATP(SUB))		
	ALERTING	→		→	180 Ringing(ACM)		
	CONN	→		→	200 OK INVITE(ANM)		
	Communication ← ACK						
	DISC	+		+	BYE(REL)		
	REL	→		→	200 OK BYE(RLC)		
	REL_COM	+					

A.1.1.2.9 Malicious Call Identification (MCID)

TP709001	SIP reference	e: 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	set to "MCID request	can successfully reply to	MCID respon	g the MCID request indicator se indicator set to "MCID			
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	SI	JT	SIP			
	SETUP	→	→	INVITE(IAM)			
			+	INFO(IDR requested)			
			→	200 OK INFO			
			→	INFO(IRS included)			
			+	200 OK INFO			
	ALERTING	+	+	180 Ringing(ACM)			
	CONN	+	+	200 OK INVITE(ANM)			
			→	ACK			
		Commu	nication				
	DISC	→	→	BYE(REL)			
	REL	+	+	200 OK BYE			
	REL_COM	→					

TP709002	SIP reference: R	SIP reference: RFC 3261 [4]			SUP reference: 5 [1], clause 7.5.2.1.1/ Q.731.7 [i.3]			
TSS reference	ISDN-(ISUP)-SIP/SS/MC	ID						
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Successful MCID reque							
	set to "MCID request" by included" and the calling	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 encapsulated							
values	INFO: The encapsulated Calling party number of to			esponse in	dicator "included" and the			
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	+		+	INVITE(IAM)			
				→	INFO(IDR requested)			
				+	200 OK INFO			
				+	INFO(IRS included)			
				→	200 OK INFO			
	ALERTING	→		→	180 Ringing(ACM)			
	CONN	→		→	200 OK INVITE(ANM)			
				→	ACK			
		С	ommunicati	on				
	DISC	+		+	BYE(REL)			
	REL	→		→	200 OK BYE			
	REL_COM	+						

TP709003	SIP reference: RFC		ij	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 will ad been received. The IUT should "MCID request" by sending	Guccessful 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 ID INFO: The encapsulated IR Calling party number of the	S contai	ns the MCII						
ISDN parameter values									
Comments	ISDN		SUT	-	SIP				
	SETUP	→		→	INVITE(IAM)				
	ALERTING	+		←	180 Ringing(ACM)				
				+	INFO(IDR requested)				
				→	200 OK INFO				
				→	INFO(IRS included)				
				←	200 OK INFO				
	CONN	+		←	200 OK INVITE(ANM)				
				→	ACK				
			Communic	cation					
	DISC	→		→	BYE(REL)				
	REL	+		+	200 OK BYE				
	REL_COM	→							
NOTE: This situa	tion may occur e.g. if the cal	l has bee	en forwarde	d before reac	hing the destination.				

TP709004	SIP reference:	RFC 3261 [4]			SUP reference: .5 [1], clause 7.5.2.1.1/ Q.731.7 [i.3]				
TSS reference	ISDN-(ISUP)-SIP/SS/M	1CID							
SIP selection criteria									
ISUP selection criteria									
Test purpose	To verify that the IUT been received. The IUT "MCID request" by sen and the calling party r	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 encapsulate	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	→		→	180 Ringing(ACM)				
				→	INFO(IDR requested)				
				+	200 OK INFO				
				+	INFO(IRS included)				
				→	200 OK INFO				
	CONN	→		→	200 OK INVITE(ANM)				
				←	ACK				
		C	ommunicatio	n					
	DISC	+		←	BYE(REL)				
	REL	→		→	200 OK BYE				
	REL_COM	←							
NOTE: This situ	ation may occur e.g. if the	e call has been t	forwarded be	fore reac	hing the destination.				

TP709005	SIP reference	: RFC 3261 [4]	Q.191	ISUP reference: 2.5 [1], clause 7.5.2.1.1/ Q.731.7 [i.3]
TSS reference	ISDN-(ISUP)-SIP/SS/I	MCID		
SIP selection criteria				
ISUP selection criteria				
Test purpose	To verify that the IUT of set to "MCID request"	by sending an IRS with	o an <mark>IDR</mark> havir n MCID respor	ng the MCID request indicator nse indicator set to "MCID ess in the access transport
SIP parameter values	INFO: The encapsulat	ed IDR contains the Mo ed IRS contains the Mo calling sub-address of	CID response i	ndicator "included", the Calling
ISDN parameter values				
Comments	ISDN	SI	UT	SIP
	SETUP	→	→	INVITE(IAM)
			←	INFO(IDR requested)
			→	200 OK INFO
			→	INFO(IRS included)
			+	200 OK INFO
	ALERTING	←	←	180 Ringing(ACM)
	CONN	+	+	200 OK INVITE(ANM)
			→	ACK
		Commu	nication	
	DISC	→	→	BYE(REL)
	REL	+	+	200 OK BYE
	REL_COM	→		

TP70906	SIP reference	: RFC 3261 [4]			ISUP reference: .5 [1], clause 7.5.2.1.2/ Q.731.7 [i.3]			
TSS reference	ISDN-(ISUP)-SIP/SS/N	MCID						
SIP selection criteria								
ISUP selection criteria	PICS 9/1							
Test purpose	MCID request - MCID To verify that the IUT r indicator set to "MCID	ejects a MCID	request by		RS with the MCID response			
SIP parameter	INFO: The encapsulate							
values	INFO: The encapsulate	INFO: The encapsulated IRS contains the MCID response indicator "not included"						
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	→		→	INVITE(IAM)			
				-	INFO(IDR requested)			
				→	200 OK INFO			
				→	INFO(IRS not included)			
				-	200 OK INFO			
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	-		-	200 OK INVITE(ANM)			
				→	ACK			
			Communica	ation				
	DISC	→		→	BYE(REL)			
	REL	+		+	200 OK BYE			
	REL_COM	→						

TP70907	SIP reference	e: RFC 3261 [4]		ISUP reference: 2.5 [1], clause 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	request I		RS with the MCID response			
SIP parameter	INFO: The encapsula	ted IDR contain	s the MC	ID Request in	dicator "requested"			
values	INFO: The encapsula	INFO: The encapsulated IRS contains the MCID response indicator "not included"						
ISDN parameter values								
Comments	ISDN		SL	JT	SIP			
	SETUP	+		+	INVITE(IAM)			
				→	INFO(IDR requested)			
				+	200 OK INFO			
				+	INFO(IRS not included)			
				→	200 OK INFO			
	ALERTING	→		→	180 Ringing(ACM)			
	CONN	→		→	200 OK INVITE(ANM)			
				+	ACK			
			Commun	nication				
	DISC	+	_	+	BYE(REL)			
	REL	→		→	200 OK BYE			
	REL_COM	+						

TP70908	SIP reference: RFC	3261 [4]		ISUP reference: Q.1912.5 [1], clause 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			S is received within timer T39 or set to "MCID requested".				
SIP parameter values	INFO: The encapsulated IDF	INFO: The encapsulated IDR contains the MCID Request indicator "requested"						
ISDN parameter values								
Comments	ISDN		SUT	SIP				
	SETUP	→	→	INVITE(IAM)				
			+	INFO(IDR requested)				
			→	200 OK INFO				
		T39	expiry					
	ALERTING	+	+	180 Ringing(ACM)				
	CONN	+	+	200 OK INVITE(ANM)				
			→	ACK				
		Comm	unication					
	DISC	→	→	BYE(REL)				
	REL	+	+	200 OK BYE				
	REL_COM	→						

A.1.1.2.10 Conference call (CONF)

TP710001	SIP r	eference: RFC 3	261 [4]	ISUP reference: Q.1912.5 [1],				
				Q.734 [i.7], clause 1.5.2.1.1.2				
TSS reference	ISDN-(ISUP)	-SIP/SS/CONF		and [m], diaded helemine				
SIP selection	10211 (1001)	- CII 700700111						
criteria								
ISUP selection								
criteria								
Test purpose			sfully begin the	conference from an active call and notify				
		arties correctly.		·				
				on "conference established" is received in				
			call at the end to	erminal and check that all network resources				
OID	are released		/IOLID 000	L d L' d BAIRAT L L				
SIP parameter values	INFO: Conte	nt-Type: application	on/ISOP; CPG e	encapsulated in the MIME body				
1 411 41 4	EAC: Decise	ONE involve com						
ISDN parameter values		CONF invoke components						
Comments	II AC. Degino	CONT TELUTITIESUI	Component					
ISDN	SUT	•	SIP 1	SIP 2				
SETUP(CRx)	→ ·	→	INVITE	011 2				
ALERTING	(-	180 Ringing					
CONN	(+	200 OK INVITI	E				
FAC(BeginCONF_ir	ıv) →	→	INFO(CPG cor	nf est)				
FAC(BeginCONF_rr	·) ((200 OK INFO					
	•	Conferen	ce communicati	ion				
DISC(CRx)	→ BYE							
RELEASE	← 200 OK BYE							
REL_COMP	→							

TP710002	SI	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1],	
TCC reference	ICDN (ICI	Q.734 [i.7], clause 1.5.2.1.1.2 ISDN-(ISUP)-SIP/SS/CONF					
TSS reference SIP selection	12014-(120	UP)-SIP/SS/CO	INF				
criteria							
ISUP selection							
criteria							
Test purpose	Verify tha	at the IUT can	SUCCE	essfully begin t	he conferenc	ce from idle state and is ab	le.
	to add a d Establish party to th in the CP0 note).	conferee to a c a conference fr he conference. I G. The conferer	onfer om IS Notify nce is	rence and notified to SIP 1. Est subscriber at S released by cal	fy the implied stablished a could be a could like the stablished a could like the stable to the stable the stable to the stable the	I parties correctly. onnection to SIP 2 and add to him/her "other party added he served user at IADN (see	this d"
SIP parameter values				•	encapsulated in	in the MIME body	
ISDN parameter	FAC: Beg	ginCONF invoke	comp	oonent			
values	FAC: Beg	ginCONF return	result	component			
		ICONF invoke c					
0	DISC: add	dCONF return r	esult (component			
Comments	10	\. I=		loip 4	1	loup o	
ISDN		SUT		SIP 1		SIP 2	
SETUP(CRx)		>	→	INVITE			
ALERTING		-	+	180 Ringing	_		
CONN		-	(200 OK INVIT			
FAC(BeginCONF_ir		7	→	INFO(CPG coi	nr est)		
FAC(BeginCONF_rr	() \	-	_	200 OK INFO			
SETUP(CRy)	-	>			-	NVITE	
ALERTING	•	-			+	180 Ringing	
CONN	•	-			+	200 OK INVITE	
FAC(AddCONF_inv		→			-	INFO(CPG conf est)	
DISC(AddCONF_rr,	CRy)	-			+	200 OK INFO	
RELEASE		→	→	INFO(CPG pa	rty add)		
REL_COMP	•	(+	200 OK INFO			
		Con	feren	ce communicati	on	•	
DISC(CRx)		>	→	BYE			
RELEASE	•	-	+	200 OK BYE			
REL_COMP	-	→					
					→	▶ BYE	
					+		
new affect	ted confere		eric n	otification indi	cator set to "o	ould be sent by the IUT to th other party added" to the nor rogress".	

TP710003	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.5.2.1.1.3				
TSS reference	ISDN-(IS	ISDN-(ISUP)-SIP/SS/CONF						•
SIP selection criteria								
ISUP selection								
criteria	\							
Test purpose	implied p Establish subscrib and che The con	oartie h a co er at ck tha feren	s correctly. onference fro SIP 1 by sen at the notifica ce is release	om IS nding ation '	DN to SIP 1. Ac him/her "other 'isolated" is rec call clearing by	dd SIP 2 to the party added" i eived in the C the served us	e co n th PG er a	conference and notify the onference and notify the CPG. Isolate a conferee at ISDN (see note).
SIP parameter values	INFO: C	onter	nt-Type: app	licatio	n/ISUP; CPG e	encapsulated in	n th	e MIME body
ISDN parameter values	FAC: Be FAC: ad DISC: ad FAC: iso FAC: iso	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						
Comments					•			
ISDN		SUT			SIP 1			SIP 2
SETUP(CRx)		^		→	INVITE			
ALERTING		4		+	180 Ringing			
CONN		+		+	200 OK INVITI	E		
FAC(BeginCONF_ir		→		→	INFO(CPG cor	nf est)		
FAC(BeginCONF _rr	·)	+		←	200 OK INFO			
SETUP(CRy)		→				-3		INVITE
ALERTING		+				•		180 Ringing
CONN	٥- ١	+				•		200 OK INVITE
FAC(AddCONF_inv		→				1		INFO(CPG conf est)
DISC(AddCONF_rr,	CRy)	+				•		200 OK INFO
RELEASE		→		→	INFO(CPG par	rty add)		
REL_COMP		+		+	200 OK INFO			
EAC(lastCant in C	ים.	_	Coni	erend	e communicati			INEO(000 :1)
FAC(IsolConf_inv,C		}				7		INFO(CPG isol)
FAC(IsolConf_rr,CR	\^)	+		_	INFO(CDC por	rty icol)		200 OK INFO
				→	INFO(CPG par 200 OK INFO	ity isol)		
			Private cor		ication ISDN w	ith SIP 1		
FAC(ReattConf_inv	CR _x)	→	i iivaic col	IIIIuI	III III III W	<u> </u>	<u> </u>	INFO(CPG reatt)
FAC(ReattConf_rr,C		/						200 OK INFO
	/			→	INFO(CPG par			LUG OIX II II U
				′	200 OK INFO	ity routty		
			Conf	_	ce communicati	on L		1
DISC(CRx)		→	00111	→	BYE			
RELEASE	← 200 OK BYE							
REL_COMP	→ 200 OK BIE							
						-3	>	BYE
						- -		200 OK BYE
to the affe	ected conf the non-	feree affect	and the gen ted conferee	eric is. The	notification inc e event indicato	in call progre dicator set to or in the CPG	ss : "oth	should be sent by the IUT her party isolated" should uld be set to "progress". the conference.

TP710004	SIP	reference	: RFC 32		SUP reference:		
					Q.1912.5 [1], Q.734 [i.7], clause 1.5.2.1.1.5		
TSS reference	ISDN-(ISUP)-SIP/SS/	CONF		Z.: 0 . [.:	,	
SIP selection	102.1 (1001	,,					
criteria							
ISUP selection							
criteria							
Test purpose	Verify that th	ne IUT car	reate a	a private comm	nunication betw	een the served user and one	
• •					correctly (see n		
SIP parameter					encapsulated in		
values		7.	• •	,	•	•	
ISDN parameter	FAC: Begin(CONF invo	oke com	onent			
values	FAC: Begin(
	FAC: addCC						
	DISC: addC						
	SETUP: spli						
	CONNECT:	splitCONI	F return i	esult componer	nt		
Comments							
ISDN		JT		SIP 1		SIP 2	
SETUP(CRx)	→		→	INVITE			
ALERTING	+	•	+	180 Ringing			
CONN	←		+	200 OK INVITI	E		
FAC(BeginCONF_i	inv) →	•	→	INFO(CPG cor	nf est)		
FAC(BeginCONF_	rr) (+	200 OK INFO			
SETUP(CRy)	→	•			→	INVITE	
ALERTING	←				+	180 Ringing	
CONN	(+	200 OK INVITE	
FAC(AddCONF_inv	v,CRy) →				→	INFO(CPG conf est)	
DISC(AddCONF_rr					+	200 OK INFO	
RELEASE	→	,	→	INFO(CPG pa	rty add)		
REL_COMP	←		+	200 OK INFO	, ,		
	l .		Conferen	ce communicati	on		
SETUP(SplitConf_	inv,CRy) →				→	INFO(CPG conf disc)	
CONN(SplitIConf_	. , ,,				-	200 OK INFO	
(- 	,,		→	INFO(CPG pa			
			-	200 OK INFO	7 -F/		
	1	Private		nication ISDN w	ith SIP 1	1	
FAC(AddCONF _inv	v,CRy) →		33		<u> </u>	INFO(CPG conf est)	
DISC(AddCONF_rr					-	200 OK INFO	
RELEASE	,orty) ₹		→	INFO(CPG pa			
REL_COMP	7			200 OK INFO	11, 444,	+	
TALE_OOM			Conferen	ce communicati	on L		
DISC(CRx)	 		→	BYE			
RELEASE	-		-	200 OK BYE			
REL_COMP	→			ZOU ON DIE			
IVEE_CONIE	7				→	BYE	
	-				-	200 OK BYE	
NOTE 4 TI			1 11			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 refer	ence: RFC 3	261 [4]	IS	SUP reference: Q.1912.5 [1],
				Q.734 [i	.7], clause 1.5.2.1.1.6
TSS reference	ISDN-(ISUP)-SIF	P/SS/CONF			27
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose					n the conference, if
	requested by the				
					conference and notify
					the CPG. Release the
			onterence is rele	eased by call cle	aring by the served user at
SIP parameter	ISDN (see note). INFO: Content-T		an/ICLID: CDC	naanaulatad in	the MIME hady
values	IINFO. Content-1	ype. applicati	UII/ISUF, CPG 6	encapsulated in	ule Milvie Dody
ISDN parameter	FAC: BeginCON	F invoke com	nonent		
values	FAC: BeginCON				
	FAC: addCONF				
	DISC: addCONF				
	FAC: dropCONF	invoke comp	onent		
	FAC: dropCONF	return result	component		
Comments			_		
ISDN	SUT		SIP 1		SIP 2
SETUP(CRx)	→	→	INVITE		
ALERTING	+	←	180 Ringing		
CONN	+	←	200 OK INVIT		
FAC(BeginCONF_		→	INFO(CPG co	nf est)	
FAC(BeginCONF_	rr) (←	200 OK INFO		
SETUP(CRy)	→			→	INVITE
ALERTING	+			(180 Ringing
CONN	((200 OK INVITE
FAC(AddCONF_in				→	INFO(CPG conf est)
DISC(AddCONF_ri	r,CRy) ←		INIEO/ODO		200 OK INFO
RELEASE	7	→ ←	INFO(CPG pa	пу ада)	
REL_COMP		1 =		·	
FAC(DropCONF _ir	ov CDv)	Conterer	nce communicati		DVE
FAC(DropCONF_ir				→	BYE 200 OK BYE
- YOUNIOHOOME_II	,01\\)	→	INFO(CPG pa		ZOU ON BTE
		 	200 OK INFO	ity uisc)	
			mmunication		
DISC(CRx)	→	→	BYE		
RELEASE	-	-	200 OK BYE		
REL_COMP	→		200 OK BIL		
	=	<u> </u>			l call release precedures 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	nce: RFC 3	261 [4]		GUP reference: Q.1912.5 [1], .7], clause 1.5.2.1.1.7				
TSS reference	ISDN-(ISUP)-SIP/	SS/CONF		-					
SIP selection	, ,								
criteria									
ISUP selection									
criteria									
Test purpose	requested by the Establish a confersubscriber at SIP	To verify that IUT can successfully disconnect a conferee from the conference, if equested by the conferee, and notify the implied parties correctly. 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. Release request by the conferee at SIP 1. The conference is released by call clearing by the served user at							
SIP parameter	INFO: Content-Typ	oe: applicati	on/ISUP: CPG	encapsulated in t	the MIME body				
values		11	,		•				
ISDN parameter values	FAC: BeginCONF FAC: addCONF in DISC: addCONF r FAC: partyDISC in	FAC: BeginCONF invoke component FAC: BeginCONF return result component FAC: addCONF invoke component DISC: addCONF return result component FAC: partyDISC invoke component FAC: partyDISC return result component							
Comments	T								
ISDN	SUT		SIP 1		SIP 2				
SETUP(CRx)	→	→	INVITE						
ALERTING	+	+	180 Ringing						
CONN	+	+	200 OK INVIT	E					
FAC(BeginCONF_i	nv) →	→	INFO(CPG co	nf est)					
FAC(BeginCONF_r		+	200 OK INFO	,					
· · ·									
SETUP(CRy)	→			→	INVITE				
ALERTING	((180 Ringing				
CONN	←			+	200 OK INVITE				
FAC(AddCONF_inv	v,CRy) →			→	INFO(CPG conf est)				
DISC(AddCONF_rr	,CRy) ←			+	200 OK INFO				
RELEASE	→	→	INFO(CPG pa	rty add)					
REL_COMP	←	+	200 OK INFO						
		Conferer	nce communicati	ion					
FAC(PartyDisc_inv		+	BYE						
FAC(PartyDisc_rr,0	CRy) →	→	200 OK BYE						
				→	INFO(CPG party disc)				
				←	200 OK INFO				
		Co	mmunication						
DISC(CRx)	→			→	BYE				
RELEASE	←			←	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 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: RFC 3	261 [4]		GUP reference: Q.1912.5 [1],
		015/00/00/15		Q.734 [i.	.7], clause 1.5.2.1.1.8
TSS reference	ISDN-(ISUP)	-SIP/SS/CONF			
SIP selection criteria					
ISUP selection					
criteria 	<u> </u>				
Test purpose	requested by conferee. Establish a conferee subscriber at released by conference or co	the served user, conference from IS t SIP 1 by sending call clearing by th	, and initiate the SDN to SIP 1. A g him/her "other e served user at	normal call releaded SIP 2 to the coparty added" in teleprotes to SDN (see note	om the conference, if ase procedure towards each conference and notify the CPG. The conference is .).
SIP parameter	INFO: Conte	nt-Type: applicati	ion/ISUP; CPG e	encapsulated in t	the MIME body
values	EAC: D = =: 0	ONE investor a			
ISDN parameter		CONF invoke com			
values		CONF return resu			
		NF invoke compo ONF return result			
Comments					
SDN	SU	JT	SIP 1		SIP 2
SETUP(CRx)	→	→	INVITE		
ALERTING	+	+	180 Ringing		
CONN	+	(200 OK INVIT	E	
FAC(BeginCONF _i			INFO(CPG co	nf est)	
FAC(BeginCONF _r	r) ←	+	200 OK INFO		
SETUP(CRy)	→			→	INVITE
ALERTING	′			′	180 Ringing
CONN	+			(200 OK INVITE
FAC(AddCONF _inv				→	INFO(CPG conf est)
DISC(AddCONF _inv	, , ,				200 OK INFO
RELEASE	,CRy) ←		INFO(CPG pa		200 OK INFO
REL_COMP	/		200 OK INFO	ity add)	+
			nce communicati	ion	
DISC(CRx)	→		BYE		
RELEASE	′		200 OK BYE		
REL COMP	→		200 01 01 0	→	INFO(CPG party disc)
IVEE_OOMI	 			-	200 OK INFO
				→ ·	BYE
	+		+	-	200 OK BYE
NOTE: The IUT :	should send R				ZUU UN DIE

TP710008	SIP re	ference: RFC 3	3261 [4]		SUP reference: Q.1912.5 [1], [i.7], clause 1.6.15			
TSS reference	ISDN-(ISUP)-S	SIP/SS/CONF						
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose					and subsequently added to			
		call, but that the	IUT sends the "	conference esta	blished" notification to the			
	held user.							
SIP parameter	INFO: Content	t-Type: applicat	ion/ISUP; CPG e	encapsulated in t	the MIME body			
values								
ISDN parameter		NF invoke com						
values		NF return resu						
		F invoke comp						
0	IDISC: addCOI	NF return result	component					
Comments	lou iz	_	loip (Tour o			
ISDN	SUT		SIP 1		SIP 2			
SETUP(CRx)	→	→	INVITE					
ALERTING	(←	180 Ringing	_				
CONN	+	←	200 OK INVIT					
FAC(BeginCONF_		→	INFO(CPG co	nf est)				
FAC(BeginCONF_	rr) ←	←	200 OK INFO					
SETUP(CRy)	→			→	INVITE			
ALERTING	+			+	180 Ringing			
CONN	+			←	200 OK INVITE			
HOLD	→			→	INFO(CPG hold)			
				(200 OK INFO			
FAC(AddCONF_inv				→	INFO(CPG conf est)			
DISC(AddCONF_ri	r,CRy) ←			+	200 OK INFO			
RELEASE	RELEASE → INFO(CPG party add)							
REL_COMP	+	+	200 OK INFO					
		Confere	nce communicati	ion				
DISC(CRx)	→	→	BYE					
RELEASE	+	(200 OK BYE					
REL_COMP	→			→	INFO(CPG party disc)			
				+	200 OK INFO			
				→	BYE			
		1		+	200 OK BYE			

TP710009		SIP reference: RFC 3261 [4] ISUP reference: Q.1912.5 [1] Q.734 [i.7], clause											
TSS reference	ISDN-(ISUP)-SI	ISDN-(ISUP)-SIP/SS/CONF											
SIP selection													
criteria													
ISUP selection													
criteria													
Test purpose		To verify that no hold and no retrieve notification is sent to the conferees when the											
	conference cont												
SIP parameter	INFO: Content-1	ype: applicati	on/ISUP; CPG ϵ	encapsulated in	the MIME body								
values													
ISDN parameter	FAC: BeginCON												
values	FAC: BeginCON												
	FAC: addCONF												
Commonst	DISC: addCONF	return result	component										
Comments	lo: :=		OID 4	ı	loip o								
ISDN	SUT		SIP 1		SIP 2								
SETUP(CRx)	→	→	INVITE										
ALERTING	+	(180 Ringing	_									
CONN	+	<u> </u>	200 OK INVIT										
FAC(BeginCONF_		→	INFO(CPG co	nt est)									
FAC(BeginCONF_	_rr) ←	←	200 OK INFO										
SETUP(CRy)	→			→	INVITE								
ALERTING	+			(180 Ringing								
CONN	((200 OK INVITE								
FAC(AddCONF _in				→	INFO(CPG conf est)								
DISC(AddCONF_r				-	200 OK INFO								
RELEASE	→	→	INFO(CPG pa	rty add)									
REL_COMP	+	+	200 OK INFO										
		Conferen	ce communicati	ion									
HOLD	→												
RETRIVE	→												
DISC(CRx)													
RELEASE	+												
REL_COMP			→	INFO(CPG party disc)									
				+	200 OK INFO								
				→	BYE								
				+	200 OK BYE								

A.1.1.2.11 Explicit Call Transfer (ECT)

TP711001	SIP referenc	e: RI	FC 3261	[4]			reference: lause 7.5.2.1.1.1 a)/		
								32.7 [i.5]	
TSS reference	ISDN-(ISUP)-SIP/SS	/ECT	Ī					• •	
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Capability of storin	g and	d sendin	g tl	ne addit	ional calling par	rty r	number in the call	
	transfer number								
	To verify that the IUT								
								r have been received er remote user in the	
	call transfer numbe								
SIP parameter	INVITE: encapsulate								
values	INVITE B SDP sende								
valuoo	INFO C: encapsulate	ad FA	C contair	าร (neneric r	notification call tra	ansf	er active, call transfer	
						ing party number			
ISDN parameter	FAC: ECT invoke red					у г			
values		DISCONNECT: ECT invoke return result component							
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3	
	SETUP	+		←	INVITE	(IAM)			
	ALERTING	→		→	180 Rir	nging(ACM)			
	CONN	→		→	200 OK	(INVITE(ANM)			
				←	ACK				
	HOLD	→		→	INVITE	(CPG hold)			
				+	200 OK	INVITE			
				→	ACK				
	SETUP	→					→	INVITE(IAM)	
	ALERTING	+						180 Ringing(ACM)	
	CONN	+					+	200 OK INVITE(ANM)	
ı							→	ACK	
	FAC(ECT invoke)	→							
ı	DISCONNECT(rr)	+				FAC ect active)			
	RELEASE	→		←	200 OK	(INFO			
	RELEASE COMPL	+						INFO (FAC ect active)	
							←	200 OK INFO	
						BYE(REL)		BYE(REL)	
					20	00 OK BYE(RLC	+	200 OK BYE(RLC	

TP711002		SIP reference: RFC 3261 [4] ISL Q.1912.5 [1]								
TSS reference	ISDN-(ISUP)-SIP/SS	ISDN-(ISUP)-SIP/SS/ECT								
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	number To verify that the IUT been received from t remote user in the categories.	is al he re	ble to sto mote use ansfer nu	re t er. T ı m k		er whoy the	nen only this CLI has e IUT to the other is when the call transfer is			
SIP parameter					e calling party number of					
values					d CPG generic notificati					
					generic notification call to					
IODAL		number derived from the calling party number of user B (SIP-I 1)								
ISDN parameter values	FAC: ECT invoke request component									
Comments	DISCONNECT: ECT invoke return result component						SIP-I 3			
Comments	ISDN 2 SETUP	+	SUT	L	SIP-I 1 INVITE(IAM)	-	SIP-1 3			
	ALERTING	→			` /	-				
		7 →			180 Ringing(ACM)	-				
	CONN	7			200 OK INVITE(ANM) ACK					
				_	ACK	-				
	HOLD	→		_	INVITE(CPG hold)	-				
	HOLD	7			200 OK INVITE	-				
		+		` →	ACK	-				
				_	AON					
	SETUP	→				-	INVITE(IAM)			
	ALERTING	+					180 Ringing(ACM)			
	CONN	+					200 OK INVITE(ANM)			
	001111	† <u> </u>					ACK			
	FAC(ECT invoke)	→				Ť				
	DISCONNECT(rr)	+		→	INFO (FAC ect active)	1				
	RELEASE	→			200 OK INFO	\top				
	RELEASE COMPL	+				→	INFO (FAC ect active)			
							200 OK INFO			
			•	•	I .					
					BYE(REL)	→	BYE(REL)			
							200 OK BYE(RLC			

TP711003		SIP reference: RFC 3261 [4]						reference: ause 7.5.2.1.1.1 b)/ 32.7 [i.5]	
TSS reference	ISDN-(ISUP)-SIP/SS	/ECT	•						
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Capability of storing transfer number To verify that the IUT number when the count the remote user. This transfer number in a	is al onne	ole to sto cted nun rmation is	re ti 1be s se	he addition of the control of the co	onal connected regeneric number IUT to the other	num er h	ber in the generic ave been received from mote user in the call	
SIP parameter									
values	200 OK INVITE: enca	NVITE B SDP sendonly, encapsulated CPG generic notification remote hold 00 OK INVITE: encapsulated ANM containing the additional connected number NFO B: encapsulated FAC contains generic notification call transfer active, call transfer							
ICDN marrameter		number derived from the additional connected of user C (SIP-I 2)							
ISDN parameter values	DISCONNECT: ECT	FAC: ECT invoke request component DISCONNECT: ECT invoke return result component							
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3	
	SETUP	←			INVITE(
	ALERTING	→		→	180 Rin	ging(ACM)			
	CONN	→				INVITE(ANM)			
				+	ACK				
	HOLD	→				CPG hold)			
				←	200 OK	INVITE			
				→	ACK				
	SETUP	→					→	INVITE(IAM)	
	ALERTING	+					+	180 Ringing(ACM)	
	CONN	+					+	200 OK INVITE(ANM)	
							→	ACK	
	FAC(ECT invoke)	→							
	DISCONNECT(rr)	←				AC ect active)			
	RELEASE	→		←	200 OK	INFO			
	RELEASE COMPL	+						INFO (FAC ect active)	
							←	200 OK INFO	
				1		BYE(REL)	→	BYE(REL)	
					20			200 OK BYE(RLC	

TP711004	SIP reference							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	ISDN-(ISUP)-SIP/SS/ECT									
SIP selection											
criteria											
ISUP selection											
criteria											
Test purpose	To verify that the IUT received from the rer	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 in the call transfer number in either the FAC or CPG when the call transfer is activated.									
SIP parameter	INVITE B SDP sendo	only,	encapsul	ate	d CPG o	eneric notificatio	n re	emote hold			
values	200 OK INVITE: enca	apsul	ated ANN	Иο	ontainin	g the connected r	num	ber			
	INFO B: encapsulate	d FA	C contair	าร g	eneric r	otification call tra	nsf	er active, call transfer			
	number derived from				user C	(SIP-I 2)					
ISDN parameter	FAC: ECT invoke red										
values	DISCONNECT: ECT invoke return result component										
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3			
	SETUP	←			INVITE						
	ALERTING	→				nging(ACM)					
	CONN	→				(INVITE(ANM)					
				←	ACK						
	HOLD	→				(CPG hold)					
						INVITE					
				→	ACK						
	SETUP	→					→	INVITE(IAM)			
	ALERTING	+						180 Ringing(ACM)			
	CONN	+						200 OK INVITE(ANM)			
	001111	† <u> </u>						ACK			
	FAC(ECT invoke)	→					Ť				
	DISCONNECT(rr)	+		→	INFO (FAC ect active)					
	RELEASE	→			200 OK						
	RELEASE COMPL	+					→	INFO (FAC ect active)			
		T -						200 OK INFO			
			l	·	I		1	1 2			
						BYE(REL)	→	BYE(REL)			
					20			200 OK BYE(RLC			

TP711005	Q.1912.5 [1]						1], c	reference: lause 7.5.2.1.1.2.1/		
TSS reference	Q.732.7 [i.5] ISDN-(ISUP)-SIP/SS/ECT									
SIP selection	10211 (1001) 011 /00	,, <u>LO</u> .								
criteria										
ISUP selection criteria										
Test purpose SIP parameter	Loop prevention pr To verify that the local prevention procedure with call transfer ref To verify that the local transfer if a LOP with loop exists", and the INFO: encapsulated	al exc e by s feren al exc n loop call id	change contending LC ce for both change contending coprevention	ntr DP n c ntr ior tcl	rolling the with localls. rolling the indicar	e ECT can succe op prevention in e ECT can succe tor set to "respondence used by the	essf ndic essf nse'	ully initiate the loop ator set to "request" and ully perform a call ' is received and "no		
values							pon	se indicator: "no loop		
ISDN parameter values										
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3		
	SETUP	+		(INVITE	(IAM)				
	ALERTING	→				iging(ACM)				
	CONN	→				(INVITE(ANM)				
			•	←	ACK	` ` ` ` ` ` ` ` ` ` ` ` ` ` ` ` ` ` ` `				
	HOLD	→				(CPG hold)				
						INVITE				
			•	<u>→</u>	ACK					
	SETUP	→					→	INVITE(IAM)		
	ALERTING	+						180 Ringing(ACM)		
	CONN	+						200 OK INVITE(ANM)		
								ACK		
							T -	7.0.1		
				}	INFO(L	OP request)				
					200 OK					
						-	→	INFO(LOP request)		
							+	200 OK INFO		
			•	(INFO(L	OP response)				
					200 OK					
			+				4	INFO(LOP response)		
								200 OK INFO		
	FAC(ECT invoke)	→								
	DISCONNECT(rr)	+		>	INFO (F	AC ect active)				
	RELEASE	→	•	(200 OK	INFO				
	RELEASE COMPL	+						INFO (FAC ect active)		
							+	200 OK INFO		
						BYE(REL)	→	BYE(REL)		
					20			200 OK BYE(RLC		

TP711006	SIP referenc	C 3261 [Q.1912.5 [1]	, cla	reference: ause 7.5.2.1.1.2.2 a)/ 32.7 [i.5]			
TSS reference	ISDN-(ISUP)-SIP/SS	/ECT	•					
SIP selection criteria								
ISUP selection criteria								
Test purpose		al exc	hange co generic r	ontr oti	olling th	e ECT can succe set to "call trans	essf fer,	ully initiate a call transfer active" or "call transfer,
SIP parameter	INFO B: encapsulate							
values	INFO C: encapsulate	d FA	C contair	าร รู	jeneric r	notification call tra	nsf	er active
ISDN parameter	FAC: ECT invoke red							
values	DISCONNECT: ECT	invol		res		oonent		
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3
	SETUP	+			INVITE			
	ALERTING	→				nging(ACM)		
	CONN	→				(INVITE(ANM)		
				←	ACK			
	HOLD	→				(CPG hold)		
						INVITE		
				<u>→</u>	ACK			
	SETUP	→						INVITE(IAM)
	ALERTING	+						180 Ringing(ACM)
	CONN	+						200 OK INVITE(ANM)
							→	ACK
	FAC(ECT invoke)	→						
	DISCONNECT(rr)	+				FAC ect active)		
	RELEASE	→		←	200 OK	(INFO		
	RELEASE COMPL	+						INFO (FAC ect active)
							←	200 OK INFO
						BYE(REL)	_	BYE(REL)
					20			200 OK BYE(RLC
						ON DIE(RLC	_	ZUU ON BTE(NLC

TP711007	SIP reference	C 3261	[4]	Q.1912.5 [1]	, cla	reference: ause 7.5.2.1.1.2.2 a)/ 32.7 [i.5]					
TSS reference	ISDN-(ISUP)-SIP/SS/	ISDN-(ISUP)-SIP/SS/ECT									
SIP selection											
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 parameter set to "call transfer".										
SIP parameter	INFO C: encapsulated										
values	INFO B: encapsulated										
	INFO B encapsulated				eneric n	otification call tra	nsfe	er active			
ISDN parameter	FAC: ECT invoke req				_						
values	DISCONNECT: ECT	invo		res		ponent		1			
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3			
	SETUP	+			INVITE						
	ALERTING	→				nging(ACM)					
	CONN	→				(INVITE(ANM)					
				+	ACK						
	HOLD	→				(CPG hold)					
				+	200 OK	(INVITE					
				→	ACK						
	SETUP	→						INVITE(IAM)			
	ALERTING	+					+	180 Ringing(ACM)			
	FAC(ECT invoke)	→									
	DISCONNECT(rr)	←		→	INFO (I	FAC ect alert)					
	RELEASE	→		←	200 Ok	(INFO					
	RELEASE COMPL	←					→	INFO (CPG ect active)			
								200 OK INFO			
		1					+	200 OK INVITE(ANM)			
		1						ACK			
		T		→	INFO (FAC ect active)	Ė				
		T			200 Ok						
		1	1			חאר/חרו	_	DVE(DEL)			
		1			00	BYE(REL)		BYE(REL)			
		1				JU ON DIE(KLC	_	200 OK BYE(RLC			

TP711008	SIP reference	e: RF0	C 3261	[4]				reference:		
								ause 7.5.2.1.1.2.2 b)/ 32.7 [i.5]		
TSS reference	ISDN-(ISUP)-SIP/SS	ISDN-(ISUP)-SIP/SS/ECT								
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	call is alerting			•				is invoked while one		
	To verify that, in case	e the E	CT is in	ıvol	ked while	e one call is alert	ing,	as soon as the local		
	exchange (controlling	g the E	CI) red	eiv	es the A	INIMI, It can succe	SSTI	ully send to the other		
	remote user the FAC notification set to "o					set to call transi	er	and the generic		
SIP parameter						otification call tra	nefe	or active		
values	I I O D GIICapsulate	B encapsulated FAC contains generic notification call transfer active						A GOUVE		
ISDN parameter										
values										
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3		
	SETUP	+		←	INVITE	(IAM)				
	ALERTING	→				nging(ACM)				
	CONN	→				(INVITE(ANM)				
		← ACK								
	HOLD	→				(CPG hold)				
						INVITE				
				1	ACK					
	SETUP	→						INVITE(IAM)		
	ALERTING	←					←	180 Ringing(ACM)		
	FAC(ECT invoke)	→								
	DISCONNECT(rr)	+				FAC ect alert)				
	RELEASE	→		←	200 OK	(INFO				
	RELEASE COMPL	+						INFO (CPG ect active)		
		\perp					+	200 OK INFO		
		\perp								
		\perp						200 OK INVITE(ANM)		
							→	ACK		
		\perp				FAC ect active)	ļ			
		+		+	200 OK	(INFO	ļ			
								T		
		\perp						BYE(REL)		
					20	00 OK BYE(RLC	←	200 OK BYE(RLC		

TP711009	SIP reference	SIP reference: RFC 3261 [4] IS Q.1912.5 [1]						
TSS reference	ISDN-(ISUP)-SIP/SS	S/ECT	-					
SIP selection criteria								
ISUP selection criteria								
Test purpose	parameter when th To verify that, in cas other remote user up with the information	e ECT e the con re receiv	F is invo ECT is incept of yed in the	ked nvol the e ge	l while of ked while ANM co eneric n	one call is alerti e one call is aleri onveys the call tr umber paramete	ng ting, ans er if t	the FAC sent to the fer number parameter both the connected per are received in the
SIP parameter values	200 OK INVITE: enc connected number INFO B: encapsulate transfer number deri	ed FA	C contai	ns g	generic r	notification call tra	ansf	er and the additional er active and call
ISDN parameter values								
Comments	ISDN 2		SUT	1	SIP-I 1		1	SIP-I 3
Comments	SETUP	+	301	_	INVITE		+	011 -1 0
	ALERTING	→				nging(ACM)	+	
	CONN	→				(INVITE(ANM)	+-	
	CONIN			_	ACK	(AINVI)	+	
				_	ACK		+-	
	HOLD	_		_	INIX/ITE	(ODO 11-1)		
	HOLD	→				(CPG hold)	-	
						(INVITE	-	
				7	ACK			
							1	
	SETUP	→						INVITE(IAM)
	ALERTING	+					+	180 Ringing(ACM)
	FAC(ECT invoke)	→		_				
	DISCONNECT(rr)	+				FAC ect alert)		
	RELEASE	→		+	200 Ok	(INFO		
	RELEASE COMPL	←					_	INFO (CPG ect active)
							←	200 OK INFO
							←	200 OK INVITE(ANM)
							→	ACK
				→	INFO (FAC ect active)		
					200 Ok			
			•		•			•
						BYE(REL)	→	BYE(REL)
					20			200 OK BYE(RLC

TP711010		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	S/ECT	•					
SIP selection criteria								
ISUP selection criteria								
Test purpose	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 ANM conveys the call transfer number parameter with the information received in the connected number parameter if only the connected number is received in the ANM.							
SIP parameter values	connected number INFO B: encapsulate	200 OK INVITE: encapsulated ANM contains the connected number and the additional connected number INFO B: encapsulated FAC contains generic notification call transfer active and call transfer number derived from the connected number						
ISDN parameter values								
Comments	ISDN 2		SUT		SIP-I 1		I	SIP-I 3
Johnnerits	SETUP	+		_	INVITE(ΙΔΜ		011 -1 0
	ALERTING	→				ging(ACM)		
	CONN	→				INVITE(ANM)		
	CONN				ACK	IIIVIIE(AINIVI)		
	HOLD	→				CPG hold)		
					200 OK ACK	INVITE		
	SETUP	→					-	INVITE(IAM)
	ALERTING	-						180 Ringing(ACM)
	FAC(ECT invoke)	→						
	DISCONNECT(rr)	+				AC ect alert)		
	RELEASE	→		←	200 OK	INFO		
	RELEASE COMPL	+						INFO (CPG ect active)
							+	200 OK INFO
								200 OK INVITE(ANM)
				_	15.150 /-	10 11 1	 →	ACK
						AC ect active)	1	
				<u>←</u>	200 OK	INFO	-	
						DVE/DEL\	_	DVE/DEL)
					200	BYE(REL)		200 OK BYE(RLC
					200	ON DIE(KLC	7	ZUU UN DIE(KLU

TP711011	SIP reference: RFC 3261 [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	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	←		_	INVITE			
	ALERTING	→				nging(ACM)		
	CONN	→		→	200 OK	(INVITE(ANM)		
				+	ACK			
	HOLD	→		→	INVITE	(CPG hold)		
						INVITE		
				→	ACK			
	SETUP	→					→	INVITE(IAM)
	ALERTING	+						180 Ringing(ACM)
	CONN	←						200 OK INVITE(ANM)
							→	ACK
	FAC(ECT invoke)	→						
	DISCONNECT(rr)	+				FAC ect active)		
	RELEASE	→		+	200 OK	INFO		
	RELEASE COMPL	+						INFO (FAC ect active)
							←	200 OK INFO
				l		BYE(REL)	→	BYE(REL)
				1	20			200 OK BYE(RLC

TP711012	SIP reference			[4]		Q.1912.5 [1]	, cl	reference: auses 7.3; 7.5.2.4.1/ 32.7 [i.5]
TSS reference	ISDN-(ISUP)-SIP/SS	S/ECT						
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	To verify that the IU number if it is the ne to "national (signification).	Call transfer number - removal of own country code To 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. The nature of address indicator shall be set to "national (significant) number".						
SIP parameter	INVITE SIP-I 1: enca						er N	oA "international
values	number" with the ne							
						ontains connected	d nu	ımber NoA "international
	number" with the ne	twork	s own co	unt	ry code.		l	danka difaran arawa ata d
	number NoA "nation			ont	ains the	call transfer num	ber	derived from connected
	INFO SIP-I 3: encap			ont	aine the	call transfer num	hor	derived from calling
	party number NoA "r				anis inc	can transfer fluin	IDCI	derived from calling
ISDN parameter	party namber 140/1	lation	arriarribe	<i></i>				
values								
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3
	SETUP	+		+	INVITE			
	ALERTING	→		→	180 Rii	nging(ACM)		
	CONN	→				(INVITE(ANM)		
				+	ACK	, ,		
	HOLD	→		→	INVITE	(CPG hold)		
				←	200 Ok	INVITE		
				→	ACK			
	SETUP	→					→	INVITE(IAM)
	ALERTING	+					+	180 Ringing(ACM)
	CONN	+					+	200 OK INVITE(ANM)
							→	ACK
	FAC(ECT invoke)	→						
	DISCONNECT(rr)	+		→	INFO (FAC ect active)		
	RELEASE	→		←	200 Ok	(INFO		
	RELEASE COMPL	←						INFO (FAC ect active)
							←	200 OK INFO
						<u>-</u>		
								BYE(REL)
					20	00 OK BYE(RLC	←	200 OK BYE(RLC

A.1.1.2.12 Call Diversion (CFB, CFNR, CFU, CD)

TP712001	SIP reference: RFC 3261 [4]	Q.19	ISUP reference: 912.5 [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 ACI		
	To verify that a call can be successfully est	ablished, if div	ersion occurs. The encapsulated
	ACM contains the generic notification inc		call is diverting", the call
	diversion information and the redirection		
	Applicable redirection reason in the call div		nation:
	"busy" CFE	3(n); CFB(u,l)	
	"deflection immediate response" CD(
SIP parameter	183 Session Progress encapsulated ACM	· / /	ation indicator "call is diverting"
values	100 0033ion 1 10gress eneapsulated 7 tolvi (goriono notino	ation indicator can is diverting
ISDN parameter values	NOTIFY: Notification indicator "call is divert	ing"	
Comments	ISDN	SUT	SIP-I
	SETUP →	→	INVITE(IAM)
	NOTIFY ←	←	183 Session Progress(ACM)
	ALERTING	+	180 Ringing(CPG)
	CONNECT ←	←	200 OK INVITE(ANM)
		→	ACK
	DISCONNECT →	→	BYE(REL)
	RELEASE ←	+	200 OK BYE(RLC)
	RELEASE COMPLETE →		

TP712002	SIP reference: RFC 32	261 [4]	G	ISUP reference: 0.1912.5 [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	"Call diversion may occur" received in ACM To 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. Applicable redirection reason in the call diversion information: "busy" CFB(u,e) "no reply" CFNR "deflection during alerting" CD(a) "deflection immediate response" CD(i,e) 180 Ringing: encapsulated ACM optional backward call indicator "call diversion may occur" 183 Session Progress: encapsulated CPG contains generic notification "call is diverting",				
ISDN parameter values	call diversion information, redir	ection num	ber		
Comments	ISDN		SUT	SIP-I	
	SETUP	→			
	ALERTING	+	•		
			•		
	CONNECT	+	•	• ` '	
			-3		
	DISCONNECT	→	_3	DVF(DFL)	
	RELEASE	+		\ /	
	RELEASE COMPLETE			ZUU UN DIE(NLU)	
	RELEASE COMPLETE	7			

TP712003	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1]		
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Redirection number - presentation allowed - according to the notification subscription option To verify that the originating exchange makes the redirection number available to the calling access signalling system, if the notification subscription option of the call diversion information is coded "010 presentation allowed with redirection number". The redirection number restriction parameter is set to "00 presentation allowed".					
SIP parameter values	183 Session Progress encapsulated ACM generic notification indicator "call is diverting" 200 OK INVITE: encapsulated ANM redirection number restriction "presentation allowed"					
ISDN parameter values	CONNECT: redirection numbe	r				
Comments	ISDN		SUT		SIP-I	
	SETUP	→		→	INVITE(IAM)	
	NOTIFY	+		+	183 Session Progress(ACM)	
	ALERTING	+		+	180 Ringing(CPG)	
	CONNECT	+		+	200 OK INVITE(ANM)	
				→	ACK	
	,					
	DISCONNECT	→		→	BYE(REL)	
	RELEASE	+		+	200 OK BYE(RLC)	
	RELEASE COMPLETE	→				

TP712004	SIP reference: RFC 3261	[4]	Q.1912	ISUP reference: 2.5 [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 encapsula	183 Session Progress encapsulated ACM call diversion information notification						
values	subscription option "presentation	allowed w	ithout redirec	tion number"				
ISDN parameter values	NOTIFY: notification indicator "ca ALERTING: no redirection number CONNECT: no redirection number	er	ed"					
Comments	ISDN		SUT	SIP-I				
	SETUP	→	→	INVITE(IAM)				
	NOTIFY	+	+	183 Session Progress(ACM)				
	ALERTING	+	+	180 Ringing(CPG)				
	CONNECT	+	+	200 OK INVITE(ANM)				
			→	ACK				
	DISCONNECT	→	→	BYE(REL)				
	RELEASE	+	(200 OK BYE(RLC)				
	RELEASE COMPLETE	→						

TP712005	SIP reference: RFC 3	261 [4]		ISUP reference:
			Q.1912	2.5 [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 - prese	ntation restric	ted - accord	ding to redirection number
	restriction parameter			_
	To verify that the originating e	xchange does	not make th	e redirection number available to
	the calling access signalling s	ystem, if the re	edirection nu	mber restriction parameter
	indicates "01 Presentation res			
	The notification subscription of			nformation is coded "010
	Presentation allowed with red			
SIP parameter	183 Session Progress encaps			
values	subscription option "presentat			
			ion number	restriction "presentation restricted"
ISDN parameter	CONNECT: no redirection nur	mber		
values				
Comments	ISDN	S	SUT	SIP-I
	SETUP	→	→	INVITE(IAM)
	NOTIFY	+	+	183 Session Progress(ACM)
	ALERTING	←	←	180 Ringing(CPG)
	CONNECT	+	+	200 OK INVITE(ANM)
			→	ACK
			•	
	DISCONNECT	→	→	BYE(REL)
	RELEASE	+	+	200 OK BYE(RLC)
	RELEASE COMPLETE	→		

TP712006	SIP reference: RFC 326	1 [4]	Q.1912	ISUP reference: 2.5 [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 - presenta	tion res	ricted - no red	irection number restriction
	parameter received			
				e redirection number available to
	the calling access signalling syst	em, if no	redirection nun	nber restriction parameter is
	received.			-f
	The notification subscription opti			nformation is coded "010
CID noromotor				aformation polification
SIP parameter values	183 Session Progress encapsula subscription option "presentation"			
values	200 OK INVITE: encapsulated A			
ISDN parameter	CONNECT: redirection number	I VIVI WILLIC	at redirection in	diffice restriction parameter
values	CONTROL Teallection number			
Comments	ISDN		SUT	SIP-I
	SETUP	→	→	INVITE(IAM)
	NOTIFY	+	+	183 Session Progress(ACM)
	ALERTING	+	+	180 Ringing(CPG)
	CONNECT	+	+	200 OK INVITE(ANM)
			→	ACK
			<u> </u>	
	DISCONNECT	→	→	BYE(REL)
	RELEASE	+	+	200 OK BYE(RLC)
	RELEASE COMPLETE	→		

TP712007	SIP reference: RFC	3261 [4]	Q	ISUP reference: .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 the calling access signallin (see note).	exchange does n ng system, if the l	ot make ar ast divertin	ny redirection number available g exchange does not send one		
SIP parameter values	183 Session Progress encapsulated ACM call diversion information notification 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 nu CONNECT: no redirection nu					
Comments	ISDN	SU	Т	SIP-I		
	SETUP	→				
	0L 101	7	→	INVITE(IAM)		
	NOTIFY	+		INVITE(IAM) 183 Session Progress(ACM)		
				` '		
			+	183 Session Progress(ACM) 183 Session Progress(CPG) 180 Ringing(CPG)		
	NOTIFY	+	+	183 Session Progress(ACM) 183 Session Progress(CPG)		
	NOTIFY ALERTING	-	+ + +	183 Session Progress(ACM) 183 Session Progress(CPG) 180 Ringing(CPG)		
	NOTIFY ALERTING	-	+ + +	183 Session Progress(ACM) 183 Session Progress(CPG) 180 Ringing(CPG) 200 OK INVITE(ANM)		
	NOTIFY ALERTING CONNECT	÷ + + + + + + + + + + + + + + + + + + +	+ + + +	183 Session Progress(ACM) 183 Session Progress(CPG) 180 Ringing(CPG) 200 OK INVITE(ANM) ACK		

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 - restrictive notification subscription option To verify that the originating exchange handle number according to the contents of the most the call diversion information, if the forward ("presentation allowed" in the redirection number of the call diversion of the call diversion in the redirection number of the call diversions of the call diversion in the redirection number of the call diversions of the call diversion in t	es the prese t restrictive ded-to user	entation of the redirection notification subscription option of allows presentation of the number			
SIP parameter	183 Session Progress encapsulated ACM ca					
values ISDN parameter	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"					
values	ALERTING: no redirection number CONNECT: no redirection number					
Comments	ISDN St	IT	SIP-I			
Commonto	SETUP →	<u>→</u>	INVITE(IAM)			
	NOTIFY		183 Session Progress(ACM)			
	1101111	+	183 Session Progress(CPG)			
	ALERTING	+	180 Ringing(CPG)			
	CONNECT	+	200 OK INVITE(ANM)			
		→	ACK			
	DISCONNECT →	→	BYE(REL)			
	RELEASE ←	+	200 OK BYE(RLC)			
	RELEASE COMPLETE →					
forwardin	messages each containing the call diversion in logs have occurred (from option B - immediate re ation takes place).					

TP712009	SIP r	eference: RF	C 32	261 [4]		1], (reference: clause 2.5.2.5.1.1/ /32 [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		sfully establish a		
SIP parameter	183 Session	Progress: end	caps	ulated ACM ger	neric notification	"cal	I is diverted", redirection
values	information, i	redirection nu	mbe	r			
ISDN parameter values							
Comments	ISDN 2	SUT		SIP-I 1			SIP-I 3
			←	INVITE(IAM)			
		CDIV					
			→	183 Session P	rogress(ACM)		
						→	INVITE(IAM)
						4	180 Ringing(ACM)
			→	180 Ringing(A	CM)		
					· · · · · · · · · · · · · · · · · · ·	←	200 OK INVITE(ANM)
			_	200 OK INVITI	E(ANM)	→	ACK
			←	ACK			
			+	BYE(REL)	·	→	BYE(REL)
			→	200 OK BYE(F	RLC	+	200 OK BYE(RLC

TP712010	SIP reference: RFC 326	l [4 <u>]</u>		Q.19	ISUP reference: 12.5 [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 redire	ction nun	nber re	estriction indicator in the ACM,
SIP parameter values	200 OK INVITE: encapsulated Al	VM redi	rection nu	mber r	estriction "presentation allowed"
ISDN parameter values					
Comments	ISDN		SUT		SIP-I
	SETUP	+		+	INVITE(IAM)
	ALERTING	→		→	180 Ringing(ACM)
	CONNECT	↑		→	200 OK INVITE(ANM)
				+	ACK
	DISCONNECT	+		+	BYE(REL)
	RELEASE	→		→	200 OK BYE(RLC)
	RELEASE COMPLETE	+			

TP712011	SIP reference: RFC 32	261 [4]	Q.19	ISUP reference: 012.5 [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 the redirection numb (pres. restricted) To verify that the IUT includes CPG, ANM or CON set to "pre	the redirectio	n number r	estriction indicator in the ACM,
SIP parameter values	200 OK INVITE: encapsulated	ANM redirection	on number i	restriction "presentation restricted"
ISDN parameter values				
Comments	ISDN	SI	JT	SIP-I
	SETUP	+	+	INVITE(IAM)
	ALERTING	→	→	180 Ringing(ACM)
	CONNECT	→	→	200 OK INVITE(ANM)
			+	ACK
	DISCONNECT	+	+	BYE(REL)
	RELEASE	→	→	200 OK BYE(RLC)
	RELEASE COMPLETE	+		

TP712012	SIP	referenc	e: RFC 32	261 [4]	Q.1912.5 [1], cl	reference: lause 2.5.2.5.1.2 b) 2)/ .732 [i.4]
TSS reference	ISDN-(ISUF)-SIP/SS	/CDIV			
SIP selection						
criteria						
ISUP selection criteria						
Test purpose	Verify that the number accordion.	he IUT se cording to	ts the add the "serv	dress presentati red user release	es his/her number to	tor of the original called o the diverted-to user"
SIP parameter	INVITE SIP	-l 3:enca	sulated l	AM original call	ed number presenta	ation allowed
values				_		
ISDN parameter						
values						
Comments	ISDN 2	S	UT	SIP-I 1		SIP-I 3
			←	INVITE(IAM)		
		C	ΝV			
			→	183 Session F	Progress(ACM)	
					-3	INVITE(IAM)
					€	- 180 Ringing(ACM)
			→	180 Ringing(A	.CM)	
				0 0	,	
					•	200 OK INVITE(ANM)
			→	200 OK INVIT		ACK
				ACK	,	
		1	I	l .		L
			+	BYE(REL)	-	BYE(REL)
			→	200 OK BYE(F	RLC €	200 OK BYE(RLC

TP712013	SIP	refer	ence: RF	C 32	261 [4]	IS		reference: 912.5 [1]
TSS reference	ISDN-(ISUF	ISDN-(ISUP)-SIP/SS/CDIV						
SIP selection criteria								
ISUP selection criteria								
Test purpose	Verify that the number accordion.	Redirecting number generated by the diverting exchange Verify that the IUT sets the address presentation restricted indicator of the redirecting number according to the "served user releases his/her number to the diverted-to user" option. The redirecting indicator in the redirection information shall be set to "011 Call diverted".						
SIP parameter values	INVITE SIP	INVITE SIP-I 3: redirecting number, redirection information						
ISDN parameter values								
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3
				+	INVITE(IAM)			
			CDIV					
				→	183 Session P	rogress(ACM)		
							→	INVITE(IAM)
							+	180 Ringing(ACM)
				→	180 Ringing(A	CM)		
							←	200 OK INVITE(ANM)
					200 OK INVITI	E(ANM)	→	ACK
				←	ACK			
				+	BYE(REL)		→	BYE(REL)
				→	200 OK BYE(F	RLC	+	200 OK BYE(RLC

TP712014	SIP	reference: RF	C 32		912.5 [1], c	P reference: ·lause 2.5.2.5.1.2 b) 5)/ 0.732 [i.4]		
TSS reference	ISDN-(ISUP)-SIP/SS/CDI\	/					
SIP selection criteria								
ISUP selection criteria								
Test purpose	To verify the indicator red - not requi-	ISDN user part preference indicator in the diverting exchange To verify that the IUT can successfully divert a call and that ISDN user part preference indicator received in the forward call indicators with the value "ISDN user part: not required all the way" shall be changed to "ISDN user part preferred all the way"; preferred all the way" shall be left unchanged; required all the way" shall be left unchanged.						
SIP parameter		INVITE SIP-I 3 : encapsulated IAM forward call indicator ISDN user part required all the						
values	way	•						
ISDN parameter values								
Comments	ISDN 2	SUT		SIP-I 1		SIP-I 3		
			+	INVITE(IAM)				
		CDIV						
			→	183 Session Progres	s(ACM)			
						➤ INVITE(IAM)		
					•	€ 180 Ringing(ACM)		
			→	180 Ringing(ACM)				
						200 OK INVITE(ANM)		
			→	200 OK INVITE(ANM	l) -	→ ACK		
			+	ACK				
				BYE(REL)		→ BYE(REL)		
			→	200 OK BYE(RLC	•	€ 200 OK BYE(RLC		

TP712015	SIP reference: RFC 326	l [4]	ISUP reference: Q.1912.5 [1], clause 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	To verify that the IUT includes an	Call diversion may occur in the diverting exchange To verify that the IUT includes an optional backward call indicator with the indication "call diversion may occur" in the ACM in case of CFNR, CD(a), CFB(u,e) and CD(i,e)					
SIP parameter	180 Ringing: encapsulated ACM	called party	status indic	ator "subscriber free" optional			
values	backward call indicator "call diver	sion may o	ccur"	·			
ISDN parameter values	ALERTING: no mapping of option	nal backwar	d call indicat	tor value			
Comments	ISDN	S	UT	SIP-I			
	SETUP	→	→	INVITE(IAM)			
	ALERTING	+	+	180 Ringing(ACM)			
	CONNECT	+	(200 OK INVITE(ANM)			
			→	ACK			
		•	•				
	DISCONNECT	→	→	BYE(REL)			
	RELEASE	+	(200 OK BYE(RLC)			
	RELEASE COMPLETE	+					

A.1.1.2.13 Call HOLD (HOLD)

TP713001	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/	/HOLD					
SIP selection criteria							
ISUP selection criteria							
Test purpose	Call hold after answ To verify that a call ca that notifications are "progress".	an be placed on ho	ld and can	be retriev	ed again by the local user and ent indicator set to		
SIP parameter values							
ISDN parameter values							
Comments	ISDN		SUT		SIP		
	SETUP	→		→	INVITE(IAM)		
	ALERTING	+		+	180 Ringing(ACM)		
	CONN	+		+	200 OK INVITE(ANM)		
				→	ACK		
		Co	mmunication	n			
	HOLD	→		→	INFO(CPG hold)		
				+	200 OK INFO		
	RETRIVE	→		→	INFO(CPG retrieve)		
				+	200 OK INFO		
		Co	mmunicatio	n			
	DISC	→		→	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/	HOLD				
SIP selection criteria						
ISUP selection criteria						
Test purpose	Call hold after answer To verify that a call call and that notifications	n be placed on h	old and car	n be retriev	red again by the remote user	
SIP parameter						
values						
ISDN parameter						
values						
Comments	ISDN		SUT		SIP	
	SETUP	→		→	INVITE(IAM)	
	ALERTING	+		+	180 Ringing(ACM)	
	CONN	+		+	200 OK INVITE(ANM)	
				→	ACK	
		C	ommunicat	ion		
	HOLD	+		+	INFO(CPG hold)	
				→	200 OK INFO	
	RETRIVE	+		+	INFO(CPG retrieve)	
				→	200 OK INFO	
		С	ommunicat	ion		
	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/	HOLD					
SIP selection criteria							
ISUP selection criteria	PICS 8/1						
Test purpose		oing call can be	placed o	n HOLD af		alerting has commenced and tions are sent with CPG	
SIP parameter values							
ISDN parameter values							
Comments	ISDN		SU	T		SIP	
	SETUP	→			→	INVITE(IAM)	
	ALERTING	+		,	(180 Ringing(ACM)	
	HOLD	→			→	INFO(CPG hold)	
					(200 OK INFO	
	RETRIVE	→		-	→	INFO(CPG retrieve)	
					(200 OK INFO	
	CONN	+			(200 OK INVITE(ANM)	
				-	→	ACK	
			Commun	ication			
	DISC	→			→	BYE(REL)	
	REL	+		,	(200 OK BYE	
	REL_COM	→					

TP713004	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses 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, requested by the remote user To verify that an incoming call can be placed on hold and can 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)		
				→	200 OK INFO		
	RETRIVE	+		+	INFO(CPG retrieve)		
				→	200 OK INFO		
	CONN	+		+	200 OK INVITE(ANM)		
				→	ACK		
		C	commun	ication			
	DISC	→		→	BYE(REL)		
	REL	+		+	200 OK BYE		
	REL_COM	→	•				

TP713005	SIP reference	: RFC 3261 [4]		_	SUP reference: 12.5 [1], clause 2.3/ Q.764 [i.12]
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 in hold service.		-		rved user user who activated the Call
SIP parameter values					
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP	→		→	INVITE(IAM)
	ALERTING	+		+	180 Ringing(ACM)
	CONN	+		+	200 OK INVITE(ANM)
				→	ACK
		(Communicati	on	
	HOLD	→		→	INFO(CPG hold)
				+	200 OK INFO
	DISC	→		→	BYE(REL)
	REL	+		+	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/F	HOLD	•						
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 the held state can be released by the user who did not activate the Call hold service							
SIP parameter values									
ISDN parameter values									
Comments	ISDN		SUT		SIP				
	SETUP	→		→	INVITE(IAM)				
	ALERTING	←		+	180 Ringing(ACM)				
	CONN	-		+	200 OK INVITE(ANM)				
				→	ACK				
		C	Communication	cation					
	HOLD	→		→	INFO(CPG hold)				
				+	200 OK INFO				
	DISC	+		+	BYE(REL)				
	REL	→		+	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/H	OLD							
SIP selection criteria									
ISUP selection criteria									
Test purpose		Call hold after alerting, release of the call by the local served user To verify that a held call can be released by the user who activated the Call hold service without retrieving the call.							
SIP parameter values									
ISDN parameter values									
Comments	ISDN		SUT		SIP				
	SETUP	+		+	INVITE(IAM)				
	ALERTING	→		→	180 Ringing(ACM)				
	HOLD	→		→	INFO(CPG hold)				
				+	200 OK INFO				
	DISC	→		→	CANCEL/BYE				
	RELEASE	+		+	200 OK CANCEL/BYE				
	REL_COMP	→		+	487 Request Terminated				
				→	ACK				

TP713008	SIP reference:	RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause 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, release of the call by the non-served user To verify that a call in the held state can be released by the user who did not activate the Call hold service.						
SIP parameter values							
ISDN parameter values							
Comments	ISDN		SU	IT		SIP	
	SETUP	+			+	INVITE(IAM)	
	ALERTING	→			→	180 Ringing(ACM)	
	HOLD	+			+	INFO(CPG hold)	
					→	200 OK INFO	
	DISC	+			+	CANCEL	
	RELEASE	→			→	200 OK CANCEL/BYE	
	REL_COMP	+			→	487 Request Terminated	
					+	ACK	

A.1.1.2.14 Call Waiting (CW)

TP714001	SIP reference: RF0	C 3261 [4]	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.		ully establishe	d if the A	CM indicates that it is a waiting						
SIP parameter	180 Ringing: encapsulated	180 Ringing: encapsulated ACM contains the Generic notification parameter value "call is									
values	a waiting call"				·						
ISDN parameter values											
Comments	ISDN		SUT		SIP						
	SETUP	→		→	INVITE(IAM)						
	ALERTING	+		+	180 Ringing(ACM)						
	CONN	+		+	200 OK INVITE(ANM)						
				→	ACK						
			Communicat	ion							
	DISC	→		→	BYE(REL)						
	REL	+		+	200 OK BYE						
	REL_COM	→									

TP714002	SIP reference: RFC	3261 [4]	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 CPG To verify that a call can be successfully established if the CPG indicates that it is a waiting call.										
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	→		→	INVITE(IAM)						
				+	183 Session Progress(ACM)						
	ALERTING	+		+	180 Ringing(CPG)						
	CONN	+		+	200 OK INVITE(ANM)						
				→	ACK						
		C	Communic	ation							
	DISC	→		→	BYE(REL)						
	REL	+		+	200 OK BYE						
	REL_COM	→									

TP714003	SIP reference: F	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/CV	٧						
SIP selection criteria								
ISUP selection criteria								
Test purpose		be successfu ification) and	lly established if he is curren	tly busy, k	er has subscribed to the call out answers the waiting call.			
SIP parameter	180 Ringing: encapsulat	ted ACM cont	ains the Gene	ric notifica	ation parameter value "call is			
values	a waiting call"				•			
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	+		+	INVITE(IAM)			
	ALERTING	→		→	180 Ringing(ACM)			
	CONN	→		→	200 OK INVITE(ANM)			
				+	ACK			
			Communication	n				
	DISC	+		+	BYE(REL)			
	REL				200 OK BYE			
	REL_COM	+	•					

TP714004	SIP reference: RFC 3261 [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: c	ause 2	1						
Comments	ISDN		SUT		SIP				
	SETUP	+		+	INVITE(IAM)				
	ALERTING	→		→	180 Ringing(ACM)				
	CONN	→		→	200 OK INVITE(ANM)				
				+	ACK				
		С	ommunicati	on					
	SETUP	←		+	INVITE(IAM)				
	ALERTING	→		→	180 Ringing(ACM waiting call)				
	RELEASE COMPLETE	→		→	480 Temporarily unavailable(REL#21)				
				+	ACK				
			ommunicati	on					
	DISC	←		+	(/				
	REL				200 OK BYE				
	REL_COM	←							

A.1.1.2.15 Three Party Service (3PTY)

TP715001	SIP refe	erence		ISUP reference: 2.5 [1], clauses 2.4; 2.2.1/ Q.734.2 [i.8]							
TSS reference	ISDN-(ISUP)-S	ISDN-(ISUP)-SIP/SS/3PTY									
SIP selection											
criteria											
ISUP selection											
criteria											
Test purpose	To verify that the successfully join remote parties of the IUT should conference estimates.	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	→		→	INVITE(IAM)						
	ALERTING	+		←	180 Ringing(ACM)						
	CONN	+		←	200 OK INVITE(ANM)						
	HOLD	→		→	INVITE(CPG hold)						
				(200 OK INVITE						
				→	ACK						
	SETUP	→				→	INVITE(IAM)				
	ALERTING	+				+	180 Ringing(ACM)				
	CONN	+				+	200 OK INVITE(ANM)				
	FAC(est3pty)	→		→	INFO(CPG conf est)						
				←	200 OK INFO						
						→	INFO(CPG conf est)				
						+	200 OK INFO				
				3	PTY communication						
	DISC	+		←	BYE(REL)						
	RELEASE	→		→	200 OK BYE						
	REL_COM	+				→	INFO(CPG conf disc)				
						+	200 OK INFO				
	DISC	→				→	BYE(REL)				
	RELEASE	+				+	200 OK BYE				
	REL_COM	→									

TP715002	SIP refe	erence	e: RFC 3261	[4]	ISUP reference: 5 [1], clause 2.5.2.1.1.3 a)/ Q.734.2 [i.8]						
TSS reference	ISDN-(ISUP)-SIP/SS/3PTY										
SIP selection											
criteria											
ISUP selection											
criteria											
Test purpose	To verify that th private commun	Served user creates a private communication with a remote user 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 CPG messages to the two users.									
SIP parameter values											
ISDN parameter											
values											
Comments	ISDN		SUT	SIP-I 1			SIP-I 2				
	SETUP	→	→	INVITE(I	AM)						
	ALERTING	+	←		jing(ACM)						
	CONN	+	+	200 OK I	NVITE(ANM)						
			Communicat	tion							
	HOLD	→	→		CPG hold)						
			+	200 OK I	NVITE						
			→	ACK							
	SETUP	→				→	INVITE(IAM)				
	ALERTING	←				←	180 Ringing(ACM)				
	CONN	←				←	200 OK INVITE(ANM)				
	FAC(est3pty)	→	→		PG conf est)						
			+	200 OK I	NFO						
						→	INFO(CPG conf est)				
						+	200 OK INFO				
			;		munication						
	FAC(end3pty)	→	→		PG conf disc)						
	FAC(ret res)	+	+	200 OK I	NFO						
						→	INFO(CPG conf disc)				
						←	200 OK INFO				
					nmunication IS	<u>DN - :</u>	SIP-I 2				
	DISC	←	+	BYE(RE							
	RELEASE	→	→	200 OK I	BYE						
	REL_COM	←									
	DISC	→				→	BYE(REL)				
	RELEASE	+			·	+	200 OK BYE				
	REL_COM	→									

TP715003	SIP refe	erence: F	ISUP reference: .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										
criteria										
Test purpose	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										
ISDN parameter										
/alues										
Comments	ISDN		SUT	SIP-I 1			SIP-I 2			
	SETUP	→	→	INVITE(IAM)						
	ALERTING	←	←		ing(ACM)					
	CONN	←	←	200 OK INVITE(ANM)						
	HOLD	→	→		CPG hold)					
			←	200 OK I	NVITE					
			→	ACK						
	SETUP	→				→	INVITE(IAM)			
	ALERTING	+				+	180 Ringing(ACM)			
	CONN	(+	200 OK INVITE(ANM)			
	FAC(est3pty)	→	→		G conf est)					
			←	200 OK I	NFO					
						→	INFO(CPG conf est)			
						←	200 OK INFO			
			3	PTY com	munication					
	DISC	→				→	BYE(REL)			
	RELEASE	+				+	200 OK BYE			
	REL_COM	→	→		G conf disc)					
			+	200 OK I						
			→	INFO(CP	G hold)					
			+	200 OK I						
	DISC	+	+	BYE(REI	_)					
	RELEASE	→	→	200 OK E	BYE					
	REL_COM	+		1		1	1			

TP715004	SIP refe	SIP reference: RFC 3261 [4]					ISUP reference: Q.1912.5 [1], clause 2.5.2.1.1.3/ Q.734.2 [i.8]			
TSS reference	ISDN-(ISUP)-SIP/SS/3PTY									
SIP selection										
criteria										
ISUP selection criteria										
Test purpose	To verify that the two remote The IUT should indicator (dependent)	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)	→ →		INVITE(IA	AM)					
	ALERTING	←		<u> </u>	180 Ringi					
	CONN	+	(€	E	200 OK II	NVITE(ANM)				
	HOLD	→		→	INVITE(C					
				<u> </u>	200 OK II	NVITE				
				→	ACK					
	SETUP(CRy)	→					→	INVITE(IAM)		
	ALERTING	+					+	180 Ringing(ACM)		
	CONN	+					←	200 OK INVITE(ANM)		
	FAC(est3pty)	→		→		G conf est)				
			•	<u> </u>	200 OK II	NFO				
							→	INFO(CPG conf est)		
							←	200 OK INFO		
					PTY comn					
	DISC(CRx)	→		→	BYE(REL					
	RELEASE	+	•	<u> </u>	200 OK B	YE				
	REL_COM	→					→	INFO(CPG conf disc)		
							+	200 OK INFO		
	DISC(CRy)	→					→	BYE(REL)		
	RELEASE	+					+	200 OK BYE		
	REL_COM	→		_						

TP715005		eference: RFC 3261 [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	To verify that the after receiving on the order of the second of the sec	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	'			<i>,</i>						
values										
ISDN parameter										
values							1			
Comments	ISDN		SUT	SIP-I 1			SIP-I 2			
	SETUP	→	→							
	ALERTING	+	+		ging(ACM)					
	CONN	←	(200 OK	INVITE(ANM)					
	HOLD	→	→		CPG hold)					
			+		INVITE					
			→	ACK						
	SETUP	→				→	INVITE(IAM)			
	ALERTING	←				←	180 Ringing(ACM)			
	CONN	+				←	200 OK INVITE(ANM)			
	FAC(est3pty)	→	→		PG conf est)					
			+	200 OK	INFO					
						→	INFO(CPG conf est)			
						+	200 OK INFO			
				3 PTY com	munication		•			
	DISC	+				(BYE(REL)			
	RELEASE	→				→	200 OK BYE			
	REL_COM	+	→	INFO(CO	GP (conf disc)					
			+							
			→	INFO(CO	GP (hold)					
			+							
	DISC(CRx)	→	→							
	RELEASE	+	-							
	REL COM	→								
NOTE: The "ren		tion sh	ould be sen	t in a CPG to	o the other rem	ote u	ser, followed by the			

NOTE: The "remote hold" notification should be sent in a **CPG** to the other remote user, followed by the "conference disconnected" notification in a separate **CPG**.

TP715006	SIP reference: R	FC 3261 [4]	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	Т	SIP		
	SETUP	+		+	INVITE(IAM)		
	ALERTING	→		→	180 Ringing(ACM)		
	CONN	→		→	200 OK INVITE(ANM)		
			Commun	ication			
				+	INVITE(CPG hold)		
				→			
				-	ACK		
	NOTIFY(conf est)	+		+			
	,			→			
	3 PTY communication						
	NOTIFY(conf disc)	+		+	INFO(CPG conf disc)		
	,			→			
	NOTIFY(hold)	+		←			
	\ /			→			
	DISC	+		+			
	REL	→		→	\ /		
	REL_COM	+			-		

Annex B (informative): Bibliography

- ITU-T Recommendation Q.761: "Signalling System No. 7 ISDN User Part functional description".
- ITU-T Recommendation Q.762: "Signalling System No. 7 ISDN User Part general functions of messages and signals".
- ITU-T Recommendation Q.763: "Signalling System No. 7 ISDN User Part formats and codes".
- ITU-T Recommendation Q.1902.1: "Bearer Independent Call Control protocol (Capability Set 2): Functional description".
- ITU-T Recommendation Q.1902.2: "Bearer Independent Call Control protocol (Capability Set 2) and Signalling System No.7 ISDN User Part: General functions of messages and parameters".
- ITU-T Recommendation Q.1902.3: "Bearer Independent Call Control protocol (Capability Set 2) and Signalling System No.7 ISDN User Part: Formats and codes".
- IETF RFC 3267: "Real-Time Transport Protocol (RTP) Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs".
- 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						
V3.1.1	June 2010	Publication				