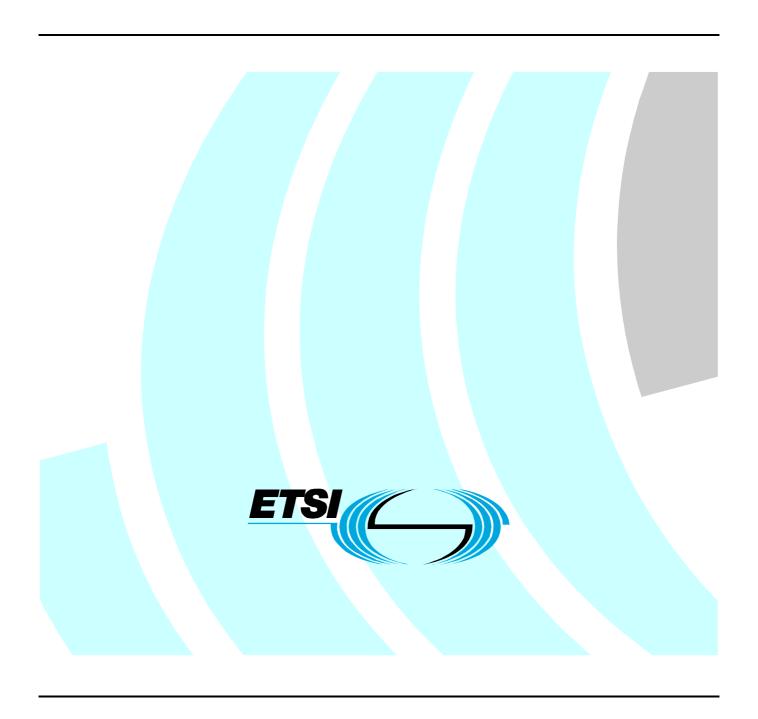
ETSITS 102 709-2 V3.2.1 (2011-03)

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

RTS/INT-00048

Keywords
BICC, ISUP, SIP, testing, TSS&TP

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Foreword

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

The present document is part 2 of a multi-part deliverable covering 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.

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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]".
- [12] Void.

2.2 Informative references

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

[i.1] Void. [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)". ITU-T Recommendation Q.732: "Call diversion services". [i.4] ITU-T Recommendation Q.732.7: "Explicit Call Transfer". [i.5]ITU-T Recommendation Q.733: "Stage 3 description for call completion supplementary services [i.6] using Signalling System No. 7: Terminal portability (TP)". ITU-T Recommendation Q.734: "Stage 3 description for multiparty supplementary services using [i.7] Signalling System No. 7: Conference calling". ITU-T Recommendation Q.734.2: "Three-party service". [i.8] [i.9] Void. ITU-T Recommendation Q.737: "User-to-user signalling (UUS)". [i.10] [i.11] ITU-T Recommendation Q.784: "ISUP basic call test specification". ITU-T Recommendation Q.764: "Signalling System No. 7 - ISDN User Part signalling [i.12] 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". [i.15] IETF RFC 4040: "RTP Payload Format for a 64 kbit/s Transparent Call". [i.16] Void.

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

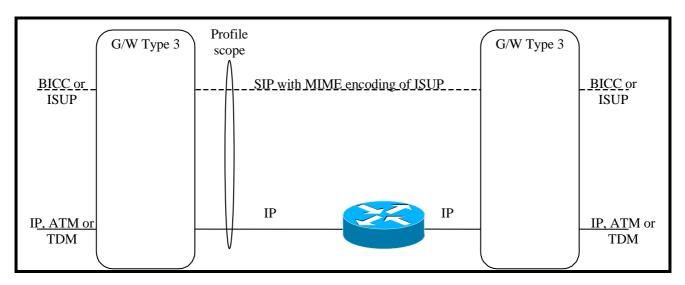


Figure 1: Profile scope for SIP with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateways

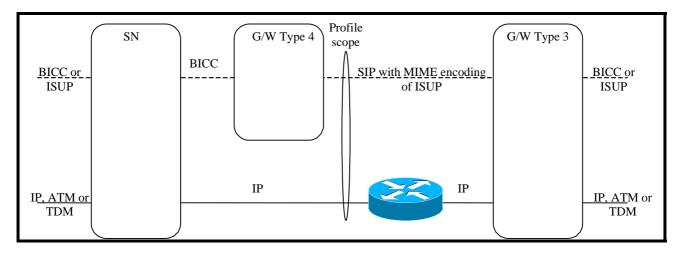


Figure 2: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 3 and 4 gateways

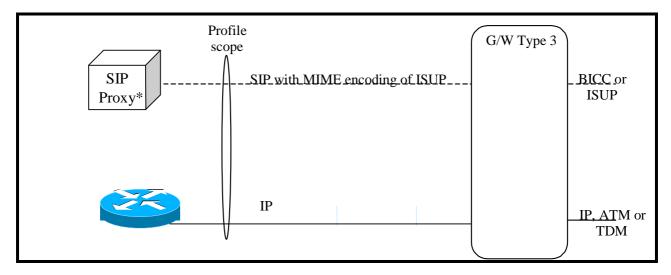


Figure 3: Profile scope for SIP with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateways

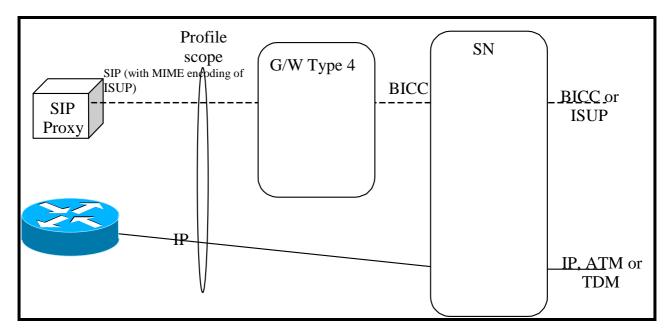


Figure 4: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 4 gateway

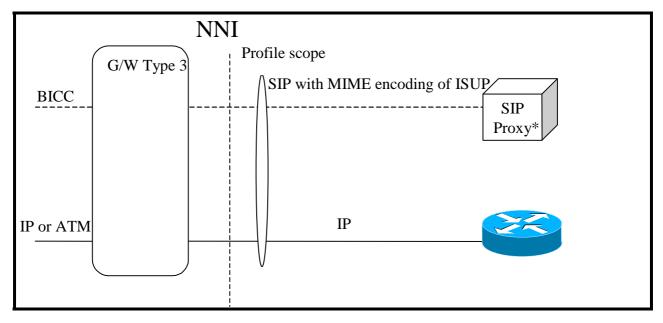


Figure 5: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateway

3.2 Abbreviations

ACM

Three-Party

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

Address Complete Message

ANswer Message **ANM Abstract Service Primitive ASP** ATC Abstract Test Case **ATM** Abstract Test Method Access Transport Parameter **ATP** Abstract Test Suite **ATS AVP** Attribute-Value Pairs BCBearer Capability **BCI Backward Call Indicators BICC** Bearer Independent Call Control protocol BLA BLocking Acknowledgement message **BLO** BLOcking message Country Code CCCompletion of Communication to Busy Subscriber **CCBS** Call Deflection CD Call DIVersion **CDIV CFB** Call Forwarding Busy **CFN** ConFusioN message **CFNR** Communications Forwarding No Reply **CFU** Call Forwarding Unconditional Circuit Group Blocking **CGB CGBA** Circuit Group Blocking Acknowledgement message Circuit Group Unblocking message **CGU** Circuit Group Unblocking Acknowledgement message **CGUA CLIP** Calling Line Identification Presentation Calling Line Identification Restriction **CLIR COnnected Line** COL COnnected Line identification Presentation **COLP COLR** COnnected Line identification Restriction CONnect message CON **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

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 Call HOLD

IAM Initial Address Message
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
OBCI Optional Backward Call Indicators
O-IWU Outgoing InterWorking Unit
OLE Originating Local Exchange

O-MGCF Outgoing Media Gateway Control Function

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

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

PT Pay load Type

PTC Parallel Test Component REL RELease message

RES RESUME

RLC ReLease Complete message

RSC ReSet Circuit RTP Real Time Protocol

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

SIP-I Session Initiation Protocol with encapsulated ISUP

SN Subscriber Number SS Supplementary Services

SUB SUBaddressing
SUS SUSPEND
SUT System Under Test

TMR Transmission Medium Requirement

TP Test Purpose TSS Test Suite Structure **UPA** User Part Available message UPT User Part Test message Uniform Resource Identifier URI USI User Service Information parameter USR User-to User message UUS User to User Signalling

4 Test Suite Structure (TSS)

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

SIP -ISUP basic call		
	Sending of the Initial Address Message (IAM)	TP101xxx
	Sending of the Subsequent Address Message (SAM)	TP102xxx
	Sending of COT	TP103xxx
	Receipt of the Address Complete Message (ACM)	TP104xxx
	Receipt of the Call Progress Message (CPG)	TP105xxx
	Receipt of the ANswer Message (ANM)	TP106xxx
	Receipt of the CONnect message (CON)	TP107xxx
	Receipt of the RELease message (REL)	TP108xxx
	Autonomous release at I-IWU	TP109xxx
	Receipt of the BYE, CANCEL message/sending of a REL	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

5 Test Purposes (TP)

5.1 Introduction

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

5.1.1 Test Purpose (TP) naming convention

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

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

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

5.1.2 Source of test purpose definition

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

5.1.3 Test purpose structure

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

5.2 Test purposes for the basic cal

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

5.2.1.1 Sending of the Initial Address Message (IAM)

TP101001	SIP reference: RFC 3261 [4]]		I	SUP reference:			
				Q.1912.5 [1], clause 6.1.2 (i,1)					
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 PI	CS 4/5							
criteria									
ISUP selection	NOT PICS 1/6								
criteria									
Test purpose	Ensure that if the SUT upon	receipt of	of the first	INVITE	with su	ufficient digits, with a SDP			
	offer:								
	 the SUT shall delete μ-l 	law (PCN	/IU), if pre	sent, fro	m the	media description that it will			
	send back in the SDP a	ınswer;							
	 the SUT shall immediat 	ely send	out the IA	λM.					
SIP parameter	SIP INVITE: Audio RTP/AVI	P 0 8							
values	200 OK: Audio RTP/AVP 8								
ISUP parameter	IAM USI: A-law or absent								
values									
Comments	SIP-I		SU	IT		ISUP			
	INVITE(IAM)	→			→	IAM			
	180 Ringing(ACM)	+			←	ACM			
	200 OK INVITE(ANM) ← ANM ACK →								
	Conversation								
	BYE(REL)	→			→	REL			
	200 OK BYE(RLC)	+		•	+	RLC			

TP101002	SIP reference: RFC 3	3261 [4]			15	SUP reference:		
				Q.19	12.5	[1], clause 6.1.2 (i,2ai)		
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)							
SIP selection	PICS 4/4 AND PICS 4/5							
criteria								
ISUP selection	PICS 1/4 AND NOT PICS 1/6	AND PIC	S 4/1					
criteria								
Test purpose	Ensure that if the SUT upon r							
	offer 100rel extensions and p							
	 the SUT shall delete μ-la 	w (PCMU)	, if pre	sent, from	the r	nedia description that it will		
	send back in the SDP an	,						
						vith the coding of the Nature		
	of Connection Indicators		: "CO	T to be exp	ecte	ed".		
SIP parameter	SIP INVITE: Audio RTP/AVP	0 8						
values	200 OK: Audio RTP/AVP 8							
ISUP parameter			exped	ted, USI: A	\-law	or absent		
values	<u> </u>	ontinuity						
Comments	SIP-I		SL			ISUP		
	INVITE(IAM)	→			→	IAM		
	183 Session Progress	+						
	PRACK	→						
	200 OK PRACK	+						
	UPDATE	→			→	COT		
	200 OK UPDATE	+						
			conditi	ons met				
180 Ringing(ACM)								
	←	ANM						
	ACK	→						
			onver					
	BYE(REL)	→			<u>→</u>	REL		
	200 OK BYE(RLC)	+			<u>←</u>	RLC		

TP101003	SIP reference: RFC 3	3261 [4]			ISUP referen			
					12.5 [1], clause	6.1.2 (i,2ai)		
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)							
SIP selection	PICS 4/4 AND PICS 4/5							
criteria								
ISUP selection	PICS 1/4 AND NOT PICS 1/6	AND P	ICS 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 							
CID naramatar	of Connection Indicators SIP INVITE: Audio RTP/AVP	•	ter. CO	i to be exp	ecteu .			
SIP parameter values	200 OK: Audio RTP/AVP 8	08						
ISUP parameter		OT to I	a avnac	tod IISI: A	-law or absent			
values		ontinui		ica, ooi. A	-law or absent			
Comments	SIP-I		SL	IT	ISUP			
	INVITE(IAM)	→			→ IAM			
	183 Session Progress	+			2 17 (17)			
	PRACK	→						
	200 OK PRACK	+						
	UPDATE	→			→ COT			
	200 OK UPDATE	+						
	Preconditions met							
	180 Ringing(ACM) ← ACM							
	200 OK INVITE(ANM) ← ANM							
	ACK	→						
			Conver	sation				
	BYE(REL)	→			→ REL			
	200 OK BYE(RLC)	←		•	F RLC			

TP101004	SIP reference: RFC 3	3261 [4]		Q 10		SUP reference: [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								
criteria		1100 1/1/11/15 1/0							
ISUP selection	PICS 1/5 AND NOT PICS 1/6	AND PIC	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 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						circuit or continuity check			
values	perf COT Continuity Indicator: cor					SI: 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	+							
Preconditions met									
	180 Ringing(ACM)	+			-	ACM			
	200 OK INVITE(ANM)	+			-	ANM			
	ACK	→							
		l	Convers	sation					
	BYE(REL)	→			<u>→</u>	REL			
	200 OK BYE(RLC)	+			(RLC			

TP101005	SIP reference: RFC	3261 [4]		Q.1	-	SUP reference: [1], clause 6.1.2 (i,2aii)	
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5						
criteria							
ISUP selection criteria	PICS 1/5 AND NOT 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 SDI 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 we 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 Require header: media description that it will with the Continuity check	
SIP parameter	SIP INVITE: Audio RTP/AVP						
values	200 OK: Audio RTP/AVP 8						
ISUP parameter	IAM Continuity Indicator: con	tinuity c	heck re	quired or	this	circuit or continuity check	
values	perf COT Continuity Indicator: co					SI: A-law or absent	
Comments	SIP-I		SL	JΤ		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	→					
			Conver	sation			
	BYE(REL)	→			→	REL	
	200 OK BYE(RLC)	+			+	RLC	

TP101006	SIP reference: RFC	3261 [4]			SUP 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	NOT PICS 1/6 AND PICS 4/	NOT PICS 1/6 AND PICS 4/1						
criteria								
Test purpose	Ensure that if the SUT upon							
	offer 100rel extensions and							
			/IU), if pre	sent, fror	m the	media description that it will		
	send back in the SDP a	,						
	 the IAM shall be deferred 		II precond	litions ha	ve be	en met.		
SIP parameter	SIP INVITE: Audio RTP/AVI	≥08						
values	200 OK: Audio RTP/AVP 8							
ISUP parameter	IAM USI: A-law or absent							
values						_		
Comments	SIP-I		SL	JT		ISUP		
	INVITE(IAM)	→						
	183 Session Progress	+						
	PRACK	→						
	200 OK PRACK	+						
	UPDATE	→			→	IAM		
	200 OK UPDATE	+						
		I	Preconditi	ons met				
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
	ACK	→						
			Conver	sation				
	BYE(REL)	→			→	REL		
	200 OK BYE(RLC)	←			+	RLC		

TP101007	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 6.1.2 (i,2b)				
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)								
SIP selection	PICS 4/4 AND PICS 4/5					,			
criteria									
ISUP selection	NOT PICS 1/6 AND PICS 4/	1							
criteria									
Test purpose	send back in the SDP ar	precondi aw (PCM nswer;	tions exte IU), if pre	ensions i sent, fro	n the S m the	SIP Require header: media description that it will			
	 the IAM shall be deferred 		l precond	litions ha	ve be	en met.			
SIP parameter	SIP INVITE: Audio RTP/AVP	08							
values	200 OK: Audio RTP/AVP 8								
ISUP parameter	IAM USI: A-law or absent								
values						I			
Comments	SIP-I	<u> </u>	SL	JT		ISUP			
	INVITE(IAM)	→							
	183 Session Progress	+							
	PRACK	→							
	200 OK PRACK	+							
	UPDATE	→			→	IAM			
	200 OK UPDATE	+							
			recondit	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	.]			SUP reference:				
	Q.1912.5 [1], clause 6.1.2 (i,1)									
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)									
SIP selection criteria	NOT PICS 4/4 AND NOT 4/5									
ISUP selection criteria	PICS 1/6	PICS 1/6								
Test purpose	Ensure that if the SUT upon roffer:	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer:								
		present in the offer of the media description, that it will send it back in the SDP answer;								
SIP parameter	SIP INVITE: Audio RTP/AVP	,	out the h	MIVI.						
values	200 OK: Audio RTP/AVP 0									
ISUP parameter	IAM USI: μ-law									
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				

TP101009	SIP reference: RFC 3	261 [4]		ISUP reference:					
		4.1. 1.1.1.1.1		5 [1], clause 6.1.2 (i,2ai)					
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)								
SIP selection	PICS 4/4 AND PICS 4/5								
criteria									
ISUP selection	PICS 1/4 AND PICS 1/6 AND	PICS 4/1							
criteria									
Test purpose	 offer 100rel extensions and p the SUT shall delete A-la present in the offer of the answer; the IAM shall be sent out 	 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	nnection Indica	tors parameter:	"COT to be expected" COT;					
values	Continuity Indicator: continui	ty	•	-					
Comments	SIP-I	,	SUT	ISUP					
	INVITE(IAM)	→	→	IAM					
	183 Session Progress	+							
	PRACK	→							
	200 OK PRACK	+							
	UPDATE	→	→	COT					
	200 OK UPDATE	+							
		Precond	litions met						
	180 Ringing(ACM)	+	(ACM					
	200 OK INVITE(ANM)	+	+	ANM					
	ACK	→							
		Conv	ersation						
	BYE(REL)	→	→	REL					
	200 OK BYE(RLC)	+	+	RLC					

TP101010	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 6.1.2 (i,2ai)					
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)								
SIP selection	PICS 4/4 AND PICS 4/5								
criteria									
ISUP selection	PICS 1/4 AND PICS 1/6 AND	PICS 4	/1						
criteria									
Test purpose	 offer 100rel extensions and p the SUT shall delete A-la present in the offer of the answer; the IAM shall be sent out 	present in the offer of the media description, that it will send it back in the SDP							
SIP parameter	SIP INVITE: Audio RTP/AVP								
values	200 OK: Audio RTP/AVP 0	-							
ISUP parameter	IAM USI: μ-law; Nature of Co	nnection	n Indicato	rs paramete	er: " C	OT to be expected" COT;			
values	Continuity Indicator: continui	ity		-		-			
Comments	SIP-I		SU	T	I	SUP			
	INVITE(IAM)	→		-	→	AM			
	183 Session Progress	+							
	PRACK	→							
	200 OK PRACK	+							
	UPDATE	→		-	→ (COT			
	200 OK UPDATE	+							
		P	reconditi	ons met					
	180 Ringing(ACM)	+		•	E /	ACM			
	200 OK INVITE(ANM)	+		•	E /	ANM			
	ACK	→							
			Convers	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								
criteria	1100 11 11100 110								
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	ontinuity che	ck required on this circuit"				
values	or continuity check pe				-				
	COT Continuity Indicator: cor	ntinuity ch	eck s	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	→							
			onvers	sation					
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP101012	SIP reference: RFC 326		Q.1912.5	SUP reference: [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									
criteria										
ISUP selection criteria	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 chec continuity check perform COT Continuity Indicator: contin	ed on previou	us circuit	k required on this circuit"						
Comments	SIP-I	SL		ISUP						
	INVITE(IAM)		→	IAM						
	183 Session Progress	:								
	PRACK -									
	200 OK PRACK	_								
	UPDATE =	>	→	COT						
	200 OK UPDATE	=								
		Precondit	ions met							
	180 Ringing(ACM)		+	ACM						
	200 OK INVITE(ANM)	-	+	ANM						
	ACK =									
		Conver	sation							
	BYE(REL)		→	REL						
	200 OK BYE(RLC)	-	+	RLC						

TP101013	SIP reference: RFC]	ISUP reference:							
T00 (Q.1912.5 [1], clause 6.1.2 (i,2b)									
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)									
SIP selection	PICS 4/4 AND PICS 4/5	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-l present in the offer of th answer; 	 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 deferred until all preconditions have been met. 								
SIP parameter	SIP INVITE: Audio RTP/AVF									
values	200 OK: Audio RTP/AVP 0									
ISUP parameter	IAM USI: μ-law									
values	·									
Comments	SIP-I		SU	IT		ISUP				
	INVITE(IAM)	→								
	183 Session Progress	+								
	PRACK	→								
	200 OK PRACK	+								
	UPDATE	→			→	IAM				
	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				

TP101014	SIP reference: RFC	3261 [4		ISUP reference: Q.1912.5 [1], clause 6.1.2 (i,2b)						
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)									
SIP selection	PICS 4/4 AND PICS 4/5									
criteria										
ISUP selection criteria	PICS 1/6 AND PICS 4/1									
Test purpose	 offer 100rel extensions and the SUT shall delete A-lapresent in the offer of the answer; 	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		SUT		ISUP					
	INVITE(IAM)	→								
	183 Session Progress	←								
	PRACK	→								
	200 OK PRACK	+								
	UPDATE	→		→	IAM					
	200 OK UPDATE	+								
			Preconditions me	et						
	180 Ringing(ACM)	←		←	ACM					
	200 OK INVITE(ANM)	+		+	ANM					
	ACK	→								
			Conversation							
	BYE(REL)	→		→	REL					
	200 OK BYE(RLC)	+		+	RLC					

TP101015	SIP reference: RFC	3261 [4]	Q.191		SUP reference: 1], clauses 6.1.3.2, 6.1.3.3, 6.1.3.4					
TSS reference	SIP-ISUP/Basic call/Sending	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)								
SIP selection criteria										
ISUP selection criteria	NOT PICS 1/9 AND NOT PI	CS 4/4 and N	OT PICS 4/5							
Test purpose SIP parameter	 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. 									
values										
ISUP parameter values										
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	→		→	IAM					
	180 Ringing(ACM)	←		+	ACM					
	200 OK INVITE(ANM) ← ANM									
	ACK →									
		Co	nversation							
	BYE(REL)	→		→	REL					
	200 OK BYE(RLC)	+		+	RLC					

TP101015A	SIP reference: RF	C 3261 [4]	ES 283 02	ISUP reference: ES 283 027 [11], clause 7.2.3.1.2.5 TS 129 164 [10], clause 6.2.4.1.3.2				
TSS reference	SIP-ISUP/Basic call/Sendi	ing of the Initial Ad	dress message (IAM)				
SIP selection criteria	Based on table 1A							
ISUP selection criteria								
SIP Parameter values ISUP Parameter values	Ensure that the SUT in the media description defined a_b_m_LINE_VALUE: and sends an IAM message, w to TMR_VALUE derived for the USI is set in parallel to	Mapping of SDP into the TMR Ensure that the SUT in the Idle state on receipt of an INVITE message containing the media description defined in table 1 with the "a =" "b =" and "m=" lines set to a_b_m_LINE_VALUE: and the media description does not match the TMR and USI value sends an IAM message, with the Transmission Medium Requirement (TMR) parameter set to TMR_VALUE derived from the media description. The Information transfer capability in the USI is set in parallel to the TMR value. INVITE; a_b_m_LINE_VALUE						
Comments	SIP		UT	ISUP				
Commonto	INVITE (IAM)	→	→	IAM				
	180 Ringing (ACM)	+	-	ACM				
	rearranging (rearry		ng tone	7.0				
	200 OK INVITE (ANM) ← ANM ACK							
		Conv	ersation	•				
	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], clause 6.1.3.5									
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	USI= unrestricted digital information	ISUP_TMR = 64 kbits/s unrestricted				
	ISDN_BC_ITR = 64 kbits/s unrestricted					
VA_04	No USI contained in the encapsulated IAM	ISUP_TMR = speech				
VA_05	No USI contained in the encapsulated IAM	ISUP_TMR = 3,1 kHz audio				
VA_06	No USI contained in the encapsulated IAM	ISUP_TMR = 64 kbits/s unrestricted				

TP101017	SIP reference: RFC	3261 [4]		ISUP reference: 2.5 [1], clause 6.1.3.5				
TSS reference	SIP-ISUP/Basic call/Sendin	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	encapsulated IAM message	Ensure that the SUT in the Idle state on receipt of a INVITE message with an encapsulated IAM message the HLC shall be taken from the encapsulated ISUP: • sends an IAM message, with the HLC taken from the encapsulated ISUP.						
SIP parameter	INVITE;							
values	·							
ISUP parameter values	IAM; Access transport pa	rameter HLC: H	ILC_VALUE; USI					
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				

	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	g of the li	nitial Address	Message	(IAM)		
SIP selection	NOT PICS 4/4 and NOT PIC	S 4/5					
criteria							
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		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		

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	S 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				Trace -			
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]	Q.		SUP reference: 2.5 [1], clause 6.1.3.1			
TSS reference	SIP-ISUP/Basic call/Sending	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)						
SIP selection	PICS 1/9 AND NOT PICS 4/4			J - (
criteria								
ISUP selection	PICS 1/7							
criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a Called party number contained in the user info component of the Request-URI. Send an IAM Message with the called party number coded as follows: Nature of address indicator: Analyse the information contained in received URI with user=phone, and if it is in the format: +CC NDC SN where CC is not the country code of the network in which the next hop terminates, then set Nature of Address indicator to "International number", remove "+" and use the remaining digits to fill the Address signals. Internal Network Number Indicator: routing to internal network number not allowed. Numbering plan Indicator 001 ISDN (Telephony) numbering plan. Address Signals CC NDC SN.							
SIP parameter								
values								
ISUP parameter	IAM: Called party number							
values					1			
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			

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	of the Ir	nitial Addr	ess Mes	sage (IAM)	
SIP selection	NOT PICS 4/4 AND NOT PIC	S 4/5 A	ND PICS	1/9			
criteria							
ISUP selection							
criteria							
Test purpose	Ensure that the SUT on recei						
	a-Law, then independent fro	om the r	eceived	order of	prefe	rence:	
	 the G.711 a-law codec s 	hall be r	eturned i	n the SD	P 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							
values							
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	

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	Iress 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			
				SIP Supported header, then
	independent from the receive			
				with the coding of the Nature
	of Connection Indicators			
	 the G.711 a-law codec sł 	hall 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			
ISUP parameter		COT to be expe	cted, USI: A-lav	v or absent
values		ontinuity		
Comments	SIP-I		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	→		
		Conve	rsation	
	BYE(REL)	→	→	REL
	200 OK BYE(RLC)	+	+	RLC

TP101023	SIP reference: RFC 3	3261 [4]		I	SUP reference:		
			EN 3	383 0	01 [2], clause 6.1.3.5.2.2		
TSS reference	SIP-ISUP/Basic call/Sending	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)					
SIP selection	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/					
criteria							
Test purpose	Ensure that the SUT on recei						
	a-Law 100rel extensions and			n the	SIP Require header, then		
	independent from the recei	ved order of p	reference:				
					with the coding of the Nature		
	of Connection Indicators						
	 the G.711 a-law codec sl 	nall be returned	in the SDF	ansv	wer as preferred codec.		
SIP parameter	Offer: m=audio 4711 RTF						
values	Answer: m=audio 4712 RTF						
ISUP parameter		OT to be exp	ected, USI:	A-lav	v or absent		
values	·	ontinuity					
Comments	SIP-I		SUT		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	→					
		Conv	ersation				
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP101024	SIP reference: RFC 3	3261 [4]		-	SUP reference:
T00 (f (1 1 1 1			01 [2], clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/Sending			ess Message	(IAM)
SIP selection	PICS 4/4 AND PICS 4/5 AND) PICS 1/9)		
criteria	DIGG 4/5 AND NOT DIGG 4/6	AND DIC	20.4/4		
ISUP selection criteria	PICS 1/5 AND NOT PICS 1/6	AND PIC	S 4/1		
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for μ-Law and a-Law 100rel extensions and preconditions extensions in the SIP Supported header, then independent from the received order of preference: • 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 sl 		urned in	the SDP ans	wer as preferred codec.
SIP parameter	Offer: m=audio 4711 RTF				
values	Answer: m=audio 4712 RTF				
ISUP parameter					his circuit or continuity
values					circuit, USI: A-law or absent
		continuity		successful	
Comments	SIP-I		SU		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	→			
		(Convers	ation	
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+	-	←	RLC

TP101025	SIP reference: RFC 3	3261 [4]			ISUP reference:
			i i	EN 383 0	01 [2], clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/Sending		Address M	lessage	(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	l/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 Continuity check
	indicator "continuity che		on this c	ircuit" o	r "continuity check
	performed on previous			000	
OID	the G.711 a-law codec sl		ed in the	SDP ans	wer as preferred codec.
SIP parameter values	Offer: m=audio 4711 RTI				
	Answer: m=audio 4712 RTI		- ale na anc	!a.l a.a. 4	his singuit on southwrite.
ISUP parameter values	IAM Continuity Indicator: c	continuity cn	eck requ	irea on t	his circuit or continuity circuit, USI: A-law or absent
values		ontinuity ch			Circuit, USI. A-law or absent
Comments	SIP-I		SUT	- CSSIGI	ISUP
Comments	INVITE(IAM)	→	001	→	IAM
	183 Session Progress	É		 	I William
	PRACK	→			
	200 OK PRACK	+			
	UPDATE	→		→	COT
	200 OK UPDATE	+			
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	-		+	ANM
	ACK	→			
		Cor	versation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		+	RLC

TP101026	SIP reference: RFC 3261 [4]		FN 383	ISUP reference: 001 [2], clause 6.1.3.5.2.2			
TSS reference	SIP-ISUP/Basic call/Sending	of the li	nitial Addı				
SIP selection		SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM) PICS 4/4 AND PICS 4/5 AND PICS 1/9					
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]			ISUP reference:				
					001 [2], clause 6.1.3.5.2.2			
TSS reference		SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)						
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 4/4 AND PICS 4/5 AND PICS 1/9						
criteria								
ISUP selection	PICS 1/5 AND NOT PICS 1/6	6 AND N	OT PICS	4/1				
criteria								
Test purpose	Ensure that the SUT on rece							
	a-Law 100rel extensions and				e SIP Require header, then			
	independent from the recei		-					
	 the shall be deferred unt 							
		 the G.711 a-law codec shall be returned in the SDP answer as preferred codec. 						
SIP parameter	Offer: m=audio 4711 RT		-					
values	Answer: m=audio 4712 RT	P/AVP 8	0					
ISUP parameter								
values		1						
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	→						
			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/Sendir	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)						
SIP selection	NOT PICS 4/4 AND NOT F	PICS 4/5 AN	ND PICS	1/9				
criteria								
ISUP selection criteria	PICS 1/7							
Test purpose	μ-Law, then independent	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 F	Offer: m=audio 4711 RTP/AVP 8						
values	Answer: m=audio 4711 R	RTP/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	ation	•			
	BYE(REL)	→			→	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP101029	SIP reference: RFC 3261 [4]				-	SUP reference:		
						01 [2], clause 6.1.3.5.2.2		
TSS reference	SIP-ISUP/Basic call/Sending			ress Messa	age (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							
	μ-Law 100rel extensions and					SIP Supported header, then		
	independent the normal offe							
						with the coding of the Nature		
	of Connection Indicators							
	 the G.711 a-law codec sł 		eturned i	n the SDP	ansv	ver.		
SIP parameter	Offer: m=audio 4711 RTF	-						
values	Answer: m=audio 4711 RTF							
ISUP parameter				ted, USI: A	\-law	v or absent		
values		continuity						
Comments	SIP-I		SL			ISUP		
	INVITE(IAM)	→			<u>→</u>	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	→						
			Conver	sation				
	BYE(REL)	→			→	REL		
	200 OK BYE(RLC)	-			←	RLC		

TP101030	SIP reference: RFC 3261 [4]		ISUP reference:				
				EN:	383 0	01 [2], clause 6.1.3.5.2.2	
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)						
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 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 shall be returned in the SDP answer.						
SIP parameter	Offer: m=audio 4711 RTF						
values	Answer: m=audio 4711 RTF						
ISUP parameter	1		•	ted, USI	: A-lav	v or absent	
values	,	ontinui	•			lious	
Comments	SIP-I		SU			ISUP	
	INVITE(IAM)	→			→	IAM	
	183 Session Progress	(
	PRACK	→					
	200 OK PRACK	(207	
	UPDATE	→			→	COT	
	200 OK UPDATE	←	11.1				
	100 D: : (1011)		reconditi	ons met	-	1.014	
	180 Ringing(ACM)	(+	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
	ACK	→					
	D)(E(DEL)		Conver	sation		551	
	BYE(REL)	→			→	REL	
	200 OK BYE(RLC)	←			←	RLC	

TP101031	SIP reference: RFC	3261 [4]				ISUP reference:		
				EN	383 0	01 [2], clause 6.1.3.5.2.2		
TSS reference	SIP-ISUP/Basic call/Sending	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)						
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 4/4 AND PICS 4/5 AND PICS 1/9						
criteria								
ISUP selection	PICS 1/5 AND NOT PICS 1/6	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1						
criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no							
	μ-Law 100rel extensions and preconditions extensions in the SIP Supported header, 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 ch performed on previous			tnis circ	uit o	continuity check		
	 the G.711 a-law codec s 			n tha SDI	D and	wor		
SIP parameter	Offer: m=audio 4711 RT			ii iile SDI	ans	wei.		
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	IT		ISUP		
	INVITE(IAM)	→			→	IAM		
	183 Session Progress	+						
	PRACK	→						
	200 OK PRACK	+						
	UPDATE	→			→	COT		
	200 OK UPDATE	+						
		Preconditions met						
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
	ACK	→						
		<u> </u>	Convers	sation				
	BYE(REL)	→			→	REL		
	200 OK BYE(RLC)	+			←	RLC		

TP101032	SIP reference: RFC 3261 [4]					ISUP reference:		
				EN 3	83 0	01 [2], clause 6.1.3.5.2.2		
TSS reference	SIP-ISUP/Basic call/Sending	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)						
SIP selection		PICS 4/4 AND PICS 4/5 AND PICS 1/9						
criteria								
ISUP selection	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1							
criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no							
	μ-Law 100rel extensions and preconditions extensions in the SIP Require 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 che			tnis circu	IIt" O	r continuity cneck		
	 performed on previous the G.711 a-law codec s 			a tha CDE	000	wor		
SIP parameter	the G.711 a-law codec sOffer: m=audio 4711 RT			Title SDF	ans	wei.		
values	Answer: m=audio 4711 RT							
ISUP parameter				required	on t	this circuit or continuity		
values						circuit, USI: A-law or absent		
				success				
Comments	SIP-I		SU	Т		ISUP		
	INVITE(IAM)	→			→	IAM		
	183 Session Progress	+						
	PRACK	→						
	200 OK PRACK	+						
	UPDATE	→			→	COT		
	200 OK UPDATE	+						
	Preconditions met							
	180 Ringing(ACM)	+			<u>+</u>	ACM		
	200 OK INVITE(ANM)	+			-	ANM		
	ACK	→						
		<u> </u>	Convers	sation				
	BYE(REL)	→			<u>→</u>	REL		
	200 OK BYE(RLC)	←			+	RLC		

TP101033	SIP reference: RFC	3261 [4]				SUP reference:		
				EN 38	83 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	PICS 4/4 AND PICS 4/5 AND 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		-		Tiours		
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:				
					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 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					a SDP offer for a-Law and no			
	μ-Law 100rel extensions and							
	independent the normal off							
	 the IAM shall be deferred 							
	 the G.711 a-law codec s 		turned ii	n the SDP ans	swer.			
SIP parameter		Offer: m=audio 4711 RTP/AVP 8						
values	Answer: m=audio 4711 RT	P/AVP 8						
ISUP parameter								
values								
Comments	SIP-I		SU	T	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			

TP101035	SIP reference: RFC 3	261 [4]	ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2					
TSS reference	SIP-ISUP/Basic call/Sending	of the Initial Add	ress Message (IAM)				
SIP selection	NOT PICS 4/4 AND NOT PIC	S 4/5 AND PICS	S 1/9					
criteria								
ISUP selection criteria	PICS 1/7							
Test purpose	Ensure that the SUT on receip	ot of an INVITE r	message with a	SDP offer m line without				
	a-law codec:							
	• the u-law codec shall be	the u-law codec shall be rejected.						
SIP parameter	Offer: m=audio 4711 RTF	Offer: m=audio 4711 RTP/AVP 0						
values	m=audio 4712 RTF	P/AVP 8						
	Answer: m=audio 0 RTP/AV	/P 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]				I	SUP reference:		
	EN 383 001 [2], clause 6.1.3.5.2.2							
TSS reference	SIP-ISUP/Basic call/Sending	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)						
SIP selection	PICS 4/4 AND PICS 4/5 AN	D 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 rece							
						in the SIP Supported header:		
						with the coding of the Nature		
	of Connection Indicators			T to be ex	xpect	ed";		
	the u-law codec shall							
SIP parameter	Offer: m=audio 4711 RT	-						
values	m=audio 4712 RT							
	Answer: m=audio 0 RTP/A							
ISUP parameter			•	ted, USI	: A-lav	v or absent		
values		continui						
Comments	SIP-I		SU	Т		ISUP		
	INVITE(IAM)	→			→	IAM		
	183 Session Progress	←						
	PRACK	→						
	200 OK PRACK	←						
	UPDATE	→			→	СОТ		
	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		

TP101037	SIP reference: RFC 3261 [4]		ISUP reference:			
			EN 383 0	01 [2], clause 6.1.3.5.2.2		
TSS reference	SIP-ISUP/Basic call/Sending	of the Initial Add	Iress 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					
	a-law codec 100rel extension	ns and precondit	ions extensions	s in the SIP Require header:		
				with the coding of the Nature		
	of Connection Indicators	parameter: "CO	T to be expect	ed";		
	 the u-law codec shall b 	e rejected.				
SIP parameter	Offer: m=audio 4711 RTF	P/AVP 0				
values	m=audio 4712 RTF	P/AVP 8				
	Answer: m=audio 0 RTP/A\					
ISUP parameter		COT to be expe	cted , USI: A-lav	v or absent		
values	COT Continuity Indicator: c	ontinuity				
Comments	SIP-I	SI	JT	ISUP		
	INVITE(IAM)	→	→	IAM		
	183 Session Progress	-				
	PRACK	→				
	200 OK PRACK	+				
	UPDATE	→	→	COT		
	200 OK UPDATE	←				
		Precondit	tions met			
	180 Ringing(ACM)	+	+	ACM		
	200 OK INVITE(ANM)	+	+	ANM		
	ACK	→				
		Conve	rsation			
	BYE(REL)	→	→	REL		
	200 OK BYE(RLC)	+	+	RLC		

TP101038	SIP reference: RFC	3261 [4]		ENI		SUP reference:
	EN 383 001 [2], clause 6.1.3.5.2.2					
TSS reference	SIP-ISUP/Basic call/Sending			ress Mess	sage	(IAM)
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 1/	9			
criteria						
ISUP selection criteria	PICS 1/5 AND NOT PICS 1/6	S AND PIO	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 extension	ns and pr	econditi	ons exter	nsions	s in the SIP Supported header:
						with the Continuity check
	indicator "continuity che			this circu	uit" o	"continuity check
	performed on previous					
	 the u-law codec shall b 	e rejecte	d.			
SIP parameter	Offer: m=audio 4711 RTI					
values	m=audio 4712 RTI	P/AVP 8				
	Answer: m=audio 0 RTP/A					
ISUP parameter						his circuit or continuity
values						circuit, USI: A-law or absent
	·	continuit		success	sful	
Comments	SIP-I		SU	ΙΤ		ISUP
	INVITE(IAM)	→			→	IAM
	183 Session Progress	←				
	PRACK	→				
	200 OK PRACK	+				
	UPDATE	→			→	COT
	200 OK UPDATE	←				
		Pr	econditi	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				SUP reference:	
				EN:	383 0	01 [2], clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/Sending	of the In	itial Addı	ress Mess	sage	(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	6 AND PI	ICS 4/1			
criteria						
Test purpose	Ensure that the SUT on rece					
	a-law codec 100rel extensio					
						with the Continuity check
	indicator "continuity che			this circ	uit" o	r "continuity check
	performed on previous					
oin 1	the u-law codec shall be a second to the u-					
SIP parameter	Offer: m=audio 4711 RT					
values	m=audio 4712 RT					
ICUD maramatar	Answer: m=audio 0 RTP/A				1 4	his singuit on southwrite.
ISUP parameter values						his circuit or continuity
values				succes		circuit, USI: A-law or absent
Comments	SIP-I		SU		siui	ISUP
Comments	INVITE(IAM)	→	- 30	' 1	→	IAM
	183 Session Progress	-				IZAIVI
	PRACK	→				
	200 OK PRACK	-				
	UPDATE	→			→	COT
	200 OK UPDATE	+				
		P	reconditi	ons met		
	180 Ringing(ACM)	+			←	ACM
	200 OK INVITE(ANM)	+			+	ANM
	ACK	→				
			Convers	sation		
	BYE(REL)	→			→	REL
	200 OK BYE(RLC)	+	·		+	RLC

TP101040	SIP reference: RFC 3	3261 [4]			-	SUP reference:
						01 [2], clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/Sending			ress Messa	age (IAM)
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 1	1/9			
criteria						
ISUP selection	NOT PICS 1/6 AND NOT PIC	S 4/1				
criteria						
Test purpose	Ensure that the SUT on receip					
						in the SIP Supported header:
	 the IAM shall be deferred 	d until al	I precond	litions have	e bee	en met;
	 the u-law codec shall be 					
SIP parameter	Offer: m=audio 4711 RTF	,,,,,,	•			
values	m=audio 4712 RTF	P/AVP 8	3			
	Answer: m=audio 0 RTP/A\	/P 0				
ISUP parameter						
values						
Comments	SIP-I		SU	IT		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

TP101041	SIP reference: RFC 3261 [4]		ISUP reference:			
				EN	383 0	01 [2], clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/Sendin	g of the Ir	nitial Addı	ress Mes	sage	(IAM)
SIP selection	PICS 4/4 AND PICS 4/5 AN	ID PICS 1	/9			
criteria						
ISUP selection	NOT PICS 1/6 AND NOT P	ICS 4/1				
criteria						
Test purpose	Ensure that the SUT on rec					
						s in the SIP Require header:
	 the IAM shall be deferred 	ed until al	I precond	litions ha	ive be	en met;
	 the u-law codec shall 	be reject	ed.			
SIP parameter	Offer: m=audio 4711 R	TP/AVP 0)			
values	m=audio 4712 R	TP/AVP 8	1			
	Answer: m=audio 0 RTP/	AVP 0				
ISUP parameter						
values						
Comments	SIP-I		SU	IT		ISUP
	INVITE(IAM)	→				
	183 Session Progress	+				
	PRACK	→				
	200 OK PRACK	+				
	UPDATE	→			→	IAM
	200 OK UPDATE	←				
		F	reconditi	ons met		
	180 Ringing(ACM)	+			←	ACM
	200 OK INVITE(ANM)	←			←	ANM
	ACK	→				
			Convers	sation		
	BYE(REL)	→			→	REL
	200 OK BYE(RLC)	+	•	•	+	RLC

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

TP101043	SIP reference: RFC 3261	1 [4] ISUP reference:						
			EN	383 001 [2], clause 6.1.3.5.2.2				
TSS reference	SIP-ISUP/Basic call/Sending of the	Initial /	Address Mes	ssage (IAM)				
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS	3 1/9 AN	ND PICS 4/1	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 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= 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:			
			EN	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	1/9 AN	ND PICS 4/19				
criteria							
ISUP selection	NOT PICS 1/6						
criteria							
Test purpose	Ensure that the SUT on receipt of a	ın INVI	TE message	with a SDP offer with more than			
	one media streams 100rel extensi	ons an	d preconditio	ns extensions in the SIP Require			
	header and based on operator po	licy th	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]	_	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	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 recei			SDP offer with more than			
	one media streams and bas						
	 if the SDP offer contains 						
			ıdio streams sh	all be considered; the other			
	streams shall be rejected	l;					
	 if the SDP offer contains 			ms, the IWU shall only			
	consider one, and reject		S.				
SIP parameter	Offer: m=audio 4711 RTF	P/AVP 8					
values	m= audio 4712 RT						
	m= video 4713 RT	P/AVP 31					
	Answer: m=audio 4711 RTF						
	m=audio 0 RTP/A\	-					
	m=video 0 RTP/A\	/P 31					
ISUP parameter							
values				I			
Comments	SIP-I	SL		ISUP			
	INVITE(IAM)	→	→	IAM			
	180 Ringing(ACM)	+	←	ACM			
	200 OK INVITE(ANM)	+	←	ANM			
	ACK	→					
		Conver	sation				
	BYE(REL)	→	→	REL			
	200 OK BYE(RLC)	+	+	RLC			

TP101046	SIP reference: RFC 3	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	NOT PICS 4/4 AND NOT PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19								
criteria									
ISUP selection	PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1								
criteria									
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than								
		one media streams 100rel extensions and preconditions extensions in the SIP Supported							
	header and based on operation			DIGG : 1	ed at the fall block				
	 the IAM shall be sent out of Connection Indicators 	immedia naramet	ately on the er: " COT t o	BICC 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	audio type i	media strea	ms, the IWU shall only				
	consider one, and reject				•				
SIP parameter	Offer: m=audio 4711 RTF	P/AVP 8							
values	m= audio 4712 RT								
	m= video 4713 RTP/AVP 31								
	Answer: m=audio 4711 RTF	-							
	m=audio 0 RTP/A\ m=video 0 RTP/A\								
ISUP parameter			a avnactor	1 11Cl· Δ-lav	w or absent				
values		continuit		a, 001. A-la	W OI absent				
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	183 Session Progress	+							
	PRACK	→							
	200 OK PRACK	+							
	UPDATE	→		→	COT				
	200 OK UPDATE	+							
		Preco	onditions m	et					
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK	→							
			Conversat						
	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 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							
				ons ex	tensions in the SIP Require			
	header and based on opera			-: -i -	with the ending of the Netwo			
	 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	the other st			•			
SIP parameter	Offer: m=audio 4711 RT							
values	m= audio 4712 RT							
	m= video 4713 RT	P/AVP 31						
	A	D/A\/D 0						
	Answer: m=audio 4711 RTI m=audio 0 RTP/A							
	m=video 0 RTP/A\							
ISUP parameter			expected, US	I: A-lav	v or absent			
values		continuity	mpoorou, oo					
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	→		1				
			onversation					
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	←		←	RLC			

TP101048	SIP reference: RFC	3261 [4]		ISUP reference:				
			EN 383 (001 [2], clause 6.1.3.5.2.2				
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)							
SIP selection	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 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							
ISUP parameter values	Answer: m=audio 4711 RT m=audio 0 RTP/A m=video 0 RTP/A IAM Continuity Indicator:	m=audio 0 RTP/AVP 8 m=video 0 RTP/AVP 31						
Comments	COT Continuity Indicator: SIP-I	continuity chec	UT UT	ISUP				
Comments	INVITE(IAM)	→	• • • • • • • • • • • • • • • • • • •	IAM				
	183 Session Progress	+	7	IAW				
	PRACK	→						
	200 OK PRACK	+						
	UPDATE	→	→	СОТ				
	200 OK UPDATE	+						
		Precondition	ns met					
	180 Ringing(ACM)	+	+	ACM				
	200 OK INVITE(ANM)	+	+	ANM				
	ACK	→						
		· · · · · · · · · · · · · · · · · · ·	rsation					
	BYE(REL)	→	→	REL				
	200 OK BYE(RLC)	←	←	RLC				

TP101049	SIP reference: RFC	3261 [4]		ISUP reference:				
			EN 383	001 [2], clause 6.1.3.5.2.2				
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)							
SIP selection	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=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]			ISUP reference:	
					001 [2], clause 6.1.3.5.2.2	
TSS reference	SIP-ISUP/Basic call/Sendin					
SIP selection	PICS 4/4 AND PICS 4/5 AN	ID PICS 1/	9 AND NOT	PICS 4/19	9	
criteria		100 111				
ISUP selection	NOT PICS 1/6 AND NOT P	ICS 4/1				
criteria	Francis that the OUT are	-1-1-1-1	NN //TE		ODD -#	
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams 100rel extensions and preconditions extensions in the SIP Supported header and based on operator policy then: the IAM shall be deferred until all preconditions have been met; if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected; if the SDP offer contains several audio type media streams, the IWU shall only					
	consider one, and reject					
SIP parameter values	Offer: m=audio 4711 RTP/AVP 8 m= audio 4712 RTP/AVP 8 m= video 4713 RTP/AVP 31 Answer: m=audio 4711 RTP/AVP 8 m=audio 0 RTP/AVP 8 m=video 0 RTP/AVP 31					
ISUP parameter values		<u> v.</u>				
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	→	Onn. 15 1'		-	
	DVE(DEL)		Conversatio		REL	
	BYE(REL)	→		→	RLC	
	200 OK BYE(RLC)			7	INLO	

TP101051	SIP reference: RF0	C 3261 [4]			ISUP reference:	
					001 [2], clause 6.1.3.5.2.2	
TSS reference	SIP-ISUP/Basic call/Sendir	ng of the Ir	itial Address	Message	(IAM)	
SIP selection	PICS 4/4 AND PICS 4/5 AN	ND PICS 1	/9 AND NOT	PICS 4/19	9	
criteria						
ISUP selection	NOT PICS 1/6 AND NOT F	7ICS 4/1				
criteria	Francisco that the CLIT are year		INIV/ITE manage		CDD affair with many them	
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams 100rel extensions and preconditions extensions in the SIP Require header and based on operator policy then: the IAM shall be 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					
				edia strea	ims, the IVVO shall only	
SIP parameter	consider one, and reje Offer: m=audio 4711 R					
values	m= audio 4712 F m= video 4713 F	RTP/AVP 8 RTP/AVP 3	3 31			
	Answer: m=audio 4711 R m=audio 0 RTP/					
	m=video 0 RTP/	AVP 31				
ISUP parameter values						
Comments	SIP-I		SUT		ISUP	
	INVITE(IAM)	→				
	183 Session Progress	+				
	PRACK	→				
	200 OK PRACK	+				
	UPDATE	→		→	IAM	
	200 OK UPDATE	+				
		Pred	onditions me	t		
	180 Ringing(ACM)	+		←	ACM	
	200 OK INVITE(ANM)	+		←	ANM	
	ACK	→				
			Conversation	n		
	BYE(REL)	→		→	REL	
	200 OK BYE(RLC)	←		←	RLC	

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 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						
		Conve	rsation				
	BYE(REL)	→	→	REL			
	200 OK BYE(RLC)	+	+	RLC			

TP102002	SIP reference: RFC 3	261 [4]		_	SUP reference:			
TCC reference	Q.1912.5 [1], clause 6.2 b) SIP-ISUP/Basic call/Sending of the Subsequent Address Message (SAM)							
TSS reference		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]		Q.1	ISUP reference: 912.5 [1], clause 6.3				
TSS reference	SIP-ISUP/Basic call/COT	SIP-ISUP/Basic call/COT						
SIP selection criteria	PICS 4/4 AND PICS 4/5							
ISUP selection criteria	PICS 1/4 AND PICS 4/1	PICS 1/4 AND PICS 4/1						
Test purpose	Ensure that the when the SUT determines that all the preconditions on the incoming SIP side have been met and any continuity procedures on the outgoing BICC side have been successfully completed: • the SUT shall send the COT message where the Continuity Indicator in the COT message shall be set to "Continuity".							
SIP parameter values								
ISUP parameter values	COT continuity indicator: C	ontinuity	1					
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			

TP103002	SIP reference: RF	SIP reference: RFC 3261 [4]			ISUP reference:			
		• •		Q.19	912.5 [1], clause 6.3			
TSS reference	SIP-ISUP/Basic call/COT	SIP-ISUP/Basic call/COT						
SIP selection	PICS 4/4 AND PICS 4/5							
criteria								
ISUP selection	PICS 1/5 AND PICS 4/1							
criteria								
Test purpose	Ensure that the when the SUT determines that all the preconditions on the incoming SIP side have been met and any continuity procedures on the outgoing ISUP side have been successfully completed: • the I-IWU shall send the COT message where the Continuity Indicator in the COT message shall be set to "Continuity check successful".							
SIP parameter	_							
values								
ISUP parameter	COT continuity indicator: 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 3261 [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: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 6.5 1)			
TSS reference	SIP-ISUP/Basic call/Receip	t of the A	ddress comple	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		'					
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 →						
			Conversatio				
	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 326	1 [4]			ISUP reference: 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 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 values	ACM: Called party status "no inc CPG; event information param			Alertir	ng			
Comments	SIP-I		SUT		ĬSUP			
	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				ISUP reference: 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 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]		ISUP reference:					
				912.5 [1], clause 6.7					
TSS reference	SIP-ISUP/Basic call/Receipt of the Answer message (ANM).								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Ensure that the SUT, having rece	ived the ACI	M message, on	receipt of an ANM message:					
	 sends a 200 OK INVITE; 								
	the received ANM is encapsulate								
	The bearer path shall be connected	ed in both di	rections when b	ooth of the following conditions					
	are satisfied:								
	the BICC outgoing bearer ser			Recommendation					
	Q.1902.4 [i.14]) is successful								
	the I-IWU determines (using)								
		stied on the	SIP side for ses	ssion establishment to proceed					
	(if applicable).	tha "Dan aall	h	a the few yeard divertion.					
	In addition, if BICC is performing of Outgoing bearer set-up procedure								
	bearer path shall be connected in								
	and the I-IWU determines (through								
	preconditions have been met for t			Tri O 3312 [3]) that sumolent					
SIP parameter	200 OK INVITE with encapsulated		o proceed.						
values	200 OK IIVITE With cheapsdiates	3 7 (1 4 1V)							
ISUP parameter	ANM								
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					

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								
Test purpose	SDP offer was received in th	e initial	I INVITE. E	nsure th	nat the	SUT, on receipt of an CON		
	message:							
	sends a 200 OK INVITE;							
	• the received CON is enca							
	are satisfied:	ected i	in both dired	ctions w	hen b	oth of the following conditions		
	 the BICC outgoing bearer Q.1902.4 [i.14]) is succes 				TU-T I	Recommendation		
					d in R	FC 3312 [5]) that sufficient		
						ssion establishment to proceed		
	(if applicable).					·		
	In addition, if BICC is performi							
	Outgoing bearer set-up proceed							
	bearer path shall be connecte							
	and the I-IWU determines (thr					RFC 3312 [5]) that sufficient		
	preconditions have been met	for the	session to p	oroceed	l			
SIP parameter								
values								
ISUP parameter values								
Comments	SIP-I		SUT			ISUP		
Comments	INVITE(IAM)	→	301		→	IAM		
	` ,	7			7	CON		
	200 OK INVITE(CON) ACK	<u> </u>			_	CON		
	ACK	7	Conversa	tion				
	BYE(REL)	→	Conversa	IIIOH	→	REL		
	200 OK BYE(RLC)				7	RLC		
	200 ON BTE(RLC)	7-			~	INLO		

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	Cit 1001 / Baolo Gall/ (Coolpt C	71 (110 11	010000 1110	occago (i	<u> </u>				
criteria									
ISUP selection criteria									
Test purpose SIP parameter values	ISUP circuit is available f	JP REL uests thor re-se opropria	: ne disconr election, a ate SIP sta	nection o	f the ir RLC is	nessage, sending out an IAM nternal bearer path. When the returned to the ISUP side; SIP_FAILURE_VA with the			
ISUP parameter values	REL; cause value: CV_ISUP (PIXIT)								
Comments	SIP-I		SU	Т		ISUP			
	INVITE(IAM)	→			→	IAM			
	SIP_FAILURE_VA(REL)	←			+	REL			
	ACK	→			→	RLC			

Table 1

		test purpose TP108001
	←SIP Message SIP_FAILURE_VA	← REL Cause Indicators parameter CV_ISUP
VA_1	404 Not Found	Cause Value No. 1 ("unallocated (unassigned) number")
VA_2	500 Server Internal Error	Cause Value No. 2 ("no route to network")
VA_3	500 Server Internal Error	Cause Value No. 3 ("no route to destination")
VA_4	500 Server Internal Error	Cause Value No. 4 ("Send special information tone")
VA_5	404 Not Found	Cause Value No. 5 ("Misdialled trunk prefix")
VA_6	500 Server Internal Error	Cause Value No. 8 ("Pre-emption")
VA_7	500 Server Internal Error	Cause Value No. 9 ("Pre-emption-circuit reserved for reuse")
VA_8	486 Busy Here	Cause Value No. 17 ("user busy")
VA_9	480 Temporarily unavailable	Cause Value No. 18 ("no user responding")
VA_10	480 Temporarily unavailable	Cause Value No. 19 ("no answer from the user")
VA_11	480 Temporarily unavailable	Cause Value No. 20 ("subscriber absent")
VA_12	480 Temporarily unavailable	Cause Value No. 21 ("all rejected")
VA_13	410 Gone	Cause Value No. 22 ("number changed")
VA_14	480 Temporarily unavailable	Cause Value No. 25 ("Exchange routing error")
VA_15	502 Bad Gateway	Cause Value No. 27 ("destination out of order")
VA_16	484 Address Incomplete	Cause Value No. 28 ("invalid number format (address
_	· ·	incomplete")
VA_17	500 Server Internal Error	Cause Value No. 29 ("facility rejected")
VA_18	480 Temporarily unavailable	Cause Value No. 31 ("normal unspecified")
_		(Class default)
VA_19	486 Busy here if Diagnostics indicator includes the (CCBS indicator = CCBS possible)	Cause Value in the Class 010 (No circuit/channel available, Cause Value No. 34)
\/A 00	else 480 Temporarily unavailable 500 Server Internal Error	Course Value in the Class 040 (resource unavailable Course
VA_20	500 Server Internal Error	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38 to 47) (47 is class default)
VA_21	500 Server Internal Error	Cause Value No. 50 ("requested facility not subscribed")
VA_21	500 Server Internal Error (SIP-I only)	Cause Value No. 55 ("incoming calls barred within CUG")
VA_23	500 Server Internal Error	Cause Value No. 57 ("bearer capability not authorized")
VA_24	500 Server Internal Error	Cause Value No. 58 ("bearer capability not presently")
VA_25	500 Server Internal Error	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_26	500 Server Internal Error	Cause Value in the Class 100 (service or option not implemented Cause Value No. 65 79) (79 is class default)
VA_27	500 Server Internal Error	Cause Value No. 87 ("user not member of CUG")
VA_28	500 Server Internal Error	Cause Value No. 88 ("incompatible destination")
VA 29	500 Server Internal Error	Cause Value No. 90 ("Non-existent CUG")
VA_30	404 Not Found	Cause Value No. 91 ("invalid transit network selection")
VA_31	500 Server Internal Error	Cause Value No. 95 ("invalid message") (Class default)
VA_32	500 Server Internal Error	Cause Value No. 97 ("Message type non-existent or not implemented")
VA_33	500 Server Internal Error	Cause Value No. 99 ("information element/parameter non-existent or not implemented")
VA_34	480 Temporarily unavailable	Cause Value No. 102 ("recovery on timer expiry")
VA_35	500 Server Internal Error	Cause Value No. 103 ("Parameter non-existent or not implemented, pass on")
VA_36	500 Server Internal Error	Cause Value No. 110 ("Message with unrecognized Parameter, discarded")
VA_37	500 Server Internal Error	Cause Value No. 111 ("protocol error, unspecified") (Class default)
VA_38	480 Temporarily unavailable	Cause Value No. 127 ("interworking unspecified") (Class default)

TP108002	SIP reference: RFC 32	61 [4]		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	L_ VA (1 1/(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	on reen reason can recomplete	<u> </u>	10000111	occago (· ()				
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	Т		ISUP			
	INVITE(IAM)	→			→	IAM			
	180 Ringing(ACM)	+			+	ACM			
	SIP_FAILURE_VA(REL)	+			+	REL			
	ACK	→	•		→	RLC			

TP108004	SIP reference: RFC 32	61 [4]			I:	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 17(11)							
ISUP parameter	REL; cause value: CV_ISUP (F	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]			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, 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	→							
			Conversa	ition					
	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	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_ISU	P (PIXIT))						
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	200 OK INVITE(CON)	(←	CON				
ACK →									
			Conversa	ition	-				
	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]				Į.	SUP reference:
				•	2.191 :	2.5 [1], clause 6.11.2
TSS reference	SIP-ISUP/Basic call/Receip	ot of the R	lelease m	essage (F	REL)	
SIP selection	PICS 4/21					
criteria						
ISUP selection						
criteria						
Test purpose	message, on receipt of an	ISUP REL equests t	with cau	se value 2	23 the	nessage, sending out an IAM SUT shall: destination according the
SIP parameter values						
ISUP parameter	REL; cause value: 23					
values						
Comments	SIP-I		SU	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		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	480 Temporarily unavailable (REL)	+		+	RSC				
	ACK	→		→	RLC				

TP109002	SIP reference: RFC 3261 [4	SUP 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)	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/Autonomo	ous release a	: I-IWU				
SIP selection	PICS 3/4						
criteria							
ISUP selection							
criteria							
Test purpose	Ensure that the SUT on receip	t of subsequ	ent 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 320	61 [4]		ISUP reference: Q.1912.5 [1], clause 6.11.3					
TSS reference	SIP-ISUP/Basic call/Autonomous release at I-IWU								
SIP selection criteria									
ISUP selection criteria									
Test purpose	Ensure that the SUT in congestion on receipt of INVITE message: • sends a 480 Temporarily Unavailable message.								
SIP parameter values			J						
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 to the BICC/ISUP compatibility procedure for unknown parameters: • sends 500 Server Internal Error.						
SIP parameter values							
ISUP parameter values	Unknown parameter in ACM: Paran	neter cor	mpatibility	"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 released due to expiry of T7 within the BICC/ISUP procedures: • sends 484 Address Incomplete.						
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 releas	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							
ISUP parameter values							
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		→	IAM		
	180 Ringing(ACM)	+		+	ACM		
			T9 expiry	1			
	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/Receip	ot of the BY	E-CANC				
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 "Netv	work bey	ond an in	iterwo	orking point"	
Comments	SIP-I		SU	T		ISUP	
	INVITE(IAM)	→			→	IAM	
	180 Ringing(ACM)	+			+	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
	ACK →						
	Conversation						
	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 BYE	-CANCEL me	essage					
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Ensure that the SUT on rece	eipt of SIP (CANCEL, the	I-IWU sh	nall send an ISUP REL with the				
	cause value # 31 to the ISU	cause value # 31 to the ISUP side.							
SIP parameter	CANCEL without encapsular	ted ISUP m	essage						
values									
ISUP parameter	REL: Cause value #31, Loca	ation "Netw	ork beyond ar	n interwo	orking point"				
values									
Comments	SIP-I		SUT		ISUP				
Ì	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 message							
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, Locat	ion "Ne	twork bey	ond an in	iterwo	rking point"		
Comments	SIP-I		SU	T		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]	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 BYE message if the SUT has already received an ACK for the 200 OK INVITE message which had it sent. 								
SIP parameter values									
ISUP parameter values									
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	←		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK	→							
		C	Conversation						
	BYE(REL)	+		+	RSC				
	200 OK BYE(RLC)	→		→	RLC				

TP111002	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	king message (CGB) with the	he indica	ation hardware failure oriented			
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose					message relating to the call has			
	already been received on re							
	 a BYE message if the S which had it sent. 	SUT has alread	/ received a	in ACK f	or the 200 OK INVITE 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	→						
		Co	nversation					
	BYE	+		+	GRS			
	200 OK BYE	→		→	GRA			

TP111003	SIP reference: RFC 3261 [4] ISUP reference: Q.1912.5 [1], clause 6.11.4							
TSS reference	SIP-ISUP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented							
CID coloction	(GRS) or Circuit group blocking	messa	age (CGB	with the	ndica	tion nardware failure oriented		
SIP selection criteria								
ISUP selection criteria								
Test purpose	Ensure that the SUT, when at least one backward ISUP message relating to the call has already been received on receipt of a CGB message, with the Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented", sends: • a BYE message if the SUT has already received an ACK for the 200 OK INVITE message which had it sent.							
SIP parameter values								
ISUP parameter values	Circuit Group Supervision Mess	age Ty	pe Indica	tor "hard	ware fa	ailure oriented"		
Comments	SIP-I		SI	JT		ISUP		
	INVITE(IAM)	→			→	IAM		
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
	ACK	→						
			Conve	sation				
	BYE	+		•	+	CGB(hardware failure)		
	200 OK BYE	→		•	→	CGBA		

TP111004	SIP reference: RFC 3	3261 [4]		Į.	SUP reference:		
			Q.19	912.5	[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 20	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		
				←	RSC		
	ACK	→		→	RLC		
	BYE(REL)	+	<u> </u>				
	200 OK BYE(RLC)	→					

TP111005	SIP reference: RFC 3	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	(GRS) or Circuit group blocking message (CGB) with the indication 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								
		receives the AC	K for the 20	10 OK	INVITE before sending the				
OID 1	BYE.								
SIP parameter									
values:									
ISUP parameter									
values:	CID I		CLIT	1	TICLID				
Comments:	SIP-I	→	SUT	→	ISUP				
	INVITE(IAM)	+		7	IAM				
	180 Ringing(ACM)				ACM				
	200 OK INVITE(ANM)	+		+	ANM				
		+			CDC				
	ACK	→		←	GRS				
	ACK			7	GRA				
	BYE	-							
	200 OK BYE	→							

TP111006	SIP reference: RFC 3	3261 [4]		I;	SUP reference:			
		Q.1912.5 [1], clause 6.11.4						
TSS reference	SIP-ISUP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message							
	(GRS) or Circuit group blockir	ng message (CG	B) with the in	ndica	tion hardware failure oriented			
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the SUT, when at already been received on received							
	Message Type Indicator code	d as "hardware	failure oriente	ed", s	sends			
	200 OK INVITE if the SUT ha	s not yet receive	ed an ACK fo	r the	200 OK INVITE:			
	 the SUT shall wait until it BYE. 	receives the AC	K for the 200	OK	INVITE before sending the			
SIP parameter								
values								
ISUP parameter values	Circuit Group Supervision Me	ssage Type Indi	cator "hardw	are fa	ailure oriented"			
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 326	61 [4]		-	SUP reference: [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 GRS message sends: • a 500 Server Internal Error on the SIP side.								
SIP parameter values									
ISUP parameter values									
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM) 180 Ringing(ACM)	→		→ ←	IAM ACM				
	500 Server Internal Error	+		+	GRS				
	ACK	→		→	GRA				

TP111009	SIP reference: RFC 326	1 [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	already been received on receipt Message Type Indicator coded a	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		SUT		ISUP					
	INVITE(IAM)	→		→	IAM					
	180 Ringing(ACM)	←		+	ACM					
	500 Server Internal Error	+		+	CGB(hardware failure)					
	ACK → CGBA									

TP111010	SIP reference: RFC 3	3261 [4]	ISUP reference:			
				[1], clauses 6.11.4 and 5		
TSS reference	SIP-ISUP/Basic call/Receipt of					
	(GRS) or Circuit group blocking	ng message (CG	B) with the indica	ation hardware failure oriented		
SIP selection						
criteria						
ISUP selection criteria						
Test purpose	Ensure that the SUT after receiving more than one INVITE sending an IAM message for each call association on receipt of a GRS message were the Range and Status Parameter value is bigger than "1": the SUT shall send a BYE requests for each call association.					
SIP parameter values						
ISUP parameter values						
Comments	SIP-I		SUT	ISUP		
	INVITE(IAM) 1	→	→	IAM		
	180 Ringing(ACM)	+	←	ACM		
	200 OK INVITE(ANM)	+	←	ANM		
	ACK	→				
	INVITE(IAM) 2	→	→	IAM		
	180 Ringing(ACM)	+	<u> </u>	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)	→	SUT	→	ISUP IAM	
	180 Ringing(ACM) 200 OK INVITE(ANM)	+		+	ACM ANM	
	ACK		Conversation			
	INFO(SUS) 200 OK INFO	← →		+	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		SUT		ISUP	
	INVITE(IAM)	→		→	IAM	
	180 Ringing(ACM)	←		+	ACM	
	200 OK INVITE(ANM)	←		+	ANM	
	ACK	←				
	Conversation					
	INFO(SUS)	+		+	SUS(network)	
	200 OK INFO	→				
			T6 is started			
			T6 is expired			
	BYE(REL)	←		→	REL	
	200 OK BYE(RLC)	←		←	RLC	

5.2.1.13 Receipt of the RESume message (RES) network initiated

TP113001	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 6.10		
TSS reference	SIP-ISUP/Basic call/					
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Ensure that the SUT, on receipt of a RESUME message containing the suspend/resume indicator set to "network initiated": the RES is transferred in an INFO message.					
SIP parameter values						
ISUP parameter	RES; Suspend indicator: 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 3261 [4] ISUP reference:						
					[1], clause A.1.1.3		
TSS reference	ISUP-SIP/ISUP Messages for special consideration/Confusion message						
SIP selection							
criteria							
ISUP selection							
criteria							
SIP parameter values ISUP 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 INFO with encapsulated CFN						
Comments	SIP-I				ISUP		
Comments	INVITE	→		→	IAM		
	183 Session Progress(CFN)	-		-	CFN		
	180 Ringing(ACM)	`		÷	ACM		
	200 OK INVITE(ANM)	÷		<u>`</u>	ANM		
	ACK	→			/ Al Al Al		
	/ COT		Communicatio	n l			
	BYE(REL)	→	Communicatio	<u>'''</u> →	REL		
	200 OK BYE(RLC)	-		-	RLC		

5.2.1.15 Segmentation

TP115001	SIP reference: RF	C 3261	[4]	ISUP reference: Q.1912.5 [1], clause A.1.1.3.1			
TSS reference	ISUP-SIP/ISUP Messages	for spec	cial conside			of a user part test message	
SIP selection criteria		•			•	1	
ISUP selection criteria							
Test purpose	Ensure that a call can be successfully completed if segmentation applies in backward direction.						
SIP parameter values	180 Ringing - encapsulated ACM: Optional Backward call indicator absent or set to "no additional information will be sent" No action takes place on the SIP side						
ISUP parameter values	ACM: optional backward comessage SGM: optional parameters		ator: additio	nal inforr	nation	will be sent in a segmentation	
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	

5.2.2 Interworking from ISUP to SIP-I

5.2.2.1 Sending of the INVITE message

TP301001	SIP reference: RFC 3261	[4]		SUP reference:				
			Q.191	2.5 [1], clause 7.1 1 a)				
TSS reference	ISUP-SIP/Basic call/Sending of the	e INVITE m	essage					
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the SUT in Idle state,	on receipt o	f an IAM messag	e containing the complete				
	called party number and the ser	called party number and the sending complete indication:						
	 sends the INVITE message with the encapsulated IAM in the MIME body. 							
SIP parameter								
values								
ISUP parameter	IAM; Called party number: with s	sending com	plete indication					
values								
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)				

TP301002	SIP reference: RFC 326	31 [4]			IS	SUP reference:		
				Q	.1912	.5 [1], clause 7.1 1 b)		
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Ensure that the SUT in Idle state, 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 con	nplete n	umber					
Comments	ISUP/BICC		SU	Т		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)		

TP301003	SIP reference: RFC 326	I [4]		15	SUP reference:				
			Q	.1912	.5 [1], clause 7.1 1 c)				
TSS reference	ISUP-SIP/Basic call/Sending of tl	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: com	olete numbe	r						
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: .5 [1], clause 7.1 1 d)
TSS reference	ISUP-SIP/Basic call/Sending of the	INVITE n	nessage		
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT in Idle state, o called party number where the en timer T _{OIW1} after the receipt of the sends the INVITE message.	d of addre	ess signallin	g is de	
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM -				
		T _{OI} \	N1 expiry		
				→	INVITE(IAM)
	ACM			+	180 Ringing(ACM)
	ANM +			+	200 OK INVITE(ANM)
				→	ACK
		Con	versation		
	REL →			→	BYE(REL)
	RLC +			←	200 OK BYE(RLC)

TP301005	SIP reference: RFC 326	1 [4]		_	SUP reference: 2.5 [1], clause 7.1 A)		
TSS reference	ISUP-SIP/Basic call/Sending of t	he INV	ITE message				
SIP selection criteria	-		-				
ISUP selection criteria	PICS 1/5						
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message with the 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]		I	SUP reference:				
	Q.1912.5 [1], clause 7.1 A)							
TSS reference	ISUP-SIP/Basic call/Sending of the INV	ITE messa	age					
SIP selection criteria	NOT PICS 4/4 AND NOT PICS 4/5 ANI	NOT PICS 4/4 AND NOT PICS 4/5 AND NOT 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				
		Conversa	ation					
	REL →		→	BYE(REL)				
	RLC ←		+	200 OK BYE(RLC)				

TP301007	SIP reference: RFC 3261 [4]		IS	SUP reference:			
	Q.1912.5 [1], clause 7.1 A)						
TSS reference	ISUP-SIP/Basic call/Sending of the INVIT	E message					
SIP selection	NOT PICS 4/4 AND NOT PICS 4/5 AND N	NOT PICS 4/15					
criteria							
ISUP selection	PICS 1/5						
criteria							
Test purpose	Ensure that the SUT in Idle state, on recei						
	party number containing the Continuity C						
	Indicators parameter which is set to "cont						
	sends the INVITE after the receipt of			ge with the Continuity			
0.0	Indicators parameter "continuity che	ck successful".					
SIP parameter							
values							
ISUP parameter							
values	laus			lain i			
Comments	ISUP	SUT		SIP-I			
	IAM →						
	COT →		<u>→</u>	INVITE(IAM)			
	ACM ←		+	180 Ringing(ACM)			
	ANM ←		+	200 OK INVITE(ANM)			
			→	ACK			
	C	Conversation					
	REL →		→	BYE(REL)			
	RLC ←		+	200 OK BYE(RLC)			

TP301008	SIP reference: RFC 326	1 [4]	[4] ISUP reference:					
			Q.1912.5 [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]	ISUP reference: Q.1912.5 [1], clause 7.1 A)					
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message							
SIP selection	NOT PICS 4/4 AND NOT PICS 4/5							
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	SUT	SIP-I				
	IAM -	→						
		T8 (expiry					
	REL	E						
	RLC -	→						

TP301010	SIP reference: RFC 3261 [4]			SUP reference: 2.5 [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 PICS	4/15					
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	5	'	,	,			
ISUP parameter values							
Comments	ISUP		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)		

TP301011	SIP reference: RFC 326	61 [4]			15	SUP reference:	
				C	2.1912	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						
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.						
OID	successful" was received	and th	e reques	sted precon	dition	s are met in the SIP network.	
SIP parameter values							
ISUP parameter							
values							
Comments	ISUP			SUT		SIP-I	
Comments	IAM	→		,01	→	INVITE(IAM)	
					+	183 Session Progress	
					→	PRACK	
					+	200 OK PRACK	
	COT(successful)	→			→	UPDATE	
	,				+	200 OK UPDATE	
			Precond	ditions met			
	ACM	+			+	180 Ringing(ACM)	
	ANM	+			4	200 OK INVITE(ANM)	
					1	ACK	
			Conve	ersation			
	REL	→			1	BYE(REL)	
	RLC	+			+	200 OK BYE(RLC)	

TP301012	SIP reference: RFC 326	61 [4]		I;	SUP reference:			
				Q.191	2.5 [1], clause 7.1 B)			
TSS reference	ISUP-SIP/Basic call/Sending of			ge				
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							
	indicator in the Nature of Conne				e IAM is set to indicate			
	"continuity check performed of							
	 sends an INVITE message 	with p	recondition u	sing the SDF	offer in the INVITE. The SDP			
	offer or answer carrying the							
					er set to "continuity check			
OID 1	successful" was received a	and th	ne requested	precondition	s are met in the SIP network.			
SIP parameter								
values								
ISUP parameter								
values	ICLID		OUT	<u> </u>	loip i			
Comments	ISUP		SUT		SIP-I			
	IAM	→		→	INVITE(IAM)			
				<u> </u>	183 Session Progress			
				→	PRACK			
	007/			<u> </u>	200 OK PRACK			
	COT(successful)	→		→	UPDATE			
					200 OK UPDATE			
	1011		Precondition		100 D: : (1011)			
	ACM	<u>+</u>		-	180 Ringing(ACM)			
	ANM	+		-	200 OK INVITE(ANM)			
				→	ACK			
			Conversa					
	REL	<u>→</u>		→	BYE(REL)			
	RLC	+		-	200 OK BYE(RLC)			

TP301013	SIP reference: RFC 326	1 [4]	0.10	ISUP reference: 12.5 [1], clause 7.1 B)				
TSS reference	ISLIP-SIP/Basic call/Sending of t	ISUP-SIP/Basic call/Sending of the INVITE message						
SIP selection criteria		PICS 4/4 AND PICS 4/5 AND PICS 4/15						
ISUP selection criteria	PICS 1/5 AND PICS 4/2							
Test purpose	The SUT in Idle state, receives an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit" and sends an INVITE message with precondition using the SDP offer in the INVITE. The Continuity message is received with the Continuity Indicators parameter set to "continuity check failed". The call has been cleared before an early dialogue has been established. Ensure that the SUT: • sends CANCEL on the SIP side.							
SIP parameter values								
ISUP parameter values								
Comments	ISUP	,	SUT	SIP-I				
	IAM	→	→	INVITE(IAM)				
			+	100 Trying				
	COT(unsuccessful)	→	→	O7 10 = =				
			+					
			+	TOT TROGUEST TOTTIMISTES				
			→	ACK				

TP301014	SIP reference: RFC 326	61 [4]		I;	SUP reference:			
				Q.191	2.5 [1], clause 7.1 B)			
TSS reference	ISUP-SIP/Basic call/Sending of	ISUP-SIP/Basic call/Sending of the INVITE message						
SIP selection	PICS 4/4 AND PICS 4/5 AND PI	ICS 4/15						
criteria								
ISUP selection	PICS 1/5 AND PICS 4/2							
criteria								
Test purpose	The SUT in Idle state, receives a							
	the Nature of Connection Indica							
	this circuit" and sends an INVI							
	INVITE. The ISUP Timer T8 exp	oires . The	call has bee	n cleared	d before an early dialogue has			
	been established. Ensure that the	ne 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]	IS	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 r	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	ISUP-SIP/Basic call/Sending of the INVITE message					
SIP selection criteria	NOT PICS 4/15						
ISUP selection criteria	PICS 1/4						
SIP parameter values ISUP parameter values	shall be received; o APM with Action indic	s delayed un with the Con ator set to "0 er control tur	til all the following tinuity Indicators Connected" - for anelling) where th	g conditions are satisfied: parameter set to "continuity" the forward bearer set-up cases the incoming Connect Type is			
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]		SUP reference:
		Q.191	2.5 [1], clause 7.1 C)
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE	message	
SIP selection	NOT PICS 4/15		
criteria			
ISUP selection	PICS 1/4		
criteria			
Test purpose	Ensure that the SUT in Idle state, on receipt	of an IAM messag	e indicating "COT to be
	expected":		
	The sending of the INVITE delays until		
	 Continuity message, with the Co 	ntinuity Indicators	parameter set to "continuity"
	shall be received;		
	 Bearer Set-up Connect indication 	n - for the backwar	d bearer set-up case was
	received.		
SIP parameter			
values			
ISUP parameter			
values			
Comments	ISUP	SUT	SIP-I
	IAM →		
	COT(successful) →	→	INVITE(IAM)
	ACM ←	+	180 Ringing(ACM)
	ANM ←	+	200 OK INVITE(ANM)
		→	ACK
	Col	nversation	
	REL →	→	BYE(REL)
	RLC ←	+	200 OK BYE(RLC)

TP301018	SIP reference: RFC 326		Q.1912.	SUP reference: 5 [1], clause 7.1 C) 2.4				
TSS reference	ISUP-SIP/Basic call/Sending of the	ISUP-SIP/Basic call/Sending of the INVITE message						
SIP selection criteria	NOT PICS 4/15							
ISUP selection criteria	PICS 1/4	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(AN							
	→ ACK							
		Conv	ersation					
	REL	→	→	BYE(REL)				
	RLC	+	+	200 OK BYE(RLC)				

TP301019	SIP reference: RFC 326	61 [4]		-	ISUP reference: 2.5 [1], clause 7.1 C)		
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message						
SIP selection criteria	NOT PICS 4/15						
ISUP selection criteria	PICS 1/4	PICS 1/4					
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected": • sends not the INVITE if the Continuity message was not received, i.e. the BICC timer T8 expires: • send REL with Cause Value 41 (temporary failure) 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]			SUP reference: 2.5 [1], clause 7.1 D)			
TSS reference	ISUP-SIP/Basic call/Sending of the IN	ISUP-SIP/Basic call/Sending of the INVITE message					
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND 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" 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	SUT		SIP-I			
	IAM →		→	INVITE(IAM)			
			+	183 Session Progress			
			→	PRACK			
			+	200 OK PRACK			
	COT(successful)		→	UPDATE			
	,		+	200 OK UPDATE			
		Precondition	s met				
	ACM +		+	180 Ringing(ACM)			
	ANM +		←	200 OK INVITE(ANM)			
			→	ACK			
		Conversat	ion				
	REL →		→	BYE(REL)			
	RLC (+	200 OK BYE(RLC)			

TP301021	SIP reference: RFC 3261 [4]			ISUP reference:			
				2.5 [1], clause 7.1 D) 2.2			
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	DIGG 4/4 AND DIGG 4/9						
ISUP selection	PICS 1/4 AND PICS 4/2						
criteria	E di di OliF: Illi di		1004				
SIP parameter values	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; • 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.						
ISUP parameter values							
Comments	ISUP/BICC	SL	JT	SIP-I			
	IAM →		→	INVITE(IAM)			
			←	183 Session Progress			
			→	PRACK			
			+	200 OK PRACK			
	COT(successful) →		→	UPDATE			
			+	200 OK UPDATE			
		Precondit	tions met				
	ACM ←		+	10011119(110111)			
	ANM ←		+	200 OK INVITE(ANM)			
			→	ACK			
		Conve	rsation				
	REL →		→	BYE(REL)			
	RLC ←		+	200 OK BYE(RLC)			

TP301022	SIP reference: RFC 3261	[4]	Q.19	ISUP reference: 012.5 [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	3 4/15				
criteria						
ISUP selection	PICS 1/4					
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 Connect indication - for the backward bearer set-up case was received.					
SIP parameter						
values						
ISUP parameter						
values						
Comments	ISUP/BICC		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	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; BNC set-up success indication for cases using bearer control tunnelling was received					
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]		15	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]			IS	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 4/1	5						
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 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 resourc	e reserv	ation was ι	unsuc	ccessful			
	REL(#47) ← CANCEL							
	RLC →			←	200 OK CANCEL			
				+	487 Request Terminated			
				→	ACK			

TP301026	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.1.1					
TSS reference	ISUP-SIP/Basic call/Sending	ISUP-SIP/Basic call/Sending of the INVITE message							
SIP selection criteria	Based on table 6		g						
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_VALUI								
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 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.1.1			
TSS reference	ISUP-SIP/Basic call/Sending	g 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	m= line			b= line	a= line				
	TMR codes	<media></media>	<transport></transport>	<fmt-list></fmt-list>	<modifier>:<bandw idth-value=""></bandw></modifier>	rtpmap: <dynamic-pt> <encoding name="">/<clock rate="">[/encoding parameters></clock></encoding></dynamic-pt>				
VA_01	"speech"	Audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)				
	"speech"	Audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000				
VA_02	"3,1 KHz audio"	Audio	RTP/AVP	0 and/or 8	AS:64	rtpmap:0 PCMU/8000 and/or rtpmap:8 PCMA/8000				
VA_03	"64 kbit/s unrestricted"	Audio	RTP/AVP	9	AS:64	rtpmap:9 G722/8000				
	"64 kbit/s unrestricted"	Audio	RTP/AVP	Dynamic PT	AS:64	rtpmap: <dynamic-pt> CLEARMODE/8000</dynamic-pt>				

Table 7

				Values for te	est purpo	ses TP3010	27			
VA		ISI	JP		SDP - a_b_m_LINE_VALUE					
		USI parameter	HLC IE in ATP		m= line		m= line		a= line	
	TMR	Information Transfer 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)	
VA_02	"speech"	"Speech"	"G.711 A-law"	Ignore	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000	
VA_03	"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_04	"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_05	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 A-law"		audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000	
VA_06	"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_07	"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_08	"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_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	"64 kbit/s unrestricted"	"Unrestricted digital inf. W/tone/ann."	N/A	Ignore	audio	RTP/AVP	9	AS:64	Rtpmap:9 G722/8000	
VA_11	"64 kbit/s unrestricted"	"Unrestricted digital information"	N/A	Ignore	Audio	RTP/AVP	Dynamic PT	AS:64	rtpmap: <dynamic-pt> CLEARMODE/8000</dynamic-pt>	

TP301027A	SIP reference: RFC 32	61 [4]	Q.19	ISUP reference: 12.5 [1], clause 7.1.2 164 [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 changir in the USI from A-law to μ-law outgoing IWU			
SIP parameter values	INVITE: IAM USI User informat	ion layer 1 pro	tocol=µ-Law	
ISUP parameter values	IAM: USI User information laye	r 1 protocol=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 INVITE	message					
SIP selection criteria								
ISUP selection criteria								
Test purpose		Ensure that the SUT is changing the received User information layer 1 protocol contained in the USI from μ -law to A -law if transcoding is performed from μ -law to A -law in the outgoing IWU						
SIP parameter values	INVITE: IAM USI User informati	on layer 1 p	rotocol= A-L	.aw				
ISUP parameter values	IAM: USI User information layer	1 protocol=	: μ-Law					
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)			

TP301028	SIP reference: RF	C 3261 [4	1	-	SUP reference: 2.5 [1], clause 7.1.2					
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message									
SIP selection		, , , , , , , , , , , , , , , , , , ,								
criteria										
ISUP selection										
criteria										
Test purpose	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 32	61 [4]	-	SUP reference:						
	Q.1912.5 [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 paramete 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=phor			•						
ISUP parameter values										
Comments	ISUP/BICC	S	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)						

TP301030	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.1.2					
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message								
SIP selection criteria	NOT PICS 1/9								
ISUP selection criteria									
Test purpose	Ensure that the SUT is mapped Called Party Number parameter to the addr-spec compo	eter of th	ne IAM and	d the and	the fo				
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 3261 [4]	-	SUP reference: 2.5 [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=phone							
ISUP parameter values								
Comments	IAM → SAM →	UT -	SIP-I					
	SAM → INVITE(IAM) ACM ← 180 Ringing(ACM) ANM ← 200 OK INVITE(ANM)							
	→ ACK Conversation							
	REL → RLC ←	→	BYE(REL) 200 OK BYE(RLC)					

TP301032	SIP reference: F	RFC 3261 [4]			ISUP reference: 12.5 [1], clause 7.1.4				
TSS reference	ISUP-SIP/Basic call/Sen	ding of the Initial	Address M	essage	(IAM)				
SIP selection criteria		-							
ISUP selection criteria	PICS 4/3	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				
		Co	nversation						
	REL	→		→	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP301033	SIP reference: RF	C 3261 [4]	l		-	SUP reference:		
<i>(</i>	Q.1912.5 [1], clause 7.1.2							
TSS reference	ISUP-SIP/Basic call/Sendi	ng of the II	NVIIE me	essage				
SIP selection	PICS 1/9							
criteria								
ISUP selection	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 = "International number" of the IAM to the addr-spec component of the To header field in the INVITE message. The format of the To header field is "+CC+NDC+SN": the forward address information is derived from the user info component of the INVITE Request-URI. INVITE: To:							
values	INVITE. TO							
ISUP parameter values								
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)		

TP301034	SIP reference: RFC	3261 [4]		_	SUP reference:				
TSS reference	ISUP-SIP/Basic call/Sending	Q.1912.5 [1], clause 7.1.2 ISUP-SIP/Basic call/Sending of the INVITE message							
SIP selection criteria	PICS 1/9	,							
ISUP selection criteria	NOT PICS 1/8	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, param	eters							
ISUP parameter values									
Comments	ISUP/BICC		SUT		SIP-I				
	IAM	→		→	INVITE(IAM)				
	ACM	-		+	180 Ringing(ACM)				
	ANM	-		+	200 OK INVITE(ANM)				
				→	ACK				
		Con	versation						
	REL	→		→	BYE(REL)				
	RLC	-		+	200 OK BYE(RLC)				

TP301035	SIP reference: RFC 32	:61 [4]		ISUP reference: 12.5 [1], clause 7.1.2					
TSS reference	ISUP-SIP/Basic call/Sending of	ISUP-SIP/Basic call/Sending of the INVITE message							
SIP selection	PICS 1/9								
criteria									
ISUP selection	PICS 1/8								
criteria									
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, Nature of address = "International number" of the IAM and the and the followed SAM: • to the addr-spec component of the To header field ; • the format of the To header field is "+CC+NDC+SN"; • the forward address information is derived from the user info component of the INVITE Request-URI.								
SIP parameter values	INVITE: To: URI, tag, paramete	ers							
ISUP parameter values									
Comments	ISUP/BICC	SU	JT	SIP-I					
	IAM	→							
	SAM	→							
	SAM	→	→	INVITE(IAM)					
	ACM	←	+	180 Ringing(ACM)					
	ANM	←	+	200 OK INVITE(ANM)					
			→	ACK					
		Conver	sation						
	REL	→	→	BYE(REL)					
	RLC	+	+	200 OK BYE(RLC)					

TP301036	SIP reference: RFC 3261 [4]			_	SUP reference:			
				Q.191	12.5 [1], clause 7.1.2			
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message							
SIP selection	PICS 1/9							
criteria								
ISUP selection	NOT PICS 1/8							
criteria								
Test purpose SIP parameter values	Ensure that the SUT is mapping the Called Called Party Number parameter, Nature of the IAM and the followed SAM: to the addr-spec component of the To the format of the To header field is "+ the forward address information is de Request-URI. INVITE: To:	of ad o hea -CC+	Idress = ader field NDC+SN	"Natio d; N";	onal (significant) number"			
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			
	Co	nvers	sation					
	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: R	FC 3261 [4]			SUP reference: 12.5 [1], clause 7.2
TSS reference	ISUP-SIP/Basic call/Rece	eipt of SAM aft	er INVITE has	been s	ent
SIP selection criteria	PICS 3/1				
ISUP selection criteria					
Test purpose	Ensure if the SUT is supp SAMs received after the				ne SIP network, subsequent d.
SIP parameter					
values					
ISUP parameter	SAM; subsequent numb	oer (PIXIT)			
values	_				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	SAM	→			
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
				→	ACK
		(Conversation	•	
	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 invite has been 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]			Į;	SUP reference:		
					Q.191	2.5 [1], clause 7.2.1		
TSS reference		ISUP-SIP/Basic call/Receipt of SAM after invite has been sent						
SIP selection	PICS 3/2 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". 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		SL	ΙΤ		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		
			Conver	sation				
	REL	→	0001		→	BYE(REL)		
	RLC	+			+	200 OK BYE(RLC)		

TP302004	SIP reference: RFC 326	1 [4]		ı	SUP reference:				
				Q.191	12.5 [1], clause 7.2.1				
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent								
SIP selection	PICS 3/2 AND 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 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 INVITE(ANM)				
		-		→	ACK				
		<u> </u>							
1	Conversation								
	REL	→	Conversation	→	BYE(REL)				

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

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

TP302007	SIP refe	erence	: RFC 3261	[4]	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 3/2 / (11)	100 0/27/1105 1/07/1105 1/10								
ISUP selection	PICS 1/5 AND	PICS 4	/2							
criteria	1.100 1/07 1112		-							
Test purpose	Check indicator check required Sends an INVI Indicators paral preconditions a On receipt of a 1) Stop timer 2) TOIW2 shal a) the Rec receive b) a new I INVITE c) the new that hav resource in ques	that the SUT in Idle state, on receipt of an IAM message containing the Continuity indicator in the Nature of Connection Indicators parameter which is set "continuity required on this circuit". an INVITE message after the reception of the Continuity message with the Continuity ors parameter set to "continuity check successful" and after the requested ditions are met in the SIP network. eipt of a SAM from the ISUP the SUT shall: p timer TOIW3 (if it is running); IW2 shall be restarted and the SUT shall invoke the following procedures: the Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call; a new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent; the new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question; all other contents of the new INVITE are interworked from the parameters of the								
SIP parameter			contains o	ligits	from the	e IAM and digits from SAM x and SAM y. The				
values ISUP parameter	IAM is also con	tained								
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/Receipt of SAM after invite has been sent							
SIP selection criteria	PICS 3/2 AND PICS 4/5 AND PICS 4/15							
ISUP selection	PICS 1/5 AND PICS 4/2							
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-I			
	IAM	→		→	INVITE1(IAM)			
	SAM x	→		_	400 Coories Drawnoo with set as a set of ACM			
	COT			<u>+</u>	183 Session Progress without encapsulated ACM			
	СОТ	→	+	<u>→</u>	UPDATE			
	CAM		+	-	200 OK UPDATE			
	SAM	→		}	INVITE2 (IAM and digits from SAM X + SAM Y)			
		-			484 Address Incomplete (1)			
	A ON 4	 		}	ACK			
	ACM	+		<u>+</u>	180 Ringing2 (ACM)			
	ANM	+		(200 OK INVITE(ANM)			
		-		→	ACK			
			Conversation					
	REL	→		→	BYE(REL)			
	RLC	+		←	200 OK BYE(RLC)			

TP302009	SIP reference: RFC 3261 [4] ISUP reference:									
		1912.5 [1], clause 7.2.1								
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent									
SIP selection	PICS 3/2 AND NOT PICS 4/15									
criteria										
ISUP selection	PICS 1/4 AND NOT 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 until all the following conditions are satisfied:									
	 Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received; 									
	 Bearer Set-up indication - for the Type is "notification not required" 			case where the incoming Connect						
	On receipt of a SAM from the BICC th	ie SUT sł								
	 Stop timer TOIW3 (if it is running); 									
	2) TOIW2 shall be restarted and the									
	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;	O-11 ID		landinahadinahan) andha massida						
	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	S	UT	SIP-I						
	IAM →									
	SAM x →									
	COT → INVITE(IAM)									
	SAM y →		-							
	ACM ←		•	180 Ringing(ACM)						
	ANM ← 200 OK INVITE(ANM)									
				→ ACK						
		Conve	ersation							
	REL →			▶ BYE(REL)						
	RLC +		•	200 OK BYE(RLC)						

TP302010	SIP reference: RFC 326	SUP reference:								
			12.5 [1], clause 7.2.1							
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent									
SIP selection	PICS 3/2 AND 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 until all the following conditions are satisfied:									
	Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received;									
	·	et to "Coi	nnected" - for th	ne forw	ard bearer set-up cases (with,					
	or without bearer control tun									
	required", and for the fast se			9						
	On receipt of a SAM from the BIG									
	1) Stop timer TOIW3 (if it is runn									
	2) TOIW2 shall be restarted and									
	 a) the Request-URI and the 		der field of the r	new IN'	VITE 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	Original IAWI.									
values										
ISUP parameter	1									
values										
Comments	ISUP/BICC		SUT		SIP-I					
	IAM	→								
	SAM x	→								
	COT	→		→	INVITE(IAM)					
	SAM y	→	<u> </u>	→	INVITE(IAM)					
	ACM ← 180 Ringing(ACM)									
	ANM	+		+	200 OK INVITE(ANM)					
				→	ACK					
			Conversation							
	REL	→		→	BYE(REL)					
	RLC	-		+	200 OK BYE(RLC)					

TP302011	SIP reference: RFC 3261 [4] ISUP reference:								
					1912.5 [1], clause 7.2.1				
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent								
SIP selection	PICS 3/2 AND NOT PICS 4/15								
criteria									
ISUP selection criteria	PICS 1/4 AND PICS 4/2								
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be								
rest purpose	expected".								
	The sending of the INVITE delays until all the following conditions are satisfied:								
					meter set to "continuity" shall be				
	received;								
	On receipt of a SAM from the BI				arer set-up case was received.				
	1) Stop timer TOIW3 (if it is run								
	2) TOIW2 shall be restarted an		UT shall ir	nvoke the fe	ollowing procedures:				
	a) the Request-URI and the	e To he	ader field	of the new	INVITE shall contain all digits				
	received so far for this c								
	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	Original II twi.								
values									
ISUP parameter									
values									
Comments	ISUP/BICC		SU	Γ	SIP-I				
	IAM	→							
	SAM x	→							
	COT → INVITE(IAM)								
	SAM y	→		-	→ INVITE(IAM)				
	ACM	+		•	180 Ringing(ACM)				
	ANM ← 200 OK INVITE(ANM)								
					→ ACK				
			Convers	ation					
	REL	→			▶ BYE(REL)				
	RLC	+		•	200 OK BYE(RLC)				

TP302012	SIP reference: RFC 3261 [4] ISUP reference:								
		Q.191	l 2.5 [1], clause 7.2.1						
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent								
SIP selection	PICS 3/2 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 delays until all the following conditions are satisfied:								
	 Continuity message, with the received; 	Continuity message, with the Continuity Indicators parameter set to "continuity" shall be							
	 BNC set-up success indicated on receipt of a SAM from the BI 			bearer cont	trol tunnelling was received				
	1) Stop timer TOIW3 (if it is run		or oriali.						
	2) TOIW2 shall be restarted an	d the SU	T shall invo	ke the follo	wing procedures:				
			der field of	the new IN'	VITE shall contain all digits				
	received so far for this c				<i></i>				
	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					T				
Comments	ISUP/BICC	<u> </u>	SUT		SIP-I				
	IAM	→							
	SAM x	→							
	COT	→		→	INVITE(IAM)				
	SAM y → INVITE(IAM)								
	ACM	+		-	180 Ringing(ACM)				
	ANM	+		←	200 OK INVITE(ANM)				
			0		ACK				
	DEL		Conversation		DVE(DEL)				
	REL	→)	BYE(REL)				
	RLC			←	200 OK BYE(RLC)				

TP302013	SIP refe	erence: RFC 3261 [4] ISUP reference: Q.1912.5 [1], clause 7.2.1						
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent							
SIP selection	PICS 3/2 AND PICS 4/5 AND PICS 4/15							
criteria								
ISUP selection	PICS 1/4 AND PICS 4/2							
criteria								
Test purpose	 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 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				fror	om the IAM and digits from SAM x and SAM y. The			
values	IAM is also con	taine	<u>d</u>					
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 IAM		SUT	→	SIP-I			
		→		7	INVITE1(IAM)			
	SAM x	7		+	183 Session Progress without encapsulated ACM			
	СОТ	→		<u>▼</u>				
	COT	7		7				
	SAM y	→		<u>~</u>				
	SAIVI Y	7		7	,			
				<u>▼</u>				
	ACM	+		7				
	ANM	+		-				
	CINIVI	_		<u>∑</u>				
		 	Conversation		AON			
	REL							
	RLC	-		7				
	KLC T ZUU ÜK BYE(KLC)							

TP302014	SIP refere	ence	: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.2.1				
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent									
SIP selection	PICS 3/2 AND PICS 4/5 AND PICS 4/15									
criteria										
ISUP selection	PICS 1/4 AND PICS 4/2									
criteria	<u> </u>									
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected".									
		me	ssage. The eve	nte:						
	Sends the INVITE message. The events: Continuity message, with the Continuity Indicators parameter set to "continuity" was 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; are indicating the successful completion of bearer set-up.									
	On receipt of a SA 1) Stop timer TO	IW3	(if it is running)	,						
	a) the Reque	est-L				I invoke the following procedures: Id of the new INVITE shall contain all digits				
			,	Call	ID or	d From boader (including tog) on the provious				
		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	INVITE2: Request URI contains digits from the IAM and digits from SAM x and SAM y. The									
values	IAM is also contain		J			· ·				
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									
						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					
	ISUP/BICC		SUT		SIP					
	IAM	<u>→</u>		→	INV	ITE1(IAM)				
	SAM	→			400	0				
	007			<u> </u>		Session Progress without encapsulated ACM				
	COT	→		<u>→</u>		DATE				
	0.4.14			<u>+</u>		OK UPDATE				
	SAM	→		<u>→</u>		ITE2 (IAM with digits from SAM X + SAM Y)				
				-		Address Incomplete (1)				
	ACM			<u>→</u>	ACŁ					
	ACM	+		<u>+</u>		Ringing2(ACM)				
	ANM		 	<u>~</u>		OK INVITE(ANM)				
			Conversation	7	ACŁ	\				
	Conversation REL → BYE(REL)									
	REL	<u>→</u>		<u>→</u>						
	RLC	~		~	∠00	OK BYE(RLC)				

TP302015	SIP refere	ence	: RFC 3261 [4]		ISUP reference: Q.1912.5 [1], clause 7.2.1			
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent							
SIP selection	PICS 3/2 AND PICS 4/5 AND PICS 4/15							
criteria								
ISUP selection	PICS 1/4 AND PI	CS 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 Connect indication - for the backward bearer set-up case 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 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 							
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>	\ \ \ \ \ \ \ \ \ \ \ \ \ \ \ \ \ \ \			
	A CM			<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 refere	SIP reference: 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 PICS 4/5 AND PICS 4/15							
criteria								
ISUP selection	PICS 1/4 AND PI	CS 4	/2					
criteria								
Test purpose	 Ensure that the SUT in Idle state, on receipt of an IAM message containing indicating "COT to be expected". Sends the INVITE message. The events: Continuity message, with the Continuity Indicators parameter set to "continuity" was received; BNC set-up success indication for cases using bearer control tunnelling was received. are indicating the successful completion of bearer set-up, On receipt of a SAM from the BICC/ISUP the SUT shall: Stop timer TOIW3 (if it is running); TOIW2 shall be restarted and the SUT shall invoke the following procedures: the Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call; a new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent; the new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question; all other contents of the new INVITE are interworked from the parameters of the 							
	original IA	M.				·		
SIP parameter			I contains digits	s fror	n the IA	M and digits from SAM x and SAM y. The		
values	IAM is also conta	ned						
ISUP parameter								
values					<u> </u>			
Comments	INVITE. The prec exchange) the co the confirmation of message are sati	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	-4 (LABA)		
	IAM	<u>→</u>		→	IINVIII	E1(IAM)		
	SAM	→			400.0	Described Described without an arrandated A OA4		
	COT			(ession Progress without encapsulated ACM		
	СОТ	→		→	UPDA			
	CANA			/		K UPDATE		
	SAM	<u>→</u>		→		E2 (IAM with digits from SAM X + SAM Y)		
				+		ddress Incomplete (1)		
	ACM			→	ACK	in sin sig(A CNA)		
	ACM	<u>←</u>		+	180 KI	nging2(ACM)		
	ANM	_		/		K INVITE(ANM)		
			Conversation	→	ACK			
	DEL		Conversation	_	DVC/5			
	REL	<u>→</u>		→	BYE(R			
	RLC	←		←	200 O	K BYE(RLC)		

TP302017	SIP reference: RFC 3261 [4]		ISUP reference: 12.5 [1], clause 7.2.1		
TSS reference	ISUP-SIP/Basic call/Receipt of SAM	after invit	e has been sent	t		
SIP selection	PICS 3/2					
criteria						
ISUP selection	PICS 1/4					
criteria						
Test purpose	The SUT in Idle state, on receipt of an IAM message sends a INVITE message. On receipt of a SAM from the BICC/ISUP the SUT shall: 1) Stop timer TOIW3 (if it is running); 2) TOIW2 shall be restarted and the SUT shall invoke the following procedures: Ensure that if timer TOIW2 has expired, subsequent SAMs received after the SUT has sent the INVITE are ignored.					
SIP parameter						
values						
ISUP parameter						
values						
Comments	ISUP/BICC	3	SUT	SIP-I		
	IAM →		→	INVITE(IAM)		
	SAM →		→	INVITE(IAM)		
		T _{oiw2}	expired			
	SAM →					
	ACM		←	180 Ringing(ACM)		
	ANM +		←	200 OK INVITE(ANM)		
			→	ACK		
		Conv	ersation			
	REL →		→	BYE(REL)		
	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 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	SUT	SIP-I					
	IAM →							
	SAM →	→	INVITE(IAM)					
	T _{oiw}	₂ expired						
	SAM →							
	ACM ←	+	180 Ringing(ACM)					
	ANM ←	+	200 OK INVITE(ANM)					
		→	ACK					
	Con	versation						
	REL →	→	BYE(REL)					
	RLC +	+	200 OK BYE(RLC)					

5.2.2.3 Sending of the ACM message

TP303001	SIP reference: RFC 326		ISUP reference: Q.1912.5 [1], clause 7.1, Q.764 [i.12], clause 2.1.4.8					
TSS reference	ISUP-SIP/Basic call/Sending of	the A	CM message					
SIP selection criteria	PICS 1/3							
ISUP selection criteria	PICS 4/9							
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number and the sending complete indication. 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		SÚT		SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM(no indication)	+			, ,			
	CPG(Alerting)	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
				→	ACK			
			Conversation	•				
	REL	→	·	→	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP303002	SIP reference: RFC 3261	[4]	ISUP reference: Q.1912.5 [1], clause 7.1, Q.764 [i.12], clause 2.1.4.8						
TSS reference	ISUP-SIP/Basic call/Sending of the ACM message								
SIP selection	PICS 1/3								
criteria ISUP selection	DIOC 4/0								
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	• the ISDN access indicator s	et to 15D	IN_ACC_IND_V	AL .					
values									
ISUP parameter values	(10) interworking indicator: INT_IND ISUP indicator: ISUP_IND_ID (P ISDN access indicator ISDN_AC	et to the v or: no indi o_VAL (Pl: IXIT)	alue in the enca cation(00) or ord	apsulated ACM CPS indicator no dinary subscriber (01) or payphone					
Comments	ISUP/BICC		SUT	SIP-I					
		→		→ INVITE(IAM)					
	ACM(no indication)	(
	CPG(Alerting)	(← 180 Ringing(ACM)					
	ANM	(← 200 OK INVITE(ANM)					
				→ ACK					
		Co	nversation						
	REL	→		→ BYE(REL)					
	RLC	(← 200 OK BYE(RLC)					

TP303003	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause 7.1, Q.764 [i.12], clause 2.1.4.8					
TSS reference	ISUP-SIP/Basic call/Sending of the	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	IAM; Called party number: comple							
values	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		SÚT Í	SIP-I				
	IAM 3		→	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 3261	[4]		SUP reference:					
				, clauses 7.1 1) d), 7.3.1, 7.4					
TSS reference	Ŭ	ISUP-SIP/Basic call/Sending of the ACM message							
SIP selection	PICS 1/3								
criteria									
ISUP selection	NOT PICS 4/9								
criteria —									
Test purpose	Ensure that the SUT in Idle state, 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 ACM message with party's category indicator s "payphone (10)", the interwo 	50.1d5 ii.6 ii.77.1 = 1.15554g5 ii.6 54.154							
SIP parameter	,								
values									
ISUP parameter	IAM; Called party number: comp		r						
values	ACM, CPS indicator no indication								
	Called party's category indicato	r: no indica	tion(00) or ordina	ry subscriber (01) or payphone					
	(10)	\/AL (DI)/I	- '\						
	interworking indicator: INT_IND ISUP indicator: ISUP_IND_ID (P		1)						
	ISDN access indicator ISDN_AC		I (DIVIT)						
Comments	ISUP/BICC		SUT	SIP-I					
		→	301						
		=	1 expiry						
	ACM(no indication)	(-	→	INVITE(IAM)					
	CPG(Alerting)	£	+	180 Ringing(ACM)					
	ANM	f-	+	200 OK INVITE(ANM)					
			→	ACK					
		Conv	ersation						
		>	→	BYE(REL)					
	RLC •	E	+	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	ne 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 interwo set to "ISUP_IND_ID", the IS			IND_VAL", the ISUP indicator "ISDN_ACC_IND_VAL".		
SIP parameter values	, , , , , , , , , , , , , , , , , , , ,					
ISUP parameter	IAM; Called party number: comp	olete numbe	r			
values	ACM, CPS indicator no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: INT_IND_VAL (PIXIT) ISUP indicator: ISUP_IND_ID (PIXIT) ISDN access indicator ISDN_ACC_IND_VAL (PIXIT)					
Comments	ISUP/BICC		SUT	SIP-I		
	-	→				
	SAM	→				
		→	→	INVITE(IAM)		
		T _{OIM}	_{/2} expiry	,		
	ACM(no indication)	(
		(+	180 Ringing(ACM)		
		(+	200 OK INVITE(ANM)		
			→	ACK		
		Conv	ersation			
	REL	→	→	BYE(REL)		
	RLC	←	+	200 OK BYE(RLC)		

TP303006	SIP reference: RFC 3261 [4]	0.404		SUP reference:			
TSS reference	Q.1912.5 [1], clauses 7.1 1) a), 7.3.1 ISUP-SIP/Basic call/Sending of the ACM message							
SIP selection criteria	PICS 1/3							
ISUP selection criteria	NOT PICS 4/9							
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number, on receipt of a 180 Ringing message: Sends the ACM message with: the CPS indicator set to the value in the encapsulated ACM; the Called party's category indicator set to the value in the encapsulated ACM; the interworking indicator set to the value in the encapsulated ACM; the ISUP indicator set to the value in the encapsulated ACM; the ISDN access indicator set to the value in the encapsulated ACM.							
SIP parameter values								
ISUP parameter	IAM; Called party number: comple	ete number						
values	ACM, Backward call indicator is set	to the valu	ue in the end	capsul	ated ACM			
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)			

TP303007	SIP reference: RFC 3	261 [4]	Q.1		SUP reference: 1], clauses 7.1 1 a), 7.3.2			
TSS reference	ISUP-SIP/Basic call/Sending of the ACM message							
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: c	omplete	number					
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)			

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/S	endin	g of the II	VIT	E message				
SIP selection criteria	PICS 1/3	PICS 1/3							
ISUP selection criteria	PICS 4/2 AND NOT P	ICS 4	/9						
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by the expiration timer T_{OIM1} 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	360 16 16 G1 _IIVB	Set to 1991 _IND_ID , the ISBN access mulcator Set to 1991v_AGO_IND_VAE .							
ISUP parameter	IAM; Called party nui	mber:	complete	e nur	mber				
values	ACM,		оор.о						
	CPS indicator no indi	ication	n (00)						
				no inc	dication(00) or ordinary subscriber (01) or payphone				
	(10)interworking indi	icator	: INT_INI	D_VA	AL (PIXIT)				
	ISUP indicator: ISUP								
	ISDN access indicate	or ISD	N_ACC_	IND_					
Comments	ISUP/BICC		SUT		SIP-I				
	IAM	→		→	INVITE(IAM)				
				←	183 Session Progress without encapsulated ACM				
	COT	→		→	UPDATE				
				+	200 OK UPDATE				
		T	_{DIW1} expir	'n					
	ACM(no indication)	+							
	CPG(Alerting, BCi)	+		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
				→	ACK				
		Co	onversation	n					
	REL	→		→	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP303012	SIP reference	RFC	3261 [4]		ISUP reference:		
					Q.1912.5 [1], clauses 7.1, 7.3.1, 7.4		
TSS reference		ISUP-SIP/Basic call/Sending of the INVITE message					
SIP selection criteria	PICS 1/3 AND PICS 3	/2 AN	D PICS 4	/5 A	ND PICS 4/4 AND PICS 4/15		
ISUP selection criteria	PICS 4/2 AND NOT P	CS 4	/9				
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 to "ISUP_IND_ID", the ISDN access indicator set to "ISDN_ACC_IND_VAL".						
SIP parameter							
values	1011 5 1 1 11						
ISUP parameter values	ACM, Backward call in						
values	Called party's of payphone (10) interworking indicator: ISUP indicator: ISDN access in	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	ISUP/BICC		SUT		SIP-I		
	IAM	→		→	INVITE(IAM)		
				←	183 Session Progress without encapsulated ACM		
	COT	→		→	UPDATE		
				+	200 OK UPDATE		
		T _C	_{DIW2} expir	У			
	ACM(no indication)	+					
	CPG(Alerting, BCi)	+		←	180 Ringing(ACM)		
	ANM	+		←	200 OK INVITE(ANM)		
				→	ACK		
		Co	nversatio	n			
	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 of	all/S	ending of the A	CM ı	
SIP selection	PICS 1/3				-
criteria					
ISUP selection	PICS 4/2 AND NO	OT P	CS 4/9		
criteria					
Test purpose					pt of an IAM message containing the complete
					k is performed (ISUP) or COT is expected (BICC)
	indication receipt			essa	ge:
			nessage with:		
					in the encapsulated ACM;
					ator set to the value in the encapsulated ACM;
					the value in the encapsulated ACM;
					e in the encapsulated ACM;
OID	o the ISDN a	acce	ss indicator se	et to 1	the value in the encapsulated ACM.
SIP parameter					
values	IAM Oallad mart				- Land
ISUP parameter values	IAM; Called party				
Comments	ISUP/BICC	all ir		tne	value in the encapsulated ACM
Comments		→	SUT	→	SIP-I
	IAM	7			INVITE(IAM)
	OOT	→		<u>←</u>	183 Session Progress without encapsulated ACM
	СОТ	7		7	UPDATE
	A O N 4				200 OK UPDATE
	ACM	<u>+</u>		(180 Ringing(ACM)
	ANM	+		<u>←</u>	200 OK INVITE(ANM)
			Convergation	7	ACK
	DEL	_	Conversation	_	DVE(DEL)
	REL	}		}	BYE(REL)
	RLC	-		-	200 OK BYE(RLC)

TP303014	SIP reference: RFC 326	1 [4]		ISUP reference:			
				.5 [1], clauses 7.1, 7.3.1, 7.4			
TSS reference	ISUP-SIP/Basic call/Sending of the	he INVITE	nessage				
SIP selection criteria	PICS 1/3 AND PICS 3/2 AND NO	OT PICS 4/	5				
ISUP selection	PICS 3/8 AND PICS 4/2 AND NO	OT 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 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} :						
	party's category indicator se "payphone (10)", the interwo	 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	Called party's category inc payphone (10) interworking indicator: INT ISUP indicator: ISUP_IND ISDN access indicator: IS	CPS indicator: no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or					
	180 Ringing						
Comments	ISUP/BICC		SUT	SIP-I			
	IAM	→					
	COT	→	-	► INVITE(IAM)			
		To	_{W2} expiry				
	ACM(no indication)	+					
	CPG(Alerting)	+	•	180 Ringing(ACM)			
	ANM	(•	200 OK INVITE(ANM)			
			-	→ ACK			
		Cor	nversation				
	REL	→		▶ BYE(REL)			
	RLC	-		200 OK BYE(RLC)			

TP303015	SIP reference: RFC 3261 [4]			SUP reference:			
			912.5 [1	1], clauses 7.1, 7.3.1; 7.4			
TSS reference	ISUP-SIP/Basic call/Sending of the ACM message						
SIP selection	PICS 1/3 AND NOT PICS 4/15						
criteria							
ISUP selection	PICS 4/2 AND NOT PICS 4/9						
criteria							
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number, the continuity check is performed (ISUP) or COT is expected (BICC) indication receipt of a 180 Ringing message: Sends the ACM message with: the CPS indicator set to the value in the encapsulated ACM; the Called party's category indicator set to the value in the encapsulated ACM; the interworking indicator set to the value in the encapsulated ACM; the ISUP indicator set to the value in the encapsulated ACM; the ISDN access indicator set to the value in the encapsulated ACM.						
SIP parameter values							
ISUP parameter	IAM; Called party number: complete n	umber					
values	ACM, Backward call indicator is set to the	ne value in the e	ncapsu	lated ACM			
Comments	ISUP/BICC	SUT		SIP-I			
	IAM →						
	COT →		→	INVITE(IAM)			
	ACM ←		←	180 Ringing(ACM)			
	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 t	he CPG mes	sage				
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	icator = no inc	dication				
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 320	61 [4]		IS	SUP reference:		
	Q.1912.5 [1], clauses 7.1, 7.3.1						
TSS reference	ISUP-SIP/Basic call/Sending of	the CPG m	essage				
SIP selection criteria							
ISUP selection criteria							
Test purpose	receipt of a 183 Session progres	Ensure that the SUT, having sent a ACM message with called party status "no indication" on ecceipt 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						
		Co	nversation	•			
	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 An	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 A		ge, on recei _l	pt of a 200 OK INVITE for this				
	call, it shall stop timer TOIW2 (if runnir	0,						
	 send ANM as determined by BICC 							
	 stop any existing awaiting answer 	indication (e.g. ringing	tone).				
SIP parameter	200 OK INVITE;							
values								
ISUP parameter	ANM;							
values								
Comments	ISUP/BICC	SUT		SIP-I				
	IAM →		→	INVITE(IAM)				
	ACM ←		+	180 Ringing(ACM)				
	ANM ←		+	200 OK INVITE(ANM)				
			→	ACK				
		Conversati	on					
	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	nnect Mess	age (CON)	
SIP selection				
criteria				
ISUP selection				
criteria				
Test purpose	Ensure that the SUT, having not sent t	he ACM me	ssage, on re	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 proce	edures.	
	Stop any existing awaiting answer indi-	cation (e.g. ı	inging tone) 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 (PIX	IT)	
values	ISUP indicator: ISUP_IND_ID (PIXIT)			
	ISDN access indicator ISDN_ACC_IN	ND_VAL (PI	XIT)	
	CPS indicator: no indication			
Comments	ISUP/BICC	SUT		SIP-I
	IAM →		→	INVITE(IAM)
	CON ←		+	200 OK INVITE(CON)
			→	ACK
		Conversation	n	
	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 32	61 [4]		_	SUP reference:		
				Q.1912	2.5 [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 3261 [4]			SUP reference:				
	_		Q.1912.	5 [1], clause 7.7.1 2) 3)				
TSS reference	ISUP-SIP/Basic call/Receipt of the Re	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. for Profile C (SIP-I), if a BYE message is sent, it shall encapsulate the received REL message. 							
SIP parameter values								
ISUP parameter values								
Comments	ISUP/BICC	SU	Γ	SIP-I				
	IAM →		→	INVITE(IAM)				
			+	100 Trying				
	REL →							
	RLC ←		→	CANCEL				
			+	200 OK INVITE(CON)				
			→	ACK				
			→	BYE(REL)				
			+	200 OK BYE(RLC)				

TP307004	SIP reference: RFC 32	261 [4]		ISUP reference: .5 [1], clause 7.7.1 2) 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 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							
SIP parameter values								
ISUP parameter values								
Comments	ISUP/BICC	,	SUT	SIP-I				
	IAM	→	→	INVITE(IAM)				
	REL	→		·				
	RLC	+						
			+	100 Trying				
			→	CANCEL				
			+	200 OK CANCEL				
			+	487 Request terminated				
			→	ACK				

TP307005	SIP reference: RF0	C 3261 [4]		0.40	ISUP reference:		
	Q.1912.5 [1], clause 7.7.1 4)						
TSS reference	ISUP-SIP/Basic call/Receip	t of the Rele	ase me	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 after a 200 OK response message has been received: • the SUT shall send a BYE request. The received REL is encapsulated in the BYE.						
SIP parameter values		The second secon					
ISUP parameter values							
Comments	ISUP/BICC		SU	Γ	SIP-I		
	IAM	→		→	INVITE(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	ANM ← 200 OK INVITE(ANM)						
					1.014		
		→ ACK					
				_			
	REL	→		→	BYE(REL)		
	RLC	-		←	200 OK BYE(RLC)		

TP307006	SIP reference: RFC 326	1 [4]	_	SUP reference:			
			Q.1912	2.5 [1], clause 7.7.1 3)			
TSS reference	ISUP-SIP/Basic call/Receipt of the	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 and before dialog has been confirmed: • the SUT shall send a CANCEL request which is answered by 200 OK CANCEL and INVITE request will be terminated by 487.						
SIP parameter	•	,					
values							
ISUP parameter values							
Comments	ISUP/BICC	SU	Т	SIP-I			
	IAM =	→	→	INVITE(IAM)			
	ACM	-	+	SIP_MESSAGE_VA			
	REL =	→					
	RLC	-					
			→	CANCEL			
			+	200 OK CANCEL			
			+	487 Request terminated			
			→	ACK			

Table 8

	Values for test purpose TP307106						
VA SIP MESSAGE_VA							
VA_1	A_1 180 Ringing(ACM)						
VA_2	VA_2 181 Call Is Being Forwarded(ACM)						
VA_3	/A_3 182 Queued(ACM)						
VA_4	4 183 Session Progress(ACM)						

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

TP308001	SIP reference: RFC 3261 [4]		ISUP reference: Q.1912.5 [1], clause 7.7.2				
TSS reference	ISUP-SIP/Basic call/Sending of the Re	ISUP-SIP/Basic call/Sending of the Release message (REL)/					
SIP							
selection							
criteria							
ISUP							
selection							
criteria							
Test	Ensure that the SUT after receiving th	e IAM sends	out an INVIT	E message and on receipt of a			
purpose	BYE message in the confirmed dialog	ue:					
	 sends a REL message constructe 	ed from the en	capsulated F	REL in the received BYE.			
SIP			•				
parameter							
values							
ISUP	REL; Cause value "Normal call clearing	ng"					
parameter		· ·					
values							
Comments	ISUP/BICC	SUT		SIP-I			
	IAM -	•	→	INVITE(IAM)			
	ACM ←	•	+	180 Ringing(ACM)			
	ANM ←		+	200 OK INVITE(ANM)			
			→	ACK			
	Co	onversation	I .				
	REL (+	BYE(REL)			
	RLC →	,	→	200 OK BYE(RLC)			

TP308002	SIP reference: RFC 3	261 [4]				SUP reference:	
						2.5 [1], clause 7.7.6	
TSS reference	ISUP-SIP/Basic call/Sending of	of the R	Release m	essage (F	REL)		
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Ensure that the SUT after rece	eiving tl	he IAM se	nds out a	n INV	ITE message. On receipt of	
	a Failure message (4xx, 5xx, 6	3xx) de	fined as S	SIP_Failur	e_VA	:	
	 sends a REL message co 	nstruct	ed from the	ne encaps	sulate	d REL.	
SIP parameter							
values							
ISUP parameter	REL; cause value: CV_ISUP	REL; cause value: CV ISUP					
values							
Comments	ISUP/BICC		SU	Т		SIP-I	
	IAM → INVITE(IAM)						
					+	100 Trying	
	REL	←			+	SIP_Failure_VA(REL)	
	RLC						

Table 9

	Values for test purpose TP308002					
VA	←REL (Cause Value) CV_ ISUP	←4XX/5XX/6XX SIP message SIP_Failure_VA				
/A_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				
√A_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				
√A_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				
/A_21	127 Interworking	504 Server timeout				
/A_22	17 User busy	600 Busy Everywhere				
/A_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 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.7.6		
TSS reference	ISUP-SIP/Basic call/Sending	of the Releas	se message ((REL)/		
SIP selection criteria	NOT PICS 4/10					
ISUP selection criteria						
Test purpose	Ensure that the SUT after receiving the IAM sends out an INVITE message, on receipt of a Failure message 487 Request terminated: • no action is taken on the ISUP if a CANCEL request was previously sent before an answer to an INVITE was received.					
SIP parameter values						
ISUP parameter values						
Comments	ISUP/BICC		SUT		SIP-I	
	IAM	→		→	INVITE(IAM)	
				+	100 Trying	
	REL → CANCÉL					
	RLC ← 200 OK CANCEL					
				+	487 Request Terminated	
				→	ACK	

TP308004	SIP reference: RFC 3261 [4]				-	SUP reference: 2.5 [1], clause 7.7.6	
TSS reference	ISUP-SIP/Basic call/Sending of	of the F	Release m				
SIP selection criteria				•	•		
ISUP selection criteria							
Test purpose	message defined as SIP MES message (4xx, 5xx, 6xx) defin	Ensure that the SUT after receiving the IAM sends out an INVITE message, a SIP message defined as SIP MESSAGE_VA has been received, on receipt of a Failure message (4xx, 5xx, 6xx) defined as SIP_Failure_VA: • sends a REL message constructed from the encapsulated REL.					
SIP parameter values							
ISUP parameter values	REL; cause value: CV_ISUP	REL; cause value: CV_ISUP					
Comments	ISUP/BICC	ISUP/BICC SUT SIP-I					
	IAM	AM → INVITE(IAM)					
	ACM	CM ← SIP MESSAGE_VA(ACM)					
	REL	4			+	SIP_Failure_VA(REL)	
	RLC	→		_	→	ACK	

Table 10

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

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						SUP reference: 2.5 [1], clause 7.7.6	
TSS reference	ISUP-SIP/Basic call/Sending of	of the F	Release m			2.0 [.], 0.00000	
SIP selection criteria	NOT PICS 4/10			3 (
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	ISUP/BICC SUT SIP-I					
	IAM	IAM → INVITE(IAM)					
	ACM ← 180 Ringing						
	REL	+		•	+	SIP_Failure_VA(REL)	
	RLC	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				

TP308006	SIP reference: RFC 3	261 [4	i]			SUP reference: 2.5 [1], clause 7.7.6	
TSS reference	ISUP-SIP/Basic call/Sending of the Release message (REL)/						
SIP selection criteria	NOT PICS 4/21	NOT PICS 4/21					
ISUP selection criteria							
Test purpose	Ensure that the SUT after receiving the IAM sends out an INVITE message. On receipt of a response message (3xx) defined as SIP_Response_VA, the SUT: • sends a REL message with the Cause value CV_ISUP.						
SIP parameter values							
ISUP parameter values	REL; cause value: CV_ISUP						
Comments	ISUP/BICC		SU	Γ		SIP-I	
	IAM	→		•	→	INVITE(IAM)	
					+	100 Trying	
	REL	+			+	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.

Autonomous release at O-IWU 5.2.2.9

5.2.2.9.1 Receipt of Reset Circuit message (RSC)

TP309001	SIP reference: RFC 3261 [4] ISUP reference:					
		_	_	Q.1912.5	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	Τ	SIP-I	
	IAM	→				
	RSC	→				
	RLC	+				

TP309002	SIP reference: RFC 3261	[4]		SUP reference:		
			Q.1912.5 [1]	, clauses 7.7.1, 7.7.4, 7.7.5		
TSS reference	ISUP-SIP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message on receipt RSC message before a SIP MESSAGE_VA response message has been received: the SUT shall hold the RSC message until a SIP response has been received; the SUT shall send a CANCEL request. The RSC is not encapsulated.					
SIP parameter values		•		·		
ISUP parameter values						
Comments	ISUP/BICC	SU	Т	SIP-I		
	IAM →		→	INVITE(IAM)		
	RSC →					
	RLC ←					
			+	SIP_MESSAGE_VA		
			→	CANCEL		
			+	200 OK CANCEL		
			+	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 3	3261 [4]			I	SUP reference:	
				Q.1912.	5 [1]	, clauses 7.7.1, 7.7.4, 7.7.5	
TSS reference	ISUP-SIP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking 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 cau	se 31				
ISUP parameter values							
Comments	ISUP/BICC		SU	Т		SIP-I	
	IAM	→			→	INVITE(IAM)	
	RSC	→					
	RLC	+					
					-	200 OK INVITE(CON)	
					→	ACK	
					→	BYE(REL#31)	
					←	200 OK BYE(RLC)	

TP309005	SIP reference: RFC 3	3261 [4]		0.4040.5		SUP reference:
	10110 010 /0 1 11/0					clauses 7.7.1, 7.7.4, 7.7.5
TSS reference	ISUP-SIP/Basic call/Receipt of					
	(GRS) or Circuit group blocking	ng mess	age (CGI	B) with the i	indic	cation hardware failure
	oriented					
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Ensure that the SUT after rece	eiving th	ne IAM wi	th the comp	olete	called party number,
	sending a INVITE message w	ith the c	complete	called party	nur nur	nber, sending a BYE
	message on receipt RSC mes					
	the SUT shall send a BYE					
SIP parameter	BYE: A REL is encapsulated v					
values	·					
ISUP parameter						
values						
Comments	ISUP/BICC		SU	T		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]	-	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 loriented					
CID coloction	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 request The RSC is not encapsulated.					
SIP parameter values						
ISUP parameter values						
Comments	ISUP/BICC		SU	Т	SIP-I	
	IAM	→		→	INVITE(IAM)	
	ACM	+		+	SIP_MESSAGE_VA(ACM)	
	RSC	→			, , ,	
	RLC	+				
				→	CANCEL	
				+	200 OK CANCEL	
				+	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	61 [4]		ISUP reference: .5 [1], clauses 7.7.1, 1), 7.7.4, 7.7.5			
TSS reference		P/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message r Circuit group blocking message (CGB) with the indication hardware failure					
SIP selection criteria							
ISUP selection criteria							
Test purpose	Ensure that the SUT after receiving the IAM but before an INVITE has been sent on receipt of GRS message: no action is required on the SIP side other than to terminate local procedures if any are in progress.						
SIP parameter values							
ISUP parameter values							
Comments	17 (14)	SL	JT	SIP-I			
	O. to	-					

TP309008	SIP reference: RFC 3	3261 [4]	l	-	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 GRS message before SIP MESSAGE_VA response message has been received: the SUT shall hold the GRS message until a SIP response has been received; the SUT shall send a CANCEL request The GRS is not encapsulated.							
SIP parameter values								
ISUP parameter values								
Comments	ISUP/BICC		SU	T	SIP-I			
	IAM	→		→	INVITE(IAM)			
	GRS	→						
	GRA	+						
				+	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 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:		
				clauses 7.7.1 3), 7.7.4, 7.7.5		
TSS reference	ISUP-SIP/Basic call/Receipt o (GRS) or Circuit group blockin 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	SU	JT	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	3261 [4	.]		IS	SUP reference:	
	Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.5						
TSS reference	ISUP-SIP/Basic call/Receipt of	of Rese	t circuit m	essage (RS	SC),	Circuit group reset message	
	(GRS) or Circuit group blocking	ng mes	sage (CGI	3) with the i	indic	ation hardware failure	
	oriented						
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Ensure that the SUT after rec						
	sending a INVITE message w						
	message on receipt GRS mes						
	 the SUT shall send a BY 	E requ	est The GF	RS is not er	cap	sulated.	
SIP parameter							
values							
ISUP parameter							
values							
Comments	ISUP/BICC		SU	Т		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 3	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	ng mes	sage (CGI	B) with the indi	cation hardware failure	
	oriented					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt GRS message after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established: • the SUT shall send a CANCEL request The GRS is not encapsulated.					
SIP parameter values						
ISUP parameter values						
Comments	ISUP/BICC		SU	Т	SIP-I	
	IAM	→		→	INVITE(IAM)	
	ACM	+		+	SIP_MESSAGE_VA(ACM)	
	GRS	→				
	GRA ←					
				→	CANCEL	
				+	200 OK CANCEL	
				+	487 Request terminated	
				→	ACK	

Table 18

	Values for test purpose TP309012					
VA	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	g message (CG	B) with the indi	cation nardware failure		
SIP selection						
criteria						
ISUP selection criteria						
Test purpose	Ensure that the SUT after receiving more than one IAM's sending an INVITE message for each call association on receipt of a GRS message were the Range Parameter value is bigger than "1": the SUT shall send a BYE requests for each call association The GRS is not encapsulated.					
SIP parameter	BYE1 contains the CSeq of IN					
values	BYE2 contains the CSeq of IN	∕ITE2				
ISUP parameter						
values						
Comments	ISUP/BICC	SU		SIP-I		
	IAM	→	→	INVITE1(IAM)		
	ACM	-	+	180 Ringing(ACM)		
	ANM	-	+	200 OK INVITE(ANM)		
			→	ACK		
	1004	_		INDUSTRICATION		
	IAM)	→	INVITE2(IAM)		
	ACM	+	-	180 Ringing(ACM)		
	ANM	+	(200 OK INVITE(ANM)		
	0.00		→	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 326	61 [4]		P reference: clauses 7.7.1, 1), 7.7.4			
TSS reference	ISUP-SIP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Ensure that the SUT after receiving the IAM but before an INVITE has been sent on receipt of CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented": • no action is required on the SIP side other than to terminate local procedures if any are in progress.						
SIP parameter values							
ISUP parameter values	CGB(hardware failure oriented)						
Comments	CGB •	> > > > + + + + + + + + + + + + + + + +	T SI	P-I			

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 an INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" before a SIP MESSAGE_VA response message has been received: • the SUT shall hold the CGB message until a SIP 200 OK response has been received; • the SUT shall send a CANCEL request The CGB is not encapsulated.						
SIP parameter values							
ISUP parameter values	CGB(hardware failure oriented)						
Comments	SIP_I INVITE(IAM) SIP_MESSAGE_VA CANCEL						
		← ← →	200 OK CANCEL 487 Request terminated ACK				

Table 19

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

TP309016	SIP reference: RFC 3261 [4]		ISUP reference: Q.1912.5 [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	JT 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 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 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 orien	ted)					
Comments	ISUP/BICC		SU	-	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: RFC 326	1 [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 on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established: • the SUT shall send a CANCEL request 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)		
	ACM •	-	+	SIP_MESSAGE_VA(ACM)		
	CGB -	>				
	CGBA ←					
			→	CANCEL		
			+	200 OK CANCEL		
			(487 Request terminated		
			→	ACK		

Table 20

Values for test purpose TP309018					
VA SIP MESSAGE_VA					
VA_1	180 Ringing				
VA 2	183 Session Progress				

TP309019	SIP reference: RFC	3261 [4]		-	SUP reference:	
TSS reference	Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.					
133 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	any message	(CGB) W	nui uie iiiai	Cation nardware failure	
SIP selection	Offertied					
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	BYE1 contains the CSeq of	INVITE1				
values	BYE2 contains the CSeq of	INVITE2				
ISUP parameter	CGB(hardware failure orien	ted)				
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	
	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 3261 [4]		ISUP reference:		
					Q.1912.5 [1], clause A.1.1.3		
TSS reference	ISUP-SIP/ISUP Messages for special consideration/Confusion message						
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number and contains an unknown parameter, sending a INVITE message with the complete called party number and encapsulated IAM as received. Ensure that when the succeeding node discards an unknown parameter and send back a Confusion message if indicated in the parameter compatibility information and the sending of a Confusion message is requested, the CFN message encapsulated in a 183 Session Progress is sent. Ensure ISUP message is transported through the SIP network encapsulated in the 183 Session Progress.						
SIP parameter values	183 Session Progress with encapsulated CFN						
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		
	Communication						
	REL → BYE(RE			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		SL	IT	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		
			Conver	sation			
	REL	→		→	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

5.2.2.12 Receipt of a Blocking message

TP312001	SIP reference: RFC 3	261 [4]	Q.191	ISUP reference: 2.5 [1], clause A.1.1.3.1		
TSS reference	ISUP-SIP/ISUP Messages for special consideration/Receipt of a Blocking message					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Ensure that the blocking/unblocking procedure can be correctly initiated. Ensure the BLO messages are not encapsulated within SIP messages					
SIP parameter values						
ISUP parameter values						
Comments	ISUP/BICC	SI	JT	SIP-I		
	BLO	→				
	BLA	+				
	UBL	→				
	UBA	+				

TP312002	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 Blocking message							
SIP selection	The same of the sa							
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the blocking from both ends; removal of blocking from one end can be correctly initiated. Ensure the BLO messages are not encapsulated within SIP messages.							
SIP parameter	,							
values								
ISUP parameter								
values								
Comments	ISUP/BICC	SU	T	SIP-I				
	BLO →							
	BLA ←							
	BLO ←							
	BLA →							
	UBL →							
	UBA ←							

TP312003	SIP reference: RFC 32	61 [4]		ISUP reference:			
	Q.1912.5 [1], clause A.1.1.3.1						
TSS reference	ISUP-SIP/ISUP Messages for s	pecia	I conside	ration/Recei	pt of a Blocking message		
SIP selection							
criteria							
ISUP selection criteria							
Test purpose	CGB and CGU sent Ensure that the SUT is able to respond on a Circuit group 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	3			•			
ISUP parameter							
values							
Comments	ISUP		SU	T	SIP-I		
	CGB	→					
	CGBA ←						
	CGU	→					
	CGUA	+					

TP312004	SIP reference: RFC	3261 [4]		JP reference: [1], clause A.1.1.3.1					
TSS reference	ISUP-SIP/ISUP Messages for special consideration/Receipt of a Blocking message								
SIP selection criteria		·	·						
ISUP selection criteria									
Test purpose	Ensure that the SUT on rece messages, discards the ISU		ch is received en	capsulated within SIP					
SIP parameter values									
ISUP parameter values									
Comments	ISUP	Sl	JT ←	SIP-I INFO(CGB)					

TP312005	SIP reference: R	FC 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	s for special co	onsiderat	tion/Receipt of	of a Blocking message	
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Ensure that a received IA	M will unblock	a remote	ely blocked c	ircuit.	
SIP parameter						
values						
ISUP parameter						
values						
Comments	ISUP		SUT		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 32	261 [4]	Q.1912	SUP reference: .5 [1], clause A.1.1.3.1 [i.11], clause 1.3.2.4					
TSS reference	ISUP-SIP/ISUP Messages for	special conside	ration/Receipt	of a user part test message					
SIP selection criteria			·	-					
ISUP selection criteria	PICS 4/22								
Test purpose	part available message.	Ensure that on receipt of a user part test message the SUT will respond by sending a user part available message. Ensure that the user part test message is not encapsulated within SIP messages.							
SIP parameter values									
ISUP parameter values									
Comments	ISUP UPT	SU →	T	SIP-I					
	UPA	′							

TP313002	SIP reference: RFC 32	61 [4]		SUP reference: 5 [1], clause A.1.1.3.1					
TSS reference	ISUP-SIP/ISUP Messages for special consideration/Receipt of a user part test messa								
SIP selection criteria									
ISUP selection criteria	PICS 4/22								
Test purpose	Ensure that the SUT is able to	send a user pa	rt test message.						
SIP parameter values									
ISUP parameter values									
Comments	ISUP	SU	JT	SIP-I					
	UPT	(
	UPA	→							

TP313003	SIP reference: RFC 3261 [4]		UP reference: [1], clause A.1.1.3.1					
TSS reference	ISUP-SIP/ISUP Messages for special cons	deration/Receipt 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	SUT	SIP-I					
	UPT ←							
	T4 expiry							
	UPT ←							
	UPA →							

5.2.2.14 Segmentation

TP314001	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause A.1.1.3.1			
TSS reference	ISUP-SIP/ISUP Me	essages for specia	al conside	ration/Rec	eipt	of a user part test message	
SIP selection criteria							
ISUP selection							
criteria							
Test purpose	Ensure that a call of direction.	can be successfull	ly complet	ed if segm	nenta	ation applies in forward	
SIP parameter	INVITE - encapsula	ated IAM: Optiona	I Forward	call indica	ator a	absent or set to "no additional	
values	information will be No action takes pla		e				
ISUP parameter	IAM: optional forwa	ard call indicator: a	additional	informatio	n wil	I be sent in a segmentation	
values	message SGM: optional para	ameters				Ç	
Comments	ISUP		SU			SIP-I	
	IAM	→					
	SGM	→			→	INVITE(IAM)	
	ACM	+			+	180 Ringing(ACM)	
	ANM	+			+	200 OK INVITE(ANM)	
					→	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	3261	[4]		I	SUP reference:				
				Q.	1912.5	[1], clauses 7.1.3, B.1				
TSS reference	ISUP-SIP-ISUP/SS/CLIP	ISUP-SIP-ISUP/SS/CLIP								
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	Calling Party number network	rk prov	rided, transf	erred in (D-MGC	CF				
	Engure that the CLIT can au	f	Illy transmit	t o coll bo	vina a	calling party number with				
	Ensure that the SUT can such the screening indicator set to									
	set to "presentation allowed"		iork provide	and th	e pres	sentation restricted indicator				
SIP parameter	Set to presentation allowed	•								
values										
ISUP parameter	IAM:									
values	Calling party number para	meter								
	Address signals = PIXIT1									
	Numbering plan indicator = '									
	Nature of address indicator :	= '0000	0011'B							
	Screening indicator = '11'B									
	presentation restricted indica	ator = p			, '00'B					
Comments	ISUP		SU	Γ		SIP-I				
	IAM	→			→	INVITE(IAM)				
	ACM	<u> </u>			+	180 Ringing(ACM)				
	ANM	-			+	200 OK INVITE(ANM)				
					→	ACK				
			Convers	ation						
	REL	<u>→</u>			→	BYE(REL)				
	RLC	+			+	200 OK BYE(RLC)				

TP401002	SIP reference: RFC	3261	[4]		ISUP reference: 5 [1], clauses 7.1.3, B.1				
TSS reference	ISUP-SIP-ISUP/SS/CLIP				· ·				
SIP selection criteria									
ISUP selection criteria									
Test purpose	Calling Party number network provided, Calling Subaddress transferred in O-MGCF								
	Ensure that the SUT can sur the screening indicator set to containing the calling sub-a	o "netv	ork provide		a calling party number with ess transport parameter				
SIP parameter values									
ISUP parameter values	Address signals = PIXIT1 Numbering plan indicator = ' Nature of address indicator : Screening indicator = '11'B presentation restricted indic Access transport parameter	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		SU	Γ	SIP-I				
	IAM	→		→	INVITE(IAM)				
	ACM	←		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
				→	ACK				
			Convers						
	REL	→		→	BYE(REL)				
	RLC	+		←	200 OK BYE(RLC)				

TP401003	SIP reference: RFC	3261	[4]		-	SUP reference:
				Q.1	912.5	5 [1], clauses 7.1.3, B.1
TSS reference	ISUP-SIP-ISUP/SS/CLIP					
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Calling Party Number user	orovide	d transferre	d in O-M	GCF	
	Ensure that the SUT can su	ccessf	ully transmit	a call ha	ving th	he calling party number with
	the screening indicator set t				nd pas	sed" and the presentation
	restricted indicator set to "pi	esenta	tion allowed	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					
	presentation restricted indic	cator =	presentation	n allowed	l, '00'E	3
Comments	ISUP		SUT			SIP-I
	IAM	→			→	INVITE(IAM)
	ACM	+			+	180 Ringing(ACM)
	ANM	+			+	200 OK INVITE(ANM)
					→	ACK
			Convers	ation		
	REL	→			→	BYE(REL)
	RLC	+			+	200 OK BYE(RLC)

TP401004	SIP reference: RFC	3261	[4]	O 191	ISUP reference: 2.5 [1], clauses 7.1.3, B.1			
TSS reference	ISUP-SIP-ISUP/SS/CLIP			Q.131	2.0 [1], clad3c3 7.1.0, D.1			
SIP selection criteria	1001 011 1001 7007 0211							
ISUP selection criteria								
Test purpose	Calling Party Number user provided and calling subaddress transferred in O-MGCF Ensure that the SUT can successfully transmit a call having a calling party number with the screening indicator set to "user provided, verified and passed" and an access transport parameter containing the calling sub-address.							
SIP parameter values								
ISUP parameter values	IAM; Calling party number para Address signals = PIXIT1 Numbering plan indicator = Nature of address indicator Screening indicator = '01'B Presentation restricted indic Access transport parameter	'001'B = '0000 ator =	presentation					
Comments	ISUP		SUT		SIP-I			
	IAM	→		→	· INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	4		+	200 OK INVITE(ANM)			
				→	ACK			
			Conversa	ation				
	REL	→		→	= - = (= -)			
	RLC	+		←	200 OK BYE(RLC)			

TP401005	SIP reference: RFC	3261	[4]			SUP reference:			
				Q.1	912.5	[1], clauses 7.1.3, B.1			
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 successfully transmit 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" and the presentation restricted indicator set to "presentation allowed".								
SIP parameter values									
ISUP parameter values	IAM;								
	Calling party number para	meter							
	Address signals = PIXIT1 Numbering plan indicator = Nature of address indicator Screening indicator = '11'B Presentation restricted indic	= '0000'		ı allowed,	'00'B				
	Generic number paramete	r							
	Address signals = PIXIT2								
	Numbering plan indicator =								
	Nature of address indicator Screening indicator = '00'B	= '0000'	0011'B						
	Presentation restricted indic	ator = p			'00'B				
Comments	ISUP		SUT	•		SIP-I			
	IAM	→				INVITE(IAM)			
	ACM	←			-	180 Ringing(ACM)			
	ANM	+			<u>+</u>	200 OK INVITE(ANM)			
					→	ACK			
			Conversa	ation					
	REL	→			<u>→</u>	BYE(REL)			
	RLC	←			←	200 OK BYE(RLC)			

TP401006	SIP reference: RFC	3261	[4]		I	SUP reference:		
				Q.1	912.5	[1], clauses 7.1.3, B.1		
TSS reference	ISUP-SIP-ISUP/SS/CLIP							
SIP selection								
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.						
	Ensure that the SUT can su- with the screening indicator additional calling party numb verified" and an access tran-	set to ' per with	'network pronter the screen	ovided", a ing indica	gene tor se	et to "user provided, not		
SIP parameter								
values								
ISUP parameter values	IAM;							
	Calling party number para	meter						
	Address signals = PIXIT1 Numbering plan indicator = ' Nature of address indicator screening indicator = '11'B		0011'B					
	Generic number paramete	r						
	Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Access transport parameter including the subaddress information							
Comments	ISUP		SUT		2	SIP-I		
	IAM	→	_		→	INVITE(IAM)		
	ACM	+			+	180 Ringing(ACM)		
	ANM	+			+	200 OK INVITE(ANM)		
					→	ACK		
			Convers	ation				
	REL	→			→	BYE(REL)		
	RLC	+			+	200 OK BYE(RLC)		

TP401007	SIP reference: RF	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses 6.1.3.6, B.1			
TSS reference	ISUP-SIP-ISUP/SS/CLIP		•		·			
SIP selection criteria								
SUP selection criteria	PICS 6/8	PICS 6/8						
Test purpose	Calling party number disca		· ·		on the I-MGCF. bilateral agreements, if the			
	.	•			•			
•	address presentation restr	•			•			
SIP parameter values ISUP parameter values	.	icted indicat			•			
/alues SUP parameter /alues	address presentation restr	icted indicat			•			
/alues SUP parameter /alues	IAM; No calling party number	icted indicat	or is set to "pi		on allowed" (see note).			
values SUP parameter values	IAM; No calling party number	parameter	or is set to "pi	resentati	on allowed" (see note).			
/alues SUP parameter /alues	IAM; No calling party number SIP-I INVITE(IAM)	parameter	or is set to "pi	resentati	ISUP			
/alues SUP parameter /alues	IAM; No calling party number SIP-I INVITE(IAM) 180 Ringing(ACM)	parameter	or is set to "pi	resentation	ISUP IAM ACM			
/alues SUP parameter /alues	IAM; No calling party number SIP-I INVITE(IAM) 180 Ringing(ACM) 200 OK INVITE(ANM)	parameter	or is set to "pi		ISUP IAM ACM			
values SUP parameter	IAM; No calling party number SIP-I INVITE(IAM) 180 Ringing(ACM) 200 OK INVITE(ANM)	parameter	or is set to "pi		ISUP IAM ACM			

TP401008	SIP reference: RFC	3261	[4]		ISUP reference:		
				Q.1912.5	[1], clauses 6.1.3.6, B.1		
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 values	IAM; No calling party number p	arame	ter				
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		→	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	→					
			Conversa	tion			
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	←		+	RLC		
	eral agreement prohibits the t ss presentation restricted ind				nber in any case. The test with ted" is a CLIR test.		

TP401009	SIP reference: RF	C 3261	[4]		ISUP reference:			
				Q.1912.5	5 [1], clauses 6.1.3.6, B.1			
TSS reference	ISUP-SIP-ISUP/SS/CLIP							
SIP selection								
criteria								
ISUP selection	PICS 6/6							
criteria								
Test purpose	available in the I-MGCF Ensure that the calling pa	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			
	ACK	→						
			Convers	ation				
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP401010	SIP reference: RF	C 3261	[4]		ISUP reference:
				Q.1912.5	[1], clauses 6.1.3.6, B.1
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Calling party number is se	ent as red	ceived		
OID	Ensure that the calling par number in the encapsulate	•	er in the sent	IAM is gener	rated from the calling party
SIP parameter					
values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	÷		+	ANM
	ACK	→			7 (1 (1))
	7.0.0	+-	Conversa	tion	
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		′	RLC

TP401011	SIP reference: RI	FC 3261 [4	-		ISUP reference: [1], clauses 6.1.3.6, B.1
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Additional calling party nu	ımber is se	ent as received		
SIP parameter values ISUP parameter	additional calling party nu	mber in th	e encapsulated	AIVI.	
values Comments	SIP-I		SUT	1	ISUP
Comments	INVITE(IAM)	→	301	→	IAM
	,	+		+	ACM
	180 Ringing(ACM) 200 OK INVITE(ANM)	+		+	ANM
	ACK	→			AINIVI
	ACK	 	Conversation		
	BYE(REL)	→	Conversation	→	REL
				· /	

TP401012	SIP reference: RF	FC 3261	[4]		SUP reference:
				Q.1912.5	[1], clauses 6.1.3.6, B.1
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Additional calling party nu	ımber is d	omitted in the l	I-MGCF	
	Ensure that if the calling			ent, then an	additional calling party
	number in a generic num				
SIP parameter	0.	number ir	ncluded in the	encapsulate	d IAM, additional calling party
values	number included.				
ISUP parameter	IAM;				
values	No calling party number	parame	eter		
	No generic number para	meter			
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
	ACK	→			
			Conversati	on	
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		+	RLC

TP401013	SIP reference: RFC	3261		Q.1912.5	SUP reference: [1], clauses 6.1.3.6, B.1, 31 [i.2], clause 3.5				
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection									
criteria									
ISUP selection criteria	PICS 1/7	PICS 1/7							
Test purpose		Convert the Calling party number into the international format in the I-MGCF							
	setting the nature of addres address presentation restrict	s indica	ator to "internation	nal numbe					
SIP parameter values				_					
ISUP parameter	IAM;								
values	Calling party number para Address signals = PIXIT1 Numbering plan indicator = Nature of address indicator Screening indicator = '11'B Presentation restricted indic	'001'B = '0000		red. '00'B					
Comments	SIP-I		SUT	34, 552	ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK	→							
			Conversation	•					
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		(RLC				

TP401014	SIP reference: RFC 3	3261	[4]		SUP reference:					
			_	Q.1912.5	[1], clauses 6.1.3.6, B.1,					
					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-	vert th	ne additiona	al calling party i	number in the generic					
					ndicator is "ISDN Telephony",					
	setting the nature of address									
	address 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)100'B							
	Screening indicator = '11'B		_							
	Presentation restricted indica		resentation	allowed, '00'B						
	Generic number parameter									
	Address signals = PIXIT2	04 ID								
	Numbering plan indicator = '0 Nature of address indicator =		1400'D							
	Screening indicator = '00'B	0000	1100 Б							
	Presentation restricted indica	tor -n	resentation	allowed '00'R						
Comments	SIP-I	101 –p	SU'		ISUP					
Comments	INVITE(IAM)	→	30	→	IAM					
	180 Ringing(ACM)	′		· +	ACM					
	200 OK INVITE(ANM)	`		+	ANM					
	ACK	`			7 11 11 11					
			Convers	ation						
	BYE(REL)	→	55516	<u> </u>	REL					
	200 OK BYE(RLC)	-		+	RLC					
	==== =	_			1					

TP401015	SIP reference: RI	reference: RFC 3261 [4]			ISUP reference: [1], clauses 6.1.3.6, B.1, 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	5 1/9						
Test purpose	Ensure that the calling pa	Discarding an incomplete calling party number in the I-MGCF Ensure that the calling party number is discarded, if it is received with the calling party number incomplete indicator set to "incomplete" (see note).						
SIP parameter values								
ISUP parameter values	IAM: No calling party number	parame	ter					
Comments	SIP-I		SU	Γ	ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	→						
			Convers	ation				
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		+	RLC			
NOTE: This test	case is only applicable with	an ITU i	mplementa	tion.	•			

TP401016	SIP reference: RFC 3	261 [[4]	Q.19	12.5	SUP reference: [1], clauses 6.1.3.6, B.1, 3.5/Q.731 [i.2]					
TSS reference	ISUP-SIP-ISUP/SS/CLIP	ISUP-SIP-ISUP/SS/CLIP									
SIP selection											
criteria											
ISUP selection	PICS 1/8	PICS 1/8									
criteria											
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) nu										
	transferred transparently.		IIIO addi	000 proo	o. nanc	on recallation maleuter entitle be					
SIP parameter	INVITE: encapsulated IAM										
values	Calling party number	para	meter								
	Address signals = PIXI										
	Numbering plan indicat		'001'B								
	Nature of address indic	cator	= '0000011'	В							
	Screening indicator = '	11'B									
	Presentation restricted	indic	ator = prese	entation a	allowe	d, '00'B					
ISUP parameter	IAM										
values	Calling party number parameter	er									
	Address signals = PIXIT1										
	Numbering plan indicator = '00										
	Nature of address indicator =	'0000	100'B								
	Screening indicator = '11'B			- 11 1	IOOID						
Comments	Presentation restricted indicate SIP-I	or = r	oresentation SUT		, 00 B	ISUP					
Comments		→	501		→						
	,,	<u>~</u> ←			7	INVITE(IAM)					
	7 (011)	-				180 Ringing(ACM)					
	ANM	~			<u>←</u>	200 OK INVITE(ANM)					
			Conversa	etion	7	ACK					
	REL	→	Conversa	111011	→	BYE(REL)					
		7 ←			7	200 OK BYE(RLC)					
	NLO	~			7	ZUU UN DIE(KLU)					

TP401017	SIP reference: RFC 326	[4]		SUP reference:							
			Q.1912.5	[1], clauses 6.1.3.6, B.1, 3.5/Q.731 [i.2]							
TSS reference:	ISUP-SIP-ISUP/SS/CLIP										
SIP selection criteria											
ISUP selection	PICS 1/8	PICS 1/8									
criteria	1.100 1/0										
Test purpose	Converting the additional calling party number to national format, if necessary in the O-MGCF										
	"additional calling party number", removed if it is the network's own	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									
SIP parameter	INVITE: encapsulated IAM										
values	Generic number parame	ter									
	Address signals = PIXIT2										
	Numbering plan indicator	= '001'B									
	Nature of address indicate	or = '0000011	'B								
	Screening indicator = '11'										
	Presentation restricted inc	licator = pres	entation allowe	d, '00'B							
ISUP parameter	IAM;										
values	Calling party number parameter	r									
	Address signals = PIXIT1										
	Numbering plan indicator = '001'E										
	Nature of address indicator = '00	00100'B									
	Screening indicator = '11'B										
	Presentation restricted indicator =	= presentatio	n allowed, '00'E	,							
	Generic number parameter										
	Address signals = PIXIT2										
	Numbering plan indicator = '001'E Nature of address indicator = '000'										
	Screening indicator = '00'B	J0100 B									
	Presentation restricted indicator =	- presentatio	n allowed '00'E								
Comments	SIP-I	SU		ISUP							
Comments	IAM →	30	<u>'</u>	INVITE(IAM)							
	ACM ←		,	180 Ringing(ACM)							
	ANM ←		+	200 OK INVITE(ANM)							
	7.4.44		<u>`</u>	ACK							
		Convers	-								
	REL →		→	BYE(REL)							
	RLC +		′	200 OK BYE(RLC)							
	I LO			1200 OK DIE(KEO)							

TP401018	SIP reference: RF0	C 3261	[4]	ISUP reference:					
				Q.1912.5	[1], clauses 6.1.3.6, B.1, 3.5/Q.731 [i.2]				
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection criteria									
ISUP selection criteria	PICS 1/7								
Test purpose	Adding a prefix to an intern	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				
	ACK	→							
			Convers	ation					
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				
NOTE: The codin	ng "unknown" is a national op	otion (@	2).						

TP401019	SIP reference: RF	C 3261	[4]	_	ISUP reference: [1], clauses 6.1.3.6, B.1, 3.5/Q.731 [i.2]			
TSS reference	ISUP-SIP-ISUP/SS/CLIP							
SIP selection 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 indicator shall be set to "network provided" if the address presentation restricted indicator in calling party number is set to "address not available" (see note).							
SIP parameter values								
ISUP parameter	IAM;							
values	Calling party number par Address signals = PIXIT1 Numbering plan indicator = Nature of address indicator Screening indicator = '11'E Presentation restricted ind	= ' *'B or = '*'B 3		ilable, '10'E				
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	→						
			Conversatio					
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	←		←	RLC			
NOTE: The codi	ing "address not available" is	a natior	nal option (@).					

TP401020	SIP reference: RFC	3261	[4]	-	SUP reference:				
				Q.1912.	5 [1], clauses 7.1.3, B.1				
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection									
criteria									
ISUP selection criteria									
Test purpose		O-MGCF: Calling party number and Additional calling party number not received							
					ereby Calling Party Number				
	parameter and the Generic								
	Sends an INVITE message								
	header field" set to "anonym				•				
SIP parameter	INVITE: No P-Asserted Ider	ntity, Fr	om Header: <u>a</u>	<u>nonymous@</u>	anonymous.invalid				
values	1004		A 1 1141 1						
ISUP parameter	IAM; no Calling party number	er and	no Additional	calling party	number present				
values		1		1	T				
Comments	ISUP		SUT		SIP-I				
	IAM	→		→	INVITE(IAM)				
	ACM	+		←	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
				→	ACK				
			Conversat	on					
	REL	→		→	BYE(REL)				
	RLC	+	-	+	200 OK BYE(RLC)				

TP401021	SIP reference: RFC	3261	[4]	-	SUP reference:		
				Q.1912.	5 [1], clauses 7.1.3, B.1		
TSS reference:	ISUP-SIP-ISUP/SS/CLIP						
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	O-MGCF: Setting of From h	eader					
SIP parameter	Ensure that when the SUT has received an IAM message whereby Calling Party Number parameter is not applicable and the Generic Number is applicable whereby the address presentation restriction parameter is set to "presentation allowed" and the Nature of Address Indicator is set to NoAS_VALUE. Sends an INVITE message without the "P-Asserted-Identity header field", a "From header field" 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 no "Privacy Header field". INVITE: no P-Asserted-Identity, no Privacy header, From header contains the value of the						
ISUP parameter values	additional calling party number IAM; no Calling party number		ent, Additiona	calling party	number present		
Comments	ISUP		SUT		SIP-I		
	IAM	→		→	INVITE(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
				→	ACK		
			Conversat	ion			
	REL	→		→	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

TP401022	SIP reference: RFC 3	3261 [4]	· ·	SUP reference:						
			Q.1912.	5 [1], clauses 7.1.3, B.1						
TSS reference	ISUP-SIP-ISUP/SS/CLIP	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; without "Privacy Header field" or "id" is not included. 									
SIP parameter	INVITE: P-Asserted-Identity d			mber, Privacy=id, From						
values	header derived from the addit			, , , , ,						
ISUP parameter values	IAM; Calling party number is p			g party number is present						
Comments	ISUP	SU	IT	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)						

TP401023	SIP reference: RFC	3261	[4]		I	SUP reference:			
				Q.1	912.5	5 [1], clauses 7.1.3, B.1			
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 He			,	uded.				
SIP parameter	INVITE: P-Asserted-Identity					mber, no Privacy header,			
values	From header derived from the								
ISUP parameter	IAM; Calling party number a	ınd Add	ditional callin	ng party n	umbe	er are present			
values			T						
Comments	ISUP		SUT			SIP-I			
	IAM	→			→	INVITE(IAM)			
	ACM	+			+	180 Ringing(ACM)			
	ANM	+			+	200 OK INVITE(ANM)			
					→	ACK			
			Convers	ation					
	REL	→			→	BYE(REL)			
	RLC	+			+	200 OK BYE(RLC)			

	Values for test purpose TP401021, TP401022, TP401023							
NoAS_VALUE	ISUP parameter values	SIP parameter values:						
VA_01	IAM	INVITE						
	NoAS_VALUE: "national (significant) number" (NDC+SN)	FHf_Addr_SPEC_ID: CC (of the country where the IWU is located) is added to the Generic Number Address Signals and then mapped to user portion of URI scheme						
VA_02	IAM NoAS_VALUE: "international number" ("+"CC+NDC+SN)	INVITE 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]	0.4	-	SUP reference:		
TCC votovonos	IOUE OID IOUE/OO/OUE			Q.1	912.5	[1], clauses 6.1.3.6, B.1		
TSS reference	ISUP-SIP-ISUP/SS/CLIP							
SIP selection criteria								
ISUP selection	DICC 4/7							
criteria	PICS 1/7							
Test purpose	Calling party derived from the P-Asserted-Identity international number							
	Ensure when no calling part							
	Privacy value "id" received.	ncapsu	lated IAM is	s not iden	tical to	o the P-Asserted-Identity, no		
	Send an IAM the calling par							
	Address Presentation Restr							
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 Calli							
values	Address signals = nu				sserte	ed-Identity		
	Screening indicator =			i				
	Number Incomplete I							
	Numbering plan indic							
	Address Presentation NoAS: "international			itor = Pres	sentat	ion allowed		
Comments	SIP-I		SU ⁻	Γ		ISUP		
	INVITE(IAM)	→			→	IAM		
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM)	+	_	-	+	ANM		
	ACK	→						
			Convers	ation				
	BYE(REL)	→			→	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP401025	SIP reference: RFC	3261	[4]	Q.19	-	SUP reference: [1], clauses 6.1.3.6, B.1			
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection criteria									
ISUP selection criteria	NOT PICS 1/7								
Test purpose	Calling party derived from the P-Asserted-Identity national (significant) number								
	Ensure when no calling party number is included in the encapsulated IAM or the calling party number in the in the encapsulated IAM is not identical to the P-Asserted-Identity, no Privacy value "id" received. Send an IAM the calling party number is derived from SIP P-Asserted-Identity. The Address Presentation Restricted Indicator is set to Presentation allowed.								
SIP parameter	INVITE: P-Asserted identity	user p	ortion is in t	he format	"+"C	C+NDC+SN, Privacy value			
values	"id" is not present								
ISUP parameter	IAM message with the Callin								
values	Address signals = nu Screening indicator = Number Incomplete I Numbering plan indic Address Presentatior NoAS: "national (sign	enetwo ndicate ator = n Resti	ork provided or = PIXIT ISDN numb ricted Indica	l pering plar	1	·			
Comments	SIP-I		SU	Γ		ISUP			
	INVITE(IAM)	→			→	IAM			
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
	ACK	→							
			Convers	ation					
	BYE(REL)	→			→	REL			
	200 OK BYE(RLC)	+			+	RLC			

TP401026	SIP reference: RFC 3	3261 [4]	_	SUP reference:				
TSS reference	ISLID SID ISLID/SS/CLID		Q.1912.5	[1], clauses 6.1.3.6, B.1				
SIP selection	ISUP-SIP-ISUP/SS/CLIP							
criteria								
ISUP selection	DIOC 4/7							
	PICS 1/7							
criteria	A -l-liti l His		tla a	n into me a tiene al monach an				
Test purpose	Additional calling party number derived from the From header international number							
				the encapsulated IAM or the				
	additional calling party number		ncapsulated IAN	I is not identical to the From				
	header field, no Privacy value	e "id" received.						
	Send an IAM the additional ca	alling party num	ber is derived from	om From header field. The				
OID	Address Presentation Restric							
SIP parameter	INVITE: P-Asserted identity u	ser portion is in	the format "+"C	C+NDC+SN, Privacy value				
values	"id" is not present							
ISUP parameter	IAM message with the Additi							
values	Address signals = num			der				
	Screening indicator =		not verified					
	Number Incomplete In		havina alaa					
	Numbering plan indica Address Presentation			sion allowed				
	NoAS: "international n		ator = Presentar	lion allowed				
Comments	SIP-I	SL	IT	ISUP				
Commonto	INVITE(IAM)	→	· ·	IAM				
	180 Ringing(ACM)	-	+	ACM				
	200 OK INVITE(ANM)	-	+	ANM				
	ACK	→						
	,	Conver	sation					
	BYE(REL)	→	→	REL				
	200 OK BYE(RLC)	+	+	RLC				

TP401027	SIP reference: RFC	3261	[4]			SUP reference:			
				Q.19	912.5	[1], clauses 6.1.3.6, B.1			
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 calling party number is included in the encapsulated IAM or the additional calling party number in the encapsulated IAM is not identical to the From								
	header field, no Privacy value "id" received. Send an IAM the additional calling party number is derived from From header field. The Address Presentation Restricted Indicator is set to Presentation allowed.								
SIP parameter	INVITE: P-Asserted identity	user p	ortion is in t	he format	: "+"C(C+NDC+SN, Privacy value			
values	"id" is not present	-							
ISUP parameter	IAM message with the Addi	itional	Calling par	ty numb	er par	rameter coded			
values	Address signals = nu	ımber d	derived from	SIP Fror	n hea	der			
	Screening indicator =	= User	provided, no	ot verified					
	Number Incomplete	Indicato	or = PIXIT						
	Numbering plan indic	cator =	ISDN numb	ering pla	n				
	Address Presentation	n Restr	icted Indica	tor = Pres	sentat	ion allowed			
	NoAS: "national (sigr	nificant) number"						
Comments	SIP-I		SUT	Γ		ISUP			
	INVITE(IAM)	^			→	IAM			
	180 Ringing(ACM)	+			←	ACM			
	200 OK INVITE(ANM)	4			+	ANM			
	ACK	1				-			
			Convers	ation					
	BYE(REL)	→			→	REL			
	200 OK BYE(RLC)	+			+	RLC			

5.3.2 Calling Line Identification Restriction (CLIR)

TP402001	SIP reference: RFC	3261 [4	i]	Q.1912.	SUP reference: 5 [1], clauses 7.1.3, B.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 provided presentation restricted is passed.							
	Ensure that the SUT can suc	ccessful	ly transmit a	call having a	a calling party number with			
	the screening indicator set to	"netwo	ork provided	and the add	lress presentation restricted			
	indicator set to "presentation	restrict	ed".		•			
SIP parameter values								
ISUP parameter	IAM;							
values	Calling party number parai	meter						
	Screening indicator = '11'B							
	Address presentation restrict			'B				
	Generic number parameter							
	Access transport parameter	er is no		ne subaddres				
Comments	ISUP		SUT		SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	+		←	180 Ringing(ACM)			
	ANM	+		-	200 OK INVITE(ANM)			
				→	ACK			
			Conversat	ion				
	REL	→		→	BYE(REL)			
	RLC	+	·	+	200 OK BYE(RLC)			

TP402002	SIP reference: RFC 3	3261 [4	1]		SUP reference:
			G	.1912.	5 [1], clauses 7.1.3, B.1,
				Q.731	[i.2], clause 4.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Restricted calling party nun	nber (ı	network provide	d) with	calling sub-address
	Ensure that the SUT can pass	s trans	parently a call har	ving a c	alling party number with the
	screening indicator set to "net	twork p	provided", the add	ress pr	esentation restricted indicator
	set to "presentation restricted	" and a	an access transp	ort para	ameter containing the calling
	sub-address.				
SIP parameter					
values					
ISUP parameter	IAM;				
values	Calling party number param	neter			
	Screening indicator = '11'B				
	Address presentation restricted				
	Generic number parameter				
	Access transport parameter	r includ	ding subaddress i	nformat	
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
		←		+	180 Ringing(ACM)
	ANM	←		+	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	(<u>-</u>	+	200 OK BYE(RLC)

TP402003	SIP reference: RFC	3261	[4]		ISUP reference:
17 402003	Sir Telefelice. Ki C	3201	[+]	0 1012	5 [1], clauses 7.1.3, B.1,
					1 [i.2], clauses 7.1.3, B.1,
TSS reference:	ISUP-SIP-ISUP/SS/CLIR			Q.13	i [i.2], clause 4.3.2.1.1
SIP selection	150P-51P-150P/55/CLIR				
0 00.000					
criteria					
ISUP selection					
criteria			,		
Test purpose	Restricted calling party nu				
					e calling party number with the
	screening indicator set to "u				
	presentation restricted indicate	ator se	t to "presen	tation restricte	ed".
SIP parameter					
values					
ISUP parameter	IAM				
values	Calling party number para	meter			
	Address signals = PIXIT1				
	Numbering plan indicator =	'001'B			
	Nature of address indicator	= '0000'	0011'B		
	Screening indicator = '01'B				
	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)
				→	ACK
			Convers	ation	
	REL	→		→	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP402004	SIP reference: RFC	3261	[4]		912.5	SUP reference: [1], clauses 7.1.3, B.1, [i.2], clause 4.5.2.1.1	
TSS reference	ISUP-SIP-ISUP/SS/CLIR						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Restricted calling party number (user provided, verified and passed) with calling sub-address 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 containing the calling sub-address.						
SIP parameter values							
ISUP parameter	IAM						
values	Calling party number para Address signals = PIXIT1 Numbering plan indicator = Nature of address indicator Screening indicator = '01'B Address presentation restric Access transport paramet	'001'B = '0000 cted pa	rameter = '01		ormati	on	
Comments	ISUP		SUT			SIP-I	
	IAM	→			→	INVITE(IAM)	
	ACM	+			+	180 Ringing(ACM)	
	ANM	+			+	200 OK INVITE(ANM)	
			Conversa	lion	→	ACK	
	DEL		Conversa	tion		DVE(DEL)	
	REL	→			<u>→</u>	BYE(REL)	
	RLC	_			+	200 OK BYE(RLC)	

TP402005	SIP reference: RFC	3261	[4]		ISUP reference: .5 [1], clauses 7.1.3, B.1, 1 [i.2], clause 4.5.2.1.1					
TSS reference	ISUP-SIP-ISUP/SS/CLIR									
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	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 para Address signals = PIXIT1 Numbering plan indicator = Nature of address indicator Screening indicator = '11'B Address presentation restric Generic number paramete Address signals = PIXIT2 Numbering plan indicator = Nature of address indicator Screening indicator = '00'B Address presentation restric ISUP	'001'B = '0000 cted pa er '001'B = '0000	rameter = '0 0011'B	11'B	SIP-I					
Comments		_	501							
	IAM	→	1	→	INVITE(IAM)					
	ACM	+		-	180 Ringing(ACM)					
	ANM	_		←	200 OK INVITE(ANM)					
			Convers		ACK					
	DEL		Convers		DVE(DEL)					
	REL	→		→	BYE(REL)					
	RLC	+		+	200 OK BYE(RLC)					

TP402006	SIP reference: RFC	3261	[4]	Q.19		SUP reference: [1], clauses 7.1.3, B.1,				
						[i.2], clause 4.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/CLIR									
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	Restricted calling party number (user provided, not verified) with calling sub-address									
	Ensure that the SUT can pass transparently 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		•	•							
values										
ISUP parameter	IAM;									
values	Calling party number para	meter								
	Address signals = PIXIT1									
	Numbering plan indicator = '									
	Nature of address indicator :	= '0000	0011'B							
	Screening indicator = '11'B									
	Address presentation restric		rameter = '0'	l'B						
	Generic number paramete	r								
	Address signals = PIXIT2	00415								
	Numbering plan indicator = '		204415							
	Nature of address indicator :	= '0000	0011'B							
	Screening indicator = '00'B			חיי						
	Address presentation restrict Access transport parameter				rmati	00				
Comments	ISUP	er incid	SUT	11622 11110	ııııaıı	SIP-I				
Comments	IAM	→	301	+	→	INVITE(IAM)				
	ACM			+		180 Ringing(ACM)				
	ANM	-		+	-	200 OK INVITE(ANM)				
	AINIVI				-	ACK				
			Conversa	tion		AOR				
	REL	→	Jonversa		→	BYE(REL)				
	RLC			+	-	200 OK BYE(RLC)				
	INLO				_	ZUU OR BTE(RLU)				

TP402007	SIP reference: RI	FC 3261 [4	1]	Q.1912.5	ISUP reference: [1], clauses 6.1.3.6, B.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				
criteria					
Test purpose	Discarding the calling p	arty numb	per if the pres	sentation i	s restricted
	Ensure that the calling pa address presentation rest				of bilateral agreements, if the con restricted".
SIP parameter				•	
values					
ISUP parameter	IAM;				
values	No Calling party number	r paramet	er		
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
	ACK	→			
			Conversation	on	
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		+	RLC

TP402008	SIP reference: RF	C 3261	[4]		I	SUP reference:
						[1], clauses 6.1.3.6, B.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 additional					
						number is discarded in case
	of bilateral agreements, if t	he addr	ess present	ation restr	icted	indicator is set to
	"presentation restricted".					
SIP parameter						
values						
ISUP parameter	IAM;					
values	No Calling party number	-	eter			
	No Generic number para	meter				
Comments	SIP-I		SU	Г		ISUP
	INVITE(IAM)	→			→	IAM
	180 Ringing(ACM)	+			←	ACM
	200 OK INVITE(ANM)	+			←	ANM
	ACK	→				
			Convers	ation		
	BYE(REL)	→			→	REL
	200 OK BYE(RLC)	+			(RLC

TP402009	SIP reference: RI	FC 3261	[4]		ISUP reference: 5 [1], clauses 6.1.3.6, B.1, 1 [i.2], clause 4.5.2.1.1
TSS reference:	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	I-MGCF: Calling party nur Ensure that the calling pa in the ISUP IAM.				nt in the IAM sulated IAM is unchanged 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	→			
			Conversa	ation	
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		+	RLC

TP402010	SIP reference: RI	FC 3261	[4]	Q.1912.5	ISUP reference: [1], clauses 6.1.3.6, B.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	g party n	umber recei	ved in the INV	ITE is sent in the IAM
SIP parameter values ISUP parameter values	Ensure that the additional unchanged sent in the ISI		party numbe	r contained in	the encapsulated IAM is
Comments	SIP-I		SU	Γ	ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
	ACK	→			
			Convers	ation	
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		+	RLC

TP402011	SIP reference: RFC 3	261 [4]	Q.19	-	SUP reference: 5 [1], clauses 7.1.3, B.1	
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 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" set to "anonymous@anonymous.invalid". • and with "Privacy Header field" set to "id".						
SIP parameter values	INVITE: P-Asserted-Identity, F	-rom i	⊣eader: an	onymous	yanc	onymous.invalid, Privacy "id"	
ISUP parameter	IAM: Calling party number. No	addi	tional callin	g party nu	mbe	r	
values				• •			
Comments	ISUP		SUT			SIP-I	
	IAM	→			→	INVITE(IAM)	
	ACM	←			+	180 Ringing(ACM)	
	ANM	(+	200 OK INVITE(ANM)	
					→	ACK	
			Convers	ation			
	REL	→			→	BYE(REL)	
	RLC	(+	200 OK BYE(RLC)	

TP402012	SIP reference: RFC	3261	[4]		ISUP reference:		
				Q.191	2.5 [1], clauses 7.1.3, B.1		
TSS reference	ISUP-SIP-ISUP/SS/CLIR						
SIP selection							
criteria							
ISUP selection							
criteria				_			
Test purpose SIP parameter	 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". INVITE: P-Asserted-Identity, From header field, Privacy "id" 						
values	,			,			
ISUP parameter	IAM: Calling party number. a	addition	nal calling pa	arty number			
values							
Comments	ISUP		SUT		SIP-I		
	IAM	→		-			
	ACM	←		+	1991119(11911)		
	ANM	←		+	=00 011		
				-	ACK		
			Conversa				
	REL	→		7	- · - (· · ·)		
	RLC	←		+	200 OK BYE(RLC)		

	Values for test purpose TP401012						
NoAS_VALUE	ISUP parameter values	SIP parameter values:					
VA_01	IAM NoAS_VALUE: "national (significant) number" (NDC+SN)	INVITE FHf_Addr_SPEC_ID: CC (of the country where the IWU is located) is added to the Generic Number Address Signals and then mapped to user portion of URI scheme					
VA_02	IAM NoAS_VALUE: "international number" ("+"CC+NDC+SN)	INVITE FHf_Addr_SPEC_ID: the complete GenericNumber Address Signals is mapped to the user portion of URI scheme used.					

TP402013	SIP reference: RFC	3261 [4]		I;	SUP reference:			
				Q.1	912.5	[1], clauses 6.1.3.6, B.1			
TSS reference	ISUP-SIP-ISUP/SS/CLIR	ISUP-SIP-ISUP/SS/CLIR							
SIP selection									
criteria									
ISUP selection	PICS 1/7								
criteria									
Test purpose	party number in the in the er Privacy value "id" received. Send an IAM the calling part	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 Calli	ng part	y number	paramet	er cod	led			
values	Address signals = nu Screening indicator =				sserte	d-Identity			
	Number Incomplete I								
	Numbering plan indic								
	Address Presentation			itor = Pre	sentat	ion restricted			
	NoAS: "international	numbe				I			
Comments	SIP-I		SU	Γ		ISUP			
	INVITE(IAM)	→			→	IAM			
	180 Ringing(ACM)	←			+	ACM			
	200 OK INVITE(ANM) ← ← ANM								
					→	ACK			
			Convers	ation					
	BYE(REL)	→			→	REL			
	200 OK BYE(RLC)	←			+	RLC			

TP402014	SIP reference: RFC	3261	[4]	Q.19	-	SUP reference: [1], clauses 6.1.3.6, B.1			
TSS reference:	ISUP-SIP-ISUP/SS/CLIR								
SIP selection									
criteria									
ISUP selection	NOT PICS 1/7								
criteria									
Test purpose	Ensure when no calling party number is included in the encapsulated IAM or the calling party number in the in the encapsulated IAM is not identical to the P-Asserted-Identity, Privacy value "id" received. Send an IAM the calling party number is derived from SIP P-Asserted-Identity. The Address Presentation Restricted Indicator is set to Presentation restricted.								
SIP parameter	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value								
values	"id" is present								
ISUP parameter	IAM message with the Calli	ng par	ty number	paramete	r coc	ded			
values	Address signals = nu Screening indicator = Number Incomplete I Numbering plan indic Address Presentation NoAS: "national (sign	e netwo Indicate Cator = In Resti	ork provided or = PIXIT ISDN numb icted Indica	l pering plar	1	·			
Comments	SIP-I		SU	Γ		ISUP			
	INVITE(IAM)	→			→	IAM			
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM) ← ANM								
	ACK →								
			Convers	ation					
	BYE(REL)	→			→	REL			
	200 OK BYE(RLC)	+			+	RLC			

TP402015	SIP reference: RFC	3261	4]		IS	SUP reference:		
				Q.19	912.5	[1], clauses 6.1.3.6, B.1		
TSS reference	ISUP-SIP-ISUP/SS/CLIR							
SIP selection criteria								
ISUP selection criteria	PICS 1/7							
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 values	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value							
ISUP parameter	"id" is present IAM message with the Addi	tional	Calling no	4		emeter coded		
values	Address signals = nu Screening indicator = Number Incomplete I Numbering plan indic Address Presentation NoAS: "international	mber of the second seco	lerived from provided, no pr = PIXIT ISDN numb icted Indica r"	n SIP Fror ot verified pering plai tor = Pres	n head n	der on restricted		
Comments	SIP-I		SU	Γ		ISUP		
	INVITE(IAM)	→			→	IAM		
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM)	-			+	ANM		
	ACK	→						
			Convers	ation				
	BYE(REL)	→			→	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP402016	SIP reference: RFC	3261	[4]	Q.1	-	SUP reference: [1], clauses 6.1.3.6, B.1	
TSS reference	ISUP-SIP-ISUP/SS/CLIR					,	
SIP selection							
criteria							
ISUP selection	NOT PICS 1/7						
criteria							
Test purpose	Ensure when no additional calling party number is included in the encapsulated IAM or the additional calling party number in the encapsulated IAM is not identical to the From header field, Privacy value "id" received. Send an IAM the additional calling party number is derived from From header field. The Address Presentation Restricted Indicator is set to Presentation restricted.						
SIP parameter	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value						
values	"id" is present	•					
ISUP parameter	IAM message with the Addi	itional	Calling par	rty numb	er pa	rameter coded	
values	Address signals = nu					der	
	Screening indicator =			ot verified	l		
	Number Incomplete						
	Numbering plan indic						
	Address Presentation			itor = Pre	sentat	tion restricted	
	NoAS: "national (sign	nificant		_		I.o	
Comments	SIP-I		SU	Γ		ISUP	
	INVITE(IAM)	→			→	IAM	
	180 Ringing(ACM)	+			+	ACM	
	200 OK INVITE(ANM) ← ← ANM						
	ACK →						
			Convers	ation	•		
	BYE(REL)	→			→	REL	
	200 OK BYE(RLC)	+			+	RLC	

5.3.3 Connected line identification presentation (COLP)

TP403001	SIP reference: RF	FC 3261	[4]	Q.19	SUP reference: 12.5 [1], clause B.2, [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 the optional forward call			call requ	esting the COLP service in
SIP parameter values					
ISUP parameter	IAM;				
values	optional forward call ind	dicators	Connected line id	entity req	uest indicator = requested
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		(ACM
	200 OK INVITE(ANM)	+		+	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		+	RLC

TP403002	SIP reference: RF	C 3261 [4	1		ISUP reference: 112.5 [1], clause B.2,			
					[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 passe							
	screening indicator set to "			" and an acce	ss transport parameter			
	containing the connected s	sub-addre	SS.					
SIP parameter								
values								
ISUP parameter	IAM;							
values	optional forward call indiconnected line identity red		otor: room	aatad				
	a)	luest indic	ator. requ	esteu				
	ANM;							
	Connected number para	meter						
	Address presentation restr		meter = '0	0'B				
	Nature of address indicator = '0000011'B							
	Numbering plan indicator =							
	Screening indicator = '01'E	3						
	Address signals = PIXIT							
	and an access transport	parametei	r containin	g the connecte	ed sub-address.			
	b)							
	CON;							
	Connected number parameter							
	Address presentation restricted parameter = '00'B							
	Nature of address indicator = '0000011'B							
	Numbering plan indicator = '001'B Screening indicator = '01'B							
	Address signals = PIXIT	,						
	and an access transport	parametei	r containin	a the connecte	ed sub-address			
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	CASE A	1		l l				
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	→						
	CASE B	<u> </u>		•	•			
	200 OK INVITE(CON)	+		+	CON			
	ACK	→						
		,	Convers	ation				
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		-	RLC			

TP403003	SIP reference: RFC	3261	[4]	(Q.19	SUP reference: 12.5 [1], clause B.2, [i.2], clause 5.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/COLP					1,			
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Connected number (user provided, not verified) without connected sub-address Ensure that the SUT passes transparently a default connected number with the screening indicator set to "network provided", a generic number containing the additional connected number with the screening indicator set to "user provided, not verified" without an access transport parameter containing the connected sub-address.								
SIP parameter									
values									
ISUP parameter	IAM;								
values	optional forward call indica	ators							
	Connected line identity reque	est ind	icator: requ	ested					
	a)		1-						
	ANM;								
	Connected number parame								
	Address presentation restrict Nature of address indicator =			10,R					
	Numbering plan indicator = '0		סווט						
	Screening indicator = '11'B	0016							
	Address signals = PIXIT								
	Additional connected num	ber pr	esent						
	Address presentation restrict			0'B					
	Nature of address indicator =		0011'B						
	Numbering plan indicator = '0	001'B							
	Screening indicator = '00'B								
	Address signals = PIXIT								
	b)								
	CON;								
	Connected number parame	eter							
	Address presentation restrict		rameter = 'C	0'B					
	Nature of address indicator =		0011'B						
	Numbering plan indicator = '0	001'B							
	Screening indicator = '11'B								
	Address signals = PIXIT								
	Additional connected number present Address presentation restricted parameter = '00'B								
				ОБ					
	Nature of address indicator = '0000011'B Numbering plan indicator = '001'B								
	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)	<u> </u>			+	ANM			
	ACK	→							
	CASE B		I			loon			
	200 OK INVITE(CON)	<u>←</u>			+	CON			
	ACK	7	Convers	otion]				
	DVE/DEL)		Convers	ation		DEI			
	BYE(REL)	<u>→</u>			→	REL RLC			
	200 OK BYE(RLC)	_			~	KLC			

TP403004	SIP reference: RFC	3261	[4]		ISUP reference: 912.5 [1], clause B.2,			
					[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							
					onnected number is removed			
	if it is the network's own cou							
	"national (significant) number				tricted indicator and the			
OID	screening indicator shall be			arentiy.				
SIP parameter values	200 OK: encapsulated ANM							
values	Connected number Address presentation			tor 1001D				
	Nature of address in							
	Numbering plan indic			Ь				
	Screening indicator :							
	Address signals = Pl		_0.					
ISUP parameter	IAM;							
values	optional forward call indic	atore						
	•							
	Connected line identity requ	iest inc	ilcator: reque	estea				
	a) ANM ;							
	Connected number param	eter						
	Address presentation restrict		rameter = '0	0'B				
	Nature of address indicator							
	Numbering plan indicator =	'001'B						
	Screening indicator = ISUP							
	Address signals = CC+PIXI	T						
	b)							
	CON;	4						
	Connected number parameter							
	Address presentation restricted parameter = '00'B							
	Nature of address indicator = '0000100'B							
	Numbering plan indicator = '001'B Screening indicator = ISUP_SI							
	Address signals = CC+PIXI							
	Generic number paramete		resent					
Comments	SIP-I		SUT	•	ISUP			
	INVITE(IAM)	→		→	IAM			
	CASE A							
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		←	ANM			
	ACK	→						
	CASE B	-		1 -				
	200 OK INVITE(CON)	←		+	CON			
	ACK	→		1:				
	D)(E(DEL)		Conversa		DE!			
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+	L	+	RLC			

TP403005	SIP reference: RFC 3	3261 [4]			SUP reference:	
						12.5 [1], clause B.2,	
TCC reference	ICUID CUD ICUID/CC/COLD			Q.	731	[i.2], clause 5.5.2.1.1	
TSS reference SIP selection	ISUP-SIP-ISUP/SS/COLP						
criteria							
ISUP selection	PICS 1/7						
criteria	1.00 1/7						
Test purpose	Converting the additional co						
	Ensure that the country code						
	"additional connected number						
	removed if it is the network's of set to "national (significant) no						
	screening indicator shall be tr				lalio	in restricted indicator and the	
SIP parameter	200 OK: encapsulated ANM of			arcinity.			
values	additional connected numb		•				
	Address presentation restricted	ed par	ameter = '0	0'B			
	Nature of address indicator =		011'B				
	Numbering plan indicator = '0	01'B					
	Screening indicator = '01'B						
ISUP parameter	Address signals = PIXIT						
values	IAM;	4					
Value	optional forward call indica						
	Connected line identity reque	st indi	cator: requ	ested			
	a) ANM;						
	Connected number parame	ter pr	esent				
	additional connected numb						
	Address presentation restricted	ed par	ameter = '0	0'B			
	Nature of address indicator =		100'B				
	Numbering plan indicator = '0	01'B					
	Screening indicator = '01'B						
	Address signals = CC+PIXIT						
	b)						
	CON;						
	Connected number parame		esent				
	additional connected numb			OID			
	Address presentation restricted			n.R			
	Nature of address indicator = Numbering plan indicator = '0		100 B				
	Screening indicator = '01'B	J. D					
	Address signals = CC+PIXIT						
Comments	SIP-I		SUT			ISUP	
		→			→	IAM	
	CASE A	<u>, </u>				14.044	
	3 3 7	(<u>←</u> ∠	ACM	
	200 OK INVITE(ANM) ← ANM ACK →						
	CASE B	-				<u> </u>	
		←			(CON	
		→					
			Conversa	ation			
	\ /	→			→	REL	
	200 OK BYE(RLC)	←			(RLC	

TP403006	SIP reference: RFC 326	l [4]		ISUP reference: 12.5 [1], clause B.2,			
				[i.2], clause 5.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/COLP						
SIP selection							
criteria							
ISUP selection	PICS 1/8 AND PICS 7/5						
criteria							
Test purpose	Adding a prefix to an internation Ensure that a prefix is added to the indicator is set to "unknown" (see	ne connecte note).	d number and	the nature of address			
SIP parameter	200 OK INVITE with encapsulate		ON				
values	Connected number parameter Address presentation restricted parameter = '00'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = PIXIT						
ISUP parameter	ANM/CON:						
values	Connected number parameter Address presentation restricted p Nature of address indicator = '000' Numbering plan indicator = '001'E Screening indicator = '11'B Address signals = Prefix+PIXIT	00010'B	00'B				
Comments	SIP-I	SU ⁻	Γ	ISUP			
	IAM →		→	INVITE(IAM)			
	CASE A						
	ACM ←		+	180 Ringing(ACM)			
	ANM ←		+	200 OK INVITE(ANM)			
			→	ACK			
	CASE B						
	CON ← 200 OK INVITE						
			→	ACK			
		Convers					
	REL →		→	BYE(REL)			
	RLC +		←	200 OK BYE(RLC)			
NOTE: The codi	ng "unknown" is a national option (@).					

TP403007	SIP reference: RFC 3261 [4]		ISUP reference:				
			Q.1912.5 [1], clause B.2,				
			Q.731 [i.2], clause 5.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/COLP						
SIP selection							
criteria	DIOC 4/0 AND DIOC 7/0						
ISUP selection	PICS 1/8 AND PICS 7/3						
criteria Test purpose	Discarding the connected number in		of hilatoral agreements				
rest purpose			ded in case of bilateral agreements, if the				
	address presentation restricted indicator						
SIP parameter	200 OK INVITE with encapsulated ANM						
values	Connected number parameter	01 00	51 1				
Value	Address presentation restricted p	arame	eter = '00'B				
	Nature of address indicator = '00						
	Numbering plan indicator = '001'						
	Screening indicator = '11'B						
	Address signals = PIXIT						
ISUP parameter	IAM						
values	optional forward call indicators						
	Connected line identity request indicator	r: requ	ested				
	a)						
	ANM						
	No Connected number parameter						
	b) CON :						
	No Connected number parameter						
Comments	ISUP	SUT	T SIP-I				
Comments	IAM →		→ INVITE(IAM)				
	CASE A		j jiivvii L(i/ (ivi)				
	ACM ←		← 180 Ringing(ACM)				
	ANM (+		€ 200 OK INVITE(ANM)				
	-		→ ACK				
	CASE B		1 1				
	CON ← 200 OK INVITE(CON)						
	→ ACK						
	Co	nvers					
	REL →		→ BYE(REL)				
	RLC 🗲		← 200 OK BYE(RLC)				
NOTE: This bila	teral agreement prohibits the transferral of	the co	onnected number in any case. The test with				
	ess presentation restricted indicator set to						
		-					

TP403008	SIP reference: RFC 3261 [4]		ISUP reference: 912.5 [1], clause B.2, 11 [i.2], clause 5.5.2.1.1					
TSS reference:	ISUP-SIP-ISUP/SS/COLP							
SIP selection								
riteria								
SUP selection	PICS 1/8 AND PICS 7/4							
riteria								
Test purpose	Discarding the additional connected numbers that the additional connected numbers of bilateral agreements, if the address pres "presentation allowed" (see note).	er in the generic entation restricte	c number is discarded in case					
SIP parameter	200 OK INVITE with encapsulated ANM or							
/alues	Additional Connected number pa Address presentation restricted para Nature of address indicator = '00000 Numbering plan indicator = '001'B Screening indicator = '00'B Address signals = PIXIT	ameter = '00'B						
SUP parameter	IAM;							
alues	optional forward call indicators							
	Connected line identity request indicator: requested							
	a)							
	ANM;							
	No Connected number parameter No Additional connected number present b) CON; No Connected number parameter No Additional connected number present							
Comments	ISUP	SUT	SIP-I					
	IAM →	→	INVITE(IAM)					
	CASE A							
	ACM ←	+	10011119					
	ANM ←	+						
		→	ACK					
	CASE B							
	CON ←	+	200 OK INVITE(CON)					
		→	ACK					
	Conv	ersation						
	REL →	→	BYE(REL)					
	RLC ←	+						
NOTE: This bila	teral agreement prohibits the transferral of the	e additional conn						
	in any case.	c additional bottl	.co.coaco a gonono					

TP403009	SIP reference: RF	C 3261	[4]		ISUP reference:				
				Q.19	12.5 [1], clause B.2,				
				Q.731	[i.2], clause 5.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/COLP								
SIP selection									
criteria	7100 1/0								
ISUP selection	PICS 1/8								
criteria	0	-l l-	(- !(
Test purpose	Converting the connected								
	Ensure that the exchange of								
					nal number" and can pass on				
SIP parameter	the address presentation re 200 OK INVITE with encap				ng indicator transparently.				
values	Connected numbe			VIN.					
values	Address presentation			stor - '00'B					
	Nature of address in								
	Numbering plan ind			Ь					
	Screening indicator		00115						
	Address signals = C		Т						
ISUP parameter	IAM;								
values	optional forward call indi	cators							
			licator: requ	ested					
	Connected line identity request indicator: requested a)								
	ÁNM								
	Connected number parar	neter							
	Address presentation restri	icted pa	rameter = '0	0'B					
	Nature of address indicator		0100'B						
	Numbering plan indicator =								
	Screening indicator = '11'B								
	Address signals = PIXIT								
	Presentation restricted indi								
	additional connected nur	nber pr	esent						
	b)								
	CON;								
	Connected number parameter								
	Address presentation restricted parameter = '00'B								
	Nature of address indicator = '0000100'B Numbering plan indicator = '001'B								
	Screening plan indicator = '001'B Screening indicator = '11'B								
	Address signals = PIXIT								
	Presentation restricted indi	cator =	'00'B						
	additional connected nur								
Comments	SIP-I	1	SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	CASE À								
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK	→							
	CASE B								
	200 OK INVITE(CON)	+		+	CON				
	ACK	→							
			Convers	ation					
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP403010	SIP reference: RFC	3261	[4]			SUP reference:					
						l2.5 [1], clause B.2, [i.2], clause 5.5.2.1.1					
TSS reference	ISUP-SIP-ISUP/SS/COLP			٠.,	٠.	[], olddoo oloiz					
SIP selection											
criteria											
ISUP selection											
criteria											
Test purpose	Handling unrequested COL										
	Ensure that the call can be				rece	eives an unsolicited COL.					
SIP parameter	200 OK INVITE with encap			N							
values	Connected number			IOOID							
	Address presentatio Nature of address in										
	Numbering plan indi			Б							
	Screening indicator										
	Address signals = P										
ISUP parameter	IAM;										
values	optional forward call indic	cators									
	Connected line identity requ	uest inc	dicator: not r	equested							
	a)										
	ANM;										
	Connected number parameter										
	Address presentation restricted parameter = '00'B										
	Nature of address indicator = '0000011'B										
		Numbering plan indicator = '001'B									
	Screening indicator = '11'B										
		Address signals = PIXIT additional connected number present									
	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 nun	a har nr	recent								
Comments	SIP-I	libei pi	SUT	·		ISUP					
	INVITE(IAM)	→	551		}	IAM					
	CASE A		1			1					
	180 Ringing(ACM)	+		—	F	ACM					
	200 OK INVITE(ANM)	+		1	-	ANM					
	ACK	→									
	CASE B										
	200 OK INVITE(CON)	+			(CON					
	ACK	→									
			Conversa	ation							
	BYE(REL)	→			}	REL					
	200 OK BYE(RLC)	+		•	<u> </u>	RLC					

TP403012	SIP reference: RFC 32		SUP reference: 12.5 [1], clause B.2,							
				[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	Ensure that an ANM or CON er									
	without changing. The connected		nchanged. The	ATP contained the						
	connected sub address is inclu									
SIP parameter values	200 OK INVITE: encapsulated	ANM or CON i	ncluded							
ISUP parameter	a)									
values	ΑNM;									
	Connected number paramete									
	Address presentation restricted		00'B							
	Nature of address indicator = '0									
	Numbering plan indicator = '00'	I'B								
	Screening indicator = '11'B									
	Address signals = PIXIT									
	and an access transport para	meter containir	ng the connecte	d sub-address.						
	b)									
	CON;	_								
	Connected number paramete Address presentation restricted		no'P							
	Nature of address indicator = '0		JU D							
	Numbering plan indicator = '00'									
	Screening indicator = '11'B	i D								
	Address signals = PIXIT									
	and an access transport para	meter containir	na the connecte	ed sub-address						
Comments	ISUP	SU		SIP-I						
	IAM =	•	→	INVITE(IAM)						
	CASE A		l .							
	ACM +		+	180 Ringing(ACM)						
	ANM		+	200 OK INVITE(ANM)						
			→	ACK						
	CASE B		I	1						
	CON		+	200 OK INVITE(CON)						
			→	ACK						
		Convers	ation							
	REL		→	BYE(REL)						
	RLC •		+	200 OK BYE(RLC)						
L	<u> </u>	I .								

TP403013	SIP reference: RFC 3	261	[4]		Q.19	SUP reference: 12.5 [1], clause B.2, [i.2], clause 5.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/COLP								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	O-MGCF: connected number								
	Ensure that an ANM or CON ewithout changing. The connected sub address is inclination.	ted n	umber is ur						
SIP parameter values	200 OK INVITE: encapsulated			cluded					
ISUP parameter values	a) ANM;								
values	Connected number paramet	۰							
	Address presentation restricte		rameter = '0	0'B					
	Nature of address indicator =			0 0					
	Numbering plan indicator = '00								
	Screening indicator = '11'B								
	Address signals = PIXIT								
	Additional connected numb								
	Address presentation restricte			0'B					
	Nature of address indicator =		011'B						
	Numbering plan indicator = '00)1'B							
	Screening indicator = '00'B Address signals = PIXIT								
	and an access transport para	amet	er containin	a the conr	necte	ad sub-address			
	b)	amet	er containin	g the com	iecie	eu sub-audress.			
	CON;								
	Connected number paramet	er							
	Address presentation restricte		rameter = '0	0'B					
	Nature of address indicator =								
	Numbering plan indicator = '00	01'B							
	Screening indicator = '11'B								
	Address signals = PIXIT								
	Additional connected numb			O.D.					
	Address presentation restricte			10.B					
	Nature of address indicator =		0118						
	Numbering plan indicator = '00' Screening indicator = '00'B	ЛБ							
	Address signals = PIXIT								
	and an access transport para	amet	er containin	a the conr	necte	ed sub-address.			
Comments	ISUP		SUT			SIP-I			
		→			→	INVITE(IAM)			
	CASE A			L		. , , ,			
	ACM	(←	180 Ringing(ACM)			
	ANM	(-	200 OK INVITE(ANM)			
					→	ACK			
	CASE B								
	CON	(←	200 OK INVITE(CON)			
					→	ACK			
			Convers	ation					
		→			→	BYE(REL)			
	RLC	(←	200 OK BYE(RLC)			

5.3.4 Connected Line Identification Restriction (COLR)

TP404001	SIP reference: RFC	3261	[4]	Q.19	ISUP reference: 012.5 [1], clause B.2, [i.2], clause 6.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/COLR			Q.731	[1.2], Clause 0.3.2.1.1				
SIP selection	ISST SIT ISST /SS/SSER								
criteria									
ISUP selection									
criteria									
Test purpose	Passing on information relating to COLR Ensure that the SUT shall pass transparently all information related to the COLR supplementary service in the address presentation restricted indicator of the connected number.								
SIP parameter									
values									
ISUP parameter	IAM;								
values	optional forward call indic Connected line identity requ a) ANM; Connected number param Address presentation restric Nature of address indicator = ' Numbering plan indicator = '01'B Address signals = PIXIT b) CON; Connected number param Address presentation restric Nature of address indicator = ' Numbering plan indicator = ' Numbering plan indicator = ' Screening indicator = '01'B Address indicator = '01'B	eter ted pa = '0000 001'B eter ted pa	rameter = '0 0011'B rameter = '0	1' B					
0	Address signals = PIXIT		CUT		loup				
Comments	SIP-I	→	SUT	→	ISUP IAM				
	INVITE(IAM) CASE A	7	L	7	IAW				
		+	1	+	IACM				
	180 Ringing(ACM) 200 OK INVITE(ANM)		-	 	ACM ANM				
	ACK				AINIVI				
	CASE B		1		1				
	200 OK INVITE(CON)	+		+	CON				
	ACK	`							
			Conversa	tion					
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)			+	RLC				

TP404002	SIP reference: RFC	3261 [[4]		ISUP referen 912.5 [1], clau 1 [i.2], clause	ıse B.2,		
TSS reference	ISUP-SIP-ISUP/SS/COLR							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Passing on information relating to COLR Ensure that the SUT shall pass transparently all information related to the COLR supplementary service in the address presentation restricted indicator of the connected number and the additional connect number in the generic number.							
SIP parameter								
values								
ISUP parameter	IAM:							
values	optional forward call indicated connected line identity reques a) ANM; Connected number paramed Address presentation restrict Nature of address indicator = 'Numbering plan indicator = '11'B Address signals = PIXIT Additional connected number Address presentation restrict Nature of address indicator = 'Numbering plan indicator = 'Numbering plan indicator = 'Oo'B Address signals = PIXIT b) CON; Connected number paramed Address presentation restrict Nature of address indicator = 'Numbering plan indicator = 'Oscreening indicator = 'Indicator = 'Indicator = 'Indicator = 'Numbering plan indicator = 'Numbering plan indicator = 'Oo'B Address signals = PIXIT	eter ted pare led pare	rameter = '0 0011'B esent rameter = '0 0011'B rameter = '0 0011'B esent rameter = '0	1' B 1' B				
0			01.17		IOUD			
Comments	SIP-I	_	SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	CASE A	,		1 -	14.014			
	180 Ringing(ACM)	(-				
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	→						
	CASE B							
	200 OK INVITE(CON)	+		+	CON			
	A O14	→						
	ACK							
i e	ACK		Conversa	ation				
	BYE(REL)	→	Conversa	ation ->	REL			

TP404003	SIP reference: RFC	3261	[4]	Q.191	SUP reference: 2.5 [1], clause B.2,			
				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 connected sub-address Ensure that the SUT can pass transparently 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	LANA.							
ISUP parameter values	IAM; optional forward call indica Connected line identity reque a) ANM; Connected number parame Address presentation restrict Nature of address indicator = '0 Screening indicator = '01'B Address signals = PIXIT access transport parameter of the connected number parameter of the	eter ted pa = '0000 001'B contain eter ted pa = '0000 001'B	rameter = '0 0011'B ning the con rameter = '0 0011'B	1' B nected sub-add				
Comments	SIP-I	CUIIIaii	SU1		ISUP			
Comments	INVITE(IAM)	→	301	→	IAM			
	CASE A	7		7	IIAIVI			
		+	1	+	IACM			
	180 Ringing(ACM) 200 OK INVITE(ANM)	-			ACM ANM			
	ACK				VIAIM			
	CASE B		<u> </u>		l			
	200 OK INVITE(CON)	+		+	CON			
	ACK	`						
	,		Conversa	ation				
	BYE(REL)	→	00.110.0	→	REL			
	200 OK BYE(RLC)			+	RLC			
	200 011 012(1120)		ļ	<u></u>	1.120			

TP404004	SIP reference: RFC 326	1 [4]	ISUP reference:							
				12.5 [1], clause B.2, [i.2], clause 6.5.2.1.1						
TSS reference	ISUP-SIP-ISUP/SS/COLR		Q./31	[1.2], Clause 6.5.2.1.1						
SIP selection	130F-31F-130F/33/COLK									
criteria										
ISUP selection	PICS 7/1									
criteria	1 100 7/1									
Test purpose	Discarding the connected number if the presentation is restricted									
	Ensure that the connected num									
	address presentation restricted i	ndicator is se	to "presentation	on restricted".						
SIP parameter	200 INVITE: encapsulated ANM	or CON								
values	No Connected number pa	arameter inclu	ded							
ISUP parameter	IAM;									
values	optional forward call indicator									
	Connected line identity request i	ndicator: requ	ested							
	a)									
	ANM;									
	Connected number parameter									
	Address presentation restricted)1'B							
	Nature of address indicator = '0000011'B									
	Numbering plan indicator = '001'B									
	Screening indicator = '11'B									
	Address signals = PIXIT									
	b)									
	CON;									
	Connected number parameter									
	Address presentation restricted parameter = '01'B									
	Nature of address indicator = '0000011'B									
	Numbering plan indicator = '001' Screening indicator = '11'B	В								
	Address signals = PIXIT									
Comments	SIP-I	SU [*]	г	ISUP						
Comments	INVITE(IAM) →	30	<u>'</u>	IAM						
	CASE A			10 000						
	180 Ringing(ACM)		+	ACM						
	200 OK INVITE(ANM)	1	+	ANM						
	ACK →			Al vivi						
	CASE B			1						
	200 OK INVITE(CON)		+	CON						
	ACK →									
	7.01	Convers	ation							
	BYE(REL) →	Jonvers	<u> </u>	REL						
	200 OK BYE(RLC)	+	+	RLC						
	200 ON DIL(NLO)			IVEO						

	SIP reference: RF0	ISUP reference:								
l				Q.1912.5 [1], clause B.2,						
				Q.731	[i.2], clause 6.5.2.1.1					
TSS reference	ISUP-SIP-ISUP/SS/COLR									
SIP selection	PICS 7/2									
criteria	_									
ISUP selection criteria										
Test purpose	Discarding the additional connected number in the generic number if the									
rest purpose	presentation is restricted		ctea mambe	ii iii tile gelle	inc number if the					
Ì			ed number i	n the generic	number is discarded in case					
Ì	of bilateral agreements, if the	he addr	ess presenta	ation restricted	d indicator is set to					
Ì	"presentation restricted".									
SIP parameter	200 INVITE: encapsulated	ANM o	r CON							
values	No Additional Conne			neter included	I					
ISUP parameter	IAM;		•							
values	optional forward call indi									
Ì	Connected line identity req	uest inc	dicator: reque	ested						
Ì	a)									
Ì	ANM;									
Ì	Connected number param									
Ì	Additional Connected number parameter									
Ì	Address presentation restricted parameter = '01'B									
Ì	Nature of address indicator = '0000011'B Numbering plan indicator = '001'B									
Ì	Screening indicator = '11'B									
Ì	Address signals = PIXIT									
Ì	, taa. eee eig. a.e									
Ì	(b)									
Ì	CON;									
Ì	Connected number parameter present									
Ì	Additional Connected number parameter									
Ì	Address presentation restricted parameter = '01'B									
Ì	Nature of address indicator = '0000011'B									
Ì	Numbering plan indicator = '001'B									
Ì	Screening indicator = '11'B Address signals = PIXIT									
Comments	SIP-I		SUT	.	ISUP					
Commicuts	INVITE(IAM)	→	301	→	IAM					
ı	CASE A		1		10 uv1					
ı	180 Ringing(ACM)	+		+	ACM					
ı	200 OK INVITE(ANM)	+		+	ANM					
ı	ACK	+		<u> </u>						
ı	CASE B		1	1						
ı	200 OK INVITE(CON)	+		+	CON					
ı	ACK	→		<u> </u>						
ı		† <u></u>	Conversa	ation						
ı	BYE(REL)	→		→	REL					
	200 OK BYE(RLC)	+	1	+	RLC					

TP404007	SIP reference: RFC	3261	[4]		1	SUP reference:				
		0_0.				12.5 [1], clause B.2,				
				C		[i.2], clause 6.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/COLR									
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	O-MGCF: Connected numb transferred	er, add	litional conn	ected nui	mber	and connected subaddress				
	Ensure that an ANM or CON without changing. The conn connected sub address is in	ected r	number is ur							
SIP parameter	200 OK INVITE: encapsulat			cluded						
values										
ISUP parameter	ANM;									
values	Connected number param									
	Address presentation restrict)1'B						
	Nature of address indicator		0011'B							
	Numbering plan indicator =	'001'B								
	Screening indicator = '11'B									
	Address signals = PIXIT		acant							
	Additional connected nun Address presentation restric			11'D						
	Nature of address indicator			ПБ						
	Numbering plan indicator =		ЮПБ							
	Screening indicator = '00'B	0010								
	Address signals = PIXIT									
	and an access transport p	aramet	er containin	g the con	necte	ed sub-address.				
	b)			J						
	CON;									
	Connected number param									
	Address presentation restrict)1'B						
	Nature of address indicator		0011'B							
	Numbering plan indicator =	'001'B								
	Screening indicator = '11'B									
	Address signals = PIXIT									
	Additional connected nun	•		MID						
	Address presentation restrict Nature of address indicator			, i D						
	Numbering plan indicator =		טווט							
	Screening indicator = '00'B	JU. D								
	Address signals = PIXIT									
	and an access transport p	aramet	er containin	g the con	necte	ed sub-address.				
Comments	ISUP		SUT			SIP-I				
	IAM	→			→	INVITE(IAM)				
	CASE A									
	ACM	+			+	180 Ringing(ACM)				
	ANM	+			+	200 OK INVITE(ANM)				
					→	ACK				
	CASE B		1							
	CON	+			<u> </u>	200 OK INVITE(CON)				
					→	ACK				
			Convers	ation						
	REL	→			<u>→</u>	BYE(REL)				
	RLC	+			+	200 OK BYE(RLC)				

5.3.5 Terminal Portability (TP)

TP405001	SIP reference:	RFC 3261	[4]		ISUP reference: 12.5 [1], clause B.13, 3 [i.6], clause 4.5.2.1				
TSS reference:	ISUP-SIP-ISUP/SS/TP								
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Ensure that SUT inform	Terminal portability, requested by the calling party Ensure that SUT informs the called party that a suspend and a resume have been requested by the calling party upon receipt of user initiated SUS and RES messages.							
SIP parameter values	INFO: Content-Type: ap	oplication/IS	SUP ; SUS a	nd RES enca	psulated in the MIME body				
ISUP parameter values									
Comments	ISUP		SUT		SIP-I				
	IAM	→		→	INVITE(IAM)				
	ACM	←		+	180 Ringing(ACM)				
	ANM	←		+	200 OK INVITE(ANM)				
				→	ACK				
			Conversa	ition					
	SUS	→		→	INFO(SUS)				
				+	200 OK INFO				
	RES	→		→	INFO(RES)				
				+	200 OK INFO				
	REL	→		→	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP405002	SIP reference: RF	C 3261	[4]	Q.19	SUP reference: 12.5 [1], clause B.13, 3 [i.6], clause 4.5.2.1
TSS reference:	ISUP-SIP-ISUP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Terminal portability, requested by the called pa	he calling	g party that a su	spend and	
SIP parameter values	INFO: Content-Type: appli	cation/IS	SUP ; SUS and	RES enca	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)
				→	ACK
			Conversatio	n 	
	SUS	+		+	INFO(SUS)
				→	200 OK INFO
	RES	+		+	INFO(RES)
				→	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP405003	SIP reference: RFC	3261	[4]	Q.19	ISUP reference: 12.5 [1], clause B.13, 3 [i.6], clause 4.5.2.1
TSS reference	ISUP-SIP-ISUP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Terminal portability, requestions that the call is releast timer T2 expires because the	sed witl	n cause #10	02 (recovery o	n timer expiry) by the SUT if
SIP parameter	INFO: Content-Type: applic				
values	BYE: Content-Type: applica				
ISUP parameter				-	-
values					
Comments	ISUP		SU	Γ	SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
				→	ACK
			Convers	ation	
	SUS	→		→	INFO(SUS)
				+	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP405004	SIP reference: RFC	3261	[4]		ISUP reference: 012.5 [1], clause B.13,				
T00 (LOUID OUR LOUID/OC/TR			Q.7.	33 [i.6], clause 4.5.2.1				
TSS reference	ISUP-SIP-ISUP/SS/TP								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Terminal portability, release	se sus	pended cal	I					
	Ensure that a suspended ca	all can b	e released,	if the remote	user releases the call.				
SIP parameter	INFO: Content-Type: application	INFO: Content-Type: application/ISUP; SUS encapsulated in the MIME body							
values	BYE: Content-Type: applica	tion/ISI	JP; REL 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)				
				→	ACK				
	Conversation								
	SUS	SUS → INFO(SUS)							
				+	200 OK INFO				
	REL	+		+	BYE(REL)				
	RLC	→		→	200 OK BYE(RLC)				

5.3.6 SUB-addressing (SUB)

TP406001	SIP referen	ce: RFC 3261 [4]	Q.19	ISUP reference: 12.5 [1], clause B.5, [i.2], clause 8.5.2.1.1/
TSS reference:	ISUP-SIP-ISUP/SS	/SUB			
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Sending the called	sub-address in th	e access trans	port para	meter
SIP parameter values	parameter in the en	capsulated IAM. /pe: multipart/mix	ed, Content-Ty	pe: applic	cation/ISUP, Content-Type:
values Comments	ISUP	<u> </u>	SUT		SIP-I
Commonto	IAM	→		→	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
				→	ACK
			Conversation)	
	REL	→		→	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP406002	SIP reference: RF	FC 3261	[4]	Q.19	ISUP reference: 12.5 [1], clause B.5, [i.2], clause 8.5.2.1.1/
TSS reference	ISUP-SIP-ISUP/SS/SUB				
SIP selection criteria					
ISUP selection					
criteria					
Test purpose	Receiving the called sub-a Ensure that the SUT can in parameter in the ISUP IAI	include tl			
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	1	
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		+	RLC

TP406003	SIP reference: RFC 3261 [4] ISUP reference: Q.1912.5 [1], clause B.5,									
					[i.2], clause 8.5.2.1.1/					
TSS reference	ISUP-SIP-ISUP/SS/SUB		•							
SIP selection criteria										
ISUP selection criteria										
Test purpose	Sending the calling sub-add	dress in	the access	ransport para	meter					
	Ensure that the SUT can include the calling sub-address in the access transport parameter in the encapsulated IAM.									
SIP parameter					cation/ISUP multipart/mixed,					
values	Content-Type: application/IS MIME body	SUP, C	ontent-Type:	application/IS	SUP; IAM encapsulated in the					
ISUP parameter										
values										
Comments	ISUP		SUT		SIP-I					
	IAM	→		→	INVITE(IAM)					
	ACM	+		-	180 Ringing(ACM)					
	ANM	ANM ← 200 OK INVITE(ANM)								
		→ ACK								
			Conversa	tion						
	REL	→		→	BYE(REL)					
	RLC	+		+	200 OK BYE(RLC)					

TP406004	SIP reference: RI	FC 3261 [4]	Q.19	ISUP reference: i12.5 [1], clause B.5, [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 IAI	include th			
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)	→		→	REL
	200 OK BYE(RLC)	+		+	RLC

5.3.7 Malicious Call Identification (MCID)

TP407001	SIP referenc	e: RFC 3261 [4]		ISUP reference: Q.1912.5 [1], clause B.4, .731.7 [i.3], clause 7.5.2.1.1				
TSS reference:	ISUP-SIP-ISUP/SS/I	MCID							
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Successful MCID red	•							
	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 values	183 Session Progress: Content-Type: application/ISUP; IDR encapsulated in the MIME body INFO: Content-Type: application/ISUP; IRS encapsulated 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	ACM ← 180 Ringing(ACM)							
	ANM								
					→ ACK				
			Conversa	tion					
	REL	→		-	→ BYE(REL)				
	RLC	+	-	•	200 OK BYE(RLC)				

TP407002	SIP reference: RFC 3	261 [4]		Q	Q.19	SUP reference: 12.5 [1], clause B.4, 7 [i.3], clause 7.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/MCID		I			<u> </u>				
SIP selection criteria										
ISUP selection criteria										
Test purpose	Ensure that the SUT can succ set to "MCID request" and pas	Successful MCID request I-MGCF Ensure that the SUT can successfully pass on an IDR having the MCID request indicator set to "MCID request" and pass on an IRS with MCID response indicator set to "MCID included" and the calling party number included. SIP-I to ISUP interworking.								
SIP parameter values	183 Session Progress: Content-Type: application/ISUP; IDR encapsulated in the MIME body INFO: Content-Type: application/ISUP; IRS encapsulated in the MIME body									
ISUP parameter values	111 C. Content Type. application	511/1001	, 1100 01	оарзана	ou iii	une winde body				
Comments	SIP-I		SI	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				
	ACK	→								
			Conver	sation						
	BYE(REL)	→			→	REL				
	200 OK BYE(RLC)	+		·	+	RLC				

TP407003	SIP reference: RFC 3261 [4]			C	ISUP reference: Q.1912.5 [1], clause B.4, Q.731.7 [i.3], clause 7.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/MCID								
SIP selection criteria									
ISUP selection									
criteria									
Test purpose	Ensure that the SUT will accept been received. The SUT should "MCID request" and pass on an	Successful MCID request - after ACM Ensure that the SUT will accept and pass on correctly an MCID request after ACM has been received. The SUT should 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 (see note).							
SIP parameter	INFO: Content-Type: application								
values	INFO: Content-Type: application								
ISUP parameter values	IRS containing the calling party								
Comments	SIP-I		S	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			
	ACK	→							
	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								
	ACK	→							
			Conve	rsation					
	BYE(REL)	→			→	REL			
	200 OK BYE(RLC)	+			+	RLC			
NOTE: This situa	ation may occur e.g. if the call ha	s been	forward	ed befor	e reac	hing the destination.			

TP407004	SIP reference: RFC	3261	[4]		ISUP reference: 912.5 [1], clause B.4, 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	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: Content-Type: application/ISUP; IDR encapsulated in the MIME								
values	body INFO: Content-Type: applic	body INFO: Content-Type: application/ISUP; IRS encapsulated 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									
				→	ACK					
			Conversation							
	REL	→		→	BYE(REL)					
	RLC	+		+	200 OK BYE(RLC)					

TP407005	SIP reference: RFC 3	261 [4]		Q.19	ISUP reference: 12.5 [1], clause B.4, 7 [i.3], clause 7.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/MCID		•						
SIP selection criteria									
ISUP selection criteria									
Test purpose	MCID request - MCID not su Ensure that the SUT rejects a indicator set to "MCID not inc	MCID r	equest by sendi	ing an I					
SIP parameter	183 Session Progress: Conter	183 Session Progress: Content-Type: application/ISUP; IDR encapsulated in the MIME							
values	body INFO: Content-Type: application	on/ISUF	P: IRS encapsul	ated in	the MIME body				
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								
	ACK	→							
			Conversation	•					
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP407006	SIP reference: RFC 3261 [4] ISUP reference:								
					12.5 [1], clause B.4,				
			Q	.731.	7 [i.3], clause 7.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/MCID								
SIP selection									
criteria									
ISUP selection	PICS 1/7								
criteria									
Test purpose SIP parameter values	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. 183 Session Progress: Content-Type: application/ISUP; IDR encapsulated in the MIME body								
	INFO: Content-Type: applicatio	n/ISUP: IRS	encapsulat	ed in	the MIME body				
ISUP parameter values	7	·	•						
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	183 Session Progress(IDR)	+		+	IDR				
	INFO(IRS)	→		1	IRS				
	200 OK INFO	+							
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM) ← ANM								
	ACK	→							
		Con	versation						
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP407007	SIP reference	ce: RFC 3261 [4	1]	ISUP reference: Q.1912.5 [1], clause B.4, 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	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: Content-Type: application/ISUP; IDR encapsulated in the MIME body INFO: Content-Type: application/ISUP; IRS encapsulated 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)				
				→ ACK				
			Conversation	n				
	REL	→		→ BYE(REL)				
	RLC	+		€ 200 OK BYE(RLC)				

TP407008	SIP reference: RFC 32	261 [4]		Q.19	ISUP reference: 112.5 [1], clause B.4, 7 [i.3], clause 7.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/MCID							
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	Successful MCID request with calling sub-address 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	body	183 Session Progress: Content-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			
	ACK	→						
			Conversation	n				
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP407009	SIP reference: RF0	3261	[4]	ISUP reference: Q.1912.5 [1], clause B.4, Q.731.7 [i.3], clause 7.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/MCID		.	<u> </u>			
SIP selection criteria							
ISUP selection criteria							
Test purpose	MCID timer (T39) expiry O-MGCF Ensure that call setup is continued (user is alerted) if no IRS is received within timer T39 expiry, after having sent the IDR with MCID request indicator set to "MCID requested". ISUP to SIP-I interworking.						
SIP parameter values	183 Session Progress: ConbodyMIME body	itent-Ty	pe: application	/ISUP; ID	R encapsulated in the MIME		
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)		
				→	ACK		
			Conversatio	n			
	REL	→		→	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

TP407010	SIP reference: RFC 32	261 [4]			Q.19	SUP reference: 12.5 [1], clause B.4, 7 [i.3], clause 7.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/MCID		<u>.</u>						
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 ISUP interworking							
SIP parameter values	183 Session Progress: Contenbody	183 Session Progress: Content-Type: application/ISUP; IDR encapsulated in the MIME							
ISUP parameter values									
Comments	SIP-I		SI	JT		ISUP			
	INVITE(IAM)	→			→	IAM			
	183 Session Progress(IDR)	+			+	IDR			
					T39 e	expiry			
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM)	200 OK INVITE(ANM) ← ANM							
	ACK	→							
			Conver	sation					
	BYE(REL)	→		•	→	REL			
	200 OK BYE(RLC)	+		•	+	RLC			

5.3.8 Call hold (HOLD)

TP408001	SIP reference: RF	C 3261 [4]	Q.733 [i.6]. c	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			Q.7 00 [0], 0	70000 2101211111, 2101211112			
SIP selection	1001 011 1001 700/11025							
criteria								
ISUP selection criteria								
Test purpose	Call hold after answer, requested by the originating user							
	Ensure that the notification messages having the ever				I retrieved are sent with CPG MGCF interworking.			
SIP parameter	INVITE: Content-Type: app							
values	· · · · · · · · · · · · · · · · · · ·		•	•	•			
ISUP parameter values								
Comments	ISUP		SU'	Т	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)			
	· -	Ì		+	200 OK INVITE			
		Ì		→	ACK			
		Ì						
	REL	→		→	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP408002	SIP reference: RFC	3261	[4]		ISUP reference:
				0 700 5: 01	Q.1912.5 [1],
				Q.733 [1.6], c	lauses 2.5.2.1.1.1; 2.5.2.1.1.2
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection					
criteria					
ISUP selection					
criteria	0 "1 11 6				
Test purpose	Call hold after answer, requ	iestea k	by the origin	nating user	
	Francisco that the motifications	- 46-4-	معادها المعا		I washing and and a safe with CDC
	messages having the even	s mai a • indica	tall is place	progress" I M	I retrieved are sent with CPG
SIP parameter	INVITE: Content-Type: app				
values	INVITE. Content-Type. app	ilication/	130F, CFG	encapsulated	III the Militie body
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	+			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		+	RLC

TP408003	SIP reference: RF	C 3261	[4]		ISUP reference: Q.1912.5 [1], lauses 2.5.2.1.1.1; 2.5.2.1.1.2			
TSS reference	ISUP-SIP-ISUP/SS/HOLD				,			
SIP selection criteria								
ISUP selection criteria								
Test purpose	Call hold after answer, requested by the terminating user Ensure that the notifications that a call is placed on hold and retrieved are sent with CPG messages having the event indicator set to "progress". O-MGCF interworking.							
SIP parameter values	INVITE: Content-Type: app	olication	/ISUP; CPG	encapsulated	in the MIME body			
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: RF	C 3261	[4]		ISUP reference: Q.1912.5 [1], lauses 2.5.2.1.1.1; 2.5.2.1.1.2		
TSS reference	ISUP-SIP-ISUP/SS/HOLD			C. CO [O], C			
SIP selection criteria							
ISUP selection criteria							
Test purpose	Call hold after answer, requested by the terminating user Ensure that the notifications that a call is placed on hold and retrieved are sent with CPG messages having the event indicator set to "progress". I-MGCF interworking.						
SIP parameter values	INVITE: Content-Type: app	olication	/ISUP; CPC	encapsulated	d in the MIME body		
ISUP parameter values							
Comments	SIP-I		SU	T	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		

TP408005	SIP reference: RFC 3261 [4]			Q.733 [i.6	ISUP reference: Q.1912.5 [1], I, 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	Ensure that when an outgo	Call hold after alerting, requested by the calling user Ensure that when an outgoing call is placed on hold and retrieved after alerting the notifications are sent with CPG messages. O-MGCF interworking.							
SIP parameter values	INVITE: Content-Type: app								
ISUP parameter values									
Comments	ISUP		SU.	Γ	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)				
			Convers	ation	\ /				
	REL	→		→	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP408006	SIP reference: RF	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			
TSS reference	ISUP-SIP-ISUP/SS/HOLD							
SIP selection criteria								
ISUP selection criteria	PICS 8/1							
Test purpose	Ensure that when an outgo	Call hold after alerting, requested by the calling user Ensure that when an outgoing call is placed on hold and retrieved after alerting the notifications are sent with CPG messages. I-MGCF interworking.						
SIP parameter values	INVITE: Content-Type: app	olication	/ISUP; CPG	encapsulated	d in the MIME body			
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], 64 [i.12], clause 2.3
TSS reference	ISUP-SIP-ISUP/SS/HOL	D			
SIP selection criteria					
ISUP selection criteria					
Test purpose	Call hold after answer, Ensure that a call in the h service. O-MGCF interwo	neld state ca			served user ser who activated the Call hold
SIP parameter values	INVITE: Content-Type: a		SUP; CPG enca	psulated	I 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		
	CPG(progress, hold)	→		→	INVITE(CPG, sendonly)
				+	200 OK INVITE
				→	ACK
			•		
	REL	→		→	BYE(REL)
	RLC	+	•	+	200 OK BYE(RLC)

TP408008	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1],					
					Q.76	64 [i.12], clause 2.3			
TSS reference	ISUP-SIP-ISUP/SS/HOLD								
SIP selection criteria									
ISUP selection criteria									
Test purpose		Call hold after answer, release of the call by the calling served user Ensure that a call in the held state can be released by the user who activated the Call hold service LMGCF interworking							
SIP parameter values	INVITE: Content-Type: app		/ISUP; CPG	encap	sulated	in the MIME body			
ISUP parameter values									
Comments	SIP-I		SUT			ISUP			
	INVITE(IAM)	→			→	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], 64 [i.12], clause 2.3				
TSS reference	ISUP-SIP-ISUP/SS/HOLI	D							
SIP selection criteria									
ISUP selection criteria									
Test purpose	Call hold after answer, Ensure that a call in the h Call hold service. O-MGC	neld state ca	n be released l		ting user ser who did not activate the				
SIP parameter		INVITE: Content-Type: application/ISUP; CPG encapsulated 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	•	, i				
	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]		-	SUP reference: Q.1912.5 [1], 64 [i.12], clause 2.3			
TSS reference	ISUP-SIP-ISUP/SS/HOLD		l			, , , , , , , , , , , , , , , , , , , ,			
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			encaps	sulated	in the MIME body			
ISUP parameter values									
Comments	SIP-I		SUT			ISUP			
	INVITE(IAM)	→			→	IAM			
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
			Conversa	ation					
	INVITE(CPG, sendonly)	+			+	CPG(progress, hold)			
	200 OK INVITE	→							
	ACK	+							
	BYE(REL)	→			→	REL			
	200 OK BYE(RLC)	+			+	RLC			

TP408011	SIP reference: R	FC 3261 [4]			ISUP reference: Q.1912.5 [1], 64 [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		user vated 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, sendonly)				
	€ 200 OK INVITE								
				→	ACK				
	REL	→		→	BYE(REL)				
	RLC	-		+	200 OK BYE(RLC)				

TP408012	SIP reference: RF	C 3261 [4]		ISUP reference: Q.1912.5 [1],				
				Q.7	64 [i.12], clause 2.3				
TSS reference	ISUP-SIP-ISUP/SS/HOLD								
SIP selection criteria									
ISUP selection criteria									
Test purpose	Ensure that a held call can	Call hold after alerting, release of the call by the calling user Ensure that a held call can be released by the user who activated the Call hold service without retrieving the call. I-MGCF interworking.							
SIP parameter values	INVITE: Content-Type: app	INVITE: Content-Type: application/ISUP; CPG encapsulated in the MIME body							
ISUP parameter values									
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		+	ACM				
			Ringing						
	INVITE(CPG, sendonly)	→		→	CPG(progress, hold)				
	200 OK INVITE	200 OK INVITE ←							
	ACK	→							
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				

5.3.9 Call Waiting (CW)

TP409001	SIP reference	e: RFC 3261	[4]	Q.19	ISUP reference: 112.5 [1], clause B.9, 5 [i.6], clause 1.5.2.1.1				
TSS reference:	ISUP-SIP-ISUP/SS/C	W							
SIP selection criteria									
ISUP selection criteria									
Test purpose	Ensure that a call car	Call waiting indication in ACM Ensure that a call can be successfully established if the ACM indicates that it this call a waiting call. O-MGCF interworking.							
SIP parameter	180 Ringing: Content	-Type: applic	ation/ISUP; ACN	1 encapsu	lated in the MIME bodyMIME				
values	body			•	·				
ISUP parameter values	ACM: Generic notifica	ation indicator	"Call is a waitin	g call"					
Comments	ISUP		SUT		SIP-I				
	IAM	→		→	INVITE(IAM)				
	ACM(waiting)	+		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
				→	ACK				
			Conversation	1					
	REL	→		→	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP409002	SIP reference: RF	C 3261	[4]	ISUP reference: Q.1912.5 [1], clause B.9, 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	CM	·						
SIP parameter values ISUP parameter values	Ensure that a call can be swaiting call. I-MGCF intervals 180 Ringing: Content-Type ACM: Generic notification	working. e: applic	ation/ISUP; AC	M encapsu	I indicates that this call is a lated in the MIME body				
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		+	ACM(waiting)				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK	ACK →							
			Conversation	n					
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP409003					ISUP reference: .1912.5 [1], clause B.9, '33 [i.6], clause 1.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/C	CW			
SIP selection criteria					
ISUP selection criteria					
Test purpose	Call waiting indication Ensure that a call can waiting call. O-MGCF	n be successfu	lly established	l if the C	PG indicates that this call is a
SIP parameter values			tion/ISUP; CP	G encap	sulated in the MIME body
ISUP parameter values	CPG: Generic notification	ation 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)
				→	ACK
			Conversation	n	
	REL	→		→	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP409004	SIP reference: RFC 32	261 [4]		_	-	SUP reference:
						12.5 [1], clause B.9,
				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					
	Ensure that a call can be succe	•	establis	hed if the C	PG	indicates that this call is a
	waiting call. I-MGCF interworki					
SIP parameter	180 Ringing: Content-Type: ap	plicatio	n/ISUP;	CPG enca	osul	ated in the MIME body
values						
ISUP parameter	CPG: Generic notification indic	ator "Ca	all is a w	aiting call"		
values						
Comments	SIP-I		SI	JT		ISUP
	INVITE(IAM)	→			→	IAM
	183 Session Progress ACM)	←			←	ACM
	180 Ringing(CPG)	+			←	CPG(waiting)
	200 OK INVITE(ANM)	+		1	(ANM
	ACK	→				
			Conver	sation		
	BYE(REL)	→			→	REL
	200 OK BYE(RLC)	+		1	←	RLC

TP409005	SIP reference: F	RFC 32	61 [4]		ISUP reference: Q.1912.5 [1], clause B.9, 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 call Ensure that the SUT pass on a REL with cause #21 (call rejected) if a busy user rejects the waiting call. O-MGCF interworking.							
SIP parameter values	body				on/ISUP; REL encapsulated in the			
ISUP parameter values	ACM or CPG: Generic n REL: Cause #21 (call re		ion indicator "Ca	all is a v	vaiting call"			
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]			ISUI	P reference:
			Q.19	912.5	5 [1], clause B.9,
], 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 wit	h ca	use #21 (call rej	ecte	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 Message
values	body				
	480 Temporarily unavailable: Content-T	ype:	application/ISU	P;R	REL encapsulated in the
	MIME body				
ISUP parameter	ACM or CPG: Generic notification indication	ator '	'Call is a waiting	g call'	"
values	REL: Cause #21 (call rejected)				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	♦		→	IAM
	180 Ringing(ACM)	←		+	ACM(waiting)
	480 Temporarily Unavailable(REL)	←		+	REL(#21)
	ACK	→		→	RLC

TP409008	SIP reference: R	FC 3261 [4]		Q.19	ISUP reference: 012.5 [1], clause B.9, 3 [i.6], clause 1.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/CW			Q.733	[1.0], Clause 1.3.2.1.1			
SIP selection criteria								
ISUP selection criteria								
Test purpose	Call waiting ignored (expiry of call waiting supervision timer) Ensure that the SUT pass on a REL with cause #19 (no answer from user, user alerted) if a busy user does not answer the waiting call. I-MGCF interworking.							
SIP parameter	180 Ringing: Content-Type: application/ISUP; ACM or CPG encapsulated in the MIME							
values	body				P; REL encapsulated in the			
ISUP parameter	ACM or CPG: Generic no	otification indic	cator "Call is	a waiting	call"			
values	REL: Cause #19 (no ans	wer from user	, user alerted	d)				
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM(waiting)			
	T9 expiry							
	→ REL(#19)							
				+	RLC			
	480 Temporarily Unavailable	+						
	ACK	→						

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

TP410001	SIP reference: RF	SIP reference: RFC 3261 [4] ISUP reference: Q.1912.5 [1], clau Q.732 [i.4], cl							
TSS reference	ISUP-SIP-ISUP/SS/Call D	iversion							
SIP selection criteria									
ISUP selection criteria									
Test purpose	"Call is diverting" indication received in 180 Ringing 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.								
SIP parameter	183 Session Progress: Co	ntent-Ty	pe: applicat	ion/ISUP; A	CM encapsulated in the MIME				
values	body								
	180 Ringing: Content-Type	e: applic	ation/ISUP;	CPG encap	sulated in the MIME body				
ISUP parameter values	ACM: BCI Called party st Generic notification Call diversion inform Redirection numbe CPG: Event indicator=aler	n mation r		dication"					
Comments	ISUP		SUT		SIP-I				
	IAM	→		→	INVITE(IAM)				
	ACM(no indication)	+		+	183 Session Progress(ACM)				
	CPG(alerting)	+		+	180 Ringing(CPG)				
	ANM	+		+	200 OK INVITE(ANM)				
				→	ACK				
			Conversa	tion					
	REL	→		→	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP410002	SIP reference: RFC 3261	[4]	ISUP reference: Q.1912.5 [1], clauses B.6, B.7, Q.732 [i.4], clause 2.5				
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion						
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	"Call is diverting" indication received in CPG Verify that a call can be successfully established, if diversion occurs. The 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 C			le call diversion information			
	180 Ringing (CPG (alerting)) is coded as if it has been mapped from the CPG. I-MCGF interworking.						
SIP parameter	183 Session Progress: Content-Ty	pe: applica	tion/ISUP; ACN	I encapsulated in the			
values	Message 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	S	UT	ISUP			
	INVITE(IAM)		→	IAM			
	183 Session Progress(ACM) ← ACM(no indication)						
	180 Ringing(CPG)						
	200 OK INVITE(ANM) ← ANM						
	ACK +						
	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], clauses B.6, B.7, Q.732 [i.4], clause 2.5.2.1.1		
TSS reference	ISUP-SIP-ISUP/SS/Call	Diversion				
SIP selection criteria						
ISUP selection criteria						
Test purpose	"Call diversion may occur" received in 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		/pe: applica	ation/ISUP;	ACM encap	osulated in the MIME body	
values		183 Session Progress: Content-Type: application/ISUP; CPG encapsulated in the MIME				
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)				
	→ ACK					
	Conversation					
	REL	→		→	BYE(REL)	
	RLC	+		+	200 OK BYE(RLC)	

TP410004	SIP reference: RFC 32	61 [4]	ISUP reference: Q.1912.5 [1], clauses B.6, B.7, Q.732 [i.4], clause 2.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion					
SIP selection						
criteria						
ISUP selection						
criteria			\R.#			
SIP parameter	"Call diversion may occur" received in ACM Verify that a call can be successfully established, if diversion may occur. The ACM indicates that "call diversion may occur" in the optional backward call indicators. The following CPG contains the generic notification indicator set to "call is diverting", the call diversion information and the redirection number, if diversion occurs. The CPG (progress) contains CV_redirection_reason in call diversion information and also Redirection number. The CPG (alerting) is coded as if it has been mapped from ACM, with RnNbRes parameter (optional). I-MCGF interworking. 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body					
values	183 Session Progress: Content-Type: application/ISUP; CPG encapsulated in the MIME body					
ISUP parameter values	ACM: BCI Called party status indicator "subscriber free", Optional backward call indicator: "Call diversion may occur" CPG: Event information=progress, Call diversion information; Generic notification; Redirection number CPG: Event information=alerting					
Comments	SIP-I		SUT	ISUP		
	INVITE(IAM)	→	→	IAM		
	180 Ringing(ACM)	+	+	ACM(free)		
	183 Session Progress(CPG)	+	+	CPG		
	183 Session Progress(CPG)	+	+	CPG(alerting)		
	200 OK INVITE(ANM)	+	+	ANM		
	ACK	→				
		Conv	ersation			
	BYE(REL)	→	→	REL		
	200 OK BYE(RLC)	+	+	RLC		

CV_redirection_reason TP410003, TP410004					
VA_1	No reply				
VA 2	Deflection during alerting				

TP410005	SIP reference: RFC	3261	[4]		ISUP reference: 2.5 [1], clauses B.6, B.7, 732 [i.4], clause 2.4.2		
TSS reference	ISUP-SIP-ISUP/SS/Call Div	ersion					
SIP selection							
criteria							
ISUP selection							
criteria —							
Test purpose	Multiple diversions -Verify diversion occur	Multiple diversions -Verify that a call can be successfully established, if multiple diversion occur					
	Several messages each cor	ntaining	the call di	version info	rmation are received, as if		
	multiple forwardings have of						
	The CV_redirection_reaso						
	The Redirection number res	striction	parameter	is passed or	l.		
	O-MCGF interworking.						
SIP parameter		tent-Ty	pe: applicat	ion/ISUP; AC	CM encapsulated in the MIME		
values	body	_					
		tent- I y	pe: applicat	ion/ISUP; Cl	PG encapsulated in the MIME		
		body 180 Ringing: Content-Type: application/ISUP; CPG encapsulated in the MIME body					
ISUP parameter	ACM: BCI Called party stat				sulated in the MIME body		
values	Generic notification	us mai	cator No in	dication			
values	Call diversion inform	ation R	edirection r	eason uncon	ditional		
	Redirection number	allonin	edirection i	eason uncon	lational		
	CPG1: Event information=p	rogress	S				
	Generic notification	. 0 9. 0 0					
	Call diversion inform	ation R	dedirection r	eason CV_re	edirection_reason		
	Redirection number						
	Redirection number	restrict	ion				
	CPG2: Event information=a	lerting,		number res			
Comments	ISUP		SUT		SIP-I		
	IAM	→		→	INVITE(IAM)		
	ACM(no indication)	+		+	183 Session Progress(ACM)		
	CPG1 ← 183 Session						
	CPG2(alerting)	+		←	180 Ringing(CPG)		
	ANM	+		←	200 OK INVITE(ANM)		
			<u> L</u>	→	ACK		
			Conversa				
	REL	→		→	BYE(REL)		
	RLC ← 200 OK BYE(RLC)						

TP410006	SIP reference: RFC 3261 [4]		SUP reference:					
			.5 [1], clauses B.6, B.7, 32 [i.4], clause 2.4.2					
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Multiple diversions -Verify that a call can be successfully established, if multiple diversion occur							
	Several messages each containing the call of	liversion inforn	nation are received, as if					
	multiple forwardings have occurred.							
	The CV_redirection_reason is used as redirection_reason	ection reason.						
	The Redirection number restriction paramete	r is passed on.						
	I-MCGF interworking.							
SIP parameter	183 Session Progress: Content-Type: applica	ation/ISUP; ACN	I encapsulated in the MIME					
values	body							
	183 Session Progress: Content-Type: applica	ation/ISUP; CPC	encapsulated in the MIME					
	body	000						
IOUD	180 Ringing: Content-Type: application/ISUP		lated in the MIME body					
ISUP parameter values	ACM: BCI Called party status indicator "No i Generic notification	ndication						
values	Call diversion information Redirection	roocon uncondi	itional					
	Redirection number	reason uncond	lional					
	CPG: Event information=progress							
	Generic notification							
	Call diversion information Redirection reason CV_redirection_reason							
	Redirection number							
	Redirection number restriction							
	CPG: Event information=alerting, Redirection	number restric	tion					
Comments	SIP-I	SUT	ISUP					
	INVITE(IAM) →	→	IAM					
	183 Session Progress(ACM) ←	+	ACM(no indication)					
	183 Session Progress(CPG) ←	+	CPG1					
	180 Ringing(CPG) ←	+	CPG2(alerting)					
	200 OK INVITE(ANM)	+	ANM					
	ACK →							
		ersation						
	BYE(REL) →	→	REL					
	200 OK BYE(RLC)	+	RLC					

CV_redirection_reason, TP410005, TP410006				
VA_1	No reply			
VA_2	_2 Deflection during alerting			
VA_3	User busy			
VA_4	Unconditional			
VA_5	Deflection immediate response			

TP410007	SIP reference: RFC	3261 [4]	Q.1912	ISUP reference: .5 [1], clauses B.6, B.7,			
TSS reference	ISUP-SIP-ISUP/SS/Call Dive	roion	Q.732	! [i.4], clause 2.5.2.2.1			
SIP selection	130F-31F-130F/33/Call Dive	3151011					
criteria							
ISUP selection							
criteria							
Test purpose	Notification procedures for a diverting call - after the diverting 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 res	triction paramet	er (in ACM/CDC	2/ANM/CON)			
	O-MCGF interworking.	triction paramet	ei (iii Acivi/ci c	B/ANW/CON).			
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP; IAM encapsulated in the MIME body 200 OK INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP; ANM encapsulated in the MIME body						
ISUP parameter	IAM: Redirecting number, O		nber. Redirectio	n information			
values	ANM: Redirection address re		,				
Comments	ISUP	SI	JT	SIP-I			
	IAM	→	→	INVITE(IAM)			
	CASE A						
	ACM	+	+	183 Session Progress(ACM,no indication)			
	CPG	+	←	180 Ringing(CPG,alerting)			
	ANM	←	+	200 OK INVITE(ANM)			
			→	ACK			
	CASE B						
	CON	+	+	200 OK INVITE(CON)			
			→	ACK			
		Conver	sation				
	REL	→	→	BYE(REL)			
	RLC	+	+	200 OK BYE(RLC)			
NOTE: Altered in	n Gateways.	•	•				

TP410008	SIP reference: RF	C 3261	[4]		I	SUP reference:		
		·				5 [1], clauses B.6, B.7, [i.4], clause 2.5.2.2.1		
TSS reference	ISUP-SIP-ISUP/SS/Call D	iversion	l					
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Notification procedures for a diverting call - after the diverting exchange							
	Verify that the IUT can su	ccessfully	pass on in	both dire	ctions	s (on the leg after the		
	diversion) all the diversion							
						is passed on in the forward		
	direction:					•		
	redirecting number (
	original called number		ote);					
	redirection information							
	It has to be checked that the following signalling information is passed on in the backwa							
	direction:							
	redirection number r	estrictio	n paramete	r (in ACM/	/CPG	G/ANM/CON).		
	I-MCGF interworking.							
SIP parameter	INVITE: Content-Type: mi		ixed, Conte	nt-Type: a	applic	cation/ISUP; IAM		
values	encapsulated in the MIME				. T	and a second and the		
			iitipart/mixe	a, Conten	t- i yp	e: application/ISUP ; ANM		
SUP parameter	encapsulated in the MIME IAM: Redirecting number,		ممالمط مینصا	or Dodin	o otio	n information		
values	ANM: Redirecting number,			ber, Redire	ectioi	n information		
Comments	SIP-I	1680100	SU1	-		ISUP		
Comments	INVITE(IAM)	→	301		→	IAM		
	CASE A	7			7	IAW		
	183 Session	+			+	ACM(no indication)		
	Progress(ACM)	_			~	ACIVI(110 Indication)		
	180 Ringing(CPG)	+			+	CPG(alerting)		
	200 OK INVITE(ANM)	+			-	ANM		
	ACK	→ ·			_	ANIVI		
	CASE B	7						
		+			+	CON		
	200 OK INVITE(CON) ACK	→			~	CON		
	AUN	7	Convers	ation				
	DVE/DEL)	→	Convers	atiOH		DEI		
	BYE(REL)	7			<u>→</u>	REL RLC		
NOTE: Alta !!	200 OK BYE(RLC)				_	IKLU		
NOTE: Altered i	n Gateways.							

TP410009	SIP reference: RFC	3261	[4]	-	SUP reference:			
		Q.1912.5 [1], clauses B.6, B.7, Q.731 [i.2], clause 3.5.2.4.1						
TSS reference	ISUP-SIP-ISUP/SS/Call Div	ersion/			[···], ·······			
SIP selection criteria								
ISUP selection criteria	PICS 10/1 AND PICS 1/7							
Test purpose	Original called number in the outgoing international gateway Verify that the outgoing international gateway checks and manipulates the original called number according to the procedures as defined for CLIP: Discarding the original called number if case of bilateral agreements. The PTC will send an IAM with OriCdNb.							
SIP parameter	INVITE: Content-Type: mult	tipart/m	ixed, Conten	t-Type: applic	ation/ISUP; IAM containing			
values	an Original called number e	ncapsu	lated in the I	/IME body				
ISUP parameter	IAM: No original called num	ber pre	esent					
values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK →							
			Conversa	tion				
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP410010	SIP reference: RF	C 3261	[4]	Q.1912	ISUP reference: .5 [1], clauses B.6, B.7,					
TSS reference	Q.731 [i.2], clause 4.5.2.1.1									
SIP selection criteria	ioer on loor/oo/can b									
ISUP selection criteria	PICS 1/7									
Test purpose	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: Converting the original called number to international format with transparent transferral of address presentation restricted indicator. The PTC will send an IAM with a national (significant) OriCdNb.									
SIP parameter					cation/ISUP; IAM containing					
values	an Original called number				e MIME body					
ISUP parameter values	IAM: Original called numb	er "Interi	national num	ber"						
Comments	SIP-I		SUT	•	ISUP					
	INVITE(IAM)	→		→	IAM					
	180 Ringing(ACM)	+		+	ACM					
	200 OK INVITE(ANM)	+		+	ANM					
	ACK	→								
			Conversa	ation						
	BYE(REL)	→		→	REL					
	200 OK BYE(RLC)	+		+	RLC					

TP410011	SIP reference: RFC	3261	[4]		ISUP reference:				
				Q.191	2.5 [1], clauses B.6, B.7,				
		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								
					nanipulates the original call	led			
	number according to the pr								
	Discarding the original								
	The PTC will send an IAM v	with an	"address no	t available" C	DriCdNb.				
SIP parameter					lication/ISUP; IAM containin	ng			
values	an Original called number of	alled n	umber enca	osulated in th	ne MIME body				
ISUP parameter	IAM: No original called num	ber pre	esent						
values									
Comments	SIP-I		SUT	•	ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK	→							
	Conversation								
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC	·			

TP410012	SIP reference: RF0	3261	[4]	ISUP reference: Q.1912.5 [1], clauses B.6, B.7, Q.731 [i.2], clause 4.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/Call Div	ersion					
SIP selection							
criteria							
ISUP selection criteria	PICS 1/8						
Test purpose	Original called number in the incoming international gateway Verify that the incoming international gateway checks and manipulates the original called number according to the procedures as defined for CLIP. Applicable tests: Converting the original called number to national format, if necessary (own country code).						
SIP parameter values	INVITE: Content-Type: mul an Original called number of				ation/ISUP ; IAM containing MIME body		
ISUP parameter values	IAM: Original called numbe	r "Natio	nal number"				
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		→	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	→					
			Conversation				
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP410013	SIP reference: RF	SIP reference: RFC 3261 [4]			1912.	SUP reference: 5 [1], clauses B.6, B.7,	
				C	2.731	[i.2], clause 4.5.2.1.1	
TSS reference	ISUP-SIP-ISUP/SS/Call Di	version					
SIP selection							
criteria							
ISUP selection	PICS 10/2 AND PICS 1/7						
criteria							
Test purpose	Redirecting number in th						
	Verify that the outgoing into					nipulates the redirecting	
	number according to the p	rocedur	es as define	ed for CLI	P:		
	Discarding the redirect	ing nur	nber if case	of bilate	ral agı	reements.	
SIP parameter	INVITE: Content-Type: mu	ltipart/m	ixed, Conte	nt-Type: a	applic	ation/ISUP; IAM containing a	
values	Redirecting number encap	sulated	in the MIME	body			
ISUP parameter	IAM: No Redirecting numb	er prese	ent				
values							
Comments	SIP-I		SUT			ISUP	
	INVITE(IAM)	→			1	IAM	
	180 Ringing(ACM)	+			+	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
	ACK →						
			Convers	ation			
	BYE(REL)	→			→	REL	
	200 OK BYE(RLC)	+			+	RLC	

TP410014	SIP reference: RF0	3261	[4]	Q.1912.	SUP reference: 5 [1], clauses B.6, B.7,					
TSS reference	ISUP-SIP-ISUP/SS/Call Div	Q.731 [i.2], clause 4.5.2.1.1								
SIP selection criteria	ioci dii ioci regredii bit	70101011								
ISUP selection criteria	PICS 1/7									
Test purpose	Redirecting number in the outgoing international gateway Verify that the outgoing international gateway checks and manipulates the redirecting number according to the procedures as defined for CLIP: Discarding the redirecting number, if the address is marked not available. The PTC will send an IAM with an "address not available" RgNb.									
SIP parameter					cation/ISUP ; IAM containing a					
values	Redirecting number encaps	ulated	in the MIME	oody	-					
ISUP parameter	IAM: No Redirecting number	er prese	ent							
values										
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	→		→	IAM					
	180 Ringing(ACM)	+		+	ACM					
	200 OK INVITE(ANM)	+		+	ANM					
	ACK									
			Conversa	tion						
	BYE(REL)	→		→	REL					
	200 OK BYE(RLC)	+		+	RLC					

TP410015	SIP reference: RFC	3261	[4]		Į.	SUP reference:					
				Q.	1912.	5 [1], clauses B.6, B.7,					
					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	SUP-SIP-ISUP/SS/Call Diversion									
SIP selection criteria											
ISUP selection criteria	PICS 1/7										
Test purpose	Redirecting number in the outgoing international gateway Verify that the outgoing international gateway checks and manipulates the redirecting number according to the procedures as defined for CLIP: Converting the redirecting number to international format with transparent transferral of address presentation restriction indicator. The PTC will send an IAM with a national significant RgNb.										
SIP parameter						ation/ISUP; IAM containing a					
values	Redirecting number "Nation										
ISUP parameter	IAM: Redirecting number "Ir					•					
values											
Comments	SIP-I		SU	Γ		ISUP					
	INVITE(IAM)	→			→	IAM					
	180 Ringing(ACM)	←			+	ACM					
	200 OK INVITE(ANM)	←			+	ANM					
	ACK	→									
			Convers	ation							
	BYE(REL)	→			→	REL					
	200 OK BYE(RLC)	+		•	+	RLC					

TP410016	SIP reference: RFC	_	[4]	ISUP reference: Q.1912.5 [1], clauses B.6, B.7, Q.732 [i.4], clause 2.5.2.3, Q.731 [i.2], clause 3.5.2.3						
TSS reference	ISUP-SIP-ISUP/SS/Call Div	ersion/								
SIP selection criteria										
ISUP selection criteria	PICS 1/8									
Test purpose	Redirecting number in the incoming international gateway Verify that the incoming international gateway checks and manipulates the redirecting number according to the procedures as defined for CLIP: Converting the redirecting number to national format, if necessary (own country code). The PTC will send an IAM with RgNb.									
SIP parameter values	INVITE: Content-Type: mult a Redirecting number "Inter									
ISUP parameter values	IAM: Redirecting number "n			ncapsulateu	III U	ne wiiwic body				
Comments	SIP-I		SUT	'	I	SUP				
	INVITE(IAM)	→		→	.	AM				
	180 Ringing(ACM)	+		+	- /	ACM				
	200 OK INVITE(ANM)	+		+	1	NMA				
	ACK	→								
			Conversa	ation						
	BYE(REL)	→		→	F	REL				
	200 OK BYE(RLC)	+		+	F	RLC				

TP410017	SIP reference: RF	FC 3261	[4]	Q.1912 Q.732	ISUP reference: .5 [1], clauses B.6, B.7, 2 [i.4], clause 2.5.2.3, 1 [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 AND 10/4								
Test purpose	Redirecting number in the incoming international gateway Verify that the incoming international gateway checks and manipulates the redirecting number according to the procedures as defined for CLIP: Adding a prefix to an international redirecting number. The PTC will send an IAM with RgNb.								
SIP parameter					cation/ISUP; IAM containing a				
values	Redirecting number encap	osulated	in the MIME I	oody					
ISUP parameter values	IAM: Redirecting number								
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK								
			Conversa	tion					
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP410018	SIP reference: RF		[4]	Q.7	ISUP reference: [2.5 [1], clauses B.6, B.7, [32 [i.4], clause 2.5.2.4, [31 [i.2], clause 3.5.2.4					
TSS reference	ISUP-SIP-ISUP/SS/Call D	iversion								
SIP selection criteria										
ISUP selection criteria	PICS 10/5 AND PICS 1/8	PICS 10/5 AND PICS 1/8								
Test purpose	Verify that the incoming in number according to the p discarding the redirect	Redirection number in the incoming international gateway Verify that the incoming international gateway checks and manipulates the redirection number according to the procedures defined for COLP: discarding the redirection number in case of bilateral agreements; removes the redirection number restriction parameter.								
SIP parameter values	number encapsulated in the 200 OK INVITE: Content-1	ne MIMÉ Γype: mι	body lltipart/mixed	d, Content-T	CM containing a Redirection Type: application/ISUP; ANM capsulated in the MIME body					
ISUP parameter values	ACM: Called party status= Generic notification Call diversion inforr No Redirection num ANM: No Redirection num	nation R nber	edirection re		nditional					
Comments	ISUP		SUT		SIP-I					
	IAM	→		→	INVITE(IAM)					
	ACM(no indication)	+		+	183 Session Progress(ACM)					
	CPG	+		+	180 Ringing(CPG)					
	ANM	+		+	200 OK INVITE(ANM)					
				→	ACK					
			Conversa	tion						
	REL	→		→	BYE(REL)					
	RLC	+		+	200 OK BYE(RLC)					

TP410019	SIP reference: RF0	3261	[4]		ISUP reference:					
					912.5 [1], clauses B.6, B.7,					
					.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										
ISUP selection criteria	PICS 1/7	PICS 1/7								
Test purpose	Redirection number in the									
					d manipulates the redirection					
	number according to the p									
		ion nui	mber to nati	onal forma	at, if necessary (own country					
	code):		m / -ti							
	1. the PTC will provide the				nternational RnNb with own CC.					
SIP parameter					ACM containing a Redirection					
values	number "International numl									
ISUP parameter	ACM: Called party status=r			T UIC WIIIVIL	_ body					
values	Generic notification	io iriaici	ation							
Valuoo	Call diversion inform	nation R	edirection re	eason unc	onditional					
	Redirection number									
Comments	ISUP		SUT		SIP-I					
	IAM	→		→	INVITE(IAM)					
	ACM(no indication)	+		+	183 Session Progress(ACM)					
	CPG	+		+	180 Ringing(CPG)					
	ANM	+		+	200 OK INVITE(ANM)					
				→	ACK					
			Conversa	tion						
	REL	→		→	BYE(REL)					
	RLC	+		+	200 OK BYE(RLC)					

TP410020	SIP reference: RFC	3261	[4]	Q.73	ISUP reference: 2.5 [1], clauses B.6, B.7, 32 [i.4], clause 2.5.2.3, 31 [i.2], clause 3.5.2.3				
TSS reference	ISUP-SIP-ISUP/SS/Call Div	ersion							
SIP selection criteria									
ISUP selection criteria	PICS 1/8								
Test purpose	Redirection number in the incoming international gateway Verify that the incoming international gateway checks and manipulates the redirection number according to the procedures defined for COLP: Converting the redirection number to international format.								
SIP parameter values		183 Session Progress: Content-Type: application/ISUP; ACM containing a Redirection number "National number" encapsulated in the MIME body							
ISUP parameter		ACM: Called party status=no indication							
values									
Comments	ISUP		SUT		SIP-I				
	IAM	→		→	INVITE(IAM)				
	ACM(no indication)	+		+	183 Session Progress(ACM)				
	CPG	+		+	180 Ringing(CPG)				
	ANM	+		+	200 OK INVITE(ANM)				
				→	ACK				
			Conversati	on					
	REL	→		→	BYE(REL)				
	RLC	+	_	+	200 OK BYE(RLC)				

TP410021	SIP reference: RFC	3261	[4]		ISUP reference: 012.5 [1], clauses B.6, B.7, 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	Redirection number in the outgoing international gateway Verify that the outgoing international gateway checks and manipulates the redirection number according to the procedures defined for COLP: Adding a prefix to an international redirection number. The PTC will provide the necessary stimulus.ACM with CDInf, GenNot = "call is diverting" and an international RnNb with foreign country code.								
SIP parameter	183 Session Progress: Con	183 Session Progress: Content-Type: application/ISUP; ACM containing a Redirection							
values	number "International numb	er" end	apsulated in	the MIME	body				
ISUP parameter	ACM: Called party status=n	o indic	ation						
values	Generic notification Call diversion inform Redirection number			ason unco	onditional				
Comments	ISUP		SUT		SIP-I				
	IAM	→		→	INVITE(IAM)				
	ACM(no indication)	+		+	183 Session Progress(ACM)				
	CPG	+		+	180 Ringing(CPG)				
	ANM	+		+	200 OK INVITE(ANM)				
				→	ACK				
			Conversat	ion					
	REL	→		→	BYE(REL)				
	RLC	+	_	+	200 OK BYE(RLC)				

5.3.11 CONF

TP411001	SIP reference: RFC 3261 [4]	Q.1912	5 [1],	erence: clause B.14, lause 1.6.15		
TSS reference	ISUP-SIP-ISUP/SS/CONF						
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Generic notification transfer "confe	rence e	stablished" and "otl	ner pai	rty 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.						
SIP parameter	INFO/INVITE: Content-Type: multi	part/mix	ed, Content-Type:	applica	ation/ISUP ; CPG		
values	encapsulated in the MIME body	-		•	•		
ISUP parameter	CPG: Generic notification: confere	nce esta	ablished				
values	CPG: Generic notification: other pa	arty add					
Comments	ISUP		SUT	SIP			
	IAM	→	-		/ITE(IAM)		
	ACM	+			Ringing(ACM)		
	ANM	+	€		OK INVITE(ANM)		
			-	AC	K		
			Conversation				
	CASE A						
	CPG(conference established)	→	-		/ITE(CPG,sendrecv)		
			•		OK INVITE(sendrecv)		
			-	AC	K		
	CASE B						
	CPG(conference established)	→	-		O(CPG)		
			•	200	OK (INFO)		
	CASE A						
	CPG(other party added)	→	-		/ITE(CPG,sendrecv)		
			•) OK INVITE(sendrecv)		
			-	AC	K		
	CASE B						
	CPG(other party added)	→	-		O(CPG)		
			•	200	OK (INFO)		
	REL	→	-		E(REL)		
	RLC	+		200	OK BYE(RLC)		

TP411002	SIP reference: R	FC 32	261 [4]		ISUP reference: Q.1912.5 [1], clause B.14, Q.734 [i.7], clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/CON	IF			
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Generic notification trans	sfer "co	onference estab	lished	d" and "other party added"
	To verify that the IUT car	n succ	essfully transfer	/deliv	er the required notifications in/from the
	CPG message:		•		·
	1. Assist a call set up fro				
			conference esta	ablish	ed" is received in the CPG from
	conferee at the ISUP				
	3. Check the notification	ı "othe	r party added" i	n the	CPG.
OID 1	I-MGCF interworking.				
SIP parameter				Conte	ent-Type: application/ISUP ; CPG
values	encapsulated in the MIM CPG: Generic notification			ll	
ISUP parameter values				nea	
Comments	CPG: Generic notification	1: Othe	SUT		ISUP
Comments	INVITE(IAM)	→	301	→	IAM
		+		+	ACM
	180 Ringing(ACM) 200 OK INVITE(ANM)	+		+	ANM
	ACK	→			ANW
	ACK	7	Conversation		
	CASE A		Conversation		
	INFO(CPG)	→		→	CPG(conference established)
	200 OK INFO	+		_	CFG(contenence established)
	CASE B	1			
	INVITE (CPG)	→		→	CPG(conference established)
	200 OK INVITE	+		Ť	er e(cornerence established)
	ACK	→			
	CASE A	-			
	INFO(CPG)	→		→	CPG(other party added)
	200 OK INFO	+		1	
	CASE B				
	INVITE (CPG)	→		→	CPG(other party added)
	200 OK INFO	+			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		+	RLC

TP411003	SIP reference: RFC 3261 [4]		Q.19 ²	12.5 [reference: 1], clause B.14,], clause 1.6.15				
TSS reference	ISUP-SIP-ISUP/SS/CONF		•		-				
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Generic notification transfer "conference	e estal	blished" and "is	olated	d"				
	To verify that the IUT can successfully to CPG message: 1. Assist a call set up from ISUP to SIF 2. Check that the notification "conferer conferee at the SIP-I; 3. Check the notification "isolated" in the O-MGCF interworking.	P-I; nce est	tablished" is red						
SIP parameter	INFO/INVITE: Content-Type: multipart/i	mived	Content-Type:	annli	cation/ISLID : CDG				
values	encapsulated in the MIME body	ilixeu,	Content-Type.	appli	caudil/130F , CFG				
ISUP parameter	CPG: Generic notification: conference	etablic	shad						
values	CPG: Generic notification: isolated	JOIGUIR	JIIGU						
Comments	ISUP		SUT		SIP-I				
Comments	IAM	→	001	→	INVITE(IAM)				
	ACM	+		+	` ` `				
	ANM	+		+	200 OK INVITE(ANM)				
	AINIVI	+		+	ACK				
	Conversation								
	CASE A								
	CPG(conference established)	→		→	INFO(CPG)				
	CFG(contenence established)	-		+	200 OK INFO				
				_	200 OK INI O				
	CPG(isolated)	→		→	INFO(CPG)				
	CFG(Isolated)	-		+	200 OK INFO				
					200 OK INFO				
	DEL	→		→	DVE(DEL)				
	REL RLC	+		+	BYE(REL) 200 OK BYE(RLC)				
	CASE B	_			200 OK BTE(RLC)				
		→			INIVITE (CDC conditions)				
	CPG(conference established)	7		→	INVITE(CPG,sendrecv) 200 OK				
				_	INVITE(sendrecv)				
		+		→	ACK				
				7	ACK				
	CDC/inclated)	→		 _	INIVITE (CDC conditions)				
	CPG(isolated)	7		→	INVITE(CPG,sendrecv)				
				+	200 OK				
		+			INVITE(sendrecv)				
		+		→	ACK				
	DEL			+-	DVE(DEL)				
	REL	→		→	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP411004	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause B.14, 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	lished	d" and "isolated"		
	CPG message: 1. assist a call set up fro	m SIF tion "o	P-I to ISUP; conference esta	ablishe	ed" is received in the CPG from		
	I-MGCF interworking.						
SIP parameter	INFO/INVITE: Content-Ty	/pe: n	nultipart/mixed,	Conte	ent-Type: application/ISUP; CPG		
values	encapsulated in the MIME	bod	y		,		
ISUP parameter	CPG: Generic notification	: conf	erence establis	hed			
values	CPG: Generic notification	: isola	ated				
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		→	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	→			7 (I VIVI		
	AOR		Conversation				
	CASE A		Conversation				
	INFO(CPG)	→		→			
	200 OK INFO	7		7	CPG(conference established)		
	200 OK INFO	~					
	INIEO(ODO)	_		_	000/: 1 / 1)		
	INFO(CPG)	→		→	CPG(isolated)		
	200 OK INFO	+					
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	+		+	RLC		
	CASE B						
	INVITE (CPG)	→		→	CPG(conference established)		
	200 OK INVITE	+					
	ACK	→					
	INVITE (CPG)	→		→	CPG(isolated)		
	200 OK INVITE	+		† -			
	ACK	→	1	1			
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	+		É	RLC		
	ZOO OK BIL(KLO)	+	-	+			
		<u> </u>	I .				

TP411005	SIP reference: RFC 3261 [4	i]		912.5	reference: [1], clause B.14, 7], clause 1.6.15			
TSS reference	ISUP-SIP-ISUP/SS/CONF							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Generic notification transfer "confere	ence estab	olished", "isola	ted" a	nd "reattached"			
	To verify that the IUT can successfu CPG message: 1. Assist a call set up from ISUP to 2. Check that the notification "confeconferee at SIP-I; 3. Check the notification "reattache O-MGCF interworking.	SIP-I; erence est	ablished" is re					
SIP parameter values	INFO/INVITE: Content-Type: multipa encapsulated in the MIME body	art/mixed,	Content-Type	appli	cation/ISUP ; CPG			
ISUP parameter	CPG: Generic notification: conference	ce establis	shed					
values	CPG: Generic notification: isolated CPG: Generic notification: reattache							
Comments	ISUP		SUT		SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	-		+	· /			
	ANM	+		+	200 OK INVITE(ANM)			
	7 11 1111			→	ACK			
	Conversation							
	CASE A		Conversation	<u>'</u>				
	CPG(conference established)	→		→	INFO(CPG)			
	CPG(conference established)	7		+	200 OK INFO			
				~	200 OK INFO			
	000(: 1 (1)			_	INEC(ODO)			
	CPG(isolated)	→		→	INFO(CPG)			
				+	200 OK INFO			
	CPG(reattached)	→		→	INFO(CPG)			
	CFG(realiached)	7		+	200 OK INFO			
					200 OK INFO			
	REL	→		→	BYE(REL)			
	RLC	7		+	200 OK BYE(RLC)			
	CASE B	_		_	ZOU ON BTE(RLC)			
				_	INDUITE (ODO dr)			
	CPG(conference established)	→			INVITE(CPG,sendrecv)			
				+	200 OK			
					INVITE(sendrecv			
				→	ACK			
	CDC(incloted)				INIVITE/ODO serializada			
	CPG(isolated)	→		→	INVITE(CPG,sendonly) 200 OK			
					INVITE(recvonly)			
				→	ACK			
	CPG(reattached)	→		→	INFO(CPG,sendrecv)			
	Or Greatiached)	7		+	200 OK			
					INVITE(sendrecv)			
				→	ACK			
				<u> </u>				
	REL	→		→	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP411006	SIP reference: RI	FC 326	1 [4]		ISUP reference:				
					Q.1912.5 [1], clause B.14, Q.734 [i.7], clause 1.6.15				
TSS reference	ISUP-SIP-ISUP/SS/CONF								
SIP selection									
criteria									
ISUP selection									
criteria	1								
Test purpose	Generic notification transf	er "con	ference estab	olished	d", "isolated" and "reattached"				
	To verify that the IUT can	succes	sfully transfe	r/deliv	er the required notifications in/from the				
	CPG message:		-		·				
	1. Assist a call set up fro								
		ation "co	onference est	ablish	ed" is received in the CPG from				
	conferee at SIP-I;								
	3. Check the notification	"reatta	ched" in the C	PG.					
OID 1	I-MGCF interworking.			<u> </u>	. T				
SIP parameter			itipart/mixed,	Conte	ent-Type: application/ISUP ; CPG				
values ISUP parameter	encapsulated in the MIME CPG: Generic notification		ongo catablia	hod.					
values	CPG: Generic notification CPG: Generic notification			ned					
values	CPG: Generic notification CPG: Generic notification								
Comments	SIP-I	Tealla	SUT		ISUP				
Comments	INVITE(IAM)	→	301	→	IAM				
	180 Ringing(ACM)	+							
	200 OK INVITE(ANM)	+		÷	ANM				
	ACK	À		\	ANN				
	Conversation								
	CASE A								
	INFO(CPG)	→		→	CPG(conference established)				
	200 OK INFO	+		† -	er e(comercines conacilement)				
	INFO(CPG)	→		→	CPG(isolated)				
	200 OK INFO	+							
	INFO(CPG)	→		→	CPG(reattached)				
	200 OK INFO	+							
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				
	CASE B								
	INVITE(CPG,sendrecv)	→		→	CPG(conference established)				
	200 OK	+							
	INVITE(sendrecv)								
	ACK	→							
	INVITE(CPG,sendonly)	→		→	CPG(isolated)				
	200 OK	-							
	INVITE(recvonly)								
	ACK	→							
	INIVITE/CDC conditions			_					
	INVITE(CPG,sendrecv)	→		→	CPG(reattached)				
	200 OK	 							
	INVITE(sendrecv) ACK	→		+					
	TON	+*+		+					
		→		→	DEI				
	IRVE(REL)								
	BYE(REL) 200 OK BYE(RLC)	-		+	REL RLC				

TP411007	SIP reference: RFC 3261 [4]		Q.191	2.5 [reference: 1], clause B.14,], clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/CONF				<u>,, c.a.a.c</u>
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Generic notification transfer "conference disconnected"	e estal	olished", "other	party	added" and "other party
	To verify that the IUT can successfully CPG message: 1. assist a call set up from ISUP to SI 2. check the notification "other party of O-MGCF interworking.	P-I;			d notifications in/from the
SIP parameter values	INFO/INVITE: Content-Type: multipart/ encapsulated in the MIME body	mixed,	Content-Type:	appli	cation/ISUP ; CPG
ISUP parameter	CPG: Generic notification: conference	establis	shed		
values	CPG: Generic notification: other party a	added			
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
				→	ACK
			Conversation	1	
	CASE A				
	CPG(conference established)	→		→	INFO(CPG)
				+	200 OK INFO
	CPG(other party added)	→		→	INFO(CPG)
				+	200 OK INFO
	CPG(other party disconnected)	→		→	INFO(CPG)
	or expansy and expansion			+	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)
	CASE B	—		Ì	200 ON BTE(NEO)
	CPG(conference established)	→		→	INVITE(CPG,sendrecv)
	or o(contenence established)			+	200 OK INVITE(sendrecv)
				→	ACK
	CPG(other party added)	→		→	INVITE(CPG, sendrecv)
				+	200 OK INVITE(sendrecv)
				→	ACK
	CPG(other party disconnected)	→		→	INVITE(CPG, sendrecv)
				+	200 OK INVITE (sendrecv)
				→	ACK
	REL	→		→	BYE(REL)
	RLC	+		7	200 OK BYE(RLC)

TP411008	SIP reference: RI	FC 32	61 [4]		ISUP reference: Q.1912.5 [1], clause B.14, 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", "other party added" and "other party 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	INFO/INVITE: Content-Ty	/pe: m	nultipart/mixed,	Conte	ent-Type: application/ISUP ; CPG			
values	encapsulated in the MIME							
ISUP parameter values	CPG: Generic notification CPG: Generic notification CPG: Generic notification	: othe	r party added					
Comments	SIP-I	. Othe	SUT	ecieu	ISUP			
Comments	INVITE(IAM)	→	301	→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		÷	ANM			
	ACK	→		_	ANIVI			
	Conversation							
	CASE A							
	INFO(CPG)	→		→	CPG(conference established)			
	200 OK INFO	′		+-	Or O(cornerence established)			
	200 01(1141 0							
	INFO(CPG)	→		→	CPG(other party added)			
	200 OK INFO	+		+-	Cr C(other party added)			
	200 01(11(1)							
	INFO(CPG)	→		→	CPG(other party disconnected)			
	200 OK INFO	+		†-	er e(emer party diecermicoted)			
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		+	RLC			
	CASE B			† <u> </u>	T.E.O			
	INVITE(CPG,sendrecv)	→		→	CPG(conference established)			
	200 OK	+						
	INVITE(sendrecv)							
	ACK	→						
	INVITE(CPG,sendrecv)	→		→	CPG(other party added)			
	200 OK	+		† -	or e(orner party added)			
	INVITE(sendrecv)							
	ACK	→						
	INVITE(CPG,sendrecv)	→		→	CPG(other party disconnected)			
	200 OK	+						
	INVITE(sendrecv)							
	ACK	→						
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		+	RLC			
		1		†				
	1	<u> </u>	<u>I</u>	1				

TP411009	SIP reference: RFC 3261 [4]	Q.1912.5	Preference: [1], clause B.14, 7], clause 1.6.15						
TSS reference	ISUP-SIP-ISUP/SS/CONF									
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	Generic notification transfer "confere									
	To verify that the IUT can successful CPG message:	ly transie	r/deliver the require	ed notifications in/from the						
	1. Assist a call set up from ISUP to	SIP-I:								
	2. Check that the notification "confe		ablished" is receive	ed in the CPG from						
	conferee at ISUP;									
	3. Release the conference.									
	O-MGCF interworking.									
SIP parameter	INFO/INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP; CPG									
values	encapsulated in the MIME body									
ISUP parameter values	CPG: Generic notification: conference	e establis	shed							
Comments	ISUP		SUT	SIP-I						
	IAM	→	→	INVITE(IAM)						
	ACM	+	+	3 3(- /						
	ANM	+	+							
			→	ACK						
			Conversation							
	CASE A									
	CPG(conference established)	→	→	INFO(CPG)						
			+	200 OK INFO						
	REL	→	→	BYE(REL)						
	RLC	+	+	200 OK BYE(RLC)						
	CASE B	\rightarrow		IN VITE (ODO						
	CPG(conference established)	→	→	=(3: 333:13:331)						
			+	_00 0.1						
				INVITE(sendrecv)						
			→	ACK						
	DEL			DVE(DEL)						
	REL	→	→	(/						
	RLC	7	+	200 OK BYE(RLC)						

TP411010	SIP reference: R	FC 32	61 [4]		ISUP reference: Q.1912.5 [1], clause B.14, Q.734 [i.7], clause 1.6.15		
TSS reference	ISUP-SIP-ISUP/SS/CON	F			Q.734 [1.7], Clause 1.0.13		
SIP selection	1001 011 1001 700/0011	<u> </u>					
criteria							
ISUP selection							
criteria							
Test purpose	Generic notification trans	fer "co	nference establ	ished	d", and disconnect the conference		
	To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message: 1. Assist a call set up from SIP-I to ISUP; 2. Check that the notification "conference established" is received in the INFO(CPG) from conferee at SIP-I; 3. Release the conference.						
SIP parameter	I-MGCF interworking.	vpe. m	ultipart/mixed (Conte	ent-Type: application/ISUP; CPG		
values	encapsulated in the MIME			01110	Type: application (i.e.)		
ISUP parameter	CPG: Generic notification			ned			
values							
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		→	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	→					
			Conversation				
	CASE A						
	INFO(CPG)	→		→	CPG(conference established)		
	200 OK INFO	+					
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	+		+	RLC		
	CASE B			\	IXEO		
	INVITE(CPG,sendrecv)	→		→	CPG(conference established)		
	200 OK	+		-	Of C(conference established)		
	INVITE(sendrecv)	•					
	ACK	→					
		_		1			
	7.011						
	BYE(REL)	→		→	REL		

5.3.12 ECT

TP412001	SIP reference: RFC 3261 [4]		.5 [1]	SUP reference: , clauses 5.4.3, 5.4.3.2, B.8, i.5], clause 7.5.2.1.1.1 a)		
TSS reference	ISUP-SIP-ISUP/SS/ECT						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Capability of sending a call transfer 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 ISUP FAC, in an INFO request for the active user. O-MGCF interworking.						
SIP parameter values	INFO: Content-Type: application/ISL	JP; FAC	encapsula	ted in	the MIME body		
ISUP parameter values	FAC: Generic notification=call transf number (PIXIT)	er active	e, Service a	ctivat	ion=call transfer, Call transfer		
Comments	ISUP		SUT		SIP-I		
	IAM	→		→	INVITE(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
				→	ACK		
	Conversation						
	FAC(call transfer active, CTNb)	→		→	INFO(FAC)		
				+	200 OK INFO		
	REL	→		→	BYE(REL)		
	RLC	←		+	200 OK BYE(RLC)		

TP412002	SIP reference: RF	C 3261 I	[4]		ISUP reference:			
11 412002	On reference. It	0 020.		Q.19 ²	12.5 [1], clauses 5.4.3, 5.4.3.2, B.8,			
				٠٠	Q.734 [i.7], clause 1.6.15			
TSS reference	ISUP-SIP-ISUP/SS/ECT		<u> </u>					
SIP selection criteria								
ISUP selection criteria								
Test purpose	Capability of sending the	call tra	nsfer numbe	er 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 end	capsu	lated in the Message body			
ISUP parameter values	FAC: Generic notification= number (PIXIT)	call trans	sfer active, So	ervice	activation=call transfer, Call transfer			
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	→						
		(Conversation	า				
	INFO(FAC)	→		→	FAC(call transfer active, CTNb)			
	200 OK INFO	+						
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP412005	SIP reference: RFC 3261 [4	ISUP reference: Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8, 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 the IUT is able to transfe sent in INFO request containing the O-MGCF interworking.	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-MGCE interworking						
SIP parameter	INVITE: Content-Type: multipart/mix	red, Con	tent-Type: ap	plication/ISUP; CPG				
values	encapsulated in the MIME body INFO: Content-Type: application/ISI new session with new INVITE to CT							
ISUP parameter	CPG: Event indicator=progress, Ge							
values	FAC: Generic notification=call trans number(PIXIT)	fer active	, Service acti	ivation=call transfer, Call trar	nsfer			
Comments	ISUP		SUT	SIP-I				
	IAM	→		→ INVITE(IAM)				
	ACM	+		★ 180 Ringing(ACM)				
	ANM	←		★ 200 OK INVITE(ANM)				
			- 1	→ ACK				
			onversation					
	CPG(hold)	→	- 1	→ INVITE(CPG, sendonly)				
				← 200 OK INVITE(recvonly)	y)			
				→ ACK				
	FAC(call transfer active, CTNb)	→		→ INFO(FAC)				
				€ 200 OK INFO				
				→ INVITE(sendrecv)				
				← 200 OK INVITE(sendred)	cv)			
			1	→ ACK				
	REL	→		→ BYE(REL)				
	RLC	+		€ 200 OK BYE(RLC)				
	INLO	_		2 200 ON BTE(NLC)				

TP412006	SIP reference: RFC 3261 [4]			Q.191	ISUP reference: 2.5 [1], clauses 5.4.3, 5.4.3.2, B.8, Q.734 [i.7], clause 1.6.15			
TSS reference	ISUP-SIP-ISUP/SS/CONF							
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	Verify that the IUT is able to	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 values	encapsulated in the MIME be INFO: Content-Type: applica	INVITE: Content-Type: multipart/mixed, 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=progress, Generic notification=hold FAC: Generic notification=call transfer active, Service activation=call transfer, Call transfer number(PIXIT)							
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	→						
		(Conversati	ion				
	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	→						
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		+	RLC			

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TP412009	SIP reference: RFC 3261 [4]		[1],	SUP reference: clauses 5.4.3, 5.4.3.2, B.8, .5], clause 7.5.2.1.1.1 a)			
TSS reference	ISUP-SIP-ISUP/SS/ECT		•					
SIP selection								
criteria								
SUP selection								
criteria								
Test purpose	Verify that the SUT is able to transful INFO request containing the LOP m	Loop prevention procedure – initiation Verify that the SUT is able to transfer the loop request received in an ISUP LOP in an INFO request containing the LOP message. SUT is able to transfer the loop response received in an ISUP LOP in an SIP INFO request containing the ISUP LOP message.						
SIP parameter	INVITE: Content-Type: multipart/mi.	xed. Cont	ent-Type: ap	plica	ation/ISUP : CPG			
values	encapsulated in the MIME body		, po. up	٠٠				
	INFO: Content-Type: application/IS	UP: FAC	encapsulated	d in t	the MIME body			
	INFO: Content-Type: application/IS							
ISUP parameter	CPG: Event indicator=progress, Ge				•			
values	LOP: request: Call transfer reference							
	LOP: response: Call transfer reference							
	FAC: Generic notification=call transfer active, Service activation=call transfer, Call transfer							
	number(PIXIT)				· ·			
Comments	ISUP		SUT		SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	+	1	(180 Ringing(ACM)			
	ANM	+	1	(200 OK INVITE(ANM)			
				→	ACK			
	Conversation							
	CPG(hold)	→		→	INVITE(CPG, sendonly)			
			•	(200 OK INVITE(recvonly)			
				→	ACK			
	LOP(request)	→	1	→	INFO(LOP)			
	201 (request)			<u>-</u>	200 OK INFO			
					200 01(1111 0			
	LOP(response)	+	1.	-	INFO(LOP)			
	20. (100)01100)	+ +		<u>`</u>	200 OK INFO			
					200 01(1111 0			
	FAC(call transfer active, CTNb)	→	 	→	INFO(FAC)			
	1770(can transfer active, CTND)	+*+		/	200 OK INFO			
		+	+		ZOO OK INFO			
		++		→	INIVITE (condract)			
		++		<u> </u>	INVITE(sendrecv) 200 OK INVITE(sendrecv)			
		+++						
			-	→	ACK			
	1	1 1			i			
		- _ 		_	5) (5 (5 5)			
	REL RLC(RLC)	→		→	BYE(REL) 200 OK BYE			

TP412010	SIP reference: RFC 3261 [4]				ISUP reference:				
				Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8, 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.	I-MGCF interworking.							
SIP parameter	INVITE: Content-Type: multi		lixea, Conte	ent- i ype	e: application/ISUP; CPG				
values	INFO: Content-Type: applica	encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body INFO: Content-Type: application/ISUP; LOP encapsulated in the MIME body							
ISUP parameter	CPG: Event indicator=progre	ess, G	eneric notifi						
values	LOP: response: Call transfer	LOP: request: Call transfer reference LOP: response: Call transfer reference FAC: Generic notification=call transfer active, Service activation=call transfer, Call transfer							
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK	→							
			Conversati	ion					
	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	+		+	AO(can transfer active, CTND)				
	ZOU ON IINFU	-							
	INVITE(sendrecv)	→							
	200 OK INVITE(sendrecv)	+							
	ACK	→							
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP412011	SIP reference: RFC 3261 [4]		0 1012		SUP reference: , clauses 5.4.3, 5.4.3.2, B.8,		
					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 values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; LOP encapsulated in the MIME body						
ISUP parameter values	CPG: Event indicator=progress, Gene LOP: request: Call transfer reference	ric no	ification=ho	ld			
Comments	ISUP		SUT		SIP-I		
	IAM	→		→	INVITE(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
				→	ACK		
		_	onversatio				
	CPG(hold)	→		→	INVITE(CPG, sendonly)		
				+	200 OK INVITE(recvonly)		
				→	ACK		
	LOP(request)	→		→	INFO(LOP)		
	20. (.04000)	Ť		′	200 OK INFO		
					200 01(1141 0		
	REL	→		→	BYE(REL)		
	RLC	+		←	200 OK BYE(RLC)		

TP412012	SIP reference: RFC	3261	[4]	Q.19	ISUP reference: 12.5 [1], clauses 5.4.3, 5.4.3.2, B.8, Q.734 [i.7], clause 1.6.15			
TSS reference	ISUP-SIP-ISUP/SS/CONF		<u>'</u>					
SIP selection criteria								
ISUP selection								
criteria								
Test purpose	Loop prevention procedure - unsuccessful on timer expiry To verify that SUT is able to transfer the loop request received in an INFO request containing the LOP message in an ISUP LOP message. Verify that the connection is unsuccessful if the loop detection procedure is unsuccessful. The connection is released from the remote end. I-MGCF interworking.							
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; LOP encapsulated in the MIME body							
ISUP parameter values	CPG: Event indicator=progr LOP: request: Call transfer i			ation=	hold			
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	→						
			Conversatio	n				
	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]			ISUP reference:					
				[1], clauses 5.4.3, 5.4.3.2, I					
			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	Lean provention procedure, augus	andul a	n timer evel	m.,					
	Verify that the SUT is able to transfer INFO request containing the LOP meloop detection procedure is unsucces O-MGCF interworking.	Loop prevention procedure - successful on timer expiry Verify that the SUT is able to transfer the loop request received in an ISUP LOP in an INFO request containing the LOP message. Verify that the connection is successful if the loop detection procedure is unsuccessful. O-MGCF interworking.							
SIP parameter	INVITE: Content-Type: multipart/mixe	d, Cont	tent-Type: app	olication/ISUP ; CPG					
values	encapsulated in the MIME body								
	INFO: Content-Type: application/ISUF								
	INFO: Content-Type: application/ISUI			I in the MIME body					
ISUP parameter	CPG: Event indicator=progress, Gene	eric noti	tication=hold						
values	LOP: request: Call transfer reference FAC: Generic notification=call transfer active, Service activation=call transfer, Call transfer								
		r active	, Service activ	vation=cali transfer, Cali trar	nster				
Comments	number(PIXIT)	1 1	SUT	SIP-I					
Comments	IAM	→		→ INVITE(IAM)					
	ACM	-		180 Ringing(ACM)					
	ANM	+		200 OK INVITE(ANM)					
	AINIVI			→ ACK					
		C	onversation	ZAOR					
	CPG(hold)	→ ``		➤ INVITE(CPG, sendonly)	١				
	Of O(fiold)	+*+		200 OK INVITE(recvonity)					
		+		ACK	у)				
		+		AOR					
	LOP(request)	→	 	→ INFO(LOP)					
	LOT (Toquest)	1		• 200 OK INFO					
		1 1							
	FAC(call transfer active, CTNb)	→		➤ INFO(FAC)					
	Trojodii dallolol dollo, O 1140)	1-		• 200 OK INFO					
		1 1							
		1 1	-	➤ INVITE(sendrecv)					
		1 1		200 OK INVITE(sendred	cv)				
		1 1		ACK	- • /				
				- ,					
	REL	→	-	➤ BYE(REL)					
	RLC	-		200 OK BYE(RLC)					
L	1								

TP412014	SIP reference: RFC	3261	[4]		ISUP reference:
				Q.191	2.5 [1], clauses 5.4.3, 5.4.3.2, B.8, Q.734 [i.7], clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/ECT			1	
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose		trans e in an	fer the loop ISUP LOP	reques messag	t received in an INFO request ge. Verify that the connection is
SIP parameter	INVITE: Content-Type: multi	part/m	ixed, Conte	ent-Type	: application/ISUP ; CPG
values	encapsulated in the MIME b	ody		•	
	INFO: Content-Type: applica				
10115	INFO: Content-Type: applica				
ISUP parameter values	CPG: Event indicator=progre			cation=I	hold
values	LOP: request: Call transfer r FAC: Generic notification=ca number(PIXIT)			Service	activation=call transfer, Call transfer
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
	ACK	→			
			Conversati	ion	
	INVITE(CPG, sendonly)	→		→	CPG(hold)
	200 OK INVITE(recvonly)	+			
	ACK	→			
	INFO(LOP)	→		→	LOP(request)
	200 OK INFO	+		- -	LOP (lequest)
	200 OK IIVI O	+			
	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

TP412015	SIP reference: RFC 3261 [4	1]		ISUP reference:], clauses 5.4.3, 5.4.3.2, B.8, [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 transfetransfer, alerting" and the service a	Facility message with generic notification sent to the remote user Verify that the SUT is able to transfer the generic notification "call transfer, active" or "call transfer, alerting" and the service activation parameter set to "call transfer" received in an ISUP FAC in a SIP INFO request containing the ISUP FAC. O-MGCF interworking.					
SIP parameter values	encapsulated in the MIME body	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP; CPG					
ISUP parameter values	CPG: Event indicator=progress, Ge FAC: Generic notification=call trans number(PIXIT)	neric noti	ification=hold				
Comments	ISUP		SUT	SIP-I			
	IAM	→	→	INVITE(IAM)			
	ACM	+	+	180 Ringing(ACM)			
	ANM	+	+	200 OK INVITE(ANM)			
			→	ACK			
		Co	onversation				
	CPG(hold)	→	→	INVITE(CPG, sendonly)			
			+	200 OK INVITE(recvonly)			
			→	ACK			
	FAC(call transfer active, CTNb)	→	→	INFO(FAC)			
			+	200 OK INFO			
			→	= (00.10.001)			
			+	=00 01(11(11)=(00)101001)			
			→	ACK			
	REL	→	→	_ : _ (: :)			
	RLC	←	+	200 OK BYE(RLC)			

TP412016	SIP reference: RFC 3261 [4]			Q.191	ISUP reference: 12.5 [1], clauses 5.4.3, 5.4.3.2, B.8, 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. I-MGCF interworking.						
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body						
ISUP parameter	CPG: Event indicator=progre						
values	FAC: Generic notification=canumber(PIXIT)	all trans	sfer active, S	Service	activation=call transfer, Call transfer		
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		→	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	←		+	ANM		
	ACK	→					
			Conversatio	n			
	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	→					
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	←		←	RLC		

TP412017	SIP reference: RFC 3261 [4		ISUP reference:			
11 412017	On relevance in o ezer [٠,	0 1912		, clauses 5.4.3, 5.4.3.2, B.8,	
					i.5], clause 7.5.2.1.1.1 a)	
TSS reference	ISUP-SIP-ISUP/SS/ECT		— — — — — — — — — — — — — — — — — — —	<u> </u>	,	
SIP selection						
criteria						
ISUP selection criteria						
Test purpose	Call progress message with generic notification sent to the remote user Verify that the transfer the CPG with the generic notification set to "call transfer, active" and the service activation parameter set to "call transfer" in a SIP-I INFO request containing the ISUP CPG message. O-MGCF interworking.					
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body					
ISUP parameter	CPG: Generic notification=call trans	fer active	e. Service ad	ctivat	tion=call transfer. Call transfer	
values	number (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)	
	7 (1 11)			→	ACK	
	Conversation					
	REL	→		→	BYE(REL)	
	RLC	+		←	200 OK BYE(RLC)	

TP412018	SIP reference: RFC	nce: RFC 3261 [4]		Q.191	ISUP reference: 12.5 [1], clauses 5.4.3, 5.4.3.2, B.8, 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		inart/m	ixed Conte	nt-Type	e: application/ISLIP : CPG			
values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body							
ISUP parameter values	CPG: Event indicator=progress, Generic notification=hold CPG: Generic notification=call transfer active, Service activation=call transfer, Call transfer number(PIXIT)							
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	←		←	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	→						
			Conversati					
	INVITE(CPG, sendonly)	→		→	CPG(hold)			
	200 OK INVITE(recvonly)	+						
	ACK	→						
	INFO(CPG)	→		→	CPG(call transfer active, CTNb)			
	200 OK INFO	+						
	INVITE(sendrecv)	→						
	200 OK INVITE(sendrecv)	+						
	ACK	→						
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP412019	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8, 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	INVITE: Content-Type: multi	ipart/m	ixed, Conter	nt-Type	e: application/ISUP ; CPG				
values	encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC(CTNb=restricted) encapsulated in the MIME body								
ISUP parameter	CPG: Event indicator=progre								
values	FAC: Generic notification=catransfer number(PIXIT)	all tran	sfer active, S	Service	activation=call transfer, no Call				
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	←		←	ACM				
	200 OK INVITE(ANM)	←		←	ANM				
	ACK	→							
		(Conversation	n					
	INVITE(CPG, sendonly)	→		→	CPG(hold)				
	200 OK INVITE(recvonly)	+							
	ACK	→							
	INFO(FAC)	→		→	FAC(call transfer active)				
	200 OK INFO	+							
	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], clauses 5.4.3, 5.4.3.2, B.8, 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 - 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	INVITE: Content-Type: multi		ixed, Conten	t-Type	e: application/ISUP ; CPG			
values	encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC(CTNb=national) encapsulated in the Message body							
ISUP parameter	CPG: Event indicator=progre							
values	FAC: Generic notification=canumber=international(PIXIT)		sfer active, S	ervice	activation=call transfer, Call transfer			
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	→						
			Conversatio	n				
	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	→						
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	←		+	RLC			

TP412021	SIP reference: RFC 3261 [4	ij		[1]	SUP reference: , clauses 5.4.3, 5.4.3.2, B.8, 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	INVITE: Content-Type: multipart/mix	ed, Con	tent-Type: ar	oplic	cation/ISUP; CPG			
values	encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC(CTNb=national) encapsulated in the MIME body							
ISUP parameter	CPG: Event indicator=progress, Generic notification=hold							
values	FAC: Generic notification=call transfer active, Service activation=call transfer, Call transfer number=international(PIXIT)							
Comments	ISUP		SUT		SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	+		←	180 Ringing(ACM)			
	ANM	+		←	200 OK INVITE(ANM)			
				→	ACK			
		Co	onversation					
	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)			

TP412022	SIP reference: RFC 3261 [4		ISUP reference: Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8, 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/ISI	JP: FAC	encapsulate	d in	the MIME body		
values		, , , , , , ,					
ISUP parameter values	FAC: Generic notification=call transfer active, Service activation=call transfer, Call transfer number(PIXIT) FAC: ATP contained the connected sub address						
Comments	ISUP		SUT		SIP-I		
	IAM	→		→	INVITE(IAM)		
	ACM	+		←	180 Ringing(ACM)		
	ANM	+		-	200 OK INVITE(ANM)		
				→	ACK		
		Cc	nversation				
	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)		
1	RLC	+		-	200 OK BYE(RLC)		

3PTY

TP413001	SIP reference: RFC 3261 [4]			ISUP reference: 912.5 [1], clause B.15, .2 [i.8], clause 2.4; 2.2.1		
TSS reference	ISUP-SIP-ISUP/SS/3PTY					
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Served user initiates 3PTY Verify that the served user with two active calls is located, can successfully join this call (remote held user) to a three-way conversation, and notify the implied remote party accordingly. The IUT should 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	INVITE: Content-Type: multipart/mixed, (Cont	ent-Type: appl	ication/ISUP : CPG		
values	encapsulated in the MIME body					
ISUP parameter	CPG: Event indicator=progress, Generic	notif	fication=hold			
values	CPG: Event indicator=progress, Generic			ence established		
Comments	ISUP		SUT	SIP-I		
		→	7	INVITE(IAM)		
	ACM	F	•	180 Ringing(ACM)		
	ANM	F	•	200 OK INVITE(ANM)		
			7	ACK		
		Co	nversation			
	CPG(hold)	→	7			
			•	200 OK INVITE(recvonly)		
			7	ACK		
	CPG(conference established)	▶	7			
			€			
			-	ACK		
		Co	nversation			
	REL -	→	-	BYE(REL)		
	RLC	-	€	 200 OK BYE(RLC) 		

TP413002	SIP reference: RFC	3261 [4]	(ISUP reference: Q.1912.5 [1], clause B.15, Q.734.2 [i.8], clause 2.4; 2.2.1		
TSS reference	ISUP-SIP-ISUP/SS/3PTY						
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Served user initiates 3PTY Verify that the served user with two active calls is located, can successfully join this call (remote held user) to a three-way conversation, and notify the implied remote party accordingly. The IUT should send a CPG message with the generic notification indicator set to "conference established" to both implied parties. The event indicator in the CPG should be set to "progress": 1. setup a call to user B; 2. put this call on hold; 3. join this call to a conference.						
SIP parameter	I-MGCF interworking INVITE: Content-Type: multi	nart/mi	ved Conter	nt-Type	a: application/ISLIP : CPG		
values	encapsulated in the MIME bo		Aca, Conto	птурс	. application/1001 , of 0		
ISUP parameter	CPG: Event indicator=progre		neric notific	ation=	hold		
values	CPG: Event indicator=progre						
Comments	SIP-I	T	SUT		ISUP		
	INVITE(IAM)	→		→	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	→					
			Conversatio	n			
	INVITE(CPG, sendonly)	→		→	CPG(hold)		
	200 OK INVITE(recvonly)	+					
	ACK	→					
	INVITE(CPG, sendrecv)	→		→	CPG(conference established)		
	200 OK INVITE(sendrecv)	+					
	ACK →						
			Conversation				
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP413003	SIP reference: RFC 3261	[4]		Q.19	ISUP reference: 12.5 [1], clause B.15, 2 [i.8], clause 2.4; 2.2.1			
TSS reference	ISUP-SIP-ISUP/SS/3PTY							
SIP selection criteria								
ISUP selection								
criteria								
SIP parameter values ISUP parameter	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": setup a call to user B; establish a conference. O-MGCF interworking. INFO/INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP; CPG encapsulated in the MIME body CPG: Event indicator=progress, Generic notification=conference established							
values					T			
Comments	ISUP		SUT		SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
				→	ACK			
		С	onversatio	n				
	CASE A							
	CPG(conference established)	→		→	INFO(CPG)			
				+	200 OK INFO			
	CASE B			1				
	CPG(conference established)	→		→	INVITE(CPG, sendrecv)			
				+	200 OK INVITE(sendrecv)			
		→ ACK						
			onversatio					
	REL	→		→	BYE(REL)			
	RLC	←		+	200 OK BYE(RLC)			

TP413004	SIP reference: RFC	3261 [4	4]	ISUP reference: Q.1912.5 [1], clause B.15, Q.734.2 [i.8], clause 2.4; 2.2.1				
TSS reference	ISUP-SIP-ISUP/SS/3PTY							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Served user initiates 3PTY Verify that the served user with two active calls is located, can successfully join this call (remote active user) to a three-way conversation, and notify the implied remote party accordingly. The IUT should send a CPG message with the generic notification indicator set to "conference established" to both implied parties. The event indicator in the CPG should be set to "progress": 1. setup a call to user B; 2. establish a conference. I-MGCF interworking.							
SIP parameter			art/mixed,	Conten	t-Type: application/ISUP ; CPG			
values ISUP parameter values	encapsulated in the MIME b CPG: Event indicator=progre		neric notifi	cation=	conference established			
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	→						
		C	onversati	on				
	INFO(CPG)	→		→	CPG(conference established)			
	200 OK INFO	+						
	CASE B							
	INVITE(CPG,sendrecv)	→		→	CPG(conference established)			
	200 OK INVITE(sendrecv)	+						
	ACK	ACK →						
			onversati					
	BYE(REL)	→		→	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP413005	SIP reference: RFC 3261 [4	ı	ISUP reference:				
		•			l2.5 [1], clause B.15,		
			Q.73	34.2	[i.8], clause 2.5.2.1.1.3 a		
TSS reference	ISUP-SIP-ISUP/SS/3PTY						
SIP selection							
criteria							
ISUP selection							
criteria	0		d dd		-1		
Test purpose	Served user creates a private com						
	Verify that a 3PTY call can successful user. The appropriate notification recommendations are successful.	ally Clea		OIIIIIIO OC an	d is sent in INVITE/INFO		
	(CPG) messages to the SIP-I.	eiveu iii	a 1501 CI	G an	d is sent in inverte, into		
	O-MGCF interworking.						
SIP parameter	INFO/INVITE: Content-Type: multipa	rt/mixed	I. Content-T	vpe:	application/ISUP : CPG		
values	encapsulated in the MIME body		., ••••••••	,,,,,,	app.::ca.:::::::::::::::::::::::::::::::		
ISUP parameter	CPG 1, 4: Event indicator=progress,	Generio	notification	=hold			
values	CPG 5: Event indicator=progress, Ge	eneric n	otification=r	etriev	re		
	CPG 2: Event indicator=progress, Ge	eneric n	otification=c	confer	ence established		
	CPG 3: Event indicator=progress, Ge	eneric n		confer			
Comments	ISUP		SUT		SIP-I		
	IAM	→		→	INVITE(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	ANM	←		+	200 OK INVITE(ANM)		
	Conversation → ACK						
	CASE A						
	CPG 1(hold)	→		→	INVITE(CPG, sendonly)		
				+	200 OK INVITE(recvonly)		
				→	ACK		
		→		_	INIVITE (CDC and draw)		
	CPG 2(conference established)	7		→	INVITE(CPG, sendrecv)		
				→	200 OK INVITE(sendrecv) ACK		
				7	ACK		
	CPG 3(conference disconnected)	→		→	INFO(CPG)		
	CFG 3(contenence disconnected)	7		+	200 OK INFO		
		+			200 OK INI O		
	CPG 4(hold)	→		→	INVITE(CPG, sendonly)		
	J. J. (11010)	-		+	200 OK INVITE(recvonly)		
				→	ACK		
				† -			
	CPG 5(retrieve)	→		→	INVITE(CPG, sendrecv)		
	,			+	200 OK INVITE(sendrecv)		
				→	ACK		
		С	onversatio	n			
	CPG 6(conference established)	→		→	INFO(CPG)		
				+	200 OK INFO		
		С	onversatio	n			
	REL	→		→	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

TP413005	SIP reference: RFC 3261 [4	ISUP reference: Q.1912.5 [1], clause B.15, Q.734.2 [i.8], clause 2.5.2.1.1.3 a			
	CASE B				
	CPG 1(hold)	→	→	INVITE(CPG, sendonly)	
			+	200 OK INVITE(recvonly)	
			→	ACK	
	CPG 2(conference established)	→	→	INVITE(CPG, sendrecv)	
	or o z(contreted established)	+*+	-	200 OK INVITE(sendrecv)	
			→	ACK	
		→	→	INIVITE (CDC condroor)	
	CPG 3(conference disconnected)	7		INVITE(CPG,sendrecv)	
			←	200 OK INVITE(sendrecv) ACK	
			7	ACK	
	CPG 4(hold)	→	→	INVITE(CPG, sendonly)	
			+	200 OK INVITE(recvonly)	
			→	ACK	
	CPG 5(retrieve)	→	→	INVITE(CPG, sendrecv)	
			+	200 OK INVITE(sendrecv)	
			→	ACK	
		Со	nversation		
	CPG 6(conference established)	→	→	INVITE(CPG,sendrecv)	
			+	200 OK INVITE(sendrecv)	
			→	ACK	
		Co	nversation		
	REL	→	→	BYE(REL)	
	RLC	+	+	200 OK BYE(RLC)	

TP413006	SIP reference: RFC	3261	[4]	Q	ISUP reference: Q.1912.5 [1], clause B.15, .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	Verify that a 3PTY call can s user. The appropriate notifical messages to the ISUP.	ucces	sfully create	private	a remote user communication with the active-held E/INFO (CPG) and is sent in CPG				
	I-MGCF interworking.								
SIP parameter	INFO/INVITE: Content-Type:	mult	ipart/mixed,	Conten	t-Type: application/ISUP ; CPG				
values	encapsulated in the MIME bo	ody							
ISUP parameter	CPG: Event indicator=progre	ess, G	eneric notific	cation=	hold				
values	CPG: Event indicator=progre	ess, G	eneric notific	cation=	retrieve				
	CPG: Event indicator=progre								
	CPG: Event indicator=progre	ess, G		cation=					
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK	→							
	Conversation								
	CASE A								
	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	+							
	NN (ITE (ODO)	_			0004 10				
	INVITE(CPG, sendonly)	→	1	→	CPG(hold)				
	200 OK INVITE(recvonly)	+	1						
	ACK	→							
	INVITE(CPG, sendrecv)	→	1	→	CPG(retrieve)				
	200 OK INVITE(sendrecv)	+	+		C. C(10th1010)				
	ACK	\	+						
	7.01	+-	Conversation	on					
	INFO(CPG)	→		→	CPG(conference established)				
	200 OK INFO	+	+		C. C(contended established)				
	200 010 1101 0	+	Conversation	on					
	BYE(REL)	→	Jonversati	<u>→</u>	REL				
	200 OK BYE(RLC)	+	+	+	RLC				
	IZUU UN DTE(NLU)				INLO				

TP413006	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause B.15, Q.734.2 [i.8], clause 2.5.2.1.1.3 a			
	CASE B						
	INVITE(CPG, sendonly)	→		→	CPG(hold)		
	200 OK INVITE(recvonly)	+					
	ACK	→					
	INVITE(CPG, sendrecv)	→		→	CPG(conference established)		
	200 OK INVITE(sendrecv)	+					
	ACK						
	INVITE(CPG,sendrecv)	→		→	CPG(conference disconnected)		
	200 OK INVITE(sendrecv)	+			, ,		
	ACK	→					
	INVITE(CPG, sendonly)	→		→	CPG(hold)		
	200 OK INVITE(recvonly)	+			, , ,		
	ACK	→					
	INVITE(CPG, sendrecv)	→		→	CPG(retrieve)		
	200 OK INVITE(sendrecv)	+			, , ,		
	ACK	→					
			Conversation				
	INVTE(CPG,sendrecv)	→		→	CPG(conference established)		
	200 OK INVITE(sendrecv)	+					
	ACK	→					
			Conversation				
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP413007	SIP reference: RFC 3261 [4	ISUP reference:					
		Q.1912.5 [1], clause B.15,					
		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 com						
	Verify that the IUT (controlling the co						
	private communication with the activ messages to the user.	e-iale us	er. The approp	onate notification is sent in CPG			
	O-MGCF interworking.						
SIP parameter	INFO/INVITE: Content-Type: multipa	art/mived	Content-Type	a: application/ISLIP : CPG			
values	encapsulated in the MIME body	arviinkeu	, Content-Type	s. application/1001, of G			
ISUP parameter	CPG: Event indicator=progress, Ger	neric notif	fication=confe	ence established			
values	CPG: Event indicator=progress, Ger						
Comments	ISUP		SUT	SIP-I			
	IAM	→	7				
	ACM	+	€	180 Ringing(ACM)			
	ANM	+	€				
			-	ACK			
	Conversation						
	CASE A						
	CPG(conference established)	→	7				
			•	200 OK INFO			
	CPG(conference disconnected)	→	1	- (/			
			€	200 OK INFO			
			onversation				
	CPG(conference established)	→	7	- \ /			
			€	200 OK INFO			
	DEI		onversation	D)(E(DEL)			
	REL	→	3				
	RLC CASE B	+	•	200 OK BYE(RLC)			
		→		INIVITE/CDC condroom			
	CPG(conference established)	7	-				
		+	3	,			
			7				
	CPG(conference disconnected)	→	-	INVITE(CPG,sendrecv)			
	Of O(contenence discontracted)						
			à				
		Co	onversation	7.3.0			
	CPG(conference established)	→	-3	INVITE(CPG,sendrecv)			
	5. 5(555.5.5.5.5.5.5.6.7.5.7.5.7.5.7.5.7.5	 -	-	, ,			
				(/			
		Co	onversation				
	REL	→	3	BYE(REL)			
	RLC	+	•				

TP413008	SIP reference: RF0	C 3261	ISUP reference: Q.1912.5 [1], clause B.15,						
	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 pri								
					PTY call can successfully create				
		the act	ive-idle use	r. The a	appropriate notification is sent in CPG				
	messages to the user.								
CID nonemater	I-MGCF interworking.	16		01-	4 T				
SIP parameter			part/mixed, (Conten	t-Type: application/ISUP ; CPG				
values ISUP parameter	encapsulated in the MIME I				andarana antahilahad				
values	CPG: Event indicator=prog								
Comments	CPG: Event indicator=prog	1655, G	SUT	Jalion=	ISUP				
Comments	INVITE(IAM)	→	301	→	IAM				
		+		+	ACM				
	180 Ringing(ACM) 200 OK INVITE(ANM)	+		+	ANM				
		→			ANIVI				
	ACK								
	Case								
	CASE A	_			ODO(f				
	INFO(CPG)	→		→	CPG(conference established)				
	200 OK INFO								
	INIEC(ODO)				000/ ()				
	INFO(CPG)	→		→	CPG(conference disconnected)				
	200 OK INFO	+							
	Conversation								
	INFO(CPG)	→		→	CPG(conference established)				
	200 OK INFO		<u> </u>						
	DVE(DEL)		Conversation		DEL				
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				
	CASE B								
	DD (TE (ODO)		Conversation	_	000/ (
	INVITE(CPG,sendrecv)	→		→	CPG(conference established)				
	200 OK INVITE(sendrecv)	+		_					
	ACK	→		_					
	INIVITE (ODC			+	ODO/serference P				
	INVITE(CPG,sendrecv)	7		7	CPG(conference disconnected)				
	200 OK INVITE(sendrecv)	+	1	_					
	ACK		Conversell						
	INIVITE (ODO du)		Conversation						
	INVITE(CPG,sendrecv)	→	1	→	CPG(conference established)				
	200 OK INVITE(sendrecv)	+	-		1				
	ACK	→	Camus == = 1'		<u> </u>				
	DVE(DEL)		Conversation		loc.				
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+	L	+	RLC				

TP413009	SIP reference: RFC 3261 [4	1]		Q.191	SUP reference: 2.5 [1], clause B.15, 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 rem Verify that the IUT (controlling the co the active-held user and retain and r messages. The IUT should send to the appropri notification indicator. The event in O-MGCF interworking.	onference notify the ate remo	e) on a 3PT other user ote users CF in the CPG	Y cal appro PG m shou	I can successfully disconnect opriately using CPG essages with a generic ald be set to "progress".			
SIP parameter	INFO/INVITE: Content-Type: multipa	art/mixed	I, Content-T	уре:	application/ISUP ; CPG			
values	encapsulated in the MIME body							
ISUP parameter	CPG: Event indicator=progress, Ger	neric noti	fication=cor	nfere	nce established			
values	CPG: Event indicator=progress, Ger	neric noti	fication=cor	nfere				
Comments	ISUP		SUT		SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
				→	ACK			
	Conversation							
	CPG(conference established)	→		→	INFO(CPG)			
				+	200 OK INFO			
	CPG(conference disconnected)	→		→	INFO(CPG)			
				+	200 OK INFO			
		С	onversatio	n				
	REL	→		→	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			
	1120	<u> </u>		_	200 011 212(1120)			
	CASE B							
	CPG(conference established)	→		→	INVITE(CPG,sendrecv)			
	er e(conterence established)			+	200 OK INVITE(sendrecv)			
				→	ACK			
					ACK			
	CPG(conference disconnected)	→		→	INVITE(CPG,sendrecv)			
	or a(contenence discontinected)	7		7	200 OK INVITE(sendrecv)			
				→	ACK			
			onvoractic	_	AUN			
	REL	→ L	onversatio		DVE(DEL)			
		7		→	BYE(REL)			
	RLC	7			200 OK BYE(RLC)			

TP413010	SIP reference: RFC	3261	[4]	Q	ISUP reference: Q.1912.5 [1], clause B.15, .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	Verify that the IUT (controlling the active-held user and retain messages. The IUT should send to the anotification indicator. The I-MGCF interworking.	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			part/mixed, Co	nten	t-Type: application/ISUP; CPG				
values	encapsulated in the MIME be	ody							
ISUP parameter	CPG: Event indicator=progre								
values	CPG: Event indicator=progre	ess, G	eneric notificat	ion=					
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK	→							
			Conversation						
	CASE A								
	INVITE(CPG,sendrecv)	→		→	CPG(conference established)				
	200 OK INVITE(sendrecv)	←							
	ACK	→							
	INVITE(CPG,sendrecv)	→		→	CPG(conference disconnected)				
	200 OK INVITE(sendrecv)	+							
	ACK	→							
			Conversation						
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				
	CASE B								
	INVITE(CPG,sendrecv)	→		→	CPG(conference established)				
	200 OK INVITE(sendrecv)	+							
	ACK	→							
	INVITE(CPG,sendrecv)	→		→	CPG(conference disconnected)				
	200 OK INVITE(sendrecv)	+							
	ACK	→							
			Conversation						
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP413011	SIP reference: RFC 3261 [4]		Q.7	Q.191	SUP reference: 2.5 [1], clause B.15, i.8], clause 2.5.2.1.1.3 b			
TSS reference	ISUP-SIP-ISUP/SS/3PTY			· ··- <u>·</u>	,			
SIP selection	25. 2 125.700,0. 11							
criteria								
ISUP selection								
criteria								
Test purpose	Served user disconnects one remot	e use	r and retain	ns 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	INFO/INVITE: Content-Type: multipart	/mixed	d, Content-T	ype:	application/ISUP ; CPG			
values	encapsulated in the MIME body		•	,,	,			
ISUP parameter	CPG: Event indicator=progress, General	ric not	ification=ho	ld				
values	CPG: Event indicator=progress, General				nce established			
	CPG: Event indicator=progress, General	ric not	ification=co	nferer	nce disconnected			
Comments	ISUP		SUT		SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		÷	200 OK INVITE(ANM)			
	1 11 41 41 41 41 41 41 41 41 41 41 41 41			\	ACK			
		_	onversatio		, tort			
	CASE A	·	l	''				
		_		_	INIVITE/CDC condents)			
	CPG(hold)	→		→	INVITE(CPG, sendonly)			
				+	200 OK INVITE(recvonly)			
				→	ACK			
	CPG(conference established)	→		→	INVITE(CPG, sendrecv)			
				+	200 OK INVITE(sendrecv)			
				→	ACK			
	CPG(conference disconnected)	→		→	INFO(CPG)			
	·			+	200 OK INFO			
	CPG(hold)	→		→	INVITE(CPG, sendonly)			
				+	200 OK INVITE(recvonly)			
				→	ACK			
				† <u> </u>				
	REL	→		→	BYE(REL)			
	RLC	′		+	200 OK BYE(RLC)			
	CASE B	<u> </u>		+	200 ON BTE(NEO)			
	CPG(hold)	→		→	INVITE(CPG, sendonly)			
	or G(noid)	7		+	200 OK INVITE(recvonly)			
		-		7				
				7	ACK			
	ODO/			+	INDUITE (ODO			
	CPG(conference established)	→		→	INVITE(CPG, sendrecv)			
				+	200 OK INVITE(sendrecv)			
				→	ACK			
	CPG(conference disconnected)	→		→	INVITE(CPG,sendrecv)			
				+	200 OK INVITE(sendrecv)			
				→	ACK			
	CPG(hold)	→		→	INVITE(CPG, sendonly)			
				+	200 OK INVITE(recvonly)			
				→	ACK			
				Ť	1			
	REL	→		→	BYE(REL)			
	RLC	′		-	200 OK BYE(RLC)			
	INLO		<u> </u>	Z -	ZOU ON DIE(NEO)			

TP413012	SIP reference: RFC	3261	ISUP reference:				
				Q.1912.5 [1], clause B.15, Q.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 o	ne re	mote user a	and reta	ains the other		
	the active-idle user and retain messages. The IUT should send to the a	and	notify the of	ther use e users	CPG messages with a generic		
	O-MGCF interworking.	event	indicator if	i the Cr	PG should be set to "progress".		
SIP parameter		multi	part/mixed.	Conten	t-Type: application/ISUP; CPG		
values	encapsulated in the MIME bo		, ,		7111		
ISUP parameter	CPG: Event indicator=progre	ss. G	eneric notific	cation=l	hold		
values	CPG: Event indicator=progre						
	CPG: Event indicator=progre						
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		→	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	<u>→</u>					
	7.01.		Conversati	on			
	CASE A						
	INVITE(CPG, sendonly)	→		→	CPG(hold)		
	200 OK INVITE(recvonly)	/			Cr G(nota)		
	ACK	→					
	ACK	7					
	INIVITE (CDC condragy)	→		→	CDC/conforces catablished)		
	INVITE(CPG, sendrecv) 200 OK INVITE(sendrecv)	+		7	CPG(conference established)		
		→					
	ACK	7					
	INITO(ODO)				000/		
	INFO(CPG)	→		→	CPG(conference disconnected)		
	200 OK INFO	~					
	INDUSTRIAL LAND				000(1.11)		
	INVITE(CPG, sendonly)	→		→	CPG(hold)		
	200 OK INVITE(recvonly)	←					
	ACK	→					
	D) (E (DEL)	_	1		551		
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)	+		+	RLC		
	CASE B	<u> </u>			oper un		
	INVITE(CPG, sendonly)	→		→	CPG(hold)		
	200 OK INVITE(recvonly)	+					
	ACK	→					
	INVITE(CPG, sendrecv)	→		→	CPG(conference established)		
	200 OK INVITE(sendrecv)	+		- * -	C. S(contended cotabilistica)		
	ACK	→					
	//OK	+		+			
	INVITE(CPG,sendrecv)	→		→	CPG(conference disconnected)		
	200 OK INVITE(sendrecv)	+		7	Or O(comerence disconnected)		
	ACK	→					
	AUR	 					
	INIVITE/CDC condonly	→		→	CPG(hold)		
	INVITE(CPG, sendonly)	7		7	CPG(hold)		
	200 OK INVITE(recvonly)	→					
	ACK	7					
	DVE(DEL)			_	DEL		
	BYE(REL)	→		→	REL		
	200 OK BYE(RLC)		<u> </u>	+	RLC		

5.3.14 User-to-user service

5.3.14.1 User-to-user service 1

TP414001	SIP referen	ce: RFC 3261 [4	ij	Q.19	ISUP reference: 12.5 [1], clause B.21, 0], clauses 1.1.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS	/UUS1			
SIP selection					
criteria					
ISUP selection criteria					
Test purpose	Service 1 implicit re	quest: User-to-u	ser informa	tion in the IA	M
	Ensure that the SUthe encapsulated IA			the User-to-us	ser service 1 implicit request in
SIP parameter	INVITE: Content-Ty	pe: multipart/mix	ced, Conter	nt-Type: appli	cation/ISUP; IAM containing
values	the user-to-user info	ormation parame	ter encaps	ulated in the I	MIME body
ISUP parameter values	IAM: User-to-user in	nformation paran	neter		
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
				→	ACK
			Conversa	tion	
	REL	→		→	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP414002	SIP reference: RF	C 3261	[4]	Q.19	SUP reference: 12.5 [1], clause B.21, 0], clauses 1.1.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Service 1 implicit request: Ensure that the SUT can see the encapsulated IAM. I-M	successf	ully transfer t		VITE ser service 1 implicit request in
SIP parameter	INVITE: Content-Type: mu	ultipart/m	nixed, Conter	t-Type: applic	cation/ISUP; IAM containing
values	the user-to-user information	n param	neter encapsi	ulated in the N	/IME body
ISUP parameter values	IAM: User-to-user informa	tion para	ameter		
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
	ACK	→			
			Conversa	tion	
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		+	RLC

TP414003	SIP reference	ce: RFC 3261 [4]	0.40	ISUP reference:
					12.5 [1], clause B.21,
TSS reference	ICLID CID ICLID/CC/	(11104		Q./3/ [I.1	0], clauses 1.1.5.2.3 and 4
	ISUP-SIP-ISUP/SS/	0051			
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Service 1 explicit re	quest: User-to-ι	ıser informa	ation in the IN	IVITE
SIP parameter values ISUP parameter values	not essential in the e INVITE: Content-Ty the user-to-user ind	encapsulated IA pe: multipart/mi icator paramete iformation parar	M. O-MGC xed, Conte r encapsula	F interworking nt-Type: appli ated in the MII	cation/ISUP; IAM containing
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
				→	ACK
			Conversa	ation	
	REL	→		→	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP414004	SIP reference: RFC	3261	[4]		Q.191	SUP reference: 2.5 [1], clause B.21,], clauses 1.1.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS1					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Service 1 explicit request: U Ensure that the SUT can su					
	essential received in the IAN				10 03	er service i explicit request
SIP parameter values	the user-to-user indicator pa 180 Ringing: Content-Type: parameter encapsulated in	aramete applica the MIN	er encapsul ation/ISUP; //E body	ated in th ACM cor	e MIN ntainin	g the user-to-user indicator
ISUP parameter values	IAM: User-to-user information request essential ACM: User-to-user indicator	-				·
Comments	ISUP		SU			SIP-I
	IAM	→			^	INVITE(IAM)
	ACM	+			+	180 Ringing(ACM)
	ANM	+			+	200 OK INVITE(ANM)
					→	ACK
			Convers	ation		
	REL	→			→	BYE(REL)
	RLC	+			+	200 OK BYE(RLC)

TP414005	SIP reference: RFC	3261	[4]		_	SUP reference: 2.5 [1], clause B.21,
], clauses 1.1.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS1				-	
SIP selection criteria						
ISUP selection criteria						
Test purpose	Service 1 explicit request: U	lser-to-	user indica	tor in the	INVIT	E
	Ensure that the SUT can su essential received in the end					
SIP parameter	INVITE: Content-Type: mult	ipart/m	ixed, Conte	nt-Type: a	applic	ation/ISUP ; IAM containing
values	the user-to-user indicator pa					
	180 Ringing: Content-Type:			ACM con	tainin	g the user-to-user indicator
	parameter encapsulated in					
ISUP parameter	IAM: User-to-user information	on para	ımeter, Use	r-to-user	indica	tor = service 1 explicit
values	request essential					
_	ACM: User-to-user indicator	set to			respo	
Comments	SIP-I		SU			ISUP
	INVITE(IAM)	→			→	IAM
	180 Ringing(ACM)	+			←	ACM
	200 OK INVITE(ANM)	+			+	ANM
	ACK	→				
			Convers	ation		
	BYE(REL)	→			→	REL
	200 OK BYE(RLC)	+			+	RLC

TP414006	SIP reference: RFC	3261	[4]		ISUP reference: 912.5 [1], clause B.21, 10], clauses 1.1.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Service 1 implicit response: Ensure that the SUT can su in the encapsulated ACM. O-MGCF interworking.				ACM user service 1 implicit response
SIP parameter values	Service 1 implicit response: INVITE: Content-Type: mult the user-to-user information	ipart/m param applica	ixed, Conte leter encaps ation/ISUP;	nt-Type: appl sulated in the	ication/ISUP; IAM containing
ISUP parameter	IAM: User-to-user information				
values	ACM: User-to-user informati				
Comments	ISUP		SUT	•	SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
				→	ACK
			Convers	ation	
	REL	→		→	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP414007	SIP reference: RFC	3261	[4]		_	SUP reference: 2.5 [1], clause B.21,
], clauses 1.1.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS1					
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Service 1 implicit response:	User-te	o-user infor	mation in	the A	CM
	- 4 44 OUT					
		ccesst	ully transfer	the User	-to-us	er service 1 implicit response
	in the encapsulated ACM.					
	I-MGCF interworking.					
SIP parameter						ation/ISUP; IAM containing
values	the user-to-user information					
				ACM cor	ntainin	g the user-to-user information
	parameter encapsulated in					
ISUP parameter	IAM: User-to-user information					
values	ACM: User-to-user informat	ion par				
Comments	SIP-I		SU			ISUP
	INVITE(IAM)	→			→	IAM
	180 Ringing(ACM)	+			+	ACM
	200 OK INVITE(ANM)	+			+	ANM
	ACK	→				
			Convers	ation		
	BYE(REL)	→			→	REL
	200 OK BYE(RLC)	+			+	RLC

TP414008	SIP reference: RFC	3261	[4]		ISUP reference: 912.5 [1], clause B.21, 10], clauses 1.1.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can su not provided in the encapsu O-MGCF interworking.	ccessf	ully transfer		CM user service 1 explicit response
SIP parameter values	the user-to-user information	param applic	neter encaps ation/ISUP;	sulated in the	lication/ISUP ; IAM containing MIME body iing the user-to-user indicator
ISUP parameter values	IAM: User-to-user information not essential ACM: User-to-user indicator	•			cator set to service 1 request
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
				→	ACK
			Convers	ation	
	REL	→		→	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP414009	SIP reference: RF	C 3261	[4]		ISUP referer 912.5 [1], clau 10], clauses 1	ise B.21,
TSS reference	ISUP-SIP-ISUP/SS/UUS1					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Service 1 explicit response Ensure that the SUT can s not provided in the ACM. I-MGCF interworking.		·			explicit response
SIP parameter values	INVITE: Content-Type: mu the user-to-user informatio 180 Ringing: Content-Type parameter encapsulated in	n param e: applica	eter encaps ation/ISUP;	sulated in the	MIME body	· ·
ISUP parameter values	IAM: User-to-user informat not essential ACM: User-to-user indicate	•				rvice 1 request
Comments	SIP-I		SUT	•	ISUP	
	INVITE(IAM)	→		7	IAM	
	180 Ringing(ACM)	+		+	- ACM	
	200 OK INVITE(ANM)	+		+	- ANM	
	ACK	→				
			Convers	ation		
	BYE(REL)	→		7	REL	
	200 OK BYE(RLC)	+		+	• RLC	

TP414010	SIP reference: RFC	3261	[4]	-	SUP reference:
					12.5 [1], clause B.21, 0], clauses 1.1.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS1		L	-	-
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can su network in the encapsulated O-MGCF interworking.		•	the User-to-us	er service 1 discarded by the
SIP parameter values	the user-to-user information	param applica	neter encaps ation/ISUP;	ulated in the N	cation/ISUP ; IAM containing //IME body ng the User-to-user indicator
ISUP parameter	IAM: User-to-user information				
values	ACM: User-to-user indicator	set to	discarded b	y the network	response
Comments	ISUP		SUT	•	SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
				→	ACK
			Conversa	ation	
	REL	→		→	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP414011	SIP reference: RFC	3261	[4]			SUP reference:
						2.5 [1], clause B.21,
				Q.737	7 [i.10], clauses 1.1.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS1					
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Ensure that the SUT can su	ccessf	ully transfer	the User-	to-use	er service 1 discarded by the
	network in the encapsulated	ACM.				
	I-MGCF interworking.					
SIP parameter	INVITE: Content-Type: mult					
values	the user-to-user information					
	180 Ringing: Content-Type:			ACM con	tainin	g the User-to-user indicator
	parameter encapsulated in					
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	SIP-I		SUT	•		ISUP
	INVITE(IAM)	→			→	IAM
	180 Ringing(ACM)	+			+	ACM
	200 OK INVITE(ANM)	+			+	ANM
	ACK	→				
			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: 2.5 [1], clauses 5.4.3, B.21, i.10], clauses 1.2.5.2.3 and 4			
TSS reference	ISUP-SIP-ISUP/SS/UUS2							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Service 2 request not essential transferred in the INVITE Ensure that the SUT can successfully transfer the User-to-user service 2 explicit request and User-to user information in the encapsulated IAM. An additional User-to-user information is sent in a USR message encapsulated in an INFO request. O-MGCF interworking.							
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP; IAM containing the user-to-user indicator and User-to-user information 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 information			-to-user in	dicator			
values	USR: User-to-user informati	ion	·					
Comments	ISUP		SUT	•	SIP-I			
	IAM	→		1	→ INVITE(IAM)			
	ACM	+		,	← 180 Ringing(ACM)			
	ANM	+		,	€ 200 OK INVITE(ANM)			
				-	→ ACK			
			Conversa	ation				
	USR	→		1	→ INFO(USR)			
					€ 200 OK INFO			
	USR	+			← INFO(USR)			
				-	→ 200 OK INFO			
	REL	→			→ BYE(REL)			
	RLC	+			€ 200 OK BYE(RLC)			

TP414102	SIP reference: RI	FC 3261 [4]	Q.1912.5	ISUP reference: 5 [1], clauses 5.4.3, B.21, 0], clauses 1.2.5.2.3 and 4			
TSS reference	ISUP-SIP-ISUP/SS/UUS2	2						
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Service 2 request not essential transferred in the IAM Ensure that the SUT can successfully transfer the User-to-user service 2 explicit request in the encapsulated IAM. An additional User-to-user information is sent in a USR message encapsulated in an INFO request.							
	O-MGCF interworking.							
SIP parameter values	INVITE: Content-Type: multipart/mixed, 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			-user indica	ator			
values	USR: User-to-user inform		,					
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	→						
			Conversation	on				
	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.19	ISUP reference: 12.5 [1], clause B.21, D], clauses 1.2.5.2.3 and 4			
TSS reference	ISUP-SIP-ISUP/SS/UUS2							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Service 2 response not provided transferred in the ACM Ensure that the SUT can successfully transfer the User-to-user service 2 explicit response not provided in the encapsulated ACM. I-MGCF interworking.							
SIP parameter values	the user-to-user information	param applic	eter encaps ation/ISUP;	sulated in the N	cation/ISUP ; IAM containing MIME body ng the user-to-user indicator			
ISUP parameter	IAM: User-to-user information			r-to-user indica	ator set to service 2 request			
values	ACM: User-to-user indicator							
Comments	ISUP		SUT	7	SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
	→ ACK							
			Convers	ation				
	REL	→		→	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP414104	SIP reference: RFC	3261	[4]	Q.1	ISUP refere 912.5 [1], cla	
				Q.737 [i.	10], clauses	1.2.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS2					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Service 2 response not prov	vided tr	ansferred ir	the ACM		
	Ensure that the SUT can sunot provided in the encapsu	ılated A	CM. I-MGC	F interworking	g.	
SIP parameter	INVITE: Content-Type: mult					; IAM containing
values	the user-to-user indicator pa					
	180 Ringing: Content-Type:			ACM contain	ing the user-t	o-user indicator
	parameter encapsulated in					
ISUP parameter	IAM: User-to-user information					
values	ACM: User-to-user indicato	r set to	service 2 n	ot provided re	esponse	
Comments	SIP-I		SU	Г	ISUP	
	INVITE(IAM)	→		→	IAM	
	180 Ringing(ACM)	+		+	ACM	
	200 OK INVITE(ANM)	+		+	ANM	
	ACK	→				
			Convers	ation		
	BYE(REL)	→		→	REL	
	200 OK BYE(RLC)	+		+	RLC	

5.3.14.3 User-to-user service 3

TP414201	SIP reference: RFC 3261 [4] ISUP reference:				
					5 [1], clauses 5.4.3, B.21,
				Q.737 [i.	10], clauses 1.2.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection					
criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can su	ıccessfı	ully transfer	the User-to-u	ser service 3 explicit request in
' '					s sent in several USR message
	encapsulated in an INFO re				Ç
	O-MGCF interworking.				
SIP parameter					ication/ISUP; IAM containing
values	the user-to-user indicator e				
	INFO: Content-Type: applic			containing the	User-to-user information
	parameter encapsulated in				
ISUP parameter	IAM: User-to-user informati		ımeter, Use	r-to-user indic	cator
values	USR: User-to-user informat	ion			
Comments	ISUP		SU'		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	+		+	
	ANM	+		+	\ /
				→	ACK
			Convers	ation	
	USR	→		→	
				+	200 OK INFO
	USR	+		+	- (/
				→	200 OK INFO
	USR	+		+	
				→	200 OK INFO
	USR	→		→	INFO(USR)
	1	<u> </u>		+	
	REL	→		→	BYE(REL)
	RLC	+		+	

TP414202	SIP reference: RI	FC 3261	[4]		ISUP reference: 5 [1], clauses 5.4.3, B.21,				
					0], clauses 1.2.5.2.3 and 4				
TSS reference	ISUP-SIP-ISUP/SS/UUS3								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose					ser service 3 explicit request in				
			Jser-to-use	r information is	s sent in several USR message				
	encapsulated in an INFO	request.							
SIP parameter	I-MGCF interworking.	ultinart/m	ivad Canta	nt Type: appli	cation/ISUP; IAM containing				
values	the user-to-user indicator				cation/150P, TAIVI containing				
valucs	INFO: Content-Type: app				Iser-to-user information				
	parameter encapsulated i			ontaining the	osci to user information				
ISUP parameter	IAM: User-to-user informa			r-to-user indica	ator				
values	USR: User-to-user inform		,						
Comments	SIP-I		SU ⁻	Γ	ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK	→							
			Convers						
	INFO(USR)	→		→	USR				
	200 OK INFO	+							
	INFO(USR)	(+	USR				
	200 OK INFO	→							
	INICO(LICE)	+		+	USR				
	INFO(USR) 200 OK INFO	7			USK				
	200 OK INFO	7			+				
	INFO(USR)	→		→	USR				
	200 OK INFO	+			OOK				
	200 011 1111 0	+			+				
	BYE(REL)	→		→	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP414203	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses 5.4.3, B.21, Q.737 [i.10], clauses 1.2.5.2.3 and 4				
TSS reference	ISUP-SIP-ISUP/SS/UUS3							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that a User-to-user request service 3 encapsulated in an INFO request during the confirmed state can successful proceeded. The User-to-user information is passed on in several encapsulated USR messages. O-MGCF interworking.							
SIP parameter	INFO: Content-Type: applic	cation/IS	SUP; FAR c	containing the u	user-to-user indicator			
values	encapsulated in the MIME			· ·				
	INFO: Content-Type: applic		SUP; FAA c	ontaining the ι	user-to-user indicator			
	encapsulated in the MIME							
	INFO: Content-Type: applic			containing the	User-to-user information			
	parameter encapsulated in							
ISUP parameter	FAR: User-to-user indicato	r service	e 3 request	not essential				
values	FAA: User-to-user indicato		e 3 respons	e provided				
•	USR: User-to-user informa	tion		_	loip i			
Comments	ISUP		SU'		SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
				→	ACK			
			Convers					
	FAR	→		→	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			
Ì	REL	→		→	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP414204	SIP reference: RI	FC 3261	[4]			ISUP reference:				
				Q.1912.5 [1], clauses 5.4.3, B.21, Q.737 [i.10], clauses 1.2.5.2.3 and 4						
TSS reference	ISUP-SIP-ISUP/SS/UUS3									
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	Ensure that a User-to-user request service 3 encapsulated in an INFO request during the confirmed state can successful proceeded. The User-to-user information is passed on in several encapsulated USR messages. I-MGCF interworking.									
SIP parameter	INFO: Content-Type: app		SUP; FAR o	ontaining	, the ι	ıser-to-user indicator				
values	encapsulated in the MIME INFO: Content-Type: app	lication/IS	SUP; FAA c	ontaining	the u	ser-to-user indicator				
	encapsulated in the MIME									
	INFO: Content-Type: app			containing	the l	Jser-to-user information				
IOUD	parameter encapsulated i				<i>.</i>					
ISUP parameter	FAR: User-to-user indicat									
values	FAA: User-to-user indicat USR: User-to-user inform		e 3 respons	e provide	ea					
Comments	SIP-I	allon	SU [*]	т	1	ISUP				
Comments			30	1						
	INVITE(IAM)	→			→	IAM				
	180 Ringing(ACM)				+	ACM				
	200 OK INVITE(ANM)	+			+	ANM				
	ACK →									
	INTER (EAR)		Convers	ation		EAD				
	INFO(FAR)	→			→	FAR				
	200 OK INFO	+								
	INFO(FAA)	+			+	FAA				
	200 OK INFO	→								
	INFO(USR)	→			→	USR				
	200 OK INFO	+								
	INFO(USR)	+			+	USR				
	200 OK INFO	→				OOK				
	INICO(LICE)	+			+	USR				
	INFO(USR) 200 OK INFO	→ →			_	USK				
	ZUU OK INFU	7								
	INFO(USR)	→			→	USR				
	200 OK INFO	+								
	BYE(REL)	→			→	REL				
	200 OK BYE(RLC)	-			-	RLC				
	1200 OK DIL(KLO)		1			INCO				

TP414205	SIP reference: RFC	3261	[4]	Q.19	SUP reference: 12.5 [1], clause B.21, 0], clauses 1.2.5.2.3 and 4				
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.								
SIP parameter values	INVITE: Content-Type: mult the user-to-user indicator pa 200 OK INVITE: Content-Ty indicator parameter encapsi	aramete pe: ap	er encapsula plication/ISU	ted in the MINP; ANM conta					
ISUP parameter	IAM: User-to-user indicator	set to s	service 3 req	uest					
values	ANM: User-to-user indicator	set to	service 3 pro	ovided respon	se				
Comments	ISUP		SUT		SIP-I				
	IAM	→		→	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
				→	ACK				
			Conversa	tion					
	REL	→		→	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP414206	SIP reference: RF	C 3261	[4]	Q.19	ISUP reference: 12.5 [1], clause B.21, D], clauses 1.2.5.2.3 and 4				
TSS reference	ISUP-SIP-ISUP/SS/UUS3								
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Ensure that the SUT can s in the encapsulated ANM. O-MGCF interworking.								
SIP parameter	INVITE: Content-Type: mu	ıltipart/m	ixed, Conte	nt-Type: applic	cation/ISUP; IAM containing				
values	the user-to-user indicator p 200 OK INVITE: Content-1 indicator parameter encap	Гуре: ар	plication/ISL	P; ANM conta					
ISUP parameter	IAM: User-to-user indicato								
values	ANM: User-to-user indicate				ise				
Comments	SIP-I		SÚT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK	→							
			Conversa	ition					
	BYE(REL)	BYE(REL) → REL							
	200 OK BYE(RLC)	+		+	RLC				

TP414207	SIP reference:	RFC 3261 [[4]	Q.1912.5	ISUP reference: 5 [1], clauses 5.4.3, B.21, 0], clauses 1.2.5.2.3 and 4			
TSS reference	ISUP-SIP-ISUP/SS/UU	S3						
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that a User-to-u confirmed state can suc				n an INFO request during the request is rejected.			
SIP parameter	INFO: Content-Type: ap		SUP; FAR c	ontaining the ι	user-to-user indicator			
values	encapsulated in the MIN							
	INFO: Content-Type: ap		SUP; FRJ co	ontaining the u	ser-to-user indicator			
		encapsulated in the MIME body						
ISUP parameter	FAR: User-to-user indic		•					
values	FRJ: User-to-user indica	ator service						
Comments	ISUP	_	SU	-	SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
				→	ACK			
			Convers					
	FAR	→		→	INFO(FAR)			
				←	200 OK INFO			
	FRJ	+		+	INFO(FRJ)			
				→	200 OK INFO			
	REL	→		→	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP414208	SIP reference: RF	C 3261	[4]	Q.1912.5	SUP reference: [1], clauses 5.4.3, B.21,], clauses 1.2.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS3	}			
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Ensure that a User-to-use	r request	service 3 e	encapsulated ir	an INFO request during the
	confirmed state can succe	essful pro	ceeded. Th	e user to user	request is rejected
SIP parameter	INFO: Content-Type: appl		SUP; FAR c	ontaining the u	ser-to-user indicator
values	encapsulated in the MIME				
	INFO: Content-Type: appl		SUP; FRJ c	ontaining the u	ser-to-user indicator
	encapsulated in the MIME				
ISUP parameter	FAR: User-to-user indicate	or service	3 request	not essential	
values	FRJ: User-to-user indicate	or service	3 response	e not provided	
Comments	SIP-I		SU	Γ	ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
	ACK	→			
			Convers	ation	
	INFO(FAR)	→		→	FAR
	200 OK INFO	+			
	INFO(FRJ)	+		+	FRJ
	200 OK INFO	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	+		(RLC

Annex A (informative): Bibliography

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Annex B (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				
V3.2.1	March 2011	Publication				