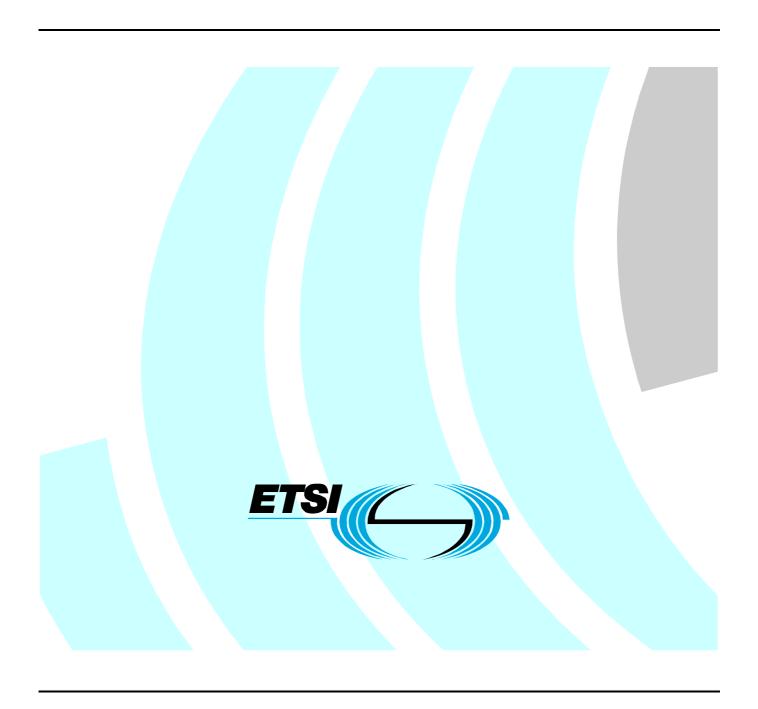
# ETSITS 186 002-3 V1.1.1 (2008-06)

Technical Specification

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



# Reference DTS/TISPAN-06014-2-NGN

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

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#### **Foreword**

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

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

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

## 1 Scope

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

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

#### 2 References

References are either specific (identified by date of publication and/or edition number or version number) or non-specific.

- For a specific reference, subsequent revisions do not apply.
- Non-specific reference may be made only to a complete document or a part thereof and only in the following cases:
  - if it is accepted that it will be possible to use all future changes of the referenced document for the purposes of the referring document;
  - for informative references.

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

For online referenced documents, information sufficient to identify and locate the source shall be provided. Preferably, the primary source of the referenced document should be cited, in order to ensure traceability. Furthermore, the reference should, as far as possible, remain valid for the expected life of the document. The reference shall include the method of access to the referenced document and the full network address, with the same punctuation and use of upper case and lower case letters.

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

#### 2.1 Normative references

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

- [1] 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 Recommendations Q.761 to Q.764 (2000): "Signalling System No.7 ISDN User Part (ISUP)".
- [4] ITU-T Recommendations Q.1902.1 to Q.1902.4 (2001): "Bearer Independent Call Control Protocol (BICC)".
- [5] ITU-T Recommendation Q.850 (1998): "Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part".
- [6] IETF RFC 3261 (2002): "SIP: Session Initiation Protocol".

[7]	IETF RFC 3312 (2002): "Integration of Resource Management and Session Initiation Protocol (SIP)".
[8]	ISO/IEC 9646-1 (1994): "Conformance testing methodology and framework - Part 1: General Concepts".
[9]	ISO/IEC 9646-3 (1992): "Conformance testing methodology and framework - Part 3: The Tree and Tabular Combined Notation".
[10]	ISO/IEC 9646-7 (1994): "Conformance testing methodology and framework - Part 7: Implementation Conformance Statements".
[11]	ITU-T Recommendation E.164: "The international public telecommunication numbering plan".
[12]	IETF RFC 3267: "Real-Time Transport Protocol (RTP) Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs".
[13]	ITU-T Recommendation Q.939: "Typical DSS 1 service indicator codings for ISDN telecommunications services".

#### 2.2 Informative references

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

Not applicable.

#### 3 Definitions and abbreviations

#### 3.1 Definitions

For the purposes of the present document, the 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 [8], ISO/IEC 9646-3 [9] and in ISO/IEC 9646-7 [10].

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

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

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

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

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

**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

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

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

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

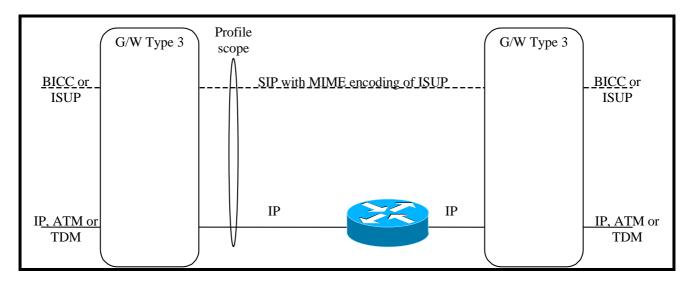


Figure 1: Profile scope for SIP with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateways

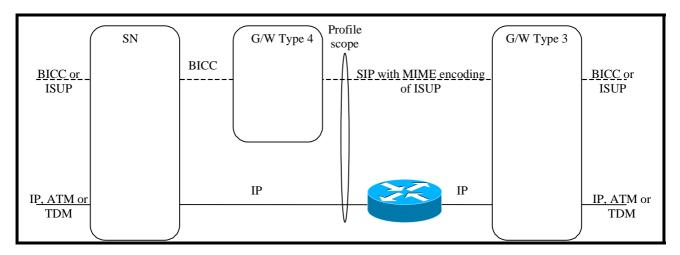


Figure 2: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 3 and 4 gateways

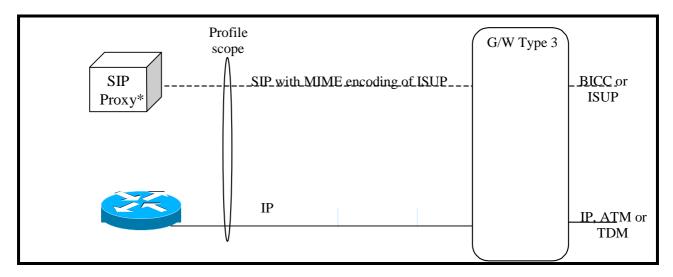


Figure 3: Profile scope for SIP with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateways

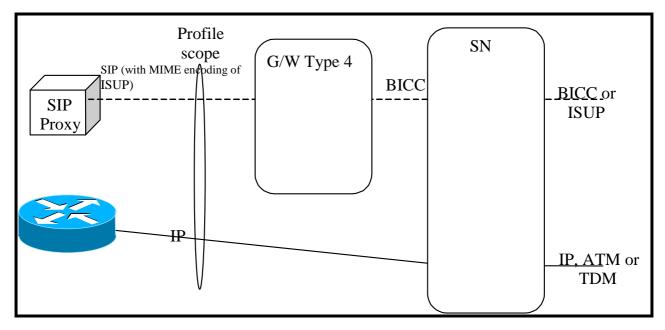


Figure 4: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 4 gateway

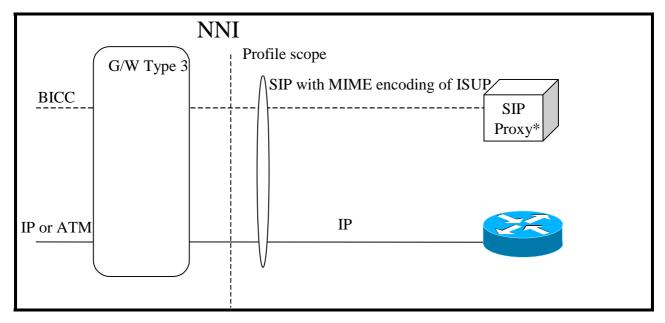


Figure 5: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateway.

#### 3.2 Abbreviations

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

ASP	Abstract Service Primitive
ATC	Abstract Test Case
ATM	Abstract Test Method
ATP	Access Transport Parameter
ATS	Abstract Test Suite
BC	Bearer Capability
BCI	Backward Call Indicators

BICCP Bearer Independent Call Control Protocol

CPS Calling Party's Category
DLE Destination Local Exchange
DSS1 Digital Subscriber System no. 1
FCI Forward Call Indicators

HLC High Layer Compatibility

ISDN Integrated Services Digital Network

ISUP ISDN User Part

MIME Multi-purpose Internet Mail Extension

MOT Means Of Testing

NCI Nature of Connection Indicators
OBCI Optional Backward Call Indicators
OLE Originating Local Exchange
OSI Open Systems Interconnection
PCO Point of Control and Observation

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

PTC Parallel Test Component SIP Session Initiation Protocol

SP Signalling Point SUT System Under Test

TMR Transmission Medium Requirement

TP Test Purpose
TSS Test Suite Structure
UNI User-Network Interface

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

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

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

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

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

## 4.6 Interworking SIP-I/ISDN Supplementary Services

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

# 5 Test Purposes (TP)

#### 5.1 Introduction

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

### 5.1.1 Test Purpose (TP) naming convention

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

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

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

## 5.1.2 Source of test purpose definition

The test purposes have been developed based on Recommendation Q.1912.5 [1].

## 5.1.3 Test purpose structure

The test purpose 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			ISUP reference:			
			Q.1912.5 clause 6.1.2 (i,1)				
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	NOT PICS 4/4 AND NOT PICS 4/5						
criteria							
ISUP selection	PICS 1/6						
criteria							
Test purpose	Ensure that if the SUT upon offer	receipt o	f the first	INVITE w	ith su	fficient digits, with a SDP	
	<ul> <li>the SUT shall delete µ-law (PCMU), if present, from the media description that it will send back in the SDP answer</li> <li>the SUT shall immediately send out the IAM</li> </ul>						
SIP parameter	SIP INVITE: Audio RTP/AVF						
values	200 OK: Audio RTP/AVP 8						
ISUP parameter values	IAM USI: A-law or absent						
Comments	SIP-I		SU <sup>*</sup>	Γ		ISUP	
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM	
	180 Ringing(ACM)	+			+	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
	ACK	<b>→</b>					
	Conversation						
BYE(REL) →					<b>→</b>	REL	
	200 OK BYE(RLC)					RLC	

TP101002	SIP reference: RF	C 3261		0.1	-	SUP reference:	
TCC reference	CID ICLID/Dasia call/ Canding of the Initial Address M					2.5 clause 6.1.2 (i,2ai)	
TSS reference SIP selection	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
criteria	PICS 4/4 AND PICS 4/5						
ISUP selection	DICC 4/4 AND NOT DICC 4	/C AND D	100 4/4				
criteria	PICS 1/4 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 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 BICC side with the coding of the Nature of Connection Indicators parameter: "COT to be expected"						
SIP parameter	SIP INVITE: Audio RTP/AVE		iter. COI	to be exp	eci	eu	
values	200 OK: Audio RTP/AVP 8	- 0 0					
ISUP parameter	IAM Continuity Indicator: CC	T to be	expected	USI: A-lav	w ∩r	ahsent	
values	COT; Continuity Indicator: c			, 001. 71 141	01	abount	
Comments	SIP-I		SU <sup>-</sup>	Т		ISUP	
	INVITE(IAM)	<b>→</b>			<del>}</del>	IAM	
	183 Session Progress	+				,	
	PRACK	<b>→</b>					
	200 OK PRACK	+					
	UPDATE	<b>→</b>			<del>}</del>	COT	
	200 OK UPDATE	+					
		F	Precondition	ons met			
	180 Ringing(ACM)	+			<del>(</del>	ACM	
	200 OK INVITE(ANM)	+			<del>(</del>	ANM	
	ACK	<b>→</b>					
			Convers	ation			
	BYE(REL)	<b>→</b>			<del>&gt;</del>	REL	
	200 OK BYE(RLC)	+		•	<del>(</del>	RLC	

TP101003	SIP reference: RFC 3261			ISUP reference:					
					Q.1912.5 clause 6.1.2 (i,2ai)				
TSS reference	SIP-ISUP/Basic call/ Sending	of the	Initial Add	Iress Mes	ssage	(IAM)/			
SIP selection	PICS 4/4 AND PICS 4/5								
criteria									
ISUP selection	PICS 1/4 AND NOT 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 Require header								
	<ul> <li>the SUT shall delete μ-law (PCMU), if present, from the media description that it will</li> </ul>								
	send back in the SDP answer								
	the IAM shall be sent out immediately on the BICC side with the coding of the Nature								
	of Connection Indicators parameter: "COT to be expected"								
SIP parameter	SIP INVITE: Audio RTP/AVP 0 8								
values	200 OK: Audio RTP/AVP 8								
ISUP parameter	IAM Continuity Indicator: COT to be expected, USI: A-law or absent								
values	COT; Continuity Indicator: co	ntinuit	<b>.</b>						
Comments	SIP-I		SL	JT		ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	183 Session Progress	+							
	PRACK	<b>→</b>							
	200 OK PRACK	+							
	UPDATE	<b>→</b>			<b>→</b>	COT			
	200 OK UPDATE	+							
			Preconditi	ons met					
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
	ACK	<b>→</b>							
			Conver	sation					
	BYE(REL)	<b>→</b>			<b>→</b>	REL			
	200 OK BYE(RLC)	+			<b>←</b>	RLC			

TP101004	SIP reference: RFC	3261		ISUP reference: Q.1912.5 clause 6.1.2 (i,2aii)					
TSS reference	SIP-ISUP/Basic call/ Sending	of the I	Initial Add	lress Mes	ssage	(IAM)/			
SIP selection	PICS 4/4 AND PICS 4/5								
criteria									
ISUP selection	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1								
criteria									
Test purpose	offer 100rel extensions and p the SUT shall delete μ-la send back in the SDP an	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer 100rel extensions and preconditions extensions in the SIP Supported header::</li> <li>the SUT shall delete μ-law (PCMU), if present, from the media description that it will send back in the SDP answer</li> <li>the IAM shall be sent out immediately on the ISUP side with the Continuity check</li> </ul>							
	indicator "continuity check required on this circuit"								
SIP parameter values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 8								
ISUP parameter						circuit, USI: A-law or absent			
values	COT Continuity Indicator: cor	ntinuity	check s	uccessfu	ıl				
Comments	SIP-I		SL	JT		ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	183 Session Progress	<b>→</b>							
	PRACK	+							
	200 OK PRACK	<b>→</b>							
	UPDATE	<b>→</b>			<b>→</b>	COT			
	200 OK UPDATE	<b>←</b>							
		F	Preconditi	ions met					
	180 Ringing(ACM)	<b>←</b>			+	ACM			
	200 OK INVITE(ANM)	<b>←</b>			+	ANM			
	ACK	<b>→</b>							
			Conver	sation					
	BYE(REL)	<b>→</b>			<b>→</b>	REL			
	200 OK BYE(RLC)	+			+	RLC			

TP101005	SIP reference: RFC 3261			ISUP reference:					
						2.5 clause 6.1.2 (i,2aii)			
TSS reference	SIP-ISUP/Basic call/ Sending	of the	Initial Add	Iress Mes	sage	(IAM)/			
SIP selection	PICS 4/4 AND PICS 4/5								
criteria									
ISUP selection	PICS 1/5 AND NOT 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 Require header								
	<ul> <li>the SUT shall delete μ-law (PCMU), if present, from the media description that it will</li> </ul>								
	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"								
SIP parameter	SIP INVITE: Audio RTP/AVP 0 8								
values	200 OK: Audio RTP/AVP 8  IAM Continuity Indicator: continuity check required on this circuit, USI: A-law or absent								
ISUP parameter						circuit, USI: A-law or absent			
values	COT Continuity Indicator: cor	itinuity			11				
Comments	SIP-I		SL	JT		ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	183 Session Progress	<b>→</b>							
	PRACK	<b>+</b>							
	200 OK PRACK	<b>→</b>							
	UPDATE	<b>→</b>			<b>→</b>	СОТ			
	200 OK UPDATE	+							
	122 21 1 (122 )		Preconditi	ons met					
	180 Ringing(ACM)	+			<b>←</b>	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
	ACK	<b>→</b>							
			Conver	sation					
	BYE(REL)	<b>→</b>			<b>→</b>	REL			
	200 OK BYE(RLC)	<b>←</b>			<b>←</b>	RLC			

TP101006	SIP reference: RFC		ISUP reference: Q.1912.5 clause 6.1.2 (i,2aii)						
TSS reference	SIP-ISUP/Basic call/ Sending	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	PICS 4/4 AND PICS 4/5								
criteria									
ISUP selection	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1								
criteria	E district Out								
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer 100rel extensions and preconditions extensions in the SIP Supported header::</li> <li>the SUT shall delete µ-law (PCMU), if present, from the media description that it will send back in the SDP answer</li> <li>the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "continuity check performed on previous circuit"</li> </ul>								
SIP parameter	SIP INVITE: Audio RTP/AVP 0.8								
values	200 OK: Audio RTP/AVP 8								
ISUP parameter	IAM Continuity Indicator: continuity check performed on previous circuit, USI: A-law or								
values	absent	-	-		-				
	COT Continuity Indicator: co	ntinuity	check s	uccessful					
Comments	SIP-I		SU	T		ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	183 Session Progress	<b>→</b>							
	PRACK	+							
	200 OK PRACK	<b>→</b>							
	UPDATE	<b>→</b>			<b>→</b>	COT			
	200 OK UPDATE	+							
			Preconditi	ons met					
	180 Ringing(ACM)	+			<del>(</del>	ACM			
	200 OK INVITE(ANM)	+			<del>(</del>	ANM			
	ACK	<b>→</b>		Ì					
			Convers	sation					
	BYE(REL)	<b>→</b>			<b>→</b>	REL			
	200 OK BYE(RLC)	+			<del>(</del>	RLC			

TP101007	SIP reference: RFC	ISUP reference: Q.1912.5 clause 6.1.2 (i,2aii)							
TSS reference	SIP-ISUP/Basic call/ Sending	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	PICS 4/4 AND PICS 4/5								
criteria									
ISUP selection	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1								
criteria	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP								
Test purpose	<ul> <li>offer 100rel extensions and preconditions extensions in the SIP Require header</li> <li>the SUT shall delete μ-law (PCMU), if present, from the media description that it will send back in the SDP answer</li> <li>the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "continuity check performed on previous circuit"</li> </ul>								
SIP parameter	SIP INVITE: Audio RTP/AVP 0 8								
values	200 OK: Audio RTP/AVP 8								
ISUP parameter	IAM Continuity Indicator: continuity check performed on previous circuit, USI: A-law or								
values	absent								
Comments	SIP-I		SU			ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	183 Session Progress	<b>→</b>							
	PRACK	+							
	200 OK PRACK	<b>→</b>							
	UPDATE	<b>→</b>			<b>→</b>	COT			
	200 OK UPDATE	+							
		F	Preconditi	ons met					
	180 Ringing(ACM)	+			<del>(</del>	ACM			
	200 OK INVITE(ANM)	+			<del>(</del>	ANM			
	ACK	<b>→</b>							
			Conver	sation					
	BYE(REL)	<b>→</b>			<b>→</b>	REL			
	200 OK BYE(RLC)	+			<del>(</del>	RLC			

TP101008	SIP reference: RFC	3261		ISUP reference: Q.1912.5 clause 6.1.2 (i,2b)					
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/								
SIP selection	PICS 4/4 AND PICS 4/5								
criteria									
ISUP selection	NOT PICS 1/6 AND PICS 4/1								
criteria									
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP								
	<ul> <li>offer 100rel extensions and preconditions extensions in the SIP Supported header::</li> <li>the SUT shall delete μ-law (PCMU), if present, from the media description that it w</li> </ul>								
	send back in the SDP answer								
	<ul> <li>the IAM shall be deferred</li> </ul>		II precond	litions have l	peen met				
SIP parameter	SIP INVITE: Audio RTP/AVP 0 8								
values	200 OK: Audio RTP/AVP 8								
ISUP parameter	IAM USI: A-law or absent								
values		1							
Comments	SIP-I		SL	JT	ISUP				
	INVITE(IAM)	<b>→</b>							
	183 Session Progress	+							
	PRACK	<b>→</b>							
	200 OK PRACK	+							
	UPDATE	<b>→</b>		-3	IAM				
	200 OK UPDATE	+							
			Preconditi	ons met					
	180 Ringing(ACM)	+		€	- ACM				
	200 OK INVITE(ANM)	<b>←</b>		€	- ANM				
	ACK	<b>→</b>							
			Conver	sation					
	BYE(REL)	<b>→</b>		7	REL				
	200 OK BYE(RLC)	<b>←</b>		•	- RLC				

TP101009	SIP reference: RF			ISUP reference: Q.1912.5 clause 6.1.2 (i,2b)						
TSS reference	SIP-ISUP/Basic call/ Sendir	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/								
SIP selection	PICS 4/4 AND PICS 4/5									
criteria										
ISUP selection	NOT PICS 1/6 AND PICS 4/1									
criteria										
Test purpose		Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP								
	offer 100rel extensions and									
			ЛU), if pre	sent, from t	he n	nedia description that it will				
	send back in the SDP a									
	the IAM shall be deferred until all preconditions have been met									
SIP parameter	SIP INVITE: Audio RTP/AVP 0 8									
values	200 OK: Audio RTP/AVP 8									
ISUP parameter	IAM USI: A-law or absent									
values										
Comments	SIP-I		SU	JT		ISUP				
	INVITE(IAM)	→								
	183 Session Progress	<b>←</b>								
	PRACK	→								
	200 OK PRACK	+								
	UPDATE	→		-	<b>→</b>	IAM				
	200 OK UPDATE	+								
			Preconditi	ons met						
	180 Ringing(ACM)	+		•	1	ACM				
	200 OK INVITE(ANM)	+		-	1	ANM				
	ACK	<b>→</b>	_							
			Conver	sation						
	BYE(REL)	<b>→</b>		-	<b>→</b>	REL				
	200 OK BYE(RLC)	+		•	<del>(</del>	RLC				

TP101010	SIP reference: RFC	3261		ISUP reference: Q.1912.5 clause 6.1.2 (i,1)					
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/								
SIP selection	NOT PICS 4/4 AND NOT 4/5								
criteria									
ISUP selection	PICS 1/6								
criteria									
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP								
	offer	ffer							
	<ul> <li>the SUT shall delete A-la</li> </ul>	the SUT shall delete A-law (PCMA) if both A-law (PCMA) and µ-law (PCMU) were							
	present in the offer of the	present in the offer of the media description, that it will send it back in the SDP answer							
	<ul> <li>the SUT shall immediate</li> </ul>	ly send	out the IA	M					
SIP parameter	SIP INVITE: Audio RTP/AVP	8 0							
values	200 OK: Audio RTP/AVP 0								
ISUP parameter	IAM USI: μ-law								
values									
Comments	SIP-I		SU	T		ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	180 Ringing(ACM)	<b>←</b>			+	ACM			
	200 OK INVITE(ANM)	<b>←</b>			<b>←</b>	ANM			
	ACK	<b>→</b>							
			Convers	sation					
	BYE(REL)	<b>→</b>		_	<b>→</b>	REL			
	200 OK BYE(RLC)	+	_		+	RLC			

TP101011	SIP reference: RFC		ISUP reference:						
		2.5 clause 6.1.2 (i,2ai)							
TSS reference	SIP-ISUP/Basic call/ Sending	of the	Initial Add	lress Mes	ssage	(IAM)/			
SIP selection	PICS 4/4 AND PICS 4/5								
criteria									
ISUP selection	PICS 1/4 AND PICS 1/6 AND PICS 4/1								
criteria									
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer 100rel extensions and preconditions extensions in the SIP Supported header::</li> <li>the SUT shall delete A-law (PCMA) if both A-law (PCMA) and μ-law (PCMU) were present in the offer of the media description, that it will send it back in the SDP answer</li> <li>the IAM shall be sent out immediately on the BICC side with the coding of the Nature of Connection Indicators parameter: "COT to be expected"</li> </ul>								
SIP parameter	SIP INVITE: Audio RTP/AVP 0 8								
values	200 OK: Audio RTP/AVP 0								
ISUP parameter	IAM USI: µ-law; Nature of Cor	nnectio	n Indicato	rs param	eter: '	"COT to be expected" COT;			
values	Continuity Indicator: continui	ty		•		-			
Comments	SIP-I		SL	JT		ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	183 Session Progress	+							
	PRACK	1							
	200 OK PRACK	+							
	UPDATE	1			<b>→</b>	COT			
	200 OK UPDATE	4							
		F	Preconditi	ons met					
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
	ACK	<b>→</b>							
			Conver	sation					
	BYE(REL)	<b>→</b>			<b>→</b>	REL			
	200 OK BYE(RLC)	+			+	RLC			

TP101012	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 6.1.2 (i,2ai)					
TSS reference	SIP-ISUP/Basic call/ Sending	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/								
SIP selection	PICS 4/4 AND PICS 4/5									
criteria										
ISUP selection	PICS 1/4 AND 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 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 BICC side with the coding of the Nature of Connection Indicators parameter: "COT to be expected"									
SIP parameter	SIP INVITE: Audio RTP/AVP 0 8									
values	200 OK: Audio RTP/AVP 0									
ISUP parameter	IAM USI: μ-law; Nature of Cor	nnectio	n Indicato	rs param	eter: '	'COT to be expected" COT;				
values	Continuity Indicator: continui	ty								
Comments	SIP-I		SL	JT		ISUP				
	INVITE(IAM)	<b>↑</b>			<b>→</b>	IAM				
	183 Session Progress	4								
	PRACK	<b>^</b>								
	200 OK PRACK	4								
	UPDATE	<b>→</b>			<b>→</b>	COT				
	200 OK UPDATE	4								
		F	Preconditi	ions met						
	180 Ringing(ACM)	+			+	ACM				
	200 OK INVITE(ANM)	+			+	ANM				
	ACK	<b>→</b>								
			Conver	sation						
	BYE(REL)	<b>→</b>		_	<b>→</b>	REL				
	200 OK BYE(RLC)	+			+	RLC				

TP101013	SIP reference: RF	C	ISUP reference: Q.1912.5 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	PICS 1/5 AND PICS 1/6 AND PICS 4/1								
criteria									
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer 100rel extensions and preconditions extensions in the SIP Supported header::</li> <li>the SUT shall delete A-law (PCMA) if both A-law (PCMA) and μ-law (PCMU) ) were present in the offer of the media description, that it will send it back in the SDP answer</li> <li>the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "continuity check required on this circuit"</li> </ul>								
SIP parameter	SIP INVITE: Audio RTP/AVP								
values	200 OK: Audio RTP/AVP 0								
ISUP parameter	IAM: USI: µ-law; Continuity check indicator "continuity check required on this circuit"								
values	COT: Continuity Indicator: co	ontinuit	y check s	uccessf	ul	-			
Comments	SIP-I		SU	Т		ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	183 Session Progress	+							
	PRACK	<b>→</b>							
	200 OK PRACK	<b>←</b>							
	UPDATE	<b>→</b>			<b>→</b>	COT			
	200 OK UPDATE	+							
			Preconditi	ons met					
	180 Ringing(ACM)	+			<b>←</b>	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
	ACK	<b>→</b>							
			Convers	sation					
	BYE(REL)	<b>→</b>			<b>→</b>	REL			
	200 OK BYE(RLC)	+			+	RLC			

TP101014	SIP reference: RFC 3261			ISUP reference:					
				Q.1912.5 clause 6.1.2 (i,2aii)					
TSS reference	SIP-ISUP/Basic call/ Sending	of the	Initial Add	lress Mes	sage	(IAM)/			
SIP selection	PICS 4/4 AND PICS 4/5								
criteria									
ISUP selection	PICS 1/5 AND PICS 1/6 AND PICS 4/1								
criteria									
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP								
	offer 100rel extensions and preconditions extensions in the SIP 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"								
SIP parameter	SIP INVITE: Audio RTP/AVP 0 8								
values	200 OK: Audio RTP/AVP 0								
ISUP parameter	IAM: USI: μ-law; Continuity check indicator "continuity check required on this circuit"								
values	COT: Continuity Indicator: co	ntinuit	1		ul				
Comments	SIP-I		SL	JT		ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	183 Session Progress	+							
	PRACK	<b>→</b>							
	200 OK PRACK	+							
	UPDATE	<b>→</b>			<b>→</b>	COT			
	200 OK UPDATE	+							
			Preconditi	ons met					
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
	ACK	<b>→</b>							
			Conver	sation					
	BYE(REL)	<b>→</b>			<b>→</b>	REL			
	200 OK BYE(RLC)	+			<b>←</b>	RLC			

TP101015	SIP reference: RFC	3261		·-	SUP reference: 2.5 clause 6.1.2 (i,2aii)			
TSS reference	SIP-ISUP/Basic call/ Sending	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5	01 1110 11	illiai / lac	noco moccago	(h uviji			
criteria								
ISUP selection	PICS 1/5 AND PICS 1/6 AND	PICS 1/5 AND PICS 1/6 AND PICS 4/1						
criteria								
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer 100rel extensions and preconditions extensions in the SIP Supported header:</li> <li>the SUT shall delete A-law (PCMA) if both A-law (PCMA) and μ-law (PCMU) ) were present in the offer of the media description, that it will send it back in the SDP answer</li> <li>the IAM shall be sent out immediately on the ISUP side with the Continuity check</li> </ul>							
	indicator "continuity che							
SIP parameter	SIP INVITE: Audio RTP/AVP			•				
values	200 OK: Audio RTP/AVP 0							
ISUP parameter	IAM: USI: μ-law; Continuity ch	neck ind	icator <b>co</b>	ntinuity check	performed on previous circuit			
values	COT: Continuity Indicator: co	ntinuity	check s	successful				
Comments	SIP-I		SL	IT	ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	183 Session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>		→	COT			
	200 OK UPDATE	+						
		P	reconditi	ions met				
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	<b>→</b>						
			Conver	sation				
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		<b>←</b>	RLC			

TP101016	SIP reference: RFC	3261		ISUP reference: Q.1912.5 clause 6.1.2 (i,2aii)				
TSS reference	SIP-ISUP/Basic call/ Sending	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5							
criteria								
ISUP selection	PICS 1/5 AND PICS 1/6 AND	PICS 4	4/1					
criteria								
Test purpose	Ensure that if the SUT upon re							
	offer 100rel extensions and p							
						and μ-law (PCMU) ) were		
						end it back in the SDP answer		
	<ul> <li>the IAM shall be sent out</li> </ul>							
	indicator "continuity che		formed o	n previou	s cir	cuit "		
SIP parameter	SIP INVITE: Audio RTP/AVP	8 0						
values	200 OK: Audio RTP/AVP 0							
ISUP parameter						performed on previous circuit		
values	COT: Continuity Indicator: co	ntinuit			l			
Comments	SIP-I		SL	JT		ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	183 Session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>			<u>→</u>	COT		
	200 OK UPDATE	+						
			Precondit	ions met				
	180 Ringing(ACM)	<b>←</b>			<del>-</del>	ACM		
	200 OK INVITE(ANM)	<b>←</b>			<b>←</b>	ANM		
	ACK	<b>→</b>						
			Conver	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	<b>←</b>			+	RLC		

TP101017	SIP reference: RFC	3261			-	SUP reference:		
	Q.1912.5 clause 6.1.2 (i,2b)							
TSS reference		SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5							
criteria								
ISUP selection	PICS 1/6 AND PICS 4/1							
criteria								
Test purpose	Ensure that if the SUT upon r							
	offer 100rel extensions and p							
						) and µ-law (PCMU) were		
						end it back in the SDP answer		
CID noromotor	the IAM shall be deferred  CIR INVITE: Audio RED(A)/R		ii precond	litions nav	e bee	en met		
SIP parameter values	SIP INVITE: Audio RTP/AVP	08						
	200 OK: Audio RTP/AVP 0							
ISUP parameter	IAM USI: μ-law							
values	SIP-I	1		ı <del>.</del>		ICLID		
Comments			SL	)		ISUP		
	INVITE(IAM)	<b>→</b>						
	183 Session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	<b>←</b>						
	UPDATE	<b>→</b>			<b>→</b>	IAM		
	200 OK UPDATE	+						
			Preconditi	ons met				
	180 Ringing(ACM)	+			<del>-</del>	ACM		
	200 OK INVITE(ANM)	+			<del>-</del>	ANM		
	ACK	<b>→</b>						
			Conver	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP101018	SIP reference: RFC 3261			ISUP reference: Q.1912.5 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	Ensure that if the SUT upon r						
	offer 100rel extensions and p						
						) and μ-law (PCMU) were	
						end it back in the SDP answer	
	<ul> <li>the IAM shall be deferred</li> </ul>	d until a	III precond	ditions hav	e bee	en met	
SIP parameter	SIP INVITE: Audio RTP/AVP	8 0					
values	200 OK: Audio RTP/AVP 0						
ISUP parameter	IAM USI: μ-law						
values							
Comments	SIP-I		SL	JT		ISUP	
	INVITE(IAM)	<b>→</b>					
	183 Session Progress	<b>←</b>					
	PRACK	<b>→</b>					
	200 OK PRACK	<b>←</b>					
	UPDATE	<b>→</b>			<b>↑</b>	IAM	
	200 OK UPDATE	<b>←</b>					
			Preconditi	ions met			
	180 Ringing(ACM)	+			+	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
	ACK	<b>→</b>					
			Conver	sation			
	BYE(REL)	<b>→</b>			<b>→</b>	REL	
	200 OK BYE(RLC)	+			+	RLC	

TP101019	SIP reference: RFC	3261			Q.19 Q.19	SUP reference: 012.5 clause 6.1.3.2 012.5 clause 6.1.3.3 012.5 clause 6.1.3.4			
TSS reference	SIP-ISUP/Basic call/ Sending	of the	Initial Add	ress Mes	ssage	(IAM)/			
SIP selection criteria									
ISUP selection criteria	NOT PICS 1/9 AND NOT PIC	CS 4/4 a	and NOT F	PICS 4/5					
Test purpose	<ul> <li>Ensure that the SUT on receipt of an INVITE message sends an IAM message, where</li> <li>the Calling party's category is generated from the Calling Party's Category present in the encapsulated IAM</li> <li>the Nature of Connection Indicators (NCI) is generated by the MGCF using the Nature of Connection Indicators received in the encapsulated IAM</li> <li>the appropriate values of the Forward Call Indicator parameter are generated by the MGCF using the Forward Call Indicators parameter present within the received encapsulated IAM</li> </ul>								
SIP parameter values									
ISUP parameter values									
Comments	SIP-I		SU	T		ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
	ACK								
			Convers	sation					
	BYE(REL)	<b>→</b>			<b>→</b>	REL			
	200 OK BYE(RLC)	+			+	RLC			

P101020	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause 6.1.3.5					
TSS reference	SIP-ISUP/Basic call/ Sending	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	NOT PICS 4/4 and NOT PIC								
criteria									
ISUP selection									
criteria									
Test purpose	Ensure that the SUT in the lo encapsulated IAM message	le state	on receip	t of a INV	/ITE m	nessage with an			
	The TMR and USI shall be ta	aken fro	m the enc	ansulated	HSHE				
	sends an IAM message, with								
	the encapsulated ISUP	1 110 110		i woalam	rtoqu	noment (Twit) taken nem			
SIP parameter	SIP INVITE								
values									
ISUP parameter values	IAM; USI; ISDN_BC_ITR; TN	ЛR							
Comments	SIP-I		SU	Т		ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	180 Ringing(ACM)	+			4	ACM			
	200 OK INVITE(ANM)	+			4	ANM			
			Convers	sation					
	BYE(REL)	<b>→</b>		·	1	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				

TP101021	SIP reference: RFC 3261			_	SUP reference: 012.5 clause 6.1.3.5			
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	NOT PICS 4/4 and NOT PIC	_		.occuge	(ii iii)			
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the SUT in the I encapsulated IAM message  sends an IAM message	the HLC sha	ll be taken f	rom the	encapsulated ISUP			
SIP parameter	INVITE;				·			
values								
ISUP parameter	IAM; Access transport par	ameter HLC	: HLC_VALI	JE; USI				
values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	<b>→</b>						
		Co	nversation					
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

	Values and selection criteria for the test purpose TP1010021
VA_01	HLC_VALUE = Telephony
	USI= speech
VA_02	HLC_VALUE = Facsimile Group 2/3 (Recommendation F.182)
	USI= 3.1 kHz audio
VA_03	HLC_VALUE == Facsimile Group 4 Class I (Recommendation F.184)
	USI= Unrestricted digital information
VA_04	HLC_VALUE == Teletex service, basic and mixed mode of operation (Recommendation
	F.230) and facsimile service Group 4, Classes II and III (Recommendation F.184)
	USI= Unrestricted digital information
VA_05	HLC_VALUE == Teletex service, basic and processable mode of operation
	(Recommendation F.220)
\/A 00	USI= Unrestricted digital information
VA_06	HLC_VALUE = Teletex service, basic mode of operation (Recommendation F.200)
\/\ 07	USI= Unrestricted digital information
VA_07	HLC_VALUE = Syntax based Videotex (Recommendations F.300 and T.102)
VA_08	USI= Unrestricted digital information  HLC_VALUE = International Videotex interworking via gateways or interworking units
VA_06	(Recommendations F.300 and T.101)
	USI= Unrestricted digital information
VA_09	HLC_VALUE = Telex service (Recommendation F.60)
V/\_00	USI= Unrestricted digital information
VA_10	HLC_VALUE = Message Handling Systems (MHS) (X.400 - Series Recommendations )
	USI= Unrestricted digital information
VA_11	HLC_VALUE = OSI application (Note 2) (X.200 - Series Recommendations )
_	USI= Unrestricted digital information
VA_12	HLC_VALUE = Audio visual (Recommendation F.721)
	USI= Unrestricted digital information

TP101022	SIP reference: RFC	3261			IS	SUP reference:		
		Q.1912.5 clause 6.1.3.9						
TSS reference	SIP-ISUP/Basic call/ Sending	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	NOT PICS 4/4 and NOT PICS	S 4/5						
criteria								
ISUP selection	PICS 4/3							
criteria								
Test purpose  SIP parameter	BICC/ISUP Hop Counter prod IAM if the Hop Counter paran The initial and successively n	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						
values								
ISUP parameter values	IAM: Hop Counter parameter	value						
Comments	SIP-I		SU	Τ		ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	180 Ringing(ACM)	<b>←</b>			+	ACM		
	200 OK INVITE(ANM)	<b>←</b>			+	ANM		
	ACK	ACK →						
			Conver	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP101023	SIP reference: RFC 3261 ISUP reference: Q.1912.5 clause 6.1.3.1							
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	PICS 1/9 AND NOT PICS 4/4 and							
criteria								
ISUP selection	NOT 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 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 values	IAM: Called party number							
Comments	SIP-I	SU		ISUP				
	INVITE(IAM) →		→	IAM				
	180 Ringing(ACM) ←		<b>←</b>	ACM				
	200 OK INVITE(ANM) ←		<b>←</b>	ANM				
	ACK →							
		Conver	sation					
	BYE(REL) →		<b>→</b>	REL				
	200 OK BYE(RLC) ←		+	RLC				

TP101024	SIP reference: RFC 3261 ISUP reference:								
				912.5 clause 6.1.3.1					
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/								
SIP selection	PICS 1/9 AND NOT PICS 4/	PICS 1/9 AND NOT PICS 4/4 and NOT PICS 4/5							
criteria									
ISUP selection	PICS 1/7								
criteria									
Test purpose	<ul> <li>Ensure that the SUT on receipt of an INVITE message with a Called party number contained in the user info component of the Request-URI</li> <li>Send an IAM Message with the called party number coded as follows</li> <li>Nature of address indicator:         <ul> <li>Analyse the information contained in received URI with user=phone, and if it is in the format:</li></ul></li></ul>								
SIP parameter values									
ISUP parameter values	IAM: Called party number								
Comments	SIP-I	SU	JT	ISUP					
	INVITE(IAM)	<b>→</b>	→	IAM					
	180 Ringing(ACM)	+	+	ACM					
	200 OK INVITE(ANM)	+	+	ANM					
	ACK	<b>→</b>							
		Conver							
	BYE(REL)	<b>→</b>	→	REL					
	200 OK BYE(RLC)	<b>←</b>	<b>+</b>	RLC					

TP101025	SIP reference: RF	C 3261			_	SUP reference:		
T00 (	OID IOLID/Di II/ Oii-	EN 383 001 clause 6.1.3.5.2.2						
TSS reference	SIP-ISUP/Basic call/ Sendir	_			ssage	(IAM)/		
SIP selection	NOT PICS 4/4 AND NOT PI	ICS 4/5 <i>P</i>	ND PICS	1/9				
criteria								
ISUP selection								
criteria								
Test purpose	Law, then independent fro	Ensure that the SUT on receipt of an INVITE message with a SDP offer for µ-Law and a-Law, then independent from the received order of preference the G.711 a-law codec shall be returned in the SDP answer as preferred codec						
SIP parameter	Offer: m=audio 4711 R			<b>0 2</b> . <b>0</b>				
values	Answer: m=audio 4712 R		-					
ISUP parameter								
values								
Comments	SIP-I		SU	Т		ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	180 Ringing(ACM)	+			<b>←</b>	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
	ACK	<b>→</b>						
			Convers	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP101026	SIP reference: RFC	3261		-	_	SUP reference:	
T00 (			A .			001 clause 6.1.3.5.2.2	
TSS reference	SIP-ISUP/Basic call/ Sending			iress Mes	sage	(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 F	PICS 4/1				
criteria							
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for μ-Law and a-						
	Law 100rel extensions and pr				the S	IP Supported header:, <b>then</b>	
	independent from the receive						
						with the coding of the Nature	
	of Connection Indicators						
	<ul> <li>the G.711 a-law codec sh</li> </ul>	nall be i	eturned i	n the SDF	ansv	wer as preferred codec	
SIP parameter	Offer: m=audio 4711 RTF	P/AVP (	8 (				
values	Answer: m=audio 4712 RTF	P/AVP 8	3 0				
ISUP parameter	IAM: Continuity Indicator: CO	T to be	expecte	d, USI: A-	law o	r absent	
values	COT: Continuity Indicator: co	ntinuity	y				
Comments	SIP-I		SU	ΙT		ISUP	
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM	
	183 Session Progress	+					
	PRACK	<b>→</b>					
	200 OK PRACK	+					
	UPDATE	<b>→</b>			<b>→</b>	COT	
	200 OK UPDATE	+					
	180 Ringing(ACM)	+			+	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
	ACK	<b>→</b>					
	-		Conver	sation	1		
	BYE(REL)	<b>→</b>			<b>→</b>	REL	
	200 OK BYE(RLC)	+			+	RLC	

TP101027	SIP reference: RFC	3261		ISUP reference:				
			EN 383	3 001 clause 6.1.3.5.2.2				
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 4/4 AND PICS 4/5 AND PICS 1/9						
criteria								
ISUP selection	PICS 1/4 AND NOT PICS 1/6	AND PICS 4/1						
criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for μ-Law and a-							
	Law 100rel extensions and pr			SIP Require header, then				
	independent from the receive							
				with the coding of the Nature				
	of Connection Indicators							
	<ul> <li>the G.711 a-law codec sh</li> </ul>	nall be returned	in the SDP ans	wer as preferred codec				
SIP parameter	Offer: m=audio 4711 RTF	P/AVP 0 8						
values	Answer: m=audio 4712 RTF	P/AVP 8 0						
ISUP parameter	IAM: Continuity Indicator: CO		ed, USI: A-law	or absent				
values	COT: Continuity Indicator: co	ntinuity						
Comments	SIP-I	S	UT	ISUP				
	INVITE(IAM)	<b>→</b>	→	IAM				
	183 Session Progress	<b>←</b>						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>	<b>→</b>	COT				
	200 OK UPDATE	+						
	180 Ringing(ACM)	+	+	ACM				
	200 OK INVITE(ANM)	+	+	ANM				
	ACK	<b>→</b>						
		Conve	ersation					
	BYE(REL)	<b>→</b>	<b>→</b>	REL				
	200 OK BYE(RLC)	<b>←</b>	<b>←</b>	RLC				

TP101028	SIP reference: RFC	3261			ISUP reference:			
					83 001 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 AND PICS 4/1							
criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for μ-Law and a-							
	Law 100rel extensions and pr				SIP Supported header:, <b>t</b>	hen		
	independent from the received order of preference							
					le with the Continuity chec	k		
	indicator "continuity che							
	<ul> <li>the G.711 a-law codec sh</li> </ul>	<ul> <li>the G.711 a-law codec shall be returned in the SDP answer as preferred codec</li> </ul>						
SIP parameter	Offer: m=audio 4711 RTF							
values	Answer: m=audio 4712 RTF							
ISUP parameter	IAM: Continuity Indicator: con				his circuit, USI: A-law or a	absent		
values	COT: Continuity Indicator: co	ntinuity	y check s	successful				
Comments	SIP-I	ļ	SL	JT	ISUP			
	INVITE(IAM)	<b>→</b>		-	IAM			
	183 Session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	1						
	UPDATE	<b>→</b>		-	COT			
	200 OK UPDATE	+						
	180 Ringing(ACM)	+		•	- ACM			
	200 OK INVITE(ANM)	+		•	- ANM			
	ACK	<b>→</b>						
			Conver	sation				
	BYE(REL)	<b>→</b>		-	REL			
	200 OK BYE(RLC)	+		•	- RLC			

TP101029	SIP reference: RFC	3261			ISUP reference:			
				EN 383	3 001 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	AND PICS	3 4/1					
criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for μ-Law and a-							
	Law 100rel extensions and pr				SIP Require header, then			
	independent from the receive							
					with the Continuity check			
	indicator "continuity che							
	<ul> <li>the G.711 a-law codec sh</li> </ul>	nall be retu	rned ir	the SDP ans	wer as preferred codec			
SIP parameter	Offer: m=audio 4711 RTF							
values	Answer: m=audio 4712 RTF							
ISUP parameter					s circuit, USI: A-law or absent			
values	COT: Continuity Indicator: co	ntinuity ch	eck s	uccessful				
Comments	SIP-I		SU	Т	ISUP			
	INVITE(IAM)	<b>→</b>		→	IAM			
	183 Session Progress	<b>←</b>						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>		<b>→</b>	COT			
	200 OK UPDATE	+						
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	<b>→</b>						
		Co	onvers	ation				
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		<b>+</b>	RLC			

TP101030	SIP reference: RFC	3261		ISUP reference: 3 001 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	1.100 1/1/11/21 100 1/0								
ISUP selection	PICS 1/5 AND NOT PICS 1/6	AND PICS 4/1							
criteria									
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for µ-Law and a-Law 100rel extensions and 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								
	<ul> <li>indicator " continuity check performed on previous circuit"</li> <li>the G.711 a-law codec shall be returned in the SDP answer as preferred codec.</li> </ul>								
OID			n the SDP ans	wer as preferred codec.					
SIP parameter values	Offer: m=audio 4711 RT Answer: m=audio 4712 RT								
ISUP parameter	IAM: Continuity Indicator: co		erformed on r	revious circuit 1191: A-law					
values	or absent	itiliaity check p	enonnea on p	orevious circuit, ooi. A-law					
Variable	COT: Continuity Indicator: co	ntinuity check	successful						
Comments	SIP-I	l sı		ISUP					
	INVITE(IAM)	<b>→</b>	<b>→</b>	IAM					
	183 Session Progress	+							
	PRACK	<b>→</b>							
	200 OK PRACK	+							
	UPDATE	<b>→</b>	→	COT					
	200 OK UPDATE	+							
	180 Ringing(ACM)	+	+	ACM					
	200 OK INVITE(ANM)	+	+	ANM					
	ACK	<b>→</b>							
		Conver	sation						
	BYE(REL)	<b>→</b>	<b>→</b>	REL					
	200 OK BYE(RLC)	<b>←</b>	←	RLC					

TP101031	SIP reference: RFC	3261			_	SUP reference:		
						001 clause 6.1.3.5.2.2		
TSS reference	SIP-ISUP/Basic call/ Sending	of the I	Initial Add	lress Mes	sage	(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 receip							
	Law 100rel extensions and pr				the SI	P Require header, <b>then</b>		
	independent from the receive							
	the IAM shall be sent out							
	indicator " continuity che							
OID	the G.711 a-law codec sh			n the SDF	ansv	ver as preferred codec		
SIP parameter values	Offer: m=audio 4711 RTF							
	Answer: m=audio 4712 RTF			- uf - u		reviews sinevit LICL A leve		
ISUP parameter values	IAM: Continuity Indicator: cor or absent	itinuity	check p	errormea	on p	revious circuit, USI. A-law		
values	COT: Continuity Indicator: co	ntinuity	v check s	uccessfu	ıl			
Comments	SIP-I		SU			ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	183 Session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>			<b>→</b>	COT		
	200 OK UPDATE	+						
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
	ACK	<b>→</b>						
			Conver	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP101032	SIP reference: RF	C 3261			ISUP refer				
						se 6.1.3.5.2.2			
TSS reference		SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	PICS 4/4 AND PICS 4/5 ANI	D PICS	1/9						
criteria									
ISUP selection	PICS 1/5 AND NOT PICS 1/	6 AND 1	NOT PICS	3 4/1					
criteria									
Test purpose	Ensure that the SUT on rece	ipt of ar	n INVITE r	nessage witl	n a SDP offer	for µ-Law and a-			
	Law 100rel extensions and p				SIP Support	ted header:, <b>then</b>			
	independent from the rece	ived or	der of pre	eference					
	<ul> <li>the shall be deferred un</li> </ul>	til all pre	econdition	s have been	met				
	<ul> <li>the G.711 a-law codec s</li> </ul>	shall be	returned i	n the SDP a	nswer as pref	ferred codec			
SIP parameter	Offer: m=audio 4711 RT	P/AVP	0 8						
values	Answer: m=audio 4712 RT	P/AVP	8 0						
ISUP parameter									
values									
Comments	SIP-I		SU	JT	ISUP				
	INVITE(IAM)	<b>→</b>							
	183 Session Progress	<b>←</b>							
	PRACK	<b>→</b>							
	200 OK PRACK	+							
	UPDATE	<b>→</b>		1	IAM				
	200 OK UPDATE	+							
	180 Ringing(ACM)	+		€	- ACM				
	200 OK INVITE(ANM)	+		€	- ANM				
	ACK	<b>→</b>							
			Conver	sation					
	BYE(REL)	<b>→</b>		-	REL				
	200 OK BYE(RLC)	+		•	• RLC				

TP101033	SIP reference: RFC 3261			ISUP reference: EN 383 001 clause 6.1.3.5.2.2			
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 1	/9				
criteria							
ISUP selection	PICS 1/5 AND NOT PICS 1/6	AND N	IOT PICS	4/1			
criteria							
Test purpose	Ensure that the SUT on receip						
	Law 100rel extensions and pr				SIP Require header, then		
	independent from the receive						
	<ul> <li>the shall be deferred until</li> </ul>						
	<ul> <li>the G.711 a-law codec sh</li> </ul>	nall be r	eturned i	n the SDP ans	swer 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	T	ISUP		
	INVITE(IAM)	<b>→</b>					
	183 Session Progress	+					
	PRACK	<b>→</b>					
	200 OK PRACK	+					
	UPDATE	<b>→</b>		<b>→</b>	IAM		
	200 OK UPDATE	<b>←</b>					
	180 Ringing(ACM)	<b>←</b>		<b>←</b>	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	<b>→</b>					
			Convers	sation			
	BYE(REL)	<b>→</b>		<b>→</b>	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP101034	SIP reference: RFC 3261			ISUP reference: EN 383 001 clause 6.1.3.5.2.2				
TSS reference	SIP-ISUP/Basic call/ Sendin	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	NOT PICS 4/4 AND NOT PI					`		
criteria								
ISUP selection	PICS 1/7							
criteria								
Test purpose	μ-Law, then independent the	Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no µ-Law, then independent the normal offer answer procedures apply  • the G.711 a-law codec shall be returned in the SDP answer						
SIP parameter	Offer: m=audio 4711 R	ΓΡ/AVP 8	3					
values	Answer: m=audio 4711 R7	TP/AVP 8	3					
ISUP parameter values								
Comments	SIP-I		SU	T		ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
	ACK	ACK →						
			Convers	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP101035	SIP reference: RFC	3261			Į	SUP reference:		
						001 clause 6.1.3.5.2.2		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 4/4 AND PICS 4/5 AND PICS 1/9						
criteria								
ISUP selection	PICS 1/4 AND NOT PICS 1/6	AND PI	CS 4/1					
criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no							
						SIP Supported header:, then		
	independent the normal offe							
						with the coding of the Nature		
	of Connection Indicators							
	<ul> <li>the G.711 a-law codec sh</li> </ul>		turned ii	n the SDP	ansv	ver		
SIP parameter	Offer: m=audio 4711 RTF							
values	Answer: m=audio 4711 RTF							
ISUP parameter	IAM: Continuity Indicator: CO		expected	<b>d</b> , USI: A-la	aw o	r absent		
values	COT: Continuity Indicator: cor	ntinuity		,				
Comments	SIP-I		SU			ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	183 Session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>			<b>→</b>	COT		
	200 OK UPDATE	+						
			econditi	ons met				
	180 Ringing(ACM)	<b>←</b>			<del>(</del>	ACM		
	200 OK INVITE(ANM)	<del>(</del>			<del>-</del>	ANM		
	ACK	<b>→</b>						
			Convers	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			<del>-</del>	RLC		

TP101036	SIP reference: RFC	3261			ISUP reference:		
					83 001 clause 6.1.3.5.2.2		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 1/9					
criteria							
ISUP selection	PICS 1/4 AND NOT PICS 1/6	AND PIC	S 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						
	independent the normal offe						
	<ul> <li>the IAM shall be sent out</li> </ul>	immediat	ely on t	the BICC sid	le with the coding of the Nature		
	of Connection Indicators	paramete	r: " <b>CO</b> 1	T to be expe	ected"		
	<ul> <li>the G.711 a-law codec sh</li> </ul>	hall be ret	urned ii	n the SDP a	nswer		
SIP parameter	Offer: m=audio 4711 RTP/AVP 8						
values	Answer: m=audio 4711 RTF						
ISUP parameter	IAM: Continuity Indicator: CO		(pecte	d, USI: A-lav	v or absent		
values	COT: Continuity Indicator: co	ntinuity					
Comments	SIP-I		SU	T	ISUP		
	INVITE(IAM)	<b>→</b>		-	IAM		
	183 Session Progress	+					
	PRACK	<b>→</b>					
	200 OK PRACK	+					
	UPDATE	<b>→</b>		7	COT		
	200 OK UPDATE	+					
		Pre	conditi	ons met			
	180 Ringing(ACM)	+		€	- ACM		
	200 OK INVITE(ANM)	+		€	- ANM		
	ACK	<b>→</b>					
		(	Convers	sation			
	BYE(REL)	<b>→</b>		-	REL		
	200 OK BYE(RLC)	+		•	- RLC		

TP101037	SIP reference: RFC 3261		ISUP reference:				
				EN 383 001 clause 6.1.3.5.2.2			
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9						
criteria							
ISUP selection	PICS 1/5 AND NOT PICS 1/6	AND F	PICS 4/1				
criteria							
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no						
	μ-Law 100rel extensions and preconditions extensions in the SIP Supported header:, <b>then</b>						
	independent the normal offe						
	the IAM shall be sent out					with the Continuity check	
	indicator "continuity che						
	<ul> <li>the G.711 a-law codec sh</li> </ul>			n the SDI	o ansv	ver	
SIP parameter	Offer: m=audio 4711 RTF		-				
values	Answer: m=audio 4711 RTF	-					
ISUP parameter	IAM: Continuity Indicator: continuity check required on this circuit, USI: A-law or absent						
values	COT: Continuity Indicator: co	ntinuit			ul	l	
Comments	SIP-I		SL	JT		ISUP	
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM	
183 Session Progress ←							
	PRACK → 200 OK PRACK ←						
	UPDATE	<b>→</b>			<b>→</b>	COT	
	200 OK UPDATE	+					
	Preconditions met						
	180 Ringing(ACM)	+			+	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
	ACK	<b>→</b>					
	Conversation						
	BYE(REL)	<b>→</b>			<b>→</b>	REL	
	200 OK BYE(RLC) ← RLC						

TP101038	SIP reference: RFC 3261		ISUP reference:				
						001 clause 6.1.3.5.2.2	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9						
criteria							
ISUP selection	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1						
criteria							
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no µ-Law 100rel extensions and preconditions extensions in the SIP Require header, <b>then</b>						
	independent the normal off						
						with the Continuity check	
	indicator "continuity che						
	the G.711 a-law codec si			n the SDP	ansv	ver	
SIP parameter	Offer: m=audio 4711 RTI		-				
values	Answer: m=audio 4711 RTI						
ISUP parameter						s circuit, USI: A-law or absent	
values	COT: Continuity Indicator: co	ntinuity			l		
Comments	SIP-I		SL	JT		ISUP	
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM	
	183 Session Progress	+					
	PRACK	<b>→</b>					
	200 OK PRACK	<b>←</b>					
	UPDATE	<b>→</b>			<b>→</b>	COT	
	200 OK UPDATE	+					
		ions met					
	180 Ringing(ACM)	1			<b>←</b>	ACM	
	200 OK INVITE(ANM)	+			<b>←</b>	ANM	
	ACK	<b>→</b>					
		Conver	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL	
	200 OK BYE(RLC)	+			<b>←</b>	RLC	

TP101039	SIP reference: RFC 3261		ISUP reference:				
				EN 383 001 clause 6.1.3.5.2.2			
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9						
criteria							
ISUP selection	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1						
criteria							
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no µ-Law 100rel extensions and preconditions extensions in the SIP Supported header:, then independent the normal offer answer procedures apply  the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "continuity check performed on previous circuit"  the G.711 a-law codec shall be returned in the SDP answer						
SIP parameter	Offer: m=audio 4711 RTI	P/AVP 8	3				
values	Answer: m=audio 4711 RTI	P/AVP 8	3				
ISUP parameter	IAM: Continuity Indicator: cor	ntinuity	check p	erformed o	on p	revious circuit, USI: A-law	
values	or absent						
	COT: Continuity Indicator: co	ntinuity					
Comments	SIP-I		SU	Т		ISUP	
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM	
	183 Session Progress	+					
	PRACK	<b>→</b>					
	200 OK PRACK	<b>←</b>					
	UPDATE	<b>→</b>			<b>→</b>	COT	
	200 OK UPDATE	<b>←</b>					
	Preconditions met						
	180 Ringing(ACM)	+			<b>←</b>	ACM	
	200 OK INVITE(ANM)	+			<b>←</b>	ANM	
	ACK	<b>→</b>					
	Conversation						
	BYE(REL)	<b>→</b>			<b>→</b>	REL	
	200 OK BYE(RLC)	+			<del>(</del>	RLC	

TP101040	SIP reference: RFC 3261			ISUP reference:				
				EN 383 001 clause 6.1.3.5.2.2				
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9							
criteria								
ISUP selection	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1							
criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no µ-Law 100rel extensions and preconditions extensions in the SIP 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 check performed on previous circuit"							
	the G.711 a-law codec sh	•		•				
SIP parameter	Offer: m=audio 4711 RTF			1 110 001	ans			
values	Answer: m=audio 4711 RTF	,,,,,,						
ISUP parameter	IAM: Continuity Indicator: cor			erformed	a no	revious circuit. USI: A-law		
values	or absent COT: Continuity Indicator: continuity check successful							
Comments	SIP-I		SU	T		ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	183 Session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>			<b>→</b>	COT		
	200 OK UPDATE	+						
	Preconditions met							
	180 Ringing(ACM)	+			<b>←</b>	ACM		
	200 OK INVITE(ANM)	+			<b>←</b>	ANM		
	ACK	<b>→</b>						
	Conversation							
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			<b>←</b>	RLC		

TP101041	SIP reference: RF	FC 3261	ISUP reference: EN 383 001 clause 6.1.3.5.2.2				
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9						
criteria							
ISUP selection	NOT PICS 1/6 AND NOT 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 deferred until all preconditions have been met  the G.711 a-law codec shall be returned in the SDP answer						
SIP parameter	Offer: m=audio 4711 R	TP/AVP 8					
values	Answer: m=audio 4711 R	TP/AVP 8					
ISUP parameter values							
Comments	SIP-I SUT ISUP						
	INVITE(IAM)	<b>→</b>		100.			
	183 Session Progress	<del>-</del>					
	PRACK	<b>→</b>					
	200 OK PRACK	+					
	UPDATE	<b>→</b>	<b>→</b>	IAM			
	200 OK UPDATE	+					
	180 Ringing(ACM)	+	+	ACM			
	200 OK INVITE(ANM)	+	+	ANM			
	ACK	<b>→</b>					
	BYE(REL)	<b>→</b>	<b>→</b>	REL			
	200 OK BYE(RLC)	+	+	RLC			

TP101042	SIP reference: RF0	C 3261			ISUP reference:				
	EN 383 001 clause 6.1.3.5.2.2								
TSS reference		SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 1	/9						
criteria									
ISUP selection	NOT PICS 1/6 AND NOT PIC	CS 4/1							
criteria									
Test purpose					a SDP offer for a-Law and no				
	μ-Law 100rel extensions and				e SIP Require header, <b>then</b>				
	independent the normal of								
	<ul> <li>the IAM shall be deferre</li> </ul>								
	<ul> <li>the G.711 a-law codec s</li> </ul>		eturned i	n the SDP and	swer				
SIP parameter	Offer: m=audio 4711 RT	-							
values	Answer: m=audio 4711 RT	P/AVP 8							
ISUP parameter									
values									
Comments	SIP-I		SU	T T	ISUP				
	INVITE(IAM)	<b>→</b>							
	183 Session Progress	+							
	PRACK	<b>→</b>							
	200 OK PRACK	+							
	UPDATE	→		→	IAM				
	200 OK UPDATE	+							
	180 Ringing(ACM)	<b>←</b>		+	ACM				
	200 OK INVITE(ANM)	<b>←</b>		+	ANM				
	ACK	<b>→</b>							
			Convers	sation					
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP101043	SIP reference: RFC	3261	_	SUP reference:				
	EN 383 001 clause 6.1.3.5.2.2							
TSS reference	SIP-ISUP/Basic call/ Sending	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	NOT PICS 4/4 AND NOT PIC	S 4/5 AND PICS	3 1/9					
criteria								
ISUP selection criteria	PICS 1/7							
Test purpose	Ensure that the SUT on recei	pt of an INVITE i	message with a	SDP offer m line without				
	a-law codec							
	the u-law codec shall be rej	ected						
SIP parameter	Offer: m=audio 4711 RTF	P/AVP 0						
values	m=audio 4712 RTF	P/AVP 8						
	Answer: m=audio 0 RTP/A\	/P 0						
ISUP parameter values								
Comments	SIP-I	SU	JT	ISUP				
	INVITE(IAM)	<b>→</b>	<b>→</b>	IAM				
	180 Ringing(ACM)	+	+	ACM				
	200 OK INVITE(ANM)	<b>←</b>	<b>←</b>	ANM				
	ACK	<b>→</b>						
		Conver	sation					
	BYE(REL)	<b>→</b>	<b>→</b>	REL				
	200 OK BYE(RLC)	+	+	RLC				

TP101044	SIP reference: RI	FC 3261			ISUP reference:		
	EN 383 001 clause 6.1.3.5.2.2						
TSS reference	SIP-ISUP/Basic call/ Sendi			ess Message	e (IAM)/		
SIP selection	PICS 4/4 AND PICS 4/5 AN	ND PICS '	1/9				
criteria							
ISUP selection	PICS 1/4 AND NOT PICS 1	/6 AND F	PICS 4/1				
criteria							
Test purpose					a SDP offer m line without		
					s in the SIP Supported header:		
					with the coding of the Nature		
	of Connection Indicator	rs parame	eter: "COT	to be expec	ted"		
	<ul> <li>the u-law codec shall</li> </ul>	be rejec	ted				
SIP parameter	Offer: m=audio 4711 R	TP/AVP (	)				
values	m=audio 4712 R	TP/AVP 8	3				
	Answer: m=audio 0 RTP/						
ISUP parameter	IAM: Continuity Indicator: C	OT to be	expected	, USI: A-law	or absent		
values	COT: Continuity Indicator: o	continuity	y				
Comments	SIP-I		SUT	Γ	ISUP		
	INVITE(IAM)	<b>→</b>		→	IAM		
	183 Session Progress	+					
	PRACK	<b>→</b>					
	200 OK PRACK	+					
	UPDATE	<b>→</b>		<b>→</b>	COT		
	200 OK UPDATE	+					
		F	Precondition	ns met			
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	→					
			Convers	ation			
	BYE(REL)	→		<b>→</b>	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP101045	SIP reference: RFC	3261			I	SUP reference:	
						001 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 receip						
	a-law codec 100rel extension						
						with the coding of the Nature	
	of Connection Indicators			to be ex	kpect(	ed"	
	the u-law codec shall be						
SIP parameter	Offer: m=audio 4711 RTF	-					
values	m=audio 4712 RTF						
	Answer: m=audio 0 RTP/A\						
ISUP parameter	IAM: Continuity Indicator: CO			d, USI: A	-law o	r absent	
values	COT: Continuity Indicator: co	ntinuity		_			
Comments	SIP-I		SU	T		ISUP	
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM	
	183 Session Progress	<b>←</b>					
	PRACK	<b>→</b>					
	200 OK PRACK	<b>←</b>					
	UPDATE	<b>→</b>			<b>→</b>	СОТ	
	200 OK UPDATE	+					
			reconditi	ons met			
	180 Ringing(ACM)	<b>←</b>			<b>←</b>	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
	ACK	<b>→</b>					
			Convers	sation			
	BYE(REL)	<b>→</b>			<b>→</b>	REL	
	200 OK BYE(RLC)	<b>←</b>			<b>←</b>	RLC	

TP101046	SIP reference: RFC	3261			_	SUP reference:
						001 clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/ Sending			Iress Mess	age	(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	S AND PI	CS 4/1			
criteria						
Test purpose	Ensure that the SUT on recei					
						in the SIP Supported header:
	<ul> <li>the IAM shall be sent out indicator "continuity che</li> </ul>					with the Continuity check
	the u-law codec shall be			tilis circui	• .	
SIP parameter	Offer: m=audio 4711 RTF		Ju			
values	m=audio 4711 RTI					
values	Answer: m=audio 0 RTP/A\					
ISUP parameter			check re	equired on	this	s circuit, USI: A-law or absent
values	COT: Continuity Indicator: co					on care, com a nam or about
Comments	SIP-I		SU			ISUP
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM
	183 Session Progress	+				
	PRACK	<b>→</b>				
	200 OK PRACK	+				
	UPDATE	<b>→</b>			<b>→</b>	COT
	200 OK UPDATE	+				
		P	reconditi	ons met		
	180 Ringing(ACM)	+			<del>(</del>	ACM
	200 OK INVITE(ANM)	+			<del>(</del>	ANM
	ACK	<b>→</b>				
			Conver	sation		
	BYE(REL)	<b>→</b>			<b>→</b>	REL
	200 OK BYE(RLC)	+			<del>(</del>	RLC

TP101047	SIP reference: RFC 3261				Į.	SUP reference:
						001 clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/ Sending	of the I	nitial Add	lress Mes	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	AND P	ICS 4/1			
criteria						
Test purpose	Ensure that the SUT on recei					
	a-law codec 100rel extension					
	<ul> <li>the IAM shall be sent out</li> </ul>					with the Continuity check
	indicator "continuity che			this circu	uit"	
	the u-law codec shall b					
SIP parameter	Offer: m=audio 4711 RTF	-				
values	m=audio 4712 RTF		}			
IOUID .	Answer: m=audio 0 RTP/A\					1 1 1 1 1 1 1 1
ISUP parameter						s circuit, USI: A-law or absent
values	COT: Continuity Indicator: co	ntinuity			JI.	lious
Comments	SIP-I		SU	) [		ISUP
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM
	183 Session Progress	<del>(</del>				
	PRACK	<b>→</b>				
	200 OK PRACK	<del>(</del>				207
	UPDATE	<b>→</b>			<b>→</b>	СОТ
	200 OK UPDATE	+				
	100 D: : (A CM)		reconditi	ons met	-	0.014
	180 Ringing(ACM)	+			+	ACM
	200 OK INVITE(ANM)	<b>+</b>			+	ANM
	ACK	<b>→</b>				
	DVE(DEL)		Convers	sation	_	DE!
	BYE(REL)	<b>→</b>			<b>→</b>	REL
	200 OK BYE(RLC)	+			+	RLC

TP101048	SIP reference: RFC	3261		EN 38	ISUP reference: 3 001 clause 6.1.3.5.2.2		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 1	1/9				
criteria							
ISUP selection criteria	PICS 1/5 AND NOT PICS 1/6	AND F	PICS 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer m line without  a-law codec 100rel extensions and preconditions extensions in the SIP Supported header:  the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "continuity check performed on previous circuit "  the u-law codec shall be rejected						
SIP parameter	Offer: m=audio 4711 RTI						
values	m=audio 4712 RTF Answer: m=audio 0 RTP/A		3				
ISUP parameter			check po	erformed on	previous circuit, USI: A-law		
values	or absent COT: Continuity Indicator: co	-	-				
Comments	SIP-I		SU	Т	ISUP		
	INVITE(IAM)	<b>→</b>		→	IAM		
	183 Session Progress	+					
	PRACK	<b>→</b>					
	200 OK PRACK	+					
	UPDATE	<b>→</b>		→	COT		
	200 OK UPDATE	+					
		F	Preconditi	ons met			
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	<b>→</b>					
			Convers	sation			
	BYE(REL)	<b>→</b>		<b>→</b>	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP101049	SIP reference: RFC	reference: RFC 3261			ISUP reference:			
					383 001 clause 6.1.3.5.2.2			
TSS reference	SIP-ISUP/Basic call/ Sending	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 1	/9					
criteria								
ISUP selection	PICS 1/5 AND NOT PICS 1/6	AND P	ICS 4/1					
criteria								
Test purpose		Ensure that the SUT on receipt of an INVITE message with a SDP offer without a-law						
	codec 100rel extensions and							
					de with the Continuity check			
	indicator "continuity che	•		n previous	circuit"			
	<ul> <li>the u-law codec shall b</li> </ul>							
SIP parameter	Offer: m=audio 4711 RTF	,,,,,,						
values	m=audio 4712 RTF		}					
	Answer: m=audio 0 RTP/A\							
ISUP parameter		ntinuity	check po	erformed o	n previous circuit, USI: A-law			
values	or absent							
	COT: Continuity Indicator: co	ntinuity			liaiin			
Comments	SIP-I		SU		ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b> IAM			
	183 Session Progress	<b>←</b>						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>		-	<b>→</b> COT			
	200 OK UPDATE	+						
			Preconditi					
	180 Ringing(ACM)	+			<b>←</b> ACM			
	200 OK INVITE(ANM)	+		•	<b>E</b> ANM			
	ACK	→						
			Convers	sation				
	BYE(REL)	→			<b>→</b> REL			
	200 OK BYE(RLC)	<b>←</b>		•	<b>E</b> RLC			

TP101050	SIP reference: RFC	3261		EN 2	ISUP reference:			
T00 (	015 101 15/5 : 11/0 1:	EN 383 001 clause 6.1.3.5.2.2  SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
TSS reference				ress Messag	ge (IAM)/			
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 1	/9					
criteria								
ISUP selection	NOT PICS 1/6 AND NOT PIC	CS 4/1						
criteria								
Test purpose	Ensure that the SUT on recei							
					ns in the SIP Supported header:			
	<ul> <li>the IAM shall be deferred</li> </ul>			litions have b	een met			
	<ul> <li>the u-law codec shall b</li> </ul>	e reject	ed					
SIP parameter	Offer: m=audio 4711 RTI	P/AVP 0						
values	m=audio 4712 RTI	P/AVP 8						
	Answer: m=audio 0 RTP/A\	VP 0						
ISUP parameter								
values								
Comments	SIP-I		SU	T	ISUP			
	INVITE(IAM)	<b>→</b>						
	183 Session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>		7	IAM			
	200 OK UPDATE	+						
		P	reconditi	ons met				
	180 Ringing(ACM)	+		+	- ACM			
	200 OK INVITE(ANM)	+		+	- ANM			
	ACK	<b>→</b>						
			Conver	sation				
	BYE(REL)	<b>→</b>		-	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP101051	SIP reference: RFC	3261			-	SUP reference:
						001 clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/ Sending			dress Mess	sage	(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 recei					
	a-law codec 100rel extension					
	<ul> <li>the IAM shall be deferred</li> </ul>			ditions hav	e bee	en met
	<ul> <li>the u-law codec shall b</li> </ul>					
SIP parameter	Offer: m=audio 4711 RTF					
values	m=audio 4712 RTF		3			
	Answer: m=audio 0 RTP/A\	√P 0				
ISUP parameter						
values						
Comments	SIP-I		SU	JT		ISUP
	INVITE(IAM)	<b>→</b>				
	183 Session Progress	+				
	PRACK	<b>→</b>				
	200 OK PRACK	+				
	UPDATE	<b>→</b>			<b>→</b>	IAM
	200 OK UPDATE	+				
		F	Preconditi	ions met		
	180 Ringing(ACM)	<b>←</b>			<b>←</b>	ACM
	200 OK INVITE(ANM)	<b>←</b>			<b>←</b>	ANM
	ACK	→				
			Conver	sation		
	BYE(REL)	<b>→</b>			<b>→</b>	REL
	200 OK BYE(RLC)	+			+	RLC

TP101052	SIP reference: RFC 3261	61 ISUP reference:							
			EI	N 383 001 clause 6.1.3.5.2.2					
TSS reference	SIP-ISUP/Basic call/ Sending of the	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	NOT PICS 4/4 AND NOT PICS 4/5	AND PI	S 1/9 AND	PICS 4/19					
criteria									
ISUP selection	NOT PICS 1/6								
criteria									
Test purpose	Ensure that the SUT on receipt of a	an INVITE	message	with a SDP offer with more than					
	one media streams and based or								
	<ul> <li>the call is refused with a 415</li> </ul>	Unsupp	orted med	ia type response					
SIP parameter	Offer: m=audio 4711 RTP/AVP 8								
values	m= audio 4712 RTP/AVP 8								
ISUP parameter									
values									
Comments	SIP-I		SUT	ISUP					
	INVITE(IAM)	INVITE(IAM) →							
	415 Unsupported media type	+	•						
	ACK	<b>→</b>							

TP101053	SIP reference: RFC 3261 ISUP reference:							
		EN 383 001 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 PIC	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND PICS 4/19						
criteria								
ISUP selection	NOT PICS 1/6							
criteria								
Test purpose	Ensure that the SUT on receipt of							
				ns extensions in the SIP Supported				
	header: and based on operator p							
	<ul> <li>the call is refused with a 41</li> </ul>	5 Unsup	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)	<b>→</b>						
	415 Unsupported media type	+						
	ACK	<b>→</b>	•					

TP101054	SIP reference: RFC 3261	SIP reference: RFC 3261 ISUP reference:					
			EN	N 383 001 clause 6.1.3.5.2.2			
TSS reference	SIP-ISUP/Basic call/ Sending of the	e Initial	Address Mes	sage (IAM)/			
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND PICS 4/19						
ISUP selection criteria	NOT PICS 1/6						
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams 100rel extensions and preconditions extensions in the SIP Require header and based on operator policy then  • the call is refused with a 415 Unsupported media type response						
SIP parameter	Offer: m=audio 4711 RTP/AVP 8		•				
values	m= audio 4712 RTP/AVP 8						
ISUP parameter values							
Comments	SIP-I		SUT	ISUP			
	INVITE(IAM)	<b>→</b>					
	415 Unsupported media type	+					
	ACK	<b>→</b>					

TP101055	SIP reference: RF	C 3261			SUP reference:			
			E	EN 383	001 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 PI	ICS 4/5 AND	PICS 1/9 ANI	D NOT	PICS 4/19			
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the SUT on rece				SDP offer with more than			
	one media streams and ba							
					dia streams and one or			
				udio s	treams shall be considered;			
	the other streams sha							
	if the SDP offer contain	ins several a	udio type m	edia st	reams, the IWU shall only			
	consider one, and reje		streams					
SIP parameter	Offer: m=audio 4711 R							
values	m= audio 4712 R							
	m= video 4713 R	TP/AVP 31						
	A	TD/AV/D 0						
	Answer: m=audio 4711 R <sup>-1</sup> m=audio 0 RTP/A							
	m=video 0 RTP/A							
ISUP parameter	III=VIGEO O RTP/F	AVESI						
values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	<b>→</b>		<u> </u>	7 4 444			
			nversation	1				
	BYE(REL)	<b>→</b>	510411011	<b>→</b>	REL			
	200 OK BYE(RLC)	+		<del>-</del>	RLC			
	LOG SIL DIL(ILLO)	•			1.120			

TP101056	SIP reference: RFC	3261			ISUP reference:	
				EN 383	3 001 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 P	ICS 1/9 AN	D NOT	Γ PICS 4/19	
criteria						
ISUP selection	PICS 1/4 AND NOT PICS 1/6	AND PICS 4	/1			
criteria						
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 Supported header: and based on operator policy then  • 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".  • if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected  • if the SDP offer contains several audio type media streams, the IWU shall only consider one, and reject the other streams  Offer: m=audio 4711 RTP/AVP 8  m= audio 4712 RTP/AVP 8  m= video 4713 RTP/AVP 31					
ISUP parameter	Answer: m=audio 4711 RTF m=audio 0 RTP/AV m=video 0 RTP/AV	'P 8 'P 31 <b>T to be expe</b>	cted, USI: /	A-law (	or absent	
values	COT: Continuity Indicator: col	ntinuity	OUT	1	LOUID	
Comments	SIP-I	<b>→</b>	SUT	<b>→</b>	ISUP IAM	
	INVITE(IAM) 183 Session Progress	<del>7</del>		7	IAW	
	PRACK	<b>→</b>				
	200 OK PRACK	<del>′</del>				
	UPDATE	<b>→</b>		<b>→</b>	COT	
	200 OK UPDATE	+		<u> </u>		
	200 011 01 27112	Precondit	ions met			
	180 Ringing(ACM)	+		+	ACM	
	200 OK INVITE(ANM)	+		+	ANM	
	ACK	<b>→</b>				
		Con	versation			
	BYE(REL)	<b>→</b>		<b>→</b>	REL	
	200 OK BYE(RLC)	+		+	RLC	

TP101057	SIP reference: R	FC 3261			ISUP reference:	
					3 001 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	ND PICS 1/9	9 AND NOT PI	CS 4/19	9	
criteria						
ISUP selection	PICS 1/4 AND NOT PICS 1	1/6 AND PI	CS 4/1			
criteria						
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams 100rel extensions and preconditions extensions in the SIP Require header and based on operator policy then  • the IAM shall be sent out immediately on the BICC side with the coding of the Nature					
	<ul> <li>of Connection Indicators parameter: "COT to be expected".</li> <li>if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected</li> <li>if the SDP offer contains several audio type media streams, the IWU shall only consider one, and reject the other streams</li> </ul>					
SIP parameter	Offer: m=audio 4711 R					
values	m= audio 4712 F m= video 4713 F	RTP/AVP 8	1			
	Answer: m=audio 4711 R m=audio 0 RTP/ m=video 0 RTP/	AVP 8 AVP 31				
ISUP parameter	IAM: Continuity Indicator: C	OT to be e	expected, USI:	A-law	or absent	
values	COT: Continuity Indicator:	continuity				
Comments	SIP-I		SUT		ISUP	
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM	
	183 Session Progress	+				
	PRACK	<b>→</b>				
	200 OK PRACK	+				
	UPDATE	<b>→</b>		<b>→</b>	COT	
	200 OK UPDATE	+				
		Preco	onditions met			
	180 Ringing(ACM)	+		+	ACM	
	200 OK INVITE(ANM)	+		+	ANM	
	ACK	<b>→</b>				
			Conversation			
	BYE(REL)	<b>→</b>		<b>→</b>	REL	
	200 OK BYE(RLC)	+		+	RLC	

TP101058	SIP reference: RF	C 3261				ISUP reference:		
						3 001 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 F	PICS 4/1					
criteria	Coord that the CLIT on your	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than						
Test purpose						ensions in the SIP Supported		
	header: and based on oper			Conditio	iis ext	ensions in the SIP Supported		
				ha 19115	) oido	with the Continuity check		
	indicator "continuity ch					with the Continuity check		
						dia streams and one or		
						streams shall be considered;		
	the other streams sha			•		,		
				type me	edia s	treams, the IWU shall only		
	consider one, and reje	ct the o	ther strea	ıms				
SIP parameter	Offer: m=audio 4711 RT	P/AVP 8	8					
values	m= audio 4712 R							
	m= video 4713 R	TP/AVP	31					
			_					
	Answer: m=audio 4711 RT		8					
	m=audio 0 RTP/A							
ISUP parameter	m=video 0 RTP/A		check re	auired (	on thi	s circuit, USI: A-law or absent		
values	COT: Continuity Indicator: co					S circuit, CCI. A law of absort		
Comments	SIP-I		SU		T .	ISUP		
	INVITE(IAM)	<b>→</b>	- 00	•	<b>→</b>	IAM		
	183 Session Progress	+			† -	7		
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>			<b>→</b>	COT		
	200 OK UPDATE	+						
		Pre	conditions	met				
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
	ACK	<b>→</b>						
			Convers	ation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP101059	SIP reference: R	FC 3261			ISUP reference:		
					3 001 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	ND PICS 1/	9 AND NOT PI	CS 4/19	e e e e e e e e e e e e e e e e e e e		
criteria							
ISUP selection	PICS 1/5 AND NOT PICS 1	1/6 AND PI	CS 4/1				
criteria							
Test purpose					a SDP offer with more than tensions in the SIP Require		
	header and based on ope			ions ex	terisions in the SIP Require		
				JP side	with the Continuity check		
	indicator "continuity c						
					dia streams and one or		
				audio s	streams shall be considered;		
	the other streams sha						
				nedia s	treams, the IWU shall only		
	consider one, and re		ner streams				
SIP parameter	Offer: m=audio 4711 R						
values	m= audio 4712 F						
	m= video 4713 F	RTP/AVP 3	1				
	Answer: m=audio 4711 R	TD/AVD 0					
	Answer: m=audio 4711 R m=audio 0 RTP/	,					
	m=video 0 RTP/						
ISUP parameter			heck required	on thi	s circuit, USI: A-law or absent		
values	COT: Continuity Indicator:				o on our, con / law or aboun		
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM		
	183 Session Progress	+					
	PRACK	<b>→</b>					
	200 OK PRACK	+					
	UPDATE	<b>→</b>		<b>→</b>	COT		
	200 OK UPDATE	+					
		Preco	onditions met				
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	<b>→</b>					
			Conversation				
	BYE(REL)	<b>→</b>		<b>→</b>	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP101060	SIP reference: RI	FC 3261		ISUP reference:				
			EN 3	383 001 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	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 sent out immediately on the ISUP side with the Continuity check indicator "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, o	only the audio	o streams shall be considered;				
	the other streams sha							
				a streams, the IWU shall only				
	consider one, and rej		eams					
SIP parameter	Offer: m=audio 4711 R							
values	m= audio 4712 F							
	m= video 4713 RTP/AVP 31							
	Answer: m=audio 4711 R m=audio 0 RTP/ m=video 0 RTP/	AVP 8 AVP 31						
ISUP parameter		ontinuity check	performed o	n previous circuit, USI: A-law				
values	or absent							
	COT: Continuity Indicator:	continuity chec	k successful					
Comments	SIP-I		SUT	ISUP				
	INVITE(IAM)	<b>→</b>	•	<b>→</b> IAM				
	183 Session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>	-	<b>→</b> COT				
	200 OK UPDATE	+						
		Precondition						
	180 Ringing(ACM)	<b>←</b>		E ACM				
	200 OK INVITE(ANM)	+	•	E ANM				
	ACK	<b>→</b>						
			ersation					
	BYE(REL)	<b>→</b>		REL				
	200 OK BYE(RLC)	←	•	F RLC				

TP101061	SIP reference: RF	C 3261			ISUP reference:			
			E	EN 383	001 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	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 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							
				edia s	treams, the IWU shall only			
010	consider one, and rej		streams					
SIP parameter values	Offer: m=audio 4711 R m= audio 4712 R m= video 4713 R	RTP/AVP 8						
	Answer: m=audio 4711 R m=audio 0 RTP// m=video 0 RTP//	AVP 8 AVP 31						
ISUP parameter	_	ontinuity che	ck performe	ed on p	previous circuit, USI: A-law			
values	or absent							
	COT: Continuity Indicator:	continuity c	heck success	ful				
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		<b>→</b>	IAM			
	183 Session Progress	+						
	PRACK	→						
	200 OK PRACK	+						
	UPDATE	<b>→</b>		→	COT			
	200 OK UPDATE	+						
			litions met					
	180 Ringing(ACM)	+		<b>←</b>	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	<b>→</b>						
		Co	nversation					
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP101062	SIP reference: R	FC 3261			ISUP reference:		
					3 001 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 AI	ND PICS 1	/9 AND NOT F	PICS 4/19	9		
criteria							
ISUP selection	NOT PICS 1/6 AND NOT F	PICS 4/1					
criteria —							
Test purpose					SDP offer with more than		
				aitions ex	tensions in the SIP Supported		
	header: and based on ope			le acces le a			
	the IAM shall be defer     if the SDR effer cents				en met dia streams and one or		
					streams shall be considered;		
	the other streams sh			auuio s	streams shan be considered,		
				madia s	treams, the IWU shall only		
	consider one, and re			ilicula 3	treams, the ivvo shall omy		
SIP parameter	Offer: m=audio 4711 F						
values	m= audio 4712 I	,					
	m= video 4713 F						
	Answer: m=audio 4711 R		}				
	m=audio 0 RTP/						
	m=video 0 RTP/	/AVP 31					
ISUP parameter							
values	0.00			1	lieur		
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>					
	183 Session Progress	+					
	PRACK	→ ←					
	200 OK PRACK	<b>→</b>		<b>→</b>	LANA		
	UPDATE	<del>7</del>		7	IAM		
	200 OK UPDATE	,					
	100 Din vin v(ACM)	<b>←</b>	conditions met		A C N A		
	180 Ringing(ACM) 200 OK INVITE(ANM)	+		<del>+</del>	ACM ANM		
	ACK	→ ·			AINIVI		
	ACK	7	Conversation	<u> </u>	+		
	BYE(REL)	<b>→</b>	CONVENSATION	<u> </u>	REL		
	200 OK BYE(RLC)	<del>7</del>		<del>7</del>	RLC		
	ZUU ON BTE(NLU)				INLO		

TP101063	SIP reference: R	FC 3261			ISUP reference:		
					001 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	ND PICS 1	/9 AND NOT F	PICS 4/19	)		
criteria							
ISUP selection	NOT PICS 1/6 AND NOT F	PICS 4/1					
criteria							
Test purpose					a SDP offer with more than		
	header and based on ope			illions ex	tensions in the SIP Require		
				hava ha			
	<ul> <li>the IAM shall be defer</li> <li>if the SDP offer conta</li> </ul>				en met dia streams and one or		
					treams shall be considered;		
	the other streams sh			auulo s	dieams shan be considered,		
				media s	treams, the IWU shall only		
	consider one, and re			ilicula 3	dealis, the ivvo shall only		
SIP parameter	Offer: m=audio 4711 R						
values	m= audio 4712 F						
	m= video 4713 F						
	Answer: m=audio 4711 R		}				
	m=audio 0 RTP/						
	m=video 0 RTP/	'AVP 31					
ISUP parameter							
values	OID I		OUT	1	Tiourn		
Comments	SIP-I	<b>→</b>	SUT		ISUP		
	INVITE(IAM)	<del>7</del>					
	183 Session Progress	<b>→</b>					
	PRACK 200 OK PRACK	<del>7</del>					
	UPDATE	<b>→</b>		<b>→</b>	IAM		
	200 OK UPDATE	<del>-</del>		7	IAW		
	200 OR OPDATE		conditions met				
	180 Pinging(ACM)	<b>+</b>	conditions met	+	ACM		
	180 Ringing(ACM) 200 OK INVITE(ANM)	+		+	ANM		
	ACK	<b>→</b>			AINIVI		
	AGR	7	Conversation	<u> </u>			
	BYE(REL)	<b>→</b>	Conversation	→	REL		
	200 OK BYE(RLC)	<del>-</del>		<del>-</del>	RLC		
	ZOO ON DIL(NLO)	•			INLO		

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

TP102001	SIP reference: RFC	3261		ISUP reference: 912.5 clause 6.2 a)				
TSS reference	SIP-ISUP/Basic call/ Sending	SIP-ISUP/Basic call/ Sending of the Subsequent Address Message (SAM)/						
SIP selection	PICS 3/4							
criteria								
ISUP selection	PICS 3/8							
criteria								
Test purpose	Ensure that the SUT receives an INVITE with the same Call-ID and From tag as a previous INVITE which was associated with a BICC/ISUP call/bearer control instance currently existing on the BICC/ISUP side whereby the number of digits in the Request-URI is 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 (	PIXIT)						
Comments	SIP-I	Sl	JT	ISUP				
	INVITE	<b>→</b>	<b>→</b>	IAM				
	INVITE	<b>→</b>	<b>→</b>	SAM				
	INVITE	<b>→</b>	<b>→</b>	SAM				
	180 Ringing	<b>←</b>	+	ACM				
	200 OK INVITE	<b>←</b>	+	ANM				
	ACK	<b>→</b>						
		Conver	sation					
	BYE(REL)	<b>→</b>	<b>→</b>	REL				
	200 OK BYE(RLC)	+	+	RLC				

TP102002	SIP reference: RFC	3261		ISUP reference: Q.1912.5 clause 6.2 b)				
TSS reference	SIP-ISUP/Basic call/ Sending of the Subsequent Address Message (SAM)/							
SIP selection	PICS 3/4	Of the Subs	equent Addre	JOS IVIC	ssage (SAIVI)/			
criteria	1 100 3/4							
ISUP selection criteria	PICS 3/8							
Test purpose	Ensure that the SUT receives an INVITE with the same Call-ID and From tag as a previous INVITE which was associated with a BICC/ISUP call/bearer control instance currently existing on the BICC/ISUP side whereby the number of digits in the Request-URI is <b>fewer</b> than the number of digits already accumulated for the call,  • then the SUT shall immediately send a <b>484 Address Incomplete</b> response for this INVITE.  • In this case no SAM is sent to BICC/ISUP procedures.							
SIP parameter values								
ISUP parameter values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>	•	<b>→</b>	IAM			
	INVITE(IAM)	<b>→</b>						
	484 Address incomplete	+		<b>→</b>	REL			
	ACK	<b>→</b>	•	+	RLC			

## 5.2.1.3 Sending of COT

TP103001	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause 6.3		
TSS reference	SIP-ISUP/Basic call/COT					
SIP selection	PICS 4/4 AND PICS 4/5					
criteria						
ISUP selection criteria	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	COT continuity indicator: Co	ntinuity	/			
values						
Comments	SIP-I		SL	JT		ISUP
	INVITE(IAM)	<b>→</b>			→	IAM
	183 Session Progress	+				
	PRACK	→				
	200 OK PRACK	+				
	UPDATE	<b>→</b>			<b>→</b>	COT
	200 OK UPDATE	+				
	180 Ringing(ACM)	+			<b>←</b>	ACM
	200 OK INVITE(ANM)	+			+	ANM
	ACK	<b>→</b>				
			Conver	sation		
	BYE(REL)	<b>→</b>			<b>→</b>	REL
	200 OK BYE(RLC)	+		•	+	RLC

TP103002	SIP reference: RF	C 3261			ISUP reference:	
				Q	.1912.5 clause 6.3	
TSS reference	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 values	COT continuity indicator: Co	ontinuity (	check suc	cessful;		
Comments	SIP-I		SL	JT	ISUP	
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM	
	183 Session Progress	+				
	PRACK	<b>→</b>				
	200 OK PRACK	+				
	UPDATE	<b>→</b>		→	COT	
	200 OK UPDATE	+				
			Precondit	ions met		
	180 Ringing(ACM)	+		<b>←</b>	ACM	
	200 OK INVITE(ANM)	+		+	ANM	
	ACK	<b>→</b>				
			Conver	sation		
	BYE(REL)	<b>→</b>		→	REL	
	200 OK BYE(RLC)	+		+	RLC	

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

TP104001	SIP reference: RFC 3261			_	SUP reference: 012.5 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)	<b>→</b>		<b>→</b>	IAM			
	183 Session Progress (ACM)	+		+	ACM(no indication)			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	<b>→</b>						
		Co	nversation					
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP104002	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause 6.5 1)		
TSS reference	SIP-ISUP/Basic call/ Recei	pt of the Ado	lress complete	e messa	ige (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		1		<del> </del>		
ISUP parameter values	ACM FCI: ISUP_ID, ISDN_ OBCI: OBCI_INBAND		),			
Comments	SIP-I		SUT		ISUP	
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM	
	180 Ringing(ACM)	+		+	ACM	
	200 OK INVITE(ANM)	+		+	ANM	
	ACK	<b>→</b>				
			Conversation			
	BYE(REL)	<b>→</b>		<b>→</b>	REL	
	200 OK BYE(RLC)	+		+	RLC	

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

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

TP105001	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause 6.6			
TSS reference	SIP-ISUP/Basic call/ Receipt of	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).					
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the <b>event information parameter event indicator</b> is set to "Alerting"  the 180 Ringing SIP response is sent  The received CPG is encapsulated in the 180 Ringing						
SIP parameter	•						
values							
ISUP parameter	ACM: Called party status "no inc	dication	"				
values	CPG; event information param	neter ev	ent indicator	: Alertir	ng		
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM		
	183 Session Progress (ACM)	+		+	ACM(no indication)		
	180 Ringing(CPG)	+		+	CPG(ALERTING)		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	<b>→</b>					
			Conversation	)			
	BYE(REL)	<b>→</b>		→	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP105002	SIP reference: RFC 32		ISUP reference: Q.1912.5 clause 6.6					
TSS reference	SIP-ISUP/Basic call/ Receipt of t	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).						
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the <b>event information parameter event indicator</b> is set to "Progress"  183 Session Progress response is sent from the I-IWU.  The received CPG is encapsulated in the 183 Session Progress							
SIP parameter					•			
values								
ISUP parameter	ACM: Called party status "no ind	ication'	1					
values	CPG; event information param	eter ev	ent indicator: I	rogre	ess			
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		1	IAM			
	183 Session Progress (ACM)	+		4	ACM(no indication)			
	183 Session (CPG)	+		4	CPG(PROGRESS)			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	<b>→</b>						
			Conversation					
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP105003	SIP reference: RFC 32		ISUP reference: Q.1912.5 clause 6.6				
TSS reference	SIP-ISUP/Basic call/ Receipt of	the Cal	l progress mess	sage (	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 "in-band information or an appropriate pattern is now available"  183 Session Progress response is sent from the I-IWU.  The received CPG is encapsulated in the 183 -session Progress						
SIP parameter					g		
values							
ISUP parameter	ACM: Called party status "no inc	dication	II .				
values	CPG; event information param pattern is now available	neter e	vent indicator:	in-bar	nd-information or an appropriate		
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM		
	183 Session Progress (ACM)	+		+	ACM(no indication)		
	183 Session (CPG)	+		+	CPG (Inbad Info available)		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	<b>→</b>					
			Conversation	•			
	BYE(REL)	<b>→</b>		<b>→</b>	REL		
	200 OK BYE(RLC)	+		+	RLC		

### 5.2.1.6 Receipt of the Answer message (ANM)

TP106001	SIP reference: RFC	3261		-	ISUP reference: 1912.5 clause 6.7
TSS reference	SIP-ISUP/Basic call/ Receipt o	f the A	nswer mess	age (ANM).	
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	<ul> <li>Ensure that the SUT, having received the ACM message, on receipt of an ANM message</li> <li>sends a 200 OK INVITE.</li> <li>The received ANM is encapsulated in the 200 OK INVITE</li> <li>The bearer path shall be connected in both directions when both of the following conditions are satisfied:</li> <li>The BICC outgoing bearer set-up procedure, (Q.1902.4) is successfully completed, and;</li> <li>The I-IWU determines (using the procedures defined in RFC 3312) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable)</li> <li>In addition, if BICC is performing the "Per-call bearer set-up in the forward direction"</li> <li>Outgoing bearer set-up procedure and the Connect Type is "notification not required", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312) that sufficient</li> </ul>				
	preconditions have been met for			oceea	
SIP parameter values	200 OK INVITE with encapsula	ated Al	VM		
ISUP parameter values	ANM				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
	ACK	<b>→</b>			
			Conversati	on	
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	+		+	RLC

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

TP107001	SIP reference: RFC	3261			ISUP reference:		
				Q.1	912.5 clause 6.4, 6.7		
TSS reference	SIP-ISUP/Basic call/ Receipt o	of the C	CONNECT m	essage (CC	DN).		
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	SDP offer was received in the message	SDP offer was received in the initial INVITE. Ensure that the SUT, on receipt of an CON					
	<ul> <li>sends a 200 OK INVITE.</li> </ul>						
	<ul> <li>The received CON is enca</li> </ul>	apsulat	ted in the 20	OK INVIT	E		
	The bearer path shall be conneare satisfied:	ected i	n both direct	ions when b	ooth of the following conditions		
	The BICC outgoing bearer	r set-u	p procedure,	(Q.1902.4)	is successfully completed, and;		
	preconditions have been s (if applicable). In addition, if BICC is performing Outgoing bearer set-up proced bearer path shall be connected and the I-IWU determines (thro	The I-IWU determines (using the procedures defined in RFC 3312) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed.					
	preconditions have been met for	or the	session to p	oceed.			
SIP parameter values							
ISUP parameter values							
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>		→	IAM		
	200 OK INVITE(CON)	<del>(</del>		+	CON		
	ACK	<b>→</b>					
			Conversat	ion			
	BYE(REL)	<b>→</b>		→	REL		
	200 OK BYE(RLC)	+		+	RLC		

### 5.2.1.8 Receipt of the REL message

TP108001	SIP reference: RFC 3	3261		-	SUP reference: 912.5 clause 6.11.2	
TSS reference	SIP-ISUP /Basic call/ Receipt of	of the Release	message	(REL)	/	
SIP selection criteria						
ISUP selection criteria						
Test purpose  SIP parameter values	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, on receipt of an ISUP REL,  • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side.  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA with the encapsulated REL message  SIP Statue-Code: SIP_FAILURE_VA (PIXIT)					
ISUP parameter values	REL; cause value: CV_ISUP (PIXIT)					
Comments	SIP-I	S	SUT		ISUP	
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM	
	SIP_FAILURE_VA(REL)	+		+	REL	
	ACK	<b>→</b>		<b>→</b>	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
V, (_10	To 17 taging of most inplote	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 20	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. 38-47) (47 is class default)
VA_21	500 Server Internal Error	Cause Value No. 50 ("requested facility not subscribed")
VA_21 VA_22	500 Server Internal Error (SIP-I only)	Cause Value No. 55 ("incoming calls barred within CUG")
VA_22 VA_23	500 Server Internal Error	Cause Value No. 57 ("bearer capability not authorized")
VA_23 VA_24	500 Server Internal Error	Cause Value No. 58 ("bearer capability not presently")
VA_24 VA_25	500 Server Internal Error	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_26	500 Server Internal Error	Cause Value in the Class 100 (service or option not implemented Cause Value No. 65 - 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_23	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 3	SIP reference: RFC 3261 ISUP reference: Q.1912.5 clause 6.11.2						
TSS reference	SIP-ISUP /Basic call/ Receipt o	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/						
SIP selection criteria								
ISUP selection criteria								
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message where the CPS indicator is set to "no indication", on receipt of an ISUP REL,  • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side.  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA with the encapsulated REL message							
SIP parameter values	SIP Statue-Code: SIP_FAILURE	_VA (PI	XIT)					
ISUP parameter values	REL; cause value: CV_ISUP (F	PIXIT)						
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	183 Session Progress(ACM) ← ACM(no indication)							
	SIP_FAILURE_VA(REL)	+		+	REL			
	ACK	<b>→</b>		<b>→</b>	RLC			

Table 2

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

TP108003	SIP reference: RFC 3261 ISUP reference: Q.1912.5 clause 6.11.2						
TSS reference	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message where the CPS indicator is set to "subscriber free", having sent a 180 Ringing message on receipt of an ISUP REL  the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side.  the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA with the encapsulated REL message						
SIP parameter values	SIP Statue-Code: SIP_FAILUR	E_VA (	PIXII)				
ISUP parameter values	REL; cause value: CV_ISUP (PIXIT)						
Comments	SIP-I	SIP-I SUT ISUP					
	INVITE(IAM)	<b>→</b>		→	IAM		
	180 Ringing(ACM) ← ACM						
	SIP_FAILURE_VA(REL)	+		+	REL		
	ACK	<b>→</b>		<b>→</b>	RLC		

TP108004	SIP reference: RFC	3261		Q	ISUP reference: 1912.5 clause 6.11.2
TSS reference	SIP-ISUP /Basic call/ Receipt	of the Re	elease me	ssage (RE	L)/
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message where the CPS indicator is set to "no indication", having received a CPG message where the event information parameter event indicator is set to "Alerting", a 180 Ringing message is sent, on receipt of an ISUP REL  the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side.  the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA with the encapsulated REL message				
SIP parameter values	SIP Statue-Code: SIP_FAILUR	E_VA (PI	XIT)		
ISUP parameter values	REL; cause value: CV_ISUP	(PIXIT)			
Comments	SIP-I		SUT	•	ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	183 Session Progress(ACM)	+		+	ACM(no indication)
	180 Ringing(CPG)	+		+	5: 5(; t==: t::: t5)
	SIP_FAILURE_VA(REL)	<b>←</b>		+	REL
	ACK	<b>→</b>		→	RLC
Comments	SIP	SUT		ISUP	
	INVITE →	<b>→</b>	IAM		
	183 Session Progress		<b>←</b> A	CM	
	180 Ringing ←	<b>←</b>	CPG		
	SIP_FAILURE_VA	+	REL		
	ACK →	<b>→</b>	RLC		

Table 3

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

TP108005	SIP reference: R	FC 3261		Q.	ISUP reference: 1912.5 clause 6.11.2
TSS reference	SIP-ISUP /Basic call/ Rece	ipt of the R	Release m	essage (RE	L)/
SIP selection					
criteria					
ISUP selection criteria					
SIP parameter values	message, having received is sent, on receipt of an ISI  the SUT immediately r	a ACM me JP REL, wherequests the le for re-sel	ssage, ha here the c e disconno lection, an	ving receive ause value o ection of the ISUP RLC	internal bearer path. When the is returned to the ISUP side.
values					
Comments	SIP-I		SUT	•	ISUP
	INVITE(IAM)	<b>→</b>		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		<b>←</b>	ANM
	ACK	<b>→</b>			
			Conversa	ation	
	BYE(REL)	+		<b>+</b>	REL
	200 OK BYE(RLC)	<b>→</b>		→	RLC

TP108006	SIP reference: RFC 3261 ISUP reference: Q.1912.5 clause 6.11.2						
TSS reference	SIP-ISUP /Basic call/ Receipt	t of the	Release messa	ge (REL)	)/		
SIP selection criteria							
ISUP selection criteria							
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a CON message, a 200 OK message is sent, on receipt of an ISUP REL, where the cause value defined as <b>CV_ISUP</b> ,  the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side.  the SUT shall send a BYE message with the encapsulated REL message						
SIP parameter values				•	-		
ISUP parameter values	REL; cause value: CV_ISUF	P (PIXIT	)				
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM		
	200 OK INVITE(CON)	+		+	CON		
	ACK	<b>→</b>					
	Conversation						
	BYE(REL)	+		+	REL		
	200 OK BYE(RLC)	<b>→</b>		<b>→</b>	RLC		

Table 4

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

TP108007	SIP reference: RFC 3261				-	SUP reference: 012.5 clause 6.11.2		
TSS reference	SIP-ISUP /Basic call/ Rece	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/						
SIP selection	PICS 4/21				, ,			
criteria								
ISUP selection								
criteria								
Test purpose	message, on receipt of an	ISUP REL version	with cau	se value :	23 the	nessage, sending out an IAM SUT shall, destination according the		
SIP parameter values								
ISUP parameter values	REL; cause value: 23							
Comments	SIP-I		SU	Τ		ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM(Destination 1)		
					+	REL(new Destination)		
					<b>→</b>	RLC		
					<b>→</b>	IAM(Destination 2)		
	180 Ringing(ACM)	<b>←</b>			+	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
	ACK	→						
			Convers	sation				
	BYE(REL)	+			+	REL		
	200 OK BYE(RLC)	→			<b>→</b>	RLC		

#### 5.2.1.9 Autonomous release at I-IWU

TP109001	SIP reference: RFC 3261	ISUP reference: Q.1912.5 clause 6.11.3								
TSS reference	SIP-ISUP/Basic call/ Autonomous rele	SIP-ISUP/Basic call/ Autonomous release at I-IWU								
SIP selection criteria										
ISUP selection criteria	PICS 4/6									
Test purpose	(because the call is not routable), the	Ensure that when a an automatic repeat attempt initiated by the SUT is not successful (because the call is not routable), the SUT shall  • send a 480 Temporarily Unavailable response to the SIP side. No actions on the ISUP								
SIP parameter values										
ISUP parameter values										
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	AM) → IAM								
	480 Temporarily unavailable (REL)	+		+	RSC					
	ACK									

TP109002	SIP reference: RFC 326	1			ISUP reference: 912.5 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 when the SUT receive information and determines that the shall send a 500 Server Interruption.	e call need	ls to be re	eleased	I based on the coding, the SUT			
SIP parameter values								
ISUP parameter values	Unknown message: Message com	patibility "F	Release c	all"				
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
			•	+	???			
	500 Server internal error(REL)	+	•	<b>→</b>	REL			
	ACK	<b>→</b>	•	+	RLC			

TP109003	SIP reference: RFC 3	261		ISUP reference: Q.1912.5 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 an 484 Address Incomplete message						
SIP parameter values		•	<u> </u>				
ISUP parameter values							
Comments	SIP-I		SUT	ISUP			
	INVITE(IAM) 484 Address incomplete ACK	→ ← →					

TP109004	SIP reference: RFC	3261		ISUP reference:					
			Q	.1912.5 clause 6.11.3					
TSS reference	SIP-ISUP/Basic call/ Autonomous release at I-IWU								
SIP selection	PICS 3/4								
criteria									
ISUP selection									
criteria									
Test purpose	Ensure that the SUT on receip	t of subseque	nt INVITE mess	sage					
	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)	<b>→</b>							
	INVITE(IAM)	<b>→</b>							
	484 Address incomplete	+							
	ACK	<b>→</b>							

TP109005	SIP reference: RFC 3	261		ISUP reference: Q.1912.5 clause 6.11.3
TSS reference	SIP-ISUP/Basic call/ Autonomo	us releas	e at I-IWU	
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that the SUT in congesti sends an 480 Temporarily Unav			message
SIP parameter values			-	
ISUP parameter values				
Comments	SIP-I		SUT	ISUP
	INVITE(IAM)	<b>→</b>		
	480 Temporarily unavailable	+		
	ACK	<b>→</b>		

TP109006	SIP reference: RFC 3	261			ISUP reference: 912.5 clause 6.11.3
TSS reference	SIP-ISUP/Basic call/ Autonomou	us release	at I-IWU	<u> </u>	312.3 Clause 0.11.3
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Ensure that the call is released of parameters  sends 500 Server Internal E		BICC/ISUP co	ompa	tibility procedure for unknown
SIP parameter values					
ISUP parameter	Unknown parameter in ACM: Pa	arameter co	ompatibility "R	Releas	se call"
values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		<b>→</b>	IAM
				<b>←</b>	ACM(???)
	500 Server internal error(REL)	+		<b>→</b>	REL
	ACK	→		+	RLC
Comments:	SIP	SUT	ISUP		
	INVITE →	<b>→</b>	IAM		
		<b>←</b>	ACM(???)	)	
	500 Server internal error ←	→	REL		
	ACK →	+	RLC		

TP109007	SIP reference: RFC 3	261			ISUP reference: 912.5 clause 6.11.3
TSS reference	SIP-ISUP/Basic call/ Autonomo	us relea	ase at I-IWU	Q.1	312.3 clause 0.11.3
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	<ul><li>Ensure that the call is released</li><li>sends 484 Address Incomp</li></ul>		expiry of T7 with	nin the	BICC/ISUP procedures
SIP parameter values	5525 .5 .7 iddi 656 iii66iii				
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
			T7 expiry		
	484 Address incomplete	+		<b>→</b>	REL
	ACK	<b>→</b>		+	RLC

TP109008	SIP reference: RFC 3	3261		Q	ISUP reference: .1912.5 clause 6.11.3
TSS reference	SIP-ISUP/Basic call/ Autonomo	us rele	ase at I-IWL	J	
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Ensure that the call is released	due ex	piry of T9 w	ithin the E	BICC/ISUP procedures
	<ul> <li>sends 480 Temporarily Una</li> </ul>	availab	le		·
SIP parameter					
values					
ISUP parameter					
values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	180 Ringing(ACM)	+		+	ACM
			T9 expir	У	
	480 Temporarily unavailable	+		→	REL
	ACK	<b>→</b>		+	RLC

## 5.2.1.10 Receipt of the Release message BYE / CANCEL

TP110001	SIP reference: R	RFC 3261			ISUP reference: 912.5 clause 6.11.1					
TSS reference	SIP-ISUP/Basic call/ Receipt of the BYE-CANCEL message									
SIP selection criteria										
ISUP selection criteria										
Test purpose	Ensure that the SUT on recause value # 16 to the ISU		BYE , the SUT	shall s	end an ISUP REL with the					
SIP parameter										
values										
ISUP parameter values	REL: Cause value #16, Loc	cation "Netw	ork beyond ar	n interwo	orking point"					
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	→		<b>→</b>	IAM					
	180 Ringing(ACM)	+		+	ACM					
	200 OK INVITE(ANM)	+		+	ANM					
	ACK	<b>→</b>								
		(	Conversation							
	BYE(REL)	→		<b>→</b>	REL					
	200 OK BYE(RLC)	+		+	RLC					

TP110002	SIP reference: RFC 3261					ISUP reference: 912.5 clause 6.11.1				
TSS reference	SIP-ISUP/Basic call/ Receipt	SIP-ISUP/Basic call/ Receipt of the BYE-CANCEL message								
SIP selection	•									
criteria										
ISUP selection criteria										
Test purpose	Ensure that the SUT on receip cause value # 31 to the ISUP		CANCE	_, the I-I\	NU sh	all send an ISUP REL with the				
SIP parameter	CANCEL without encapsulate	d ISUP	message							
values	·		_							
ISUP parameter values	REL: Cause value #31, Locati	ion "Net	work bey	ond an ir	terwo	rking point"				
Comments	SIP-I		SU	Τ		ISUP				
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM				
	180 Ringing(ACM)	+			+	ACM				
	CANCEL	<b>→</b>			<b>→</b>	REL				
	200 OK CANCEL	+			+	RLC				
	487 Request Terminated	+	-	-						
	ACK	→								

TP110003	SIP reference: RFC			_	SUP reference: 912.5 clause 6.11.1					
TSS reference	SIP-ISUP/Basic call/ Receipt	SIP-ISUP/Basic call/ Receipt of the BYA-CANCEL message								
SIP selection	•				<u> </u>					
criteria										
ISUP selection										
criteria										
Test purpose	Ensure that the SUT on receip		P BYE, the	I-IWU sha	all se	end an ISUP REL with the				
	cause value # 31 to the ISUP	side								
SIP parameter	BYE without encapsulated IS	UP mes	ssage							
values										
ISUP parameter values	REL: Cause value #31, Locat	ion "Ne	twork beyo	nd an inte	rwoı	rking point"				
Comments	SIP-I		SUT			ISUP				
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM				
	180 Ringing(ACM)	+			<del>(</del>	ACM				
	BYE	<b>→</b>			<b>→</b>	REL				
	200 OK BYE	+			<del>(</del>	RLC				
	487 Request Terminated	+								
	ACK	<b>→</b>								

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

TP111001	SIP reference: RFC 3	3261		Q.		SUP reference: 5 clause 6.11.4 and 5
TSS reference	SIP-ISUP/Basic call/ Receipt of (GRS) or Circuit group blocking					
SIP selection criteria						
ISUP selection criteria						
Test purpose	already been received on receip	pt of a F	RSC mess	age sen	ıds	nessage relating to the call has or the 200 OK INVITE message
SIP parameter values						
ISUP parameter values						
Comments	SIP-I INVITE(IAM) 180 Ringing(ACM) 200 OK INVITE(ANM) ACK BYE(REL)	+ + + +	Convers		<b>+</b>	ISUP IAM ACM ANM
	200 OK BYE(RLC)	→			<b>→</b>	RLC

TP111002	SIP reference: RF	C 3261		Į;	SUP reference:					
	Q.1912.5 clause 6.11.4 and 5									
TSS reference	SIP-ISUP/Basic call/ Receipt	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message								
	(GRS) or Circuit group block	ing messag	e (CGB) with t	he indica	tion hardware failure oriented					
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	Ensure that the SUT, when a	at least one	backward ISU	P/BICC r	message relating to the call has					
	already been received on rec	ceipt of a GI	RS message s	ends						
	<ul> <li>a BYE message if the S</li> </ul>	UT has alre	ady received a	in ACK fo	or the 200 OK INVITE message					
	which had it sent									
SIP parameter										
values										
ISUP parameter										
values										
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM					
	180 Ringing(ACM)	<b>←</b>		<b>←</b>	ACM					
	200 OK INVITE(ANM)	+		+	ANM					
	ACK	<b>→</b>								
			Conversation							
	BYE	+		+	GRS					
	200 OK BYE	<b>→</b>		<b>→</b>	GRA					

TP111003	SIP reference: RFC 3	261			Į;	SUP reference:			
	Q.1912.5 clause 6.11.4								
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message								
	(GRS) or Circuit group blocking	messa	ge (CGB) ν	vith the	indica	tion hardware failure oriented			
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	already been received on receip Message Type Indicator coded	a B 1 E moddago ii alo oo i mad amaaay roodii aa ah ii tol ah a E oo o k ii ti ii E moddago							
SIP parameter values									
ISUP parameter	Circuit Group Supervision Mess	age Ty	pe Indicato	r "hard	ware f	ailure oriented"			
values									
Comments	SIP-I		SUT			ISUP			
	INVITE(IAM)	1			<b>→</b>	IAM			
	180 Ringing(ACM)	4			+	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
	ACK	1							
			Conversa	ation					
	BYE	4		•	+	CGB(hardware failure)			
	200 OK BYE	<b>^</b>		•	<b>→</b>	CGBA			

TP111004	SIP reference: RFC 3261 ISUP reference:								
	Q.1912.5 clause 6.11.4 and 5								
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message								
	(GRS) or Circuit group blocking	message (C	GB) 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 receip								
	200 OK INVITE if the SUT has								
	<ul> <li>The SUT shall wait until it r BYE.</li> </ul>	eceives the	ACK for the 2	00 Oł	CINVITE before sending the				
SIP parameter									
values									
ISUP parameter									
values									
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		<b>→</b>	IAM				
	180 Ringing(ACM)	<del>-</del>		+	ACM				
	200 OK INVITE(ANM)	<del>-</del>		+	ANM				
				+	RSC				
	ACK	<b>→</b>		<b>→</b>	RLC				
	BYE(REL)	+							
	200 OK BYE(RLC)	<b>→</b>							

TP111005	SIP reference: RFC 3261		ISUP reference:	
		Q.1912.5 clause 6.11.4 and 5		
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message			
	(GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented			
SIP selection				
criteria:				
ISUP selection				
criteria:				
Test purpose:	Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a GRS message sends  200 OK INVITE if the SUT has not yet received an ACK for the 200 OK INVITE.			
	The SUT shall wait until it receives the ACK for the 200 OK INVITE before sending the			
CID navamatar	BYE.			
SIP parameter values:				
ISUP parameter				
values:				
Comments:	SIP-I		SUT	ISUP
Comments.	INVITE(IAM)	<b>→</b>	→ ·	
	180 Ringing(ACM)	<del>-</del>	+	,,
	200 OK INVITE(ANM)	+	+	, .e
	200 011 1111 2(7 11 111)	-		7 4 444
			<del>(</del>	GRS
	ACK	<b>→</b>	<b>→</b>	
	BYE	<del>(</del>		
	200 OK BYE	<b>→</b>		

TP111006	SIP reference: RFC 3	3261		ISUP reference:					
	Q.1912.5 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	messag	e (CGB) with	the indica	tion hardware failure oriented				
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Ensure that the SUT, when at least one backward ISUP message relating to the call has already been received on receipt of a CGB message, with the Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented", sends 200 OK INVITE if the SUT has not yet received an ACK for the 200 OK INVITE  The SUT shall wait until it receives the ACK for the 200 OK INVITE before sending the BYE.								
SIP parameter values									
ISUP parameter values	Circuit Group Supervision Mess	sage Typ	e Indicator "h	ardware f	ailure oriented"				
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	<b>←</b>		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
				<b>←</b>	CGB(hardware failure)				
	ACK	→		→	CGBA				
	BYE	+							
	200 OK BYE	→							

TP111007	SIP reference: RFC 3261			ISUP reference:					
			Q.	Q.1912.5 clause 6.11.4 and 5					
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented								
SIP selection criteria									
ISUP selection criteria									
Test purpose	Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a RSC message sends  a 500 Server Internal Error on the SIP side.								
SIP parameter values									
ISUP parameter values									
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+	•	+	ACM				
	500 Server Internal Error(REL)	+	<u> </u>	+	RSC				
	ACK	<b>→</b>		<b>→</b>	RLC				

TP111008	SIP reference: RFC 3261			ISUP reference:					
	Q.1912.5 clause 6.11.4 and 5								
TSS reference	SIP-ISUP/Basic call/ Receipt of	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message							
	(GRS) or Circuit group blocking	message (	CGB) with	the indica	ation hardware failure oriented				
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Ensure that the SUT, when at le	east one ba	ckward IS	UP/BICC	message relating to the call has				
	already been received on receip	pt of a GRS	message	sends					
	<ul> <li>a 500 Server Internal Error</li> </ul>	on the SIP	side.						
SIP parameter									
values									
ISUP parameter									
values									
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	<b>←</b>		<b>←</b>	ACM				
	500 Server Internal Error	+		+	GRS				
	ACK	→		<b>→</b>	GRA				
Comments	SIP	SUT		ISUP					
	INVITE ->		→ IAN	Λ					
	180 Ringing ←		← AC	M					
	500 Server Internal Error← ACK →	<b>←</b>	GRS → GR	:A					

TP111009	SIP reference: RFC 32	261		ISUP reference:					
			Q.1912.5 clause 6.11.4						
TSS reference	SIP-ISUP/Basic call/ Receipt of								
	(GRS) or Circuit group blocking	message	(CGB) with the	indica	tion hardware failure oriented				
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Ensure that the SUT, when at le								
	already been received on receip	t of a CGE	message, wit	h the C	Circuit Group Supervision				
	Message Type Indicator coded a	as "hardwa	re failure orie	nted", s	sends				
	<ul> <li>a 500 Server Internal Error</li> </ul>	on the SIP	side.						
SIP parameter									
values									
ISUP parameter									
values									
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		+	ACM				
	500 Server Internal Error	+		+	CGB(hardware failure)				
	ACK	<b>→</b>		<b>→</b>	CGBA				

TP111010	SIP reference: RFC	3261	ISUP reference:						
	Q.1912.5 clause 6.11.4 and 5								
TSS reference		SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message							
	(GRS) or Circuit group blocking	ng message (CG	B) with the indicate	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	<b>→</b>	<b>→</b>	IAM					
	180 Ringing(ACM)	+	+	ACM					
	200 OK INVITE(ANM)	<del>-</del>	+	ANM					
	ACK	<b>→</b>							
	INVITE(IAM) 2	<b>→</b>	<b>→</b>	IAM					
	180 Ringing(ACM)	+	+	ACM					
	200 OK INVITE(ANM)	+	+	ANM					
	ACK	<b>→</b>							
	BYE 1	+	+	GRS					
	200 OK BYE	<b>→</b>	<b>→</b>	GRA					
	BYE 2	+							
	200 OK BYE	<b>→</b>							

TP111011	SIP reference: RF	C 3261	1	SUP reference: 5 clause 6.11.4 and 5				
T00 (		(D)						
TSS reference	SIP-ISUP/Basic call/ Receipt							
010 1 1	(GRS) or Circuit group block	ing message (CC	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 CGB message, with the Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" were the Range and Status Parameter value is bigger than "1"  the SUT shall send a BYE requests for each call association.							
SIP parameter								
values								
ISUP parameter								
values								
Comments	SIP-I		SUT	ISUP				
	INVITE(IAM) 1	<b>→</b>	<b>→</b>	IAM				
	180 Ringing(ACM)	+	+	ACM				
	200 OK INVITE(ANM)	+	+	ANM				
	ACK	<b>→</b>						
	INVITE(IAM) 2	→	→	IAM				
	180 Ringing(ACM)	<b>←</b>	<b>←</b>	ACM				
	200 OK INVITE(ANM)	+	+	ANM				
	ACK	<b>→</b>						
	BYE 1	+	<b>+</b>	CGB(hardware failure)				
	200 OK BYE	→ ·	→ ·	CGBA				
	BYE 2	+	7	OGDA				
		<b>→</b>						
	200 OK BYE	7						

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

TP112001	SIP reference: RI			ISUP reference: 1912.5 clause 6.9				
TSS reference	SIP-ISUP/Basic call/ receipt of a SUSPEND message with the <b>suspend indicator</b> set to "network initiated"							
SIP selection criteria								
ISUP selection criteria								
Test purpose	<ul><li>Ensure that the SUT, on rec</li><li>"network initiated"</li><li>is transferred in an INF</li></ul>	-	SPEND messa	ge with	the <b>suspend indicator</b> set to			
SIP parameter		<b>.</b>						
values								
ISUP parameter values	SUS; Suspend indicator:	network initiat	ted					
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	<b>→</b>						
			Conversation					
	INFO(SUS)	+		+	SUS(network)			
	200 OK INFO	<b>→</b>						
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP112002	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 6.9			
TSS reference	SIP-ISUP/Basic call/ receipt of "network initiated"	a SUSF	PEND me	essage wi	th the	suspend indicator set to		
SIP selection criteria								
ISUP selection criteria	PICS 4/14							
Test purpose	Ensure that the SUT, on receipt of a SUSPEND message with the <b>suspend indicator</b> set to "network initiated"  To is started  After To is expired, the call is released							
SIP parameter	INFO: encapsulated SUS							
values								
ISUP parameter	SUS; Suspend indicator: net	work init	iated; RE	L: Cause	value	102		
values								
Comments	SIP-I		SI	UT		ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
	ACK	<b>→</b>						
			Conve	rsation				
	INFO(SUS)	+			+	SUS(network)		
	200 OK INFO	<b>→</b>				,		
			T6 is	started				
			T6 is e	expired				
	BYE(REL)	+		•	<b>→</b>	REL		
	200 OK BYE(RLC)	<b>→</b>			+	RLC		

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

TP113001	SIP reference: RFC	3261		=	SUP reference: 912.5 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: netv	vork init	iated					
values	_							
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		→	IAM			
	180 Ringing(ACM)	<b>←</b>		+	ACM			
	200 OK INVITE(ANM)	<b>←</b>		+	ANM			
	ACK	<b>→</b>						
			Conversation	on				
	INFO(SUS)	<b>←</b>		+	SUS(network)			
	200 OK INFO	<b>→</b>						
	INFO(RES)	+			RES(network)			
	200 OK INFO	<b>→</b>						
	BYE(REL)	<b>→</b>		→	REL			
	200 OK BYE(RLC)	<b>←</b>		<b>←</b>	RLC			

## 5.2.1.14 Receipt of Confusion message

TP114001	SIP reference: RFC 3	3261	Q.		P reference: 5 clause A.1.1.3					
TSS reference	ISUP-SIP/ ISUP Messages for	ISUP-SIP/ ISUP Messages for special consideration / Confusion message								
SIP selection					-					
criteria										
ISUP selection criteria										
Test purpose  SIP parameter	Ensure that the SUT after receiving the INVITE with encapsulated IAM that contains an unknown parameter, sending an IAM message as received encapsulated in the INVITE request Ensure that when the succeeding node discards an unknown parameter and send back a Confusion message if indicated in the parameter compatibility information and the sending of a Confusion message is requested, the CFN message is transported through the SIP network encapsulated in the 183 Session Progress									
values	180 Ringing containing an ACM	i with an	unknown paramei	eı						
ISUP parameter values	INFO with encapsulated CFN									
Comments	SIP-I				ISUP					
	INVITE	<b>→</b>		<b>→</b>	IAM					
	183 Session Progress(CFN)	<del>(</del>		+	CFN					
	180 Ringing(ACM)	+		+	ACM					
	200 OK INVITE(ANM)	+		+	ANM					
	ACK	<b>→</b>								
			Communication	•						
	BYE(REL)	<b>→</b>		<b>→</b>	REL					
	200 OK BYE(RLC)	+		+	RLC					

## 5.2.1.15 Segmentation

TP115001	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause A.1.1.3.1						
TSS reference	ISUP-SIP/ ISUP Messages	ISUP-SIP/ ISUP Messages for special consideration / Receipt of a user part test message								
SIP selection criteria										
ISUP selection criteria										
Test purpose	Ensure that a call can be si direction.	Ensure that a call can be successfully completed if segmentation applies in backward direction.								
SIP parameter	180 Ringing - encapsulated	ACM:	Backward c	all indicator	abs	sent or set to "no additional				
values	information will be sent"									
	No action takes place on the	e SIP s	ide							
ISUP parameter	ACM: optional forward call	indicato	r: additiona	l informatior	n wi	Il be sent in a segmentation				
values	message SGM: optional parameters									
Comments	SIP-I		SUT			ISUP				
	INVITE(IAM)	<b>→</b>		-	<del>}</del>	IAM				
	180 Ringing(ACM)	+		•	<del>(</del>	ACM				
				•	<del>(</del>	SGM				
	200 OK INVITE(ANM)	+		•	<del>(</del>	ANM				
	ACK	<b>→</b>								
			Convers	ation						
	BYE(REL)	<b>→</b>		-	<del>}</del>	REL				
	200 OK BYE	+		•	<b>(</b>	RLC				

# 5.2.2 Interworking from ISUP to SIP-I

## 5.2.2.1 Sending of the INVITE message

TP301001	SIP reference: RFC 3261				SUP reference: 12.5 clause 7.1 1 a)		
TSS reference	ISUP-SIP /Basic call/Sending of the	INVITE m	essage				
SIP selection criteria							
ISUP selection criteria							
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number and the sending complete indication  • Sends the INVITE message with the encapsulated IAM in the MIME body						
SIP parameter values			•		·		
ISUP parameter values	IAM; Called party number: with ser	nding com	plete indicat	ion			
Comments	ISUP/BICC	5	SUT		SIP-I		
	IAM →			<b>→</b>	INVITE(IAM)		
	ACM ←			<b>←</b>	180 Ringing(ACM)		
	ANM ←			+	200 OK INVITE(ANM)		
	→ ACK						
		Conv	ersation	•			
	REL →			<b>→</b>	BYE(REL)		
	RLC ←			+	200 OK BYE(RLC)		

TP301002	SIP reference: RFC 3	261		I	SUP reference:		
				Q.191	2.5 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 sta maximum number of digits us sends the INVITE message						
SIP parameter values							
ISUP parameter values	IAM; Called party number cor	nplet	e number				
Comments	ISUP/BICC		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
			Conversation	•			
	REL	<b>→</b>		<b>→</b>	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

TP301003	SIP reference: RFC 326	1		-	SUP reference: 2.5 clause 7.1 1 c)			
TSS reference	ISUP-SIP /Basic call/Sending of the INVITE message							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party  sends the INVITE message							
SIP parameter values								
ISUP parameter values	IAM; Called party number: comp	lete numbe	r					
Comments	ISUP/BICC		SUT		SIP-I			
	IAM -	<b>&gt;</b>		<b>→</b>	INVITE(IAM)			
	ACM	F		+	180 Ringing(ACM)			
	ANM	F		+	200 OK INVITE(ANM)			
		→ ACK						
		Con	versation					
	REL -	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC •	F		+	200 OK BYE(RLC)			

TP301004	SIP reference: RFC 3261		ISUP reference: Q.1912.5 clause 7.1 1 d)		
TSS reference	ISUP-SIP /Basic call/Sending of the INV	/ITE message	е		
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the <b>complete called party number</b> where the end of address signalling is determined by the expiration timer T <sub>OIW1</sub> after the receipt of the latest address message  • sends the INVITE message				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC	SUT		SIP-I	
	IAM →				
		T <sub>OIW1</sub> expir	y		
			<b>→</b>	INVITE(IAM)	
	ACM ←		+	180 Ringing(ACM)	
	ANM ←		+	200 OK INVITE(ANM)	
			→	ACK	
		Conversatio			
	REL →		→	BYE(REL)	
	RLC ←		<del>-</del>	200 OK BYE(RLC)	

TP301005	SIP reference: RFC 32	61		_	SUP reference: 12.5 clause 7.1 A)				
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message								
SIP selection	_								
criteria									
ISUP selection criteria	PICS 1/5								
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message with the 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	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
		Со	nversation						
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP301006	SIP reference: RFC 320	61		-	SUP reference: 12.5 clause 7.1 A)		
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message						
SIP selection criteria	NOT PICS 4/4 AND NOT PICS 4/5 AND 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	<b>→</b>					
	COT	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ACM	←		+	180 Ringing(ACM)		
	ANM	<del>(</del>		+	200 OK INVITE(ANM)		
	→ ACK						
			Conversation				
	REL	<b>→</b>		<b>→</b>	BYE(REL)		
	RLC	<del>(</del>		+	200 OK BYE(RLC)		

TP301007	SIP reference: RFC 3261	IS	SUP reference:					
		Q.1912.5 clause 7.1 A)						
TSS reference	ISUP-SIP/Basic call/Sending of the INVIT	ISUP-SIP/Basic call/Sending of the INVITE message						
SIP selection	NOT PICS 4/4 AND NOT PICS 4/5 AND I	NOT PICS 4/15						
criteria								
ISUP selection criteria	PICS 1/5	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 performed on previous circuit"  • Sends the INVITE after the receipt of the Continuity message with the Continuity Indicators parameter "continuity check successful"							
SIP parameter values								
ISUP parameter values								
Comments	ISUP	SUT	SIP-I					
	IAM →							
	COT →	→	INVITE(IAM)					
	ACM ←	+	180 Ringing(ACM)					
	ANM ←	+	200 OK INVITE(ANM)					
		→	ACK					
		Conversation						
	REL →	<b>→</b>	BYE(REL)					
	RLC +	<del>(</del>	200 OK BYE(RLC)					

TP301008	SIP reference: RFC 32	61		ISUP reference:			
			Q.19	912.5 clause 7.1 A)			
TSS reference	ISUP-SIP/Basic call/Sending of the	he INVITE m	essage				
	NOT PICS 4/4 AND NOT PICS 4	/5 AND NOT	PICS 4/15				
criteria							
	PICS 1/5						
criteria							
	Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit". INVITE shall not be sent if the Continuity message is received with the Continuity Indicators parameter set to "continuity check failed"						
SIP parameter							
values							
ISUP parameter							
values							
Comments	ISUP	SUT SIP-I					
	IAM	<b>→</b>					
	COT	<b>→</b>					

TP301009	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause 7.1 A)			
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message						
SIP selection criteria	NOT PICS 4/4 AND NOT PICS 4/5 AND 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 containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit". INVITE shall not be sent if the ISUP timer T8 expires. The SUT  Sends a REL message						
SIP parameter values							
ISUP parameter values							
Comments	ISUP	S	UT	SIP-I			
	IAM →						
		T8 e	xpiry				
	REL +						
	RLC →						

TP301010	SIP reference: RFC 3261		<del>-</del> '	SUP reference: 12.5 clause 7.1 B)		
TSS reference	ISUP-SIP/Basic call/Sending of the IN	VITE mess	sage			
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 4/	15				
criteria						
ISUP selection	PICS 1/5 AND PICS 4/2					
criteria						
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message where the Continuity Check indicator in the Nature of Connection Indicators parameter in the IAM is set to indicate "continuity check not required"  • sends an INVITE message without precondition using the SDP offer in the INVITE					
SIP parameter			· · · · · · · · · · · · · · · · · · ·			
values						
ISUP parameter values						
Comments	ISUP	SU	Γ	SIP-I		
	IAM →		<b>→</b>	INVITE(IAM)		
	ACM ←		+	180 Ringing(ACM)		
	ANM ←		+	200 OK INVITE(ANM)		
			<b>→</b>	ACK		
		Convers	sation			
	REL →		→	BYE(REL)		
	RLC ←		+	200 OK BYE(RLC)		

TP301011	SIP reference: RFC 32	261		(	_	SUP reference: 12.5 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 PI					
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			s parameter	in th	e IAM is set to indicate
	"continuity check required on					
						P offer in the INVITE. The SDP
	offer or answer carrying the					er set to <b>"continuity check</b>
						s are met in the SIP network
SIP parameter	Successial was received to	and the	reques	ited precond	aitiOii	s are met in the on hetwork
values						
ISUP parameter						
values						
Comments	ISUP		S	UT		SIP-I
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)
					+	183 Session Progress
					<b>→</b>	PRACK
					+	200 OK PRACK
	COT(successful)	<b>→</b>			<b>→</b>	UPDATE
					+	200 OK UPDATE
		F	Precond	itions met		
	ACM	+			+	180 Ringing(ACM)
	ANM	<b>←</b>			+	200 OK INVITE(ANM)
					<b>→</b>	ACK
			Conve	ersation		
	REL	<b>→</b>			<b>→</b>	BYE(REL)
	RLC	+			<b>←</b>	200 OK BYE(RLC)

TP301012	SIP reference: RFC 3	261		ISUP reference:	
			C	1.1912.5 clause 7.1 B)	
TSS reference	ISUP-SIP/Basic call/Sending of	the INVITE	message		
SIP selection	PICS 4/4 AND PICS 4/5 AND P	ICS 4/15			
criteria					
ISUP selection	PICS 1/5 AND PICS 4/2				
criteria					
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message where the Continuity Check indicator in the Nature of Connection Indicators parameter in the IAM is set to indicate "continuity check performed on previous circuit" sends an INVITE message with precondition using the SDP offer in the INVITE. The SDP offer or answer carrying the confirmation of a precondition being met is sent when the Continuity message with the Continuity Indicators parameter set to "continuity check successful" was received and the requested preconditions are met in the SIP network				
SIP parameter	received and the requested pred	conditions a	e met in the Si	Phetwork	
values					
ISUP parameter					
values					
Comments	ISUP		SUT	SIP-I	
	IAM	<b>→</b>		→ INVITE(IAM)	
				← 183 Session Progress	
				→ PRACK	
				← 200 OK PRACK	
	COT(successful)	<b>→</b>		→ UPDATE	
				← 200 OK UPDATE	
		Preco	nditions met		
	ACM	+		← 180 Ringing(ACM)	
	ANM	+		← 200 OK INVITE(ANM)	
				→ ACK	
			nversation		
	REL	<b>→</b>		→ BYE(REL)	
	RLC	<b>←</b>		← 200 OK BYE(RLC)	

TP301013	SIP reference: RFC 3261 ISUP reference: Q.1912.5 clause 7.1 B)						
TSS reference	ISUP-SIP/Basic call/Sending of	the IN	VITE message				
SIP selection	PICS 4/4 AND PICS 4/5 AND PI	CS 4/	15				
criteria							
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 performed on previous circuit" and sends an INVITE message with precondition using the SDP offer in the INVITE. The call has been cleared before an early dialogue has been established. Ensure that the SUT  sends CANCEL if on the SIP side the internal resource reservation was unsuccessful.  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	ISUP		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
				+	100 Trying		
	internal resource reservation was unsuccessful						
	REL(#47)	+		<b>→</b>	CANCEL		
	RLC	<b>→</b>		+	200 OK CANCEL		
				+	487 Request Terminated		
				<b>→</b>	ACK		

TP301014	SIP reference: RFC 3261				SUP reference: 112.5 clause 7.1 B)		
TSS reference	ISUP-SIP/Basic call/Sending of the	INVITE me	ssage				
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 performed on previous circuit" and sends an INVITE message with precondition using the SDP offer in the INVITE. The call has been cleared after an early dialogue with the message defined as SIP_MESSAGE_VA has been established. Ensure that the SUT  sends CANCEL if on the SIP side the internal resource reservation was unsuccessful.  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	ISUP	S	UT		SIP-I		
	IAM →			<b>→</b>	INVITE(IAM)		
				+	SIP_MESSAGE_VA		
	internal reso	urce reser	vation was ι	unsuc	ccessful		
	REL(#47) ←			<b>→</b>	CANCEL/BYE		
	RLC →			+	200 OK CANCEL/BYE		
				+	487 Request Terminated		
				<b>→</b>	ACK		

TP301015	SIP reference: RFC 3	261				SUP reference: 12.5 clause 7.1 B)	
TSS reference	ISUP-SIP/Basic call/Sending of	the IN	IVITE me	ssage			
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 call has been cleared <b>before</b> an early dialogue has been established. Ensure that the SUT  sends CANCEL if on the SIP side the internal resource reservation was unsuccessful.  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	ISUP		S	JT		SIP-I	
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)	
					<b>←</b>	100 Trying	
			rce reser	ation was	unsuc	cessful	
	REL(#47)	+			<b>→</b>	CANCEL	
	RLC	<b>→</b>			+	200 OK CANCEL	
					+	487 Request Terminated	
					<b>→</b>	ACK	

TP301016	SIP reference: RFC 32	261			SUP reference: 12.5 clause 7.1 B)		
TSS reference	ISUP-SIP/Basic call/Sending of	he INVITE	message				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PI	CS 4/15					
ISUP selection criteria	PICS 1/5 AND PICS 4/2						
Test purpose	The SUT in Idle state, receives an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit" and sends an INVITE message with precondition using the SDP offer in the INVITE. The call has been cleared after an early dialogue with the message defined as SIP_MESSAGE_VA has been established. Ensure that the SUT  • sends CANCEL if on the SIP side the internal resource reservation was unsuccessful.  • 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	ISUP		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
				<b>←</b>	SIP_MESSAGE_VA		
	internal r	esource re	servation was	unsuc	ccessful		
	REL(#47)	+		<b>→</b>	CANCEL/BYE		
	RLC	<b>→</b>	·	<del>(</del>	200 OK CANCEL/BYE		
			·	+	487 Request Terminated		
				<b>→</b>	ACK		

#### Table 5

Values	for test purpose
•	TP301014
•	TP301016
•	TP301018
•	TP301020
•	TP301031
•	TP301033
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	181 Call Is Being Forwarded
VA_3	182 Queued
VA_4	183 Session Progress without SDP

TP301017	SIP reference: RFC 3	261		=	SUP reference: 012.5 clause 7.1 B)		
TSS reference	,						
	ISUP-SIP/Basic call/Sending of						
SIP selection	PICS 4/4 AND PICS 4/5 AND P	ICS 4/1	5				
criteria							
ISUP selection criteria	PICS 1/5 AND PICS 4/2						
Test purpose	The SUT in Idle state, receives an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit" and sends an INVITE message with precondition using the SDP offer in the INVITE. The Continuity message is received with the Continuity Indicators parameter set to "continuity check failed". The call has been cleared before an early dialogue has been established. Ensure that the SUT  • sends CANCEL on the SIP side						
SIP parameter							
values							
ISUP parameter values							
Comments	ISUP		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
				+	100 Trying		
	COT(unsuccessful)	<b>→</b>		<b>→</b>	CANCEL		
				+	200 OK CANCEL		
		-	•	+	487 Request Terminated		
				<b>→</b>	ACK		

TP301018	SIP reference: RFC 320	61			SUP reference: 12.5 clause 7.1 B)			
TSS reference	ISUP-SIP/Basic call/Sending of th	ne INV	'ITE message					
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PIC	PICS 4/4 AND PICS 4/5 AND PICS 4/15						
ISUP selection criteria	PICS 1/5 AND PICS 4/2							
Test purpose  SIP parameter	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 after an early dialogue with the message defined as SIP_MESSAGE_VA has been established. Ensure that the SUT  • sends CANCEL or BYE on the SIP side							
values								
ISUP parameter values								
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
				+	SIP_MESSAGE_VA			
	COT(unsuccessful)	<b>→</b>		<b>→</b>	CANCEL			
				+	200 OK CANCEL			
				+	487 Request Terminated			
				<b>→</b>	ACK			

TP301019	SIP reference: RFC 32	261		_	SUP reference:			
	Q.1912.5 clause 7.1 B)							
TSS reference	ISUP-SIP/Basic call/Sending of	the IN∖	'ITE message					
SIP selection	PICS 4/4 AND PICS 4/5 AND PI	CS 4/1	5					
criteria								
ISUP selection	PICS 1/5 AND PICS 4/2							
criteria								
Test purpose	The SUT in Idle state, receives an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit" and sends an INVITE message with precondition using the SDP offer in the INVITE. The ISUP Timer T8 expires. The call has been cleared before an early dialogue has been established. Ensure that the SUT  • sends CANCEL on the SIP side							
SIP parameter values								
ISUP parameter								
values								
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
				+	100 Trying			
	T8 expires							
	REL(#47)	+	•	<b>→</b>	CANCEL			
	RLC	<b>→</b>		+	200 OK CANCEL			
				+	487 Request Terminated			
				<b>→</b>	ACK			

TP301020	SIP reference: RFC 3	261			SUP reference:		
				-	912.5 clause 7.1 B)		
TSS reference	ISUP-SIP/Basic call/Sending of	the INV	ITE messag	e	,		
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 ISUP Timer T8 expires. The call has been cleared after an early dialogue with the message defined as SIP_MESSAGE_VA has been established. Ensure that the SUT  • sends CANCEL or BYE on the SIP side						
SIP parameter values							
ISUP parameter values							
Comments	ISUP		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
				+	SIP_MESSAGE_VA		
	T8 expires						
	REL(#47)	+		→	CANCEL/BYE		
	RLC	<b>^</b>		+	200 OK CANCEL/BYE		
				+	487 Request Terminated		
				<b>→</b>	ACK		

TP301021	SIP reference: RFC 3261			ISUP reference: Q.1912.5 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					
Test purpose  SIP parameter values	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected"  The sending of the INVITE is delayed until all the following conditions are satisfied: Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received Bearer Set-up indication - for the forward bearer set-up case where the incoming Connect Type is "notification not required" was received					
ISUP parameter values						
Comments	BICC		SUT		SIP-I	
	IAM	<b>→</b>				
	COT(successful)	<b>→</b>		<b>→</b>	INVITE(IAM)	
	ACM	+		+	180 Ringing(ACM)	
	ANM ← 200 OK INVITE(ANM)					
				<b>→</b>	ACK	
		С	onversation			
	REL	<b>→</b>		<b>→</b>	BYE(REL)	
	RLC	+		+	200 OK BYE(RLC)	

TP301022	SIP reference: RFC 3261	· · · · · · · · · · · · · · · · · · ·	ISUP reference: 912.5 clause 7.1 C)				
TSS reference:	ISUP-SIP/Basic call/Sending of the INVIT	E message					
SIP selection criteria	NOT PICS 4/15	-					
ISUP selection criteria	PICS 1/4						
SIP parameter values ISUP parameter values	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected"  The sending of the INVITE is delayed until all the following conditions are satisfied:  Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received  APM with Action indicator set to "Connected" - for the forward bearer set-up cases (with, or without bearer control tunnelling) where the incoming Connect Type is "notification required", and for the fast set-up (backward) case						
Comments	BICC IAM → COT(successful) →	SUT	SIP-I				
	APM →	<b>→</b>	INVITE(IAM)				
	ACM ←	<del>-</del>	180 Ringing(ACM)				
	ANM ← 200 OK INVITE(ANM)						
		<b>→</b>	ACK				
	C	Conversation					
	REL →	<b>→</b>	BYE(REL)				
	RLC ←	<del>(</del>	200 OK BYE(RLC)				

TP301023	SIP reference: RFC	3261		-	SUP reference: 912.5 clause 7.1 C)		
TSS reference	ISUP-SIP/Basic call/Sending o	f the IN\	/ITE message				
SIP selection criteria	NOT PICS 4/15						
ISUP selection criteria	PICS 1/4						
SIP parameter values ISUP parameter	shall be received	E delays age, with	until all the follow the Continuity Inc	ing cor	· ·		
values							
Comments	ISUP		SUT		SIP-I		
	IAM	<b>→</b>					
	COT(successful)	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ACM         ←         180 Ringing(ACM)           ANM         ←         200 OK INVITE(ANM)						
				<b>→</b>	ACK		
			Conversation				
	REL	<b>→</b>		<b>→</b>	BYE(REL)		
	RLC	<b>←</b>		<b>←</b>	200 OK BYE(RLC)		

TP301024	SIP reference: RFC 3261		-	SUP reference: 2.5 clause 7.1 C) 2.4		
TSS reference	ISUP-SIP/Basic call/Sending of	the INVITE m	nessage			
SIP selection criteria	NOT PICS 4/15					
ISUP selection criteria	PICS 1/4					
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected"  The sending of the INVITE delays until all the following conditions are satisfied: Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received BNC set-up success indication for cases using bearer control tunnelling was received					
SIP parameter values						
ISUP parameter values						
Comments	ISUP		SUT	SIP-I		
	IAM	<b>→</b>				
	COT(successful)	<b>→</b>	→	INVITE(IAM)		
	ACM	+	+	180 Ringing(ACM)		
	ANM	+	+	200 OK INVITE(ANM)		
			<b>→</b>	ACK		
		Con	versation			
	REL	<b>→</b>	→	BYE(REL)		
	RLC	+	+	200 OK BYE(RLC)		

TP301025	SIP reference: RFC 3261	ISUP refer Q.1912.5 clau:								
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message									
SIP selection	NOT PICS 4/15									
criteria										
ISUP selection criteria	PICS 1/4									
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected"  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 →									
L										

TP301026	SIP reference: RFC 3261			SUP reference: 12.5 clause 7.1 D)					
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message								
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 4/1								
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 →		<b>→</b>	INVITE(IAM)					
			+	183 Session Progress					
			<b>→</b>	PRACK					
			+	200 OK PRACK					
	COT(successful) →		<b>→</b>	UPDATE					
			+	200 OK UPDATE					
	P	reconditions met							
	ACM ←		+	180 Ringing(ACM)					
	ANM ←		+	200 OK INVITE(ANM)					
			<b>→</b>	ACK					
		Conversation							
	REL →		<b>→</b>	BYE(REL)					
	RLC ←		+	200 OK BYE(RLC)					

TP301027	SIP reference: RFC 32	61	Q.19	ISUP reference: 112.5 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 PI	CS 4/15							
criteria									
ISUP selection	PICS 1/4 AND PICS 4/2								
criteria									
Test purpose  SIP parameter	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								
values									
ISUP parameter values									
Comments	ISUP/BICC		SUT	SIP-I					
	IAM	<b>→</b>	→	INVITE(IAM)					
			+	183 Session Progress					
			→						
			+						
	COT(successful)	<b>→</b>	→	UPDATE					
			<b>+</b>	200 OK UPDATE					
			ditions met						
	ACM	<del>-</del>	+	180 Ringing(ACM)					
	ANM	<del>-</del>	+	=======================================					
			→	ACK					
			versation						
	REL	<b>→</b>	<b>→</b>	()					
	RLC	<b>←</b>	+	200 OK BYE(RLC)					

TP301028	SIP reference: RFC 32	:61		= -	SUP reference:				
	Q.1912.5 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 PI	CS 4/15							
criteria									
ISUP selection	PICS 1/4								
criteria									
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  Bearer Set-up Connect indication - for the backward bearer set-up case was received								
ISUP parameter									
values									
Comments	ISUP/BICC		SUT		SIP-I				
	IAM	<b>→</b>		→	INVITE(IAM)				
				+	183 Session Progress				
				<b>→</b>	PRACK				
				←	200 OK PRACK				
	COT(successful)	<b>→</b>		<b>→</b>	UPDATE				
				+	200 OK UPDATE				
			conditions	s met					
	ACM	+		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
		(	Conversati	ion					
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP301029	SIP reference: RFC 326			Į;	SUP reference:			
				2.191	2.5 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	expected" sends an INVITE mess. The precondition signalling is concexchange) confirmation of a preconconfirmation of a preconconfirmation of a precondition bein satisfied when  Continuity message, with the or received	Continuity message, with the Continuity Indicators parameter set to "continuity" shall be						
SIP parameter			<u>g</u>		<b>9</b>			
values								
ISUP parameter								
values								
Comments	BICC		SUT		SIP-I			
	IAM =	<b>)</b>		<b>→</b>	INVITE(IAM)			
				+	183 Session Progress			
				<b>→</b>	PRACK			
				+	200 OK PRACK			
	COT(successful)	<b>)</b>		<b>→</b>	UPDATE			
				+	200 OK UPDATE			
		Precon	ditions met					
	ACM +	-		+	180 Ringing(ACM)			
	ANM	•		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
			ersation					
	REL -			<b>→</b>	BYE(REL)			
	RLC •	•		+	200 OK BYE(RLC)			

TP301030	SIP reference: RFC 326	61			SUP reference: 12.5 clause 7.1 D)				
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message								
SIP selection	PICS 4/4 AND PICS 4/5 AND PIC	S 4/15							
criteria									
ISUP selection	PICS 1/4 AND PICS 4/2								
criteria									
Test purpose	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 "COT to be expected", sends an INVITE message with precondition using the SDP offer in the INVITE  • Ensure that the SUT sends CANCEL if the ISUP timer T8 expires if the call has been								
	cleared <b>before</b> an early dialo	gue has bee	en established	d.					
SIP parameter values									
ISUP parameter values									
Comments	BICC	Ç	SUT		SIP-I				
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
				<b>←</b>	100 Trying				
		T8 (	expires						
	=(,, ,,	<b>←</b>		<b>→</b>	CANCEL				
	RLC	<b>→</b>		<b>←</b>	200 OK CANCEL				
				<b>←</b>	487 Request Terminated				
				<b>→</b>	ACK				

TP301031	SIP reference: RFC 3	261			SUP reference: 12.5 clause 7.1 D)					
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message									
SIP selection	PICS 4/4 AND PICS 4/5 AND P	ICS 4/	15							
criteria										
ISUP selection criteria	PICS 1/5 AND PICS 4/2									
Test purpose	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 "COT to be expected", sends an INVITE message with precondition using the SDP offer in the INVITE  • Ensure that the SUT sends CANCEL if the ISUP timer T8 expires if the call has been cleared after an early dialogue with the message defined as SIP_MESSAGE_VA has been established									
SIP parameter values										
ISUP parameter values										
Comments	BICC		SUT		SIP-I					
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)					
				+	SIP_MESSAGE_VA					
			T8 expires	•						
	REL(#47) ← CANCEL/BYE									
	RLC	1		+	200 OK CANCEL/BYE					
				+	487 Request Terminated					
				<b>→</b>	ACK					

TP301032	SIP reference: RFC 3	261				SUP reference: 912.5 clause 7.1					
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 P	ICS 4/	′15								
criteria											
ISUP selection criteria	PICS 1/5 AND PICS 4/2										
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing 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		SUT			SIP-I					
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)					
					+	100 Trying					
	internal	resou	rce reservati	on was i	unsuc	cessful					
	REL(#47)										
	RLC	<b>→</b>			<b>←</b>	200 OK CANCEL					
					+	487 Request Terminated					
					<b>→</b>	ACK					

TP301033	SIP reference: RFC 32	61			SUP reference: 912.5 clause 7.1	
TSS reference	ISUP-SIP/Basic call/Sending of the	ne INVITE	message			
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PIC	CS 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 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 after an early dialogue with the message defined as SIP_MESSAGE_VA 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		SUT		SIP-I	
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)	
				<b>←</b>	SIP_MESSAGE_VA	
	internal re	esource res	ervation was	unsuc	ccessful	
	REL(#47)	+		<b>→</b>	CANCEL/BYE	
	RLC	<b>→</b>		+	200 OK CANCEL/BYE	
			<u> </u>	+	487 Request Terminated	
				<b>→</b>	ACK	

TP301034	SIP reference: RFC	3261			_	SUP reference: 912.5 clause 7.1.1	
TSS reference	ISUP-SIP/Basic call/ Sending	of the	INVITE m	essage			
SIP selection criteria	Based on table 6						
ISUP selection criteria							
Test purpose	Ensure that the SUT in the Idle state on receipt of a IAM message, with the <b>Transmission</b> Medium Requirement (TMR) parameter set to TMR_VALUE if no USI parameter is contained in the IAM  sends an INVITE message containing the media description defined with the "a =" "b =" and "m=" lines set to a b m LINE VALUE						
SIP parameter values	INVITE : a_b_m_LINE_VALU	JE					
ISUP parameter values	IAM: TMR : ISUP_TMR						
Comments	ISUP/BICC		SU	Τ		SIP-I	
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)	
	ACM	+			+	180 Ringing(ACM)	
	ANM	+			+	200 OK INVITE(ANM)	
					<b>→</b>	ACK	
			Conver	sation			
	REL	<b>→</b>			<b>→</b>	BYE(REL)	
	RLC	+			+	200 OK BYE(RLC)	

TP301035	SIP reference: RF0			-	SUP reference: 912.5 clause 7.1.1		
TSS reference	ISUP-SIP/Basic call/ Sending	g of the	INVITE me	essage			
SIP selection criteria	Based on table 7						
ISUP selection criteria							
Test purpose	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	E					
ISUP parameter values	IAM: USI : ISUP_USI						
Comments	ISUP/BICC		SU	Ī		SIP-I	
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)	
	ACM	+			+	180 Ringing(ACM)	
	ANM	+			+	200 OK INVITE(ANM)	
					<b>→</b>	ACK	
			Convers	ation			
	REL	<b>→</b>			<b>→</b>	BYE(REL)	
	RLC	+			+	200 OK BYE(RLC)	

Table 6

	Values for test purposes TP301053									
	ISUP	SDP - a_b_m_LINE_VALUE								
	TMR parameter	m= line			b= line	a= line				
	TMR codes	<media></media>	<transport></transport>	<fmt-list></fmt-list>	<modifier>:<bandw idth-value=""></bandw></modifier>	rtpmap: <dynamic-pt> <encoding name="">/<clock rate="">[/encoding parameters&gt;</clock></encoding></dynamic-pt>				
VA_01	"speech"	Audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)				
	"speech"	Audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000				
VA_02	"3.1 KHz audio"	Audio	RTP/AVP	0 and/or 8	AS:64	rtpmap:0 PCMU/8000 and/or rtpmap:8 PCMA/8000				
VA_03	"64 kbit/s unrestricted"	Audio	RTP/AVP	9	AS:64	rtpmap:9 G722/8000				
	"64 kbit/s unrestricted"	Audio	RTP/AVP	Dynamic PT	AS:64	rtpmap: <dynamic-pt> CLEARMODE/8000</dynamic-pt>				

Table 7

			Va	alues for test pu	rposes T	P301053, TF	P301054					
VA		ISUP				SDP - a_b_m_LINE_VALUE						
		USI parameter		HLC IE in ATP	m= line			b= line	a= line			
	TMR	Information Transport Capability	User Information Layer 1 Protocol Indicator	High Layer Characteristics Identification	<media></media>	<transport></transport>	<fmt-list></fmt-list>	<modifier>:<bandwidt h-value=""></bandwidt></modifier>	rtpmap: <dynamic-pt> <encoding name="">/<clock rate="">[/encoding parameters&gt;</clock></encoding></dynamic-pt>			
VA_01	"speech"	"Speech"	"G.711 μ-law"	Ignore	audio	RTP/AVP	0 (and possibly 8) Note 1	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000) Note 1			
	"speech"	"Speech"	"G.711 A-law"	Ignore	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000			
VA_02	"3.1 KHz audio"	USI Absent		Ignore	audio	RTP/AVP	0 and/or 8 Note 1	AS:64	rtpmap:0 PCMU/8000 and/or rtpmap:8 PCMA/8000 Note 1			
VA_03	"3.1 KHz audio"	"3.1 KHz audio"	"G.711 μ-law"		audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)			
	"3.1 KHz audio"	"3.1 KHz audio"	"G.711 A-law"		audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000			
VA_04	"3.1 KHz audio"	"3.1 KHz audio"	"G.711 A-law"	"Facsimile Group 2/3"	image	tcptl	t38	AS:64	Based on T.38.			
	"3.1 KHz audio"	"3.1 KHz audio"	"G.711 A-law"	"Facsimile Group 2/3"	image	udptl	t38	AS:64	Based on T.38.			
VA_05	"3.1 KHz audio"	"3.1 KHz audio"	"G.711 μ-law"	"Facsimile Group 2/3"	image	udptl	t38	AS:64	Based on T.38.			
	"3.1 KHz audio"	"3.1 KHz audio"	"G.711 μ-law"	"Facsimile Group 2/3"	image	tcptl	t38	AS:64	Based on T.38.			
VA_06	"64 kbit/s unrestricted"	"Unrestricted digital inf. W/tone/ann."	N/A	Ignore	audio	RTP/AVP	9	AS:64	Rtpmap:9 G722/8000			
VA_07	"64 kbit/s unrestricted"	"Unrestricted digital information"	N/A	Ignore	Audio	RTP/AVP	Dynamic PT	AS:64	rtpmap: <dynamic-pt> CLEARMODE/8000 Note 2</dynamic-pt>			

TP301036	SIP reference: RFC 32	261	Q.	ISUP reference: 1912.5 clause 7.1.2					
TSS reference	ISUP-SIP/Basic call/ Sending of	the INVITE n	nessage						
SIP selection	NOT PICS 1/9								
criteria									
ISUP selection criteria									
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter of the IAM  to the addr-spec component of the <b>To header field</b> in the INVITE message								
SIP parameter values	INVITE: To:								
ISUP parameter values									
Comments	ISUP/BICC	SI	JT	SIP-I					
	IAM -	<b>&gt;</b>	<b>→</b>	INVITE(IAM)					
	ACM	F	+	180 Ringing(ACM)					
	ANM	F	+	200 OK INVITE(ANM)					
	→ ACK								
		Conve	rsation						
	REL -	<b>→</b>	→	BYE(REL)					
	RLC	F	+	200 OK BYE(RLC)					

TP301037	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause 7.1.2					
TSS reference	ISUP-SIP/Basic call/ Sending	of the	INVITE m	essage					
SIP selection criteria	NOT PICS 1/9								
ISUP selection criteria									
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter of the IAM  to the addr-spec component of the <b>To header field</b> which shall include the "user=phone" URI parameter if the To header field contains a sip: URI								
SIP parameter values	INVITE: To: sip:; user=phone								
ISUP parameter values									
Comments	ISUP/BICC		SU	T		SIP-I			
	IAM	<b>→</b>		- 1	<b>→</b>	INVITE(IAM)			
	ACM	+			<del>(</del>	180 Ringing(ACM)			
	ANM	+			<del>(</del>	200 OK INVITE(ANM)			
	→ ACK								
			Convers	ation					
	REL	<b>→</b>		1	<b>→</b>	BYE(REL)			
	RLC	+			←	200 OK BYE(RLC)			

TP301038	SIP reference: F	RFC 3261			SUP reference: 912.5 clause 7.1.2				
TSS reference	ISUP-SIP/Basic call/ Send	ding of the	INVITE message	)					
SIP selection criteria	NOT PICS 1/9								
ISUP selection criteria									
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter of the IAM and the and the followed SAM  to the addr-spec component of the <b>To header field</b>								
SIP parameter values	INVITE: To:								
ISUP parameter values									
Comments	ISUP/BICC		SUT		SIP-I				
	IAM	<b>→</b>							
	SAM	<b>→</b>							
	SAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	200 OK INVITE(ANM)								
				<b>→</b>	ACK				
			Conversation						
	REL	<b>→</b>	_	<b>→</b>	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP301039	SIP reference: RF	ISUP reference:  912.5 clause 7.1.2							
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message								
SIP selection	NOT PICS 1/9								
criteria									
ISUP selection criteria									
Test purpose	Ensure that the SUT is mapping in the Called Party Number parameter contained in the Called Party address information of the IAM and followed SAM  to the addr-spec component of the <b>To header field</b> which shall include the "user=phone" URI parameter if the To header field contains a sip: URI								
SIP parameter values	INVITE: To: sip:; user=p	hone							
ISUP parameter values									
Comments	ISUP/BICC		SUT		SIP-I				
	IAM	<b>→</b>							
	SAM	<b>→</b>							
	SAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
			Conversation						
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP301040	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause 7.1.4				
TSS reference	ISUP-SIP/Basic call/ Sen	ding of the	Initial Address	Message	(IAM)/			
SIP selection criteria								
ISUP selection criteria	PICS 4/3							
Test purpose	The O-IWU acting as an independent exchange shall perform the normal BICC/ISUP Hop Counter procedure as it constructs the outgoing encapsulated IAM.							
SIP parameter								
values								
ISUP parameter values								
Comments	ISUP/BICC		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
	→ ACK							
	Conversation							
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP301041	SIP reference: RFC 326	1		ISUP reference: Q.1912.5 clause 7.1.2					
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message								
SIP selection criteria	PICS 1/9								
ISUP selection criteria	PICS 1/8	PICS 1/8							
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, <b>Nature of address = "International number"</b> of the IAM to the addr-spec component of the <b>To header field</b> in the INVITE message.  The format of the To header field is "+CC+NDC+SN" the forward address information is derived from the user info component of the INVITE Request-URI								
SIP parameter values	INVITE: To:								
ISUP parameter values									
Comments	ISUP/BICC	SL	JT	SIP-I					
	IAM →		-	➤ INVITE(IAM)					
	ACM ←		•	180 Ringing(ACM)					
	ANM ← 200 OK INVITE(ANM)  → ACK								
	Conversation								
	REL →		-	► BYE(REL)					
	RLC ←		•	200 OK BYE(RLC)					

TP301042	SIP reference: RFC 32		ISUP reference: 1912.5 clause 7.1.2						
TSS reference	ISUP-SIP/Basic call/ Sending of	the INVITE m	essage						
SIP selection criteria	PICS 1/9								
ISUP selection criteria	NOT PICS 1/8								
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, Nature of address = "National (significant) number" of the IAM  to the addr-spec component of the To header field in the INVITE message.  The format of the To header field is "+CC+NDC+SN"  the forward address information is derived from the user info component of the INVITE Request-URI								
SIP parameter values	INVITE: To:								
ISUP parameter values									
Comments	ISUP/BICC	SU	JT	SIP-I					
	IAM -	<b>→</b>	<b>→</b>	INVITE(IAM)					
	ACM	<del>-</del>	+	180 Ringing(ACM)					
	ANM	<del>-</del>	+	200 OK INVITE(ANM)					
	→ ACK								
		Conversation							
	REL -	<b>→</b>	<b>→</b>	BYE(REL)					
	RLC	<del>(</del>	+	200 OK BYE(RLC)					

TP301043	SIP reference: RFC 3261	•	SUP reference: 912.5 clause 7.1.2				
TSS reference	ISUP-SIP/Basic call/ Sending of the	INVITE m	essage				
SIP selection	PICS 1/9						
criteria							
ISUP selection criteria	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 = "International number" of the IAM and the and the followed SAM  to the addr-spec component of the To header field.  The format of the To header field is "+CC+NDC+SN"  the forward address information is derived from the user info component of the INVITE Request-URI						
SIP parameter values	INVITE: To:						
ISUP parameter values							
Comments	ISUP/BICC	SU	T T	SIP-I			
	IAM →						
	SAM →						
	SAM →		→	INVITE(IAM)			
	ACM ←		+	180 Ringing(ACM)			
	ANM ←		+	200 OK INVITE(ANM)			
			<b>→</b>	ACK			
		Convers	sation				
	REL →		<b>→</b>	BYE(REL)			
	RLC		+	200 OK BYE(RLC)			

TP301044	SIP reference: RFC 3261		ISUP reference: Q.1912.5 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	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, Nature of address = "National (significant) number" of the IAM and the followed SAM  to the addr-spec component of the To header field.  The format of the To header field is "+CC+NDC+SN"  the forward address information is derived from the user info component of the INVITE Request-URI							
SIP parameter values	INVITE: To:							
ISUP parameter values								
Comments	ISUP/BICC	SUT		SIP-I				
	IAM →							
	SAM →							
	SAM →		→	INVITE(IAM)				
	ACM ←		+	180 Ringing(ACM)				
	ANM ←		<b>+</b>	200 OK INVITE(ANM)				
			→	ACK				
	C	onversa	tion					
	REL →		<b>→</b>	BYE(REL)				
	RLC ←		+	200 OK BYE(RLC)				

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

TP302001	SIP reference: R	RFC 3261		_	SUP reference: 1912.5 clause 7.2			
TSS reference	ISUP-SIP/Basic call/Recei	pt of SAM af	ter INVITE has	been s	ent			
SIP selection criteria	PICS 3/1							
ISUP selection criteria								
Test purpose	Ensure if the SUT is supporting en bloc addressing towards the SIP network, subsequent SAMs received after the SUT has sent the INVITE are ignored.							
SIP parameter								
values								
ISUP parameter	SAM; subsequent number	er (PIXIT)						
values								
Comments	ISUP/BICC		SUT		SIP-I			
	IAM	→		<b>→</b>	INVITE(IAM)			
	SAM	→						
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
			Conversation	-				
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP302002	SIP reference: RFC 32	:61		-	SUP reference: 912.5 clause 7.2.1				
TCC reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent								
TSS reference SIP selection	PICS 3/2	Aivi aitei	invite has been	sent					
criteria	PICS 3/2								
ISUP selection	PICS 1/5								
criteria	FICS 1/3								
Test purpose	nsure that the SUT in Idle state, on receipt of an IAM message containing the Continuity heck indicator in the Nature of Connection Indicators parameter which is set to indicate continuity check not required"  ends a INVITE message n receipt of a SAM from the ISUP the SUT shall: Stop timer TOIW3 (if it is running) TOIW2 shall be restarted and the SUT shall invoke the following procedures: a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent c) The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question d) All other contents of the new INVITE are interworked from the parameters of the original IAM								
SIP parameter values									
ISUP parameter values									
Comments	ISUP/BICC		SUT		SIP-I				
	IAM	<b>→</b>	- 55.	<b>→</b>	INVITE(IAM)				
	SAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	SAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	1.10.11.19.19(1.10.11)							
	→ ACK								
		, C	Conversation	1	-				
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP302003	SIP reference: RFC 32	261		-	SUP reference: 912.5 clause 7.2.1	
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent					
SIP selection criteria	PICS 3/2 AND NOT PICS 4/15					
ISUP selection criteria	PICS 1/5 AND PICS 4/2					
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit"  Sends the INVITE after the receipt of the Continuity message with the Continuity Indicators parameter "continuity check successful"  On receipt of a SAM from the ISUP the SUT shall:  1) Stop timer TOIW3 (if it is running)  2) TOIW2 shall be restarted and the SUT shall invoke the following procedures:  a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call  b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent  c) The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question  d) All other contents of the new INVITE are interworked from the parameters of the original IAM					
SIP parameter values						
ISUP parameter values						
Comments	ISUP/BICC		SUT		SIP-I	
	IAM	<b>→</b>				
	SAM	<b>→</b>				
	COT	<b>→</b>		<b>→</b>	INVITE(IAM)	
	SAM	<b>→</b>		<b>→</b>	INVITE(IAM)	
	ACM	+		+	180 Ringing(ACM)	
	ANM	+		+	200 OK INVITE(ANM)	
				<b>→</b>	ACK	
	Conversation					
	REL	<b>→</b>		<b>→</b>	BYE(REL)	
	RLC ← 200 OK BYE(R					

TP302004	SIP reference: RFC 32	261			SUP reference: 912.5 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	LOUID/DIGG		OUT	-	Tota i	
Comments	ISUP/BICC	<b>→</b>	SUT		SIP-I	
	IAM SAM	<u> </u>				
	COT	<u>→</u>		<b>→</b>	INVITE(IAM)	
	SAM	<u>→</u>		→ →	INVITE(IAM)	
	ACM	<del></del>		<del>-</del>	180 Ringing(ACM)	
	ANM	<del>-</del>		+	200 OK INVITE(ANM)	
	, a sivi	_		<b>→</b>	ACK	
	Conversation					
	REL	<b>→</b>	30111013011011	→	BYE(REL)	
	RLC	<del></del>		<del>,</del>	200 OK BYE(RLC)	
	INLO				200 ON DIL(NLO)	

TP302005	SIP reference: RFC 32	261		ISUP reference: Q.1912.5 clause 7.2.1			
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent						
SIP selection criteria	PICS 3/2 AND NOT PICS 4/15						
ISUP selection criteria	PICS 1/5 AND PICS 4/2						
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		SUT	SIP-I			
	IAM	<b>→</b>					
	SAM	<b>→</b>					
	COT →						

TP302006	SIP reference: RFC 32	261		-	SUP reference: 912.5 clause 7.2.1	
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent					
SIP selection criteria	PICS 3/2 AND NOT PICS 4/15					
ISUP selection criteria	PICS 1/5 AND PICS 4/2					
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 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					
SIP parameter values						
ISUP parameter values						
Comments	ISUP/BICC		SUT		SIP-I	
	IAM	<b>→</b>				
	SAM	<b>→</b>				
	T8 expires					
	REL	+				
	RLC	<b>→</b>				

TP302007	SIP ref	erenc	e: RFC 320	61	ISUP reference:			
					Q.1912.5 clause 7.2.1			
TSS reference	ISUP-SIP/Basic	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 1/5 AND P	ICS 4/	/2					
criteria								
Test purpose	Check indicator in check required Sends an INVITE Indicators param preconditions are On receipt of a Sends and Invite Indicators param preconditions are On receipt of a Sends Invite Indicators parameter sends and Invite	sure that the SUT in Idle state, on receipt of an IAM message containing the Continuity eck indicator in the Nature of Connection Indicators parameter which is set "continuity eck required on this circuit"  Inds an INVITE message after the reception of the Continuity message with the Continuity icators parameter set to "continuity check successful" and after the requested conditions are met in the SIP network  receipt of a SAM from the ISUP the SUT shall:  Stop timer TOIW3 (if it is running)  TOIW2 shall be restarted and the SUT shall invoke the following procedures:  a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call  b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent  c) The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question  d) All other contents of the new INVITE are interworked from the parameters of the						
SIP parameter	INVITE2: Reque	st URI	contains d	ligits	ts from the IAM and digits from SAM x and SAM y. The			
values	IAM is also conta			Ū	,			
ISUP parameter								
values								
Comments								
	ISUP/BICC		SUT		SIP-I			
	IAM	→		<b>→</b>	INVITE1(IAM)			
	SAM x	→						
				+	The second secon			
	COT	→		<b>→</b>				
				+				
	SAM y	<b>→</b>		<b>→</b>				
	ACM	+		+	100 1 111 1911 19 (7 10 111 )			
	ANM	+		+				
				<b>→</b>	ACK			
		С	onversatio	n				
	REL	<b>→</b>		<b>→</b>	(			
	RLC	<b>←</b>		<b>←</b>	- 200 OK BYE(RLC)			

TP302008			e: RFC 3261		ISUP reference: Q.1912.5 clause 7.2.1			
TSS reference	ISUP-SIP/Basic	call/R	eceipt of SAM	after	r invite has been sent			
SIP selection criteria	PICS 3/2 AND F	PICS 3/2 AND PICS 4/5 AND PICS 4/15						
ISUP selection	PICS 1/5 AND F	ICS 1	/2					
criteria	I IOS 1/3 AND I	100 4	12					
Test purpose	Check indicator check performe Sends an INVIT Indicators param preconditions ar On receipt of a S 1) Stop timer T 2) TOIW2 shall a) The Requ received b) A new IN INVITE is c) The new resources reserved paramete d) All other	the SUT in Idle state, on receipt of an IAM message containing the Continuity cator in the Nature of Connection Indicators parameter which is set or "continuity formed on previous circuit"  NVITE message after the reception of the Continuity message with the Continuity charameter set to "continuity check successful" and after the requested ans are met in the SIP network of a SAM from the ISUP the SUT shall:  ner TOIW3 (if it is running)  shall be restarted and the SUT shall invoke the following procedures:  Request-URI and the To header field of the new INVITE shall contain all digits eived so far for this call  ew INVITE with the same Call-ID and From header (including tag) as the previous ITE is sent  new INVITE shall contain a new SDP offer. The O-IWU may re-use any cources that have already been reserved for this call. This re-use of existing erved resources shall be reflected within the precondition attributes for the SDP ameters in question 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 cont		r cornains digita	3 11 0	in the IAM and digits from SAM X and SAM y. The			
ISUP parameter	17 (10) 13 (130 (0))	anica						
values								
Comments:	INVITE. The pre exchange) the c							
		<b>→</b>	SUT	<b>→</b>	SIP-I INVITE1(IAM)			
	IAM SAM x	<del>7</del>  →		7	INVII⊏I(IAIVI)			
	SAIVI X	7		+	183 Session Progress without encapsulated ACM			
	COT	<b>→</b>		<u>~</u>	UPDATE UPDATE			
	СОТ	7		<del>7</del>	200 OK UPDATE			
	SAM	_		<u>~</u>				
	ACM	<b>→</b>		<del>7</del>	INVITE2 (IAM and digits from SAM X + SAM Y)			
	ANM	+		<del>-</del>	180 Ringing(ACM) 200 OK INVITE(ANM)			
	AINIVI	_	<del> </del>	<u>←</u>	ACK			
			Conversedis	7	AUN			
	DEL	-	Conversation		DVE(DEL)			
	REL	<b>→</b>		<del>}</del>	BYE(REL)			
	RLC	<b>←</b>		+	200 OK BYE(RLC)			

TP302009	SIP reference: RFC 3261			ISUP reference:				
				).1912.5 clause 7.2.1				
TSS reference	ISUP-SIP/Basic call/Receipt of SAM at	fter invite	e has been ser	nt				
SIP selection criteria	PICS 3/2 AND NOT PICS 4/15							
ISUP selection criteria	PICS 1/4 AND NOT PICS 4/2							
Test purpose	Ensure that the SUT in Idle state, on reexpected"	eceipt of	an IAM messa	age indicating "COT to be				
	<ul> <li>The sending of the INVITE is delayed until all the following conditions are satisfied:</li> <li>Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received</li> <li>Bearer Set-up indication - for the forward bearer set-up case where the incoming Connect Type is "notification not required" was received</li> <li>On receipt of a SAM from the BICC the SUT shall:</li> <li>Stop timer TOIW3 (if it is running).</li> <li>TOIW2 shall be restarted and the SUT shall invoke the following procedures: <ul> <li>a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call.</li> <li>b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent</li> <li>c) The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question.</li> <li>d) All other contents of the new INVITE are interworked from the parameters of the</li> </ul> </li> </ul>							
SIP parameter values								
ISUP parameter								
values								
Comments	ISUP/BICC	S	UT	SIP-I				
	IAM →							
	SAM x →							
	COT →		<b>-</b>	NVITE(IAM)				
	SAM y →		-					
	ACM ←		+	180 Ringing(ACM)				
	ANM ←		+					
			<b>-</b>	ACK .				
		Conve	rsation					
	REL →		-	BYE(REL)				
	RLC ←		+	<del>- i - · · · · · · · · · · · · · · · · · </del>				

TP302010	SIP reference: RFC 3261		ISUP reference: 1912.5 clause 7.2.1					
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent							
SIP selection criteria	PICS 3/2 AND NOT PICS 4/15							
ISUP selection criteria	PICS 1/4 AND PICS 4/2							
Test purpose	Ensure that the SUT in Idle state, on recexpected"  The sending of the INVITE is delayed un	•						
	<ul> <li>Continuity message, with the Continuence received</li> </ul>	uity Indicators parame	eter set to " <b>continuity</b> " shall be					
	or without bearer control tunnelling) required", and for the fast set-up (ba	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  On receipt of a SAM from the BICC the SUT shall:						
	<ul> <li>2) TOIW2 shall be restarted and the SUT shall invoke the following procedures:</li> <li>a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call</li> <li>b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent</li> <li>c) The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP</li> </ul>							
	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					
Commonto	IAM →	001	011 1					
	SAM x							
	COT →	<b>→</b>	INVITE(IAM)					
	SAM y	<b>→</b>	INVITE(IAM)					
	ACM ←	+	180 Ringing(ACM)					
	ANM ←	+	200 OK INVITE(ANM)					
		<b>→</b>	ACK					
		Conversation						
	REL →	<b>→</b>	BYE(REL)					
	RLC +	+	200 OK BYE(RLC)					

TP302011	SIP reference: RFC 326	1		ISUP reference:						
		<del></del>	·	1912.5 clause 7.2.1						
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent									
SIP selection criteria	PICS 3/2 AND NOT PICS 4/15									
ISUP selection	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 Continuity Indicators parameter set to "continuity" shall be received  Bearer Set-up Connect indication - for the backward bearer set-up case was received.  On receipt of a SAM from the BICC the SUT shall:  Stop timer TOIW3 (if it is running)  TOIW2 shall be restarted and the SUT shall invoke the following procedures:  a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call  b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent  c) The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question  d) All other contents of the new INVITE are interworked from the parameters of the									
SIP parameter	original IAM									
values										
ISUP parameter										
values		1								
Comments	ISUP/BICC		SUT	SIP-I						
	1 111	<del>}</del>								
	G	<b>&gt;</b>		INDUSTRIAL O						
	38.	<del>}</del>	<b>→</b>	INVITE(IAM)						
		<del>}</del>	<b>→</b>	INVITE(IAM)						
		<b>←</b>	+	180 Ringing(ACM)						
	ANM	<b>←</b>	+	200 OK INVITE(ANM)						
			→	ACK						
		Conv	ersation							
	REL -	<b>→</b>	<b>→</b>	BYE(REL)						
	RLC •	<del>(</del>	+	200 OK BYE(RLC)						

TP302012	SIP reference: RFC 3261	SUP reference: 912.5 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	anei invite	tias been	Seni					
criteria	FIGS 3/2 AND NOT FIGS 4/13								
ISUP selection criteria	PICS 1/4 AND PICS 4/2								
Test purpose	Ensure that the SUT in Idle state, on	receipt of	on IAM mo	0000	o indicating "COT to be				
rest purpose	expected"	receipt of	an iAw me	ssayı	e malcating COT to be				
	The sending of the INVITE delays ur	itil all the f	ollowing co	nditio	ns are satisfied:				
	Continuity message, with the Correceived	ntinuity In	dicators pa	rame	ter set to "continuity" shall be				
	BNC set-up success indication for receipt of a SAM from the BICC to the set of the			r cont	trol tunnelling was received				
	1) Stop timer TOIW3 (if it is running)								
	2) TOIW2 shall be restarted and the								
	a) The Request-URI and the To received so far for this call	neader fie	la of the ne	W IIN	VITE shall contain all digits				
	b) A new INVITE with the same	Call-ID a	nd From he	ader	(including tag) as the previous				
	INVITE is sent	· · · · · · · · · · · · · · · · · · ·			(mendaming tag) as and provides				
	c) The new INVITE shall contain								
	resources that have already b								
	reserved resources shall be re	eflected w	ithin the pre	econd	lition attributes for the SDP				
	parameters in question	INIV/ITE	- !	1 6	the a mean and a mean of the a				
	d) All other contents of the new original IAM	NVIIE ar	e interworke	ea tro	m the parameters of the				
SIP parameter	Ĭ								
values									
ISUP parameter values									
Comments	ISUP/BICC	s	UT		SIP-I				
	IAM →								
	SAM x →								
	COT →			<b>→</b>	INVITE(IAM)				
	SAM y →			<b>→</b>	INVITE(IAM)				
	ACM ←			+	180 Ringing(ACM)				
	ANM ←			+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
		Conve	ersation						
	REL →			<u>→</u>	BYE(REL)				
	RLC +			+	200 OK BYE(RLC)				

TP302013	SIP re	ferer	nce: RFC 3261		ISUP reference:		
					Q.1912.5 clause 7.2.1		
TSS reference	ISUP-SIP/Basic	call/	Receipt of SAM	after	r invite has been sent		
SIP selection			4/5 AND PICS 4				
criteria							
ISUP selection	PICS 1/4 AND I	PICS	4/2				
criteria							
Test purpose	expected" Sends the INVI Continuity	TE m	essage . The ev	ents	eipt of an IAM message indicating "COT to be uity Indicators parameter set to "continuity" was		
	Type is "no are indicating the	rer Set-up indication - for the forward bearer set-up case where the incoming Connect is "notification not required" was received ating the successful completion of bearer set-up of a SAM from the BICC the SUT shall:					
	2) TOIW2 shal a) The Red received b) A new I INVITE c) The new	Response to the state of the Sut shall invoke the following procedures:  Request-URI and the To header field of the new INVITE shall contain all digits eived so far for this call new INVITE with the same Call-ID and From header (including tag) as the previous VITE is sent enew INVITE shall contain a new SDP offer. The O-IWU may re-use any ources 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	INVITE2: Requi			s froi	m the IAM and digits from SAM x and SAM y. The		
ISUP parameter values			-				
Comments	INVITE. The pre exchange) the o	econd confir n of a	dition signalling is mation of a preco precondition bei	s cor ondit	on signalling procedure using the SDP Offer in the included upon sending (within an SDP offer-answer tion being met. The SDP Offer or Answer carrying met is sent when the conditions to send a INVITE		
	ISUP/BICC		SUT		SIP-I		
	IAM	<b>→</b>		<b>^</b>	INVITE1(IAM)		
	SAM x	<b>→</b>					
				+	183 Session Progress without encapsulated ACM		
	COT	<b>→</b>		<b>→</b>	UPDATE		
		<u> </u>		+	200 OK UPDATE		
	SAM y	<b>→</b>		<b>→</b>	INVITE2 (IAM and digits from SAM X + SAM Y)		
	ACM	<del>(</del>		+	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
		ļ	<u> </u>	<b>→</b>	ACK		
	551		Conversation	_	D)(E(DEL)		
	REL	<b>→</b>		<b>→</b>	BYE(REL)		
	RLC	<b>←</b>		+	200 OK BYE(RLC)		

TP302014	SIP refe	renc	e: RFC 3261		ISUP reference:		
11 302014	On Toro		.c. 1(1 O 0201		Q.1912.5 clause 7.2.1		
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent						
SIP selection	PICS 3/2 AND PI				mivite has been sent		
criteria	1 100 3/2 AND 1 1	-	75 AND 1 100 4	713			
ISUP selection	PICS 1/4 AND PI	CS 4	/2				
criteria	1 100 1/4 / (100 1 1	00 7	/ <b>L</b>				
Test purpose	Ensure that the S	UT ir	n Idle state, on	rece	eipt of an IAM message indicating "COT to be		
	expected"		, ,		, , , , , , , , , , , , , , , , , , ,		
	Sends the INVITE	me	ssage. The eve	nts			
					uity Indicators parameter set to " <b>continuity</b> " was		
		tion i	ndicator set to	"Cor	nnected" - for the forward bearer set-up cases (with,		
	or without be	arer		ng) v	where the incoming Connect Type is "notification		
	are indicating the						
	On receipt of a S				SUT shall:		
	Stop timer TO						
	2) TOIW2 shall b	e res	started and the	SUT	T shall invoke the following procedures:		
				neac	der field of the new INVITE shall contain all digits		
			for this call	0-11	ID and From booder (including to a) so the provious		
				Call	-ID and From header (including tag) as the previous		
	INVITE is			2 nc	ew SDP offer. The O-IWU may re-use any		
					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 IA						
SIP parameter			I contains digits	s fror	m the IAM and digits from SAM x and SAM y. The		
values	IAM is also conta				·		
ISUP parameter							
values							
Comments					on signalling procedure using the SDP Offer in the		
					ncluded upon sending (within an SDP offer-answer		
					tion being met. The SDP Offer or Answer carrying		
				ng n	net is sent when the conditions to send a INVITE		
	message are sati	stiea			loip i		
	ISUP/BICC	_	SUT		SIP-I		
	IAM	<b>→</b>	<del> </del>	<b>→</b>	INVITE1(IAM)		
	SAM	7		+	192 Coppies Progress without anappoulated ACM		
	СОТ	<b>→</b>		<u>~</u>	183 Session Progress without encapsulated ACM		
	001	7		<del>7</del>	UPDATE 200 OK UPDATE		
	SAM	<b>→</b>		<u>▼</u>	INVITE2 (IAM with digits from SAM X + SAM Y)		
	ACM	+		<del>7</del>	180 Ringing(ACM)		
	ANM	+		<del>-</del>	200 OK INVITE(ANM)		
	CINIVI			<u>∑</u>	ACK		
			Conversation		non en		
	REL	<b>→</b>	CONVENSATION	<b>→</b>	BYE(REL)		
	RLC	7		<del>7</del>	200 OK BYE(RLC)		
	INLU	~		_	ZUU UN DIE(NLU)		

TP302015	SIP refe	erence: RFC 3261		ISUP reference: Q.1912.5 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	I ICO 3/2 AND I I	00 4/3 AND 1 100 .	+/13					
ISUP selection	PICS 1/4 AND PI	CS 4/2						
criteria	1 100 1/4 / ((10) 1 1	OO 4/2						
Test purpose	Ensure that the S	UT in Idle state, on	rece	ipt of an IAM message indicating "COT to be				
	expected"	- · · · · · · · · · · · · · · · · · · ·		,				
		message. The eve	ents					
	<ul> <li>Continuity m received</li> </ul>	essage, with the Co	ntinu	ity Indicators parameter set to "continuity" was				
	<ul> <li>Bearer Set-u</li> </ul>	p Connect indicatio	n - fo	r the backward bearer set-up case was received.				
		successful comple						
	On receipt of a S	AM from the BICC t	he S	UT shall:				
		IW3 (if it is running						
	2) TOIW2 shall b	e restarted and the	SUT	shall invoke the following procedures:				
			head	ler field of the new INVITE shall contain all digits				
		o far for this call	<b>~</b> "	ID 15 1 1 (' 1 1' ( ) 4 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1				
			Call	ID and From header (including tag) as the previous				
	INVITE is			ew 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			s fror	n the IAM and digits from SAM x and SAM y. The				
values	IAM is also conta			·				
ISUP parameter								
values								
Comments				n signalling procedure using the SDP Offer in the				
				cluded upon sending (within an SDP offer-answer				
				ion being met. The SDP Offer or Answer carrying				
			ing m	net is sent when the conditions to send a INVITE				
	message are sati			Tour I				
	ISUP/BICC	SUT	_	SIP-I				
	IAM	<b>→</b>	<b>→</b>	INVITE1(IAM)				
	SAM	<b>→</b>	1	400 Cassian Drawnas with set as secured to d. A CAA				
	COT	_	<b>←</b>	183 Session Progress without encapsulated ACM				
	СОТ	<b>→</b>	<b>→</b>	UPDATE				
	CAM	_		200 OK UPDATE				
	SAM	<b>→</b>	<b>→</b>	INVITE2 (IAM with digits from SAM X + SAM Y)				
	ACM	<del>-</del>	+	180 Ringing(ACM)				
	ANM	_	<b>←</b>	200 OK INVITE(ANM)				
		Conversetier	7	ACK				
	DEL	Conversation		DVE(DEL)				
	REL	<b>→</b>	<b>→</b>	BYE(REL)				
	RLC	<b>T</b>	_	200 OK BYE(RLC)				

TP302016	SIP refe	erence: RFC 3261		ISUP reference: Q.1912.5 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	FIGO 3/2 AIND FIGO 4/10 AIND FIGO 4/10								
ISUP selection	PICS 1/4 AND PI	ICS 4/2							
criteria	1.100 1, 17 1112 1 1	.00 1/2							
Test purpose	be expected" Sends the INVITI Continuity m received BNC set-up are indicating the On receipt of a S Stop timer TC TOIW2 shall I a) The Requ received s b) A new IN INVITE is c) The new I resources reserved I parametei	Indicating 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 indicating the successful completion of bearer set-up, 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: 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			jits fror	n the IAM and digits from SAM x and SAM y. The					
values	IAM is also conta	ined							
ISUP parameter									
values Comments	The O IV/III show	ld initiate the proc	anditi a	a signalling procedure using the CDD Offer in the					
Comments	INVITE. The pred exchange) the co the confirmation	The O-IWU should initiate the precondition signalling procedure using the SDP Offer in the INVITE. The precondition signalling is concluded upon sending (within an SDP offer-answer exchange) the confirmation of a precondition being met. The SDP Offer or Answer carrying the confirmation of a precondition being met is sent when the conditions to send a INVITE message are satisfied  ISUP/BICC   SUT   SIP-I							
	IAM	<b>→</b>	<b>→</b>	INVITE1(IAM)					
	SAM	<b>→</b>							
			+	183 Session Progress without encapsulated ACM					
	COT	<b>→</b>	<b>→</b>	UPDATE					
			+	200 OK UPDATE					
	SAM	<b>→</b>	<b>→</b>	INVITE2 (IAM with digits from SAM X + SAM Y)					
	ACM	+	+	180 Ringing(ACM)					
	ANM	+	+	200 OK INVITE(ANM)					
			<b>→</b>	ACK					
		Conversatio	n						
	REL	<b>→</b>	<b>→</b>	BYE(REL)					
	RLC	+	+	200 OK BYE(RLC)					

TP302017	SIP reference: RFC 3261		ISUP reference: Q.1912.5 clause 7.2.1							
TSS reference	ISUP-SIP/Basic call/Receipt of SAM	SUP-SIP/Basic call/Receipt of SAM after invite has been sent								
SIP selection criteria	PICS 3/2									
ISUP selection criteria	PICS 1/4									
Test purpose	The SUT in Idle state, on receipt of an IAM message sends a INVITE message On receipt of a SAM from the BICC/ISUP the SUT shall:  1) Stop timer TOIW3 (if it is running) 2) TOIW2 shall be restarted and the SUT shall invoke the following procedures: Ensure that if timer TOIW2 has expired, subsequent SAMs received after the SUT has sent the INVITE are ignored									
SIP parameter values										
ISUP parameter values										
Comments	ISUP/BICC		SUT	SIP-I						
	IAM →		→	INVITE(IAM)						
	SAM -		→	INVITE(IAM)						
		T <sub>oiw2</sub>	expired							
	SAM →									
	ACM ←		<b>+</b>	180 Ringing(ACM)						
	ANM ←		+							
			→	ACK						
			ersation							
	REL →		→	= : = (: :==)						
	RLC +		<b>←</b>	200 OK BYE(RLC)						

TP302018	SIP reference: RFC 3261	=	SUP reference: 912.5 clause 7.2.1						
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent								
SIP selection criteria	PICS 3/1								
ISUP selection criteria	PICS 3/8								
Test purpose	<ul> <li>The SUT in Idle state, on receipt of an IAM message</li> <li>On receipt of a SAM from the BICC/ISUP the SUT shall:</li> <li>sends a INVITE message if the minimum number of digits for routing the call has been received in the IAM and the SAM</li> <li>TOIW1 and TIOW2 shall be started and the SUT shall invoke the following procedures:</li> <li>Ensure that if timer TOIW2 has expired, subsequent SAMs received after the SUT has sent the INVITE are ignored</li> </ul>								
SIP parameter values									
ISUP parameter values									
Comments	ISUP/BICC IAM → SAM →	SUT →	SIP-I INVITE(IAM)						
	SAM →	<sub>2</sub> expired							
	ACM ← ANM ←	<b>+</b>	180 Ringing(ACM) 200 OK INVITE(ANM)						
	Con	→ versation	ACK						
	REL →	<b>→</b>	BYE(REL)						
	RLC ←	+	200 OK BYE(RLC)						

# 5.2.2.3 Sending of the ACM message

TP303001	SIP reference: RFC 3261			ference lause 2.	: Q.1912.5 clause 7.1, 1.4.8				
TSS reference	ISUP-SIP /Basic call/Sending of	ISUP-SIP /Basic call/Sending of the ACM message							
SIP selection	PICS 1/3	PICS 1/3							
criteria									
ISUP selection	PICS 4/9								
criteria									
Test purpose	Ensure that the SUT in Idle state				e containing the complete				
	called party number and the s								
	Sends the INVITE message to d			ACM mes	ssage with				
	the CPS indicator set to " r				(00)				
		y indi	cator set to "no	indicatioi	n(00)" or "ordinary subscriber				
	(01)" or "payphone (10)"		"INT IND VAL	,					
	<ul> <li>the interworking indicator</li> <li>the ISUP indicator set to "I</li> </ul>			-					
	<ul> <li>the ISDN access indicator</li> </ul>			ום אאו					
CID parameter	the ISDN access indicator	sei ic	13DN_ACC_IN	ID_VAL					
SIP parameter values									
ISUP parameter	IAM; Called party number: con	nnlata	number						
values	ACM, CPS indicator no indicat								
Values	Called party's category indica			or ordina	ry subscriber (01) or payphone				
	(10)		· · · · · · · · · · · · · · · · · · ·	, o. aa	., easee. (e., e. pa,pee				
	interworking indicator: INT_IN	ID_VA	L (PIXIT)						
	ISUP indicator: ISUP_IND_ID (								
	ISDN access indicator ISDN_	ACC_	IND_VAL (PIXIT	)					
Comments	ISUP/BICC		SUT		SIP-I				
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM(no indication)	+							
	CPG(Alerting)	+		+	180 Ringing(ACM)				
	ANM	+		<b>←</b>	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
			Conversation						
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	+		<b>←</b>	200 OK BYE(RLC)				

TP303002	SIP reference: RFC 326		ISUP reference: Q.1912.5 clause 7.1, Q.764 clause 2.1.4.8							
TSS reference		ISUP-SIP /Basic call/ Sending of the ACM message								
SIP selection criteria	PICS 1/3	PICS 1/3								
ISUP selection criteria	PICS 4/9									
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan  Sends the INVITE message to called user  Sends the ACM message with  the CPS indicator set to "no indication (00)"  the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)"  the interworking indicator set to "INT_IND_VAL"  the ISUP indicator set to "ISUP_IND_ID"  the ISDN access indicator set to "ISDN_ACC_IND_VAL"									
SIP parameter values	the ISDN access indicator se	tto lobit	_//.00_  \\D_\//	<b>-</b>						
ISUP parameter values	ACM, Backward call indicator is se indication (00)  Called party's category indicator (10) interworking indicator: INT_IND_	Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: INT_IND_VAL (PIXIT) ISUP indicator: ISUP_IND_ID (PIXIT)								
Comments	ISUP/BICC		SUT	SIP-I						
	IAM 🗦		7	_						
	ACM(no indication)									
	CPG(Alerting)	•	+	180 Ringing(ACM)						
	ANM		+							
			7							
		Conv	ersation							
	REL -	•	)	BYE(REL)						
	RLC •		+							

TP303003	SIP reference: RFC 3261			ISUP reference:						
				Q.1912.5 clause 7.1,						
				Q.764 clause 2.1.4.8						
TSS reference	ISUP-SIP /Basic call/Sending of the	ISUP-SIP /Basic call/Sending of the ACM message								
SIP selection	PICS 1/3									
criteria										
ISUP selection	PICS 4/9									
criteria										
Test purpose	Ensure that the SUT in Idle state, or									
	called party number where the end									
	called party number to indicate that	a <b>sufficie</b>	nt number of	digits has been received to						
	route the call to the called party									
	Sends the INVITE message to									
				o " no indication (00)", the <b>Called</b>						
	party's category indicator set									
				NT_IND_VAL", the ISUP indicator						
	set to "ISUP_IND_ID", the ISDI	l access i	indicator set t	to "ISDN_ACC_IND_VAL "						
SIP parameter										
values										
ISUP parameter	IAM; Called party number: comple									
values	ACM, CPS indicator no indication		. (00)							
	Called party's category indicator:	no indicat	ion(00) or ord	linary subscriber (01) or payphone						
	(10)	/AL /DIX/IT	-\							
	interworking indicator: INT_IND_\		)							
	ISUP indicator: ISUP_IND_ID (PIX ISDN access indicator ISDN_ACC	11) 110 \/A	I (DIVIT)							
Comments	ISUP/BICC		SUT	SIP-I						
Comments	IAM -		_	→ INVITE(IAM)						
	ACM(no indication)			INVITE(IAIVI)						
	CPG(Alerting)			€ 180 Ringing(ACM)						
	ANM $\leftarrow$			€ 200 OK INVITE(ANM)						
	VIAINI			→ ACK						
		Conv	ersation	7 NOIL						
	REL →	CON		→ BYE(REL)						
	RLC +			€ 200 OK BYE(RLC)						
	IKLC 7		•	E ZUU UN BIE(KLU)						

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TP303004	SIP reference: RFC 326	1	I	SUP reference:						
				clause 7.1 1) d), 7.3.1, 7.4						
TSS reference	ISUP-SIP /Basic call/Sending of the ACM message									
SIP selection	PICS 1/3									
criteria										
ISUP selection	NOT PICS 4/9									
criteria										
Test purpose	Ensure that the SUT in Idle state, called party number where the entimer T <sub>OIW1</sub> after the receipt of the	nd of addres	ss signalling is de							
	<ul> <li>Sends the INVITE message to</li> <li>Sends the ACM message with party's category indicator so</li> <li>"payphone (10)", the interwork set to "ISUP_IND_ID", the ISI</li> </ul>	the CPS in et to "no ind king indica	ndicator set to "r ication(00)" or "o ator set to " INT_	rdinary subscriber (01)" or IND_VAL", the <b>ISUP indicator</b>						
SIP parameter	,									
values										
ISUP parameter	IAM; Called party number: compl									
values	ACM, CPS indicator no indication									
	Called party's category indicator	<b>r</b> : no indicat	ion(00) or ordina	ry subscriber (01) or payphone						
	(10)									
	interworking indicator: INT_IND_		)							
	ISUP indicator: ISUP_IND_ID (PI		I (DIVIT)							
Comments	ISDN access indicator ISDN_AC		L (PIXIT)	SIP-I						
Comments	ISUP/BICC		501	5IP-I						
	IAIVI	IAM →								
		T <sub>OIW1</sub> expiry								
	ACM(no indication)		<b>→</b>	INVITE(IAM)						
	CPG(Alerting)		+	180 Ringing(ACM)						
	ANM	<u>-</u>	<b>+</b>	200 OK INVITE(ANM)						
			<del>`</del>	ACK						
			ersation							
	REL		<b>→</b>	BYE(REL)						
	RLC	<b>-</b>	+	200 OK BYE(RLC)						

TP303005	SIP reference: RFC 326	1		ISUP reference:					
			Q.19	12.5 clause 7.1, 7.3.1					
TSS reference	ISUP-SIP /Basic call/Sending of th	e ACM mes	ssage						
SIP selection	PICS 1/3								
criteria									
ISUP selection	NOT PICS 4/9								
criteria									
Test purpose	Ensure that the SUT if overlap addressing is to be used toward the SIP network, on receipt of an IAM message containing the <b>minimum number of digits required for routing the call has been received</b> (start timer TOIW2 and invoke the appropriate outgoing SIP signalling procedure)  • Sends an INVITE message to the called user and after the expiration of T <sub>OIW2</sub> • Sends the ACM message with the <b>CPS indicator</b> set to " no indication (00)", the <b>Called party's category indicator</b> set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the <b>interworking indicator</b> set to " INT_IND_VAL", the <b>ISUP indicator</b>								
	set to "ISUP_IND_ID", the ISE	ON access	indicator set to	"ISDN_ACC_IND_VAL "					
SIP parameter									
values									
ISUP parameter values	IAM; Called party number: compl								
values	Called party's category indicator (10) interworking indicator: INT_IND_	interworking indicator: INT_IND_VAL (PIXIT) ISUP indicator: ISUP_IND_ID (PIXIT)							
Comments	ISUP/BICC		SUT	SIP-I					
	IAM =								
	SAM								
		<b>&gt;</b>	<b>→</b>	INVITE(IAM)					
			2 expiry						
	ACM(no indication)	- OIW	2 1 7						
	_ (	_	+	180 Ringing(ACM)					
	(: .: -: .: .:	4	+	200 OK INVITE(ANM)					
	/ U VIVI	`	<u>`</u>	ACK					
	Conversation								
	REL =		<u>→</u>	BYE(REL)					
		-	+	200 OK BYE(RLC)					
	j= <del>-</del>			1==== (=)					

TP303006	SIP reference: RFC 32	61		ISUP reference:					
			Q.1912	.5 clause 7.1 1) a), 7.3.1					
TSS reference	ISUP-SIP /Basic call/Sending of the ACM message								
SIP selection	PICS 1/3								
criteria									
ISUP selection	NOT PICS 4/9								
criteria									
Test purpose	Ensure that the SUT in Idle state,			ge containing the complete					
	called party number, on receipt	of a 180 Ring	ing message						
	<ul> <li>Sends the ACM message with</li> </ul>								
	<ul> <li>the CPS indicator set</li> </ul>								
				alue in the encapsulated ACM					
	<ul> <li>the interworking indicates</li> </ul>								
	<ul> <li>the ISUP indicator set</li> </ul>								
	<ul> <li>the ISDN access indic</li> </ul>	cator set to th	e value in the	encapsulated ACM					
SIP parameter									
values									
ISUP parameter	IAM; Called party number: comp								
values	ACM, Backward call indicator is s								
Comments	ISUP/BICC		UT	SIP-I					
	17 (141	<b>→</b>	→	INVITE(IAM)					
	7 (011)	<del>-</del>	<b>+</b>	180 Ringing(ACM)					
	ANM	<del>-</del>	<b>←</b>	200 OK INVITE(ANM)					
			→	ACK					
		Conve	ersation						
	REL	<b>→</b>	<b>→</b>	BYE(REL)					
	RLC	<b>←</b>	+	200 OK BYE(RLC)					

TP303007	SIP reference: RFC	3261		Q.	-	SUP reference: 5 clause 7.1 1 a), 7.3.2
TSS reference	ISUP-SIP /Basic call/Sending of	of the A	CM mes	sage		
SIP selection	PICS 3/1			<u> </u>		
criteria						
ISUP selection	NOT PICS 4/9					
criteria						
Test purpose	Ensure that the SUT in Idle sta					
	called party number on receip		83 Sess	sion Progre	ss with	n encapsulated ACM
	<ul> <li>Sends the INVITE message</li> </ul>					
	<ul> <li>The encapsulated ACM me</li> </ul>	essage	is sent ι	unchanged	backy	ward
SIP parameter						
values						
ISUP parameter	IAM; Called party number: co	mplete	number			
values						
Comments	ISUP/BICC		S	SUT		SIP-I
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)
	ACM(no indication)	+			+	183 Session Progress(ACM)
	CPG(Alerting)	+			+	180 Ringing(CPG)
	ANM	+			+	200 OK INVITE(ANM)
					<b>→</b>	ACK
			Conve	ersation		
	REL	<b>→</b>			<b>→</b>	BYE(REL)
	RLC	+			+	200 OK BYE(RLC)

TP303011	SIP reference	e: RF	C 3261		ISUP reference:			
					Q.1912.5 clause 7.1, 7.3.1, 7.4			
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message							
SIP selection	PICS 1/3							
criteria								
ISUP selection	PICS 4/2 AND NOT P	ICS 4	/9					
criteria	E 4 44 OUT:							
Test purpose	called party number timer T <sub>OIW1</sub> after the performed (ISUP) or C	where receip OT is	the end on tof the late expected	f ad est (BI				
	received.	hhold	sending A	CM	M until a successful continuity indication has been  CPS indicator set to " no indication (00)", the Called			
	party's category "payphone (10)", t	indic the in	ator set to terworking	"nd <b>g in</b>	no indicator set to "No indication (00)", the Called no indication(00)" or "ordinary subscriber (01)" or indicator set to "INT_IND_VAL", the ISUP indicator sets indicator set to "ISDN_ACC_IND_VAL"			
SIP parameter								
values								
ISUP parameter	IAM; Called party nur	nber:	complete	nun	ımber			
values	ACM,		(0.0)					
	CPS indicator no ind				- didi (00) di (04)			
	(10)interworking indi				ndication(00) or ordinary subscriber (01) or payphone			
	ISUP indicator: ISUP				AL (FIAIT)			
	ISDN access indicate				D VAI (PIXIT)			
Comments	ISUP/BICC		SUT		SIP-I			
	IAM	<b>→</b>		<del>-</del>				
				<del>(</del>	\ /			
	COT	<b>→</b>		<del>&gt;</del>				
				<del>(</del>	200 OK UPDATE			
		T <sub>C</sub>	<sub>DIW1</sub> expiry	/				
	ACM(no indication)							
	CPG(Alerting, BCi)	+		<del>(</del>	180 Ringing(ACM)			
	ANM	+		<del>(</del>				
				<b>→</b>	ACK			
			nversation	1				
	REL	→		<u>→</u>	= : = (: :==)			
	RLC	<b>←</b>		<del>(</del>	200 OK BYE(RLC)			

TP303012	SIP reference	e: RF	C 3261		ISUP reference:				
					Q.1912.5 clause 7.1, 7.3.1, 7.4				
TSS reference	ISUP-SIP /Basic call/Sending of the INVITE message								
SIP selection	PICS 1/3 AND PICS 3/2 AND PICS 4/5 AND PICS 4/4 AND PICS 4/15								
criteria									
ISUP selection	PICS 4/2 AND NOT PI	CS 4	/9						
criteria	E (1 ) (1 OUT '								
Test purpose	an IAM message conta has been received (sta procedure) and the co- expiry of Toiw2  sends the ACM m party's category ir "payphone (10)", t	<ul> <li>sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "INT_IND_VAL", the ISUP indicator set</li> </ul>							
SIP parameter	IO ISOP_IND_ID	, the	ISDN acce	SS III	dicator set to "ISDN_ACC_IND_VAL"				
values									
ISUP parameter	ACM, Backward call in	dicate	or						
values	CPS indicator								
	payphone (10) interworking inc ISUP indicator: ISDN access in	Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: INT_IND_VAL (PIXIT) ISUP indicator: ISUP_IND_ID (PIXIT) ISDN access indicator ISDN_ACC_IND_VAL (PIXIT)  CPG: Event indicator = ALRTING and the BCI from the ACM encapsulated in the received							
Comments	ISUP/BICC		SUT	S	SIP-I				
	IAM	<b>→</b>			NVITE(IAM)				
			•	- 1	83 Session Progress without encapsulated ACM				
	COT	<b>→</b>	-		IPDATE				
				- 2	00 OK UPDATE				
		T <sub>C</sub>	DIW2 expiry						
	ACM(no indication)	+							
	CPG(Alerting, BCi)	+	•	- 1	80 Ringing(ACM)				
	ANM	+	•		00 OK INVITE(ANM)				
		→ ACK							
		Сс	nversation						
	REL	<b>→</b>	-		SYE(REL)				
	RLC	+	•	2	00 OK BYE(RLC)				

TP303013	SIP refer	ence: F	RFC 3261		ISUP reference: Q.1912.5 clause 7.1, 7.3.1, 7.4				
TSS reference	ISUP-SIP /Basic ca	ISUP-SIP /Basic call/Sending of the ACM message							
SIP selection criteria	PICS 1/3								
ISUP selection criteria	PICS 4/2 AND NO	T PICS	4/9						
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	numbe	r: complete	nun	ber				
values	ACM, Backward ca	all indica	ator is set to	the	value in the encapsulated ACM				
Comments	ISUP/BICC		SUT		SIP-I				
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
				<b>←</b>	183 Session Progress without encapsulated ACM				
	COT	→		<b>→</b>	UPDATE				
				<b>←</b>	200 OK UPDATE				
	, . •	CM ← 180 Ringing(ACM)							
	ANM	← 200 OK INVITE(ANM)							
				<b>→</b>	ACK				
			nversation						
		<b>→</b>		<u>→</u>	BYE(REL)				
	RLC	<b>←</b>		<del>(</del>	200 OK BYE(RLC)				

TP303014	SIP reference: RFC 320	61		ISUP reference:						
			Q.19	12.5 clause 7.1, 7.3.1, 7.4						
TSS reference	ISUP-SIP /Basic call/Sending of the INVITE message									
SIP selection	PICS 1/3 AND PICS 3/2 AND NO	T PICS 4/15	5							
criteria										
ISUP selection	PICS 3/8 AND PICS 4/2 AND NO	T 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) and the continuity check is performed (ISUP) or COT is expected (BICC). After the expiry of Toiw2  • sends the ACM message with the CPS indicator set to " no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "INT_IND_VAL", the ISUP indicator set to "ISUP_IND_ID", the ISDN access indicator set to "ISDN_ACC_IND_VAL"									
SIP parameter values										
ISUP parameter	ACM, Backward call indicator									
values	CPS indicator: no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: INT_IND_VAL (PIXIT) ISUP indicator: ISUP_IND_ID (PIXIT) ISDN access indicator: ISDN_ACC_IND_VAL (PIXIT)  CPG: Event indicator = ALRTING and the BCI from the ACM encapsulated in the received									
Comments	180 Ringing ISUP/BICC	,	SUT	SIP-I						
		<b>→</b>								
	COT	<b>→</b>	-	► INVITE(IAM)						
		T <sub>OIV</sub>	<sub>V2</sub> expiry							
	ACM(no indication)	<del>(</del>								
	CPG(Alerting)	<del>(</del>	•	180 Ringing(ACM)						
	ANM	<del>(</del>		200 OK INVITE(ANM)						
				<b>→</b> ACK						
		Conv	ersation							
		<b>→</b>		<b>▶</b> BYE(REL)						
	RLC	<b>←</b>	•	200 OK BYE(RLC)						

TP303015	SIP reference: RFC 32	261	Q.	-	SUP reference: 5 clause 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	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: com	plete numb	er							
values	ACM, Backward call indicator is			capsu	lated ACM					
Comments	ISUP/BICC		SUT		SIP-I					
	IAM	<b>→</b>								
	COT	<b>→</b>		<b>→</b>	INVITE(IAM)					
	ACM	+		+	180 Ringing(ACM)					
	ANM									
				<b>→</b>	ACK					
		Co	nversation							
	REL	<b>→</b>		<b>→</b>	BYE(REL)					
	RLC	<b>←</b>		<b>←</b>	200 OK BYE(RLC)					

## 5.2.2.4 Sending of the CPG message

TP304001	SIP reference: RFC 3	261	Q.19	ISUP reference: 912.5 clause 7.1, 7.3.1			
TSS reference	ISUP-SIP /Basic call/ Sending of the CPG message						
SIP selection	PICS 3/1	PICS 3/1					
criteria							
ISUP selection criteria	PICS 3/8						
Test purpose	Ensure that the SUT, having sent a ACM message with called party status "no indication" on receipt of a 180 Ringing with a encapsulated ISUP message  • Sends the CPG message with the <b>event indicator</b> set to "Alerting"						
SIP parameter							
values							
ISUP parameter	ACM: BCi called party status inc	dicator = no i	ndication				
values	CPG: Event Indicator = ALERTI	NG, BCi as ı	eceived from the	encapsulated ACM			
Comments	ISUP/BICC		SUT	SIP-I			
	IAM	<b>→</b>					
	SAM	<b>→</b>					
	SAM	<b>→</b>	→	INVITE(IAM)			
		T <sub>OIW2</sub> expiry					
	ACM(no indication)	<b>←</b>					
	CPG(Alerting BCi)	<b>←</b>	<b>+</b>	180 Ringing(ACM)			
	ANM	<b>←</b>	<b>+</b>	200 OK INVITE(ANM)			
			<b>→</b>	ACK			
		Cor	versation				
	REL	<b>→</b>	→	BYE(REL)			
	RLC	+	+	200 OK BYE(RLC)			

TP304002	SIP reference: RFC 32	261			ISUP reference: 12.5 clause 7.1, 7.3.1		
TSS reference	ISUP-SIP /Basic call/ Sending of	the C	CPG message	)			
SIP selection	PICS 3/1						
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 receipt of a 183 Session progress message with a encapsulated ISUP message  • Sends the CPG message with the <b>event indicator</b> set to "Alerting"					
SIP parameter							
values							
ISUP parameter							
values							
Comments	ISUP/BICC		SUT		SIP-I		
	IAM	<b>\</b>		<b>→</b>	INVITE(IAM)		
	ACM(no indication)	1		+	183 Session Progress(ACM)		
	CPG(Alerting)	+		+	180 Ringing(CPG)		
	ANM	+		+	200 OK INVITE(ANM)		
				→	ACK		
			Conversat	ion			
	REL	<b>→</b>		→	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

## 5.2.2.5 Sending of the ANM message

TP305001	SIP reference: RFC 326	1		ISUP reference: .1912.5 clause 7.5		
TSS reference	ISUP-SIP/Basic call/ Sending of the	e Answer Messa	age (ANM)/	1		
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	<ul> <li>Ensure that the SUT having sent the ACM message, on receipt of a 200 OK INVITE for this call, it shall stop timer TOIW2 (if running)</li> <li>Send ANM as determined by BICC/ISUP procedures.</li> <li>Stop any existing awaiting answer indication (e.g. ringing tone).</li> </ul>					
SIP parameter	200 OK INVÍTE;	,		,		
values	,					
ISUP parameter	ANM;					
values						
Comments	ISUP/BICC	SUT		SIP-I		
	IAM →		→	INVITE(IAM)		
	ACM ←		+	180 Ringing(ACM)		
	ANM ←		+	200 OK INVITE(ANM)		
			<b>→</b>	ACK		
		Conversation	n			
	REL →		<b>→</b>	BYE(REL)		
	RLC ←		+	200 OK BYE(RLC)		

## 5.2.2.6 Sending of the CON message

TP306001	SIP reference: RFC	3261			ISUP reference: 12.5 clause 7.5, 7.5.1			
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the SUT, having not sent the ACM message, on receipt of a 200 OK INVITE for this call, it shall stop timer TOIW2 (if running)  Send CON as determined by BICC/ISUP procedures.  Stop any existing awaiting answer indication (e.g. ringing tone) BCI encoded as received in the encapsulated CON							
SIP parameter	200 OK INVITE;							
values								
ISUP parameter	CON; interworking indicator							
values	ISUP indicator: ISUP_IND_ID							
	ISDN access indicator ISDN CPS indicator: no indication	I_ACC	C_IND_VAL (PIXIT)					
Comments	ISUP/BICC		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	CON	+		+	200 OK INVITE(CON)			
				<b>→</b>	ACK			
			Conversation					
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			
Comments	ISUP SUT		SIP		·			
	IAM →	<b>→</b>	INVITE(IAM)					
	CON ←	+	200 OK INVITE(	CON)				

## 5.2.2.7 Receipt of the Release message (REL)

TP307001	SIP reference: RFC	3261		ISUP reference: Q.1912.5 clause 7.7.1, 1)				
TSS reference	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/							
SIP selection criteria								
SUP 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	<b>→</b>						
	REL	<b>→</b>						
	RLC	+						

TP307002	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause 7.7.1 2)				
TSS reference	ISUP-SIP/Basic call/ Recei	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	<b>→</b>		<b>→</b>	INVITE(IAM)			
	REL	<b>→</b>						
	RLC	+						
				+	200 OK INVITE(CON)			
				<b>→</b>	ACK			
				<b>→</b>	BYE(REL)			
				+	200 OK BYE(RLC)			

TP307003	SIP reference: RFC 32	61		ISUP reference: 2.5 clause 7.7.1 2) 3)			
TSS reference	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/						
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose  SIP parameter	<ul> <li>Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message before a 200 OK SIP response message has been received</li> <li>The SUT shall hold the REL message. A CANCEL is sent when any SIP response was been received.</li> <li>On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent</li> </ul>						
values							
ISUP parameter values							
Comments	ISUP/BICC	SU <sup>*</sup>	Т	SIP-I			
	IAM →	•	<b>→</b>	INVITE(IAM)			
			+	100 Trying			
	REL +	•					
	RLC		<b>→</b>	CANCEL(REL)			
			+	200 OK INVITE(CON)			
			<b>→</b>	ACK			
			+	200 OK CANCEL			
			<b>→</b>	BYE(REL)			
			+	200 OK BYE(RLC)			

TP307004	SIP reference: RFC 3261		SUP reference: 2.5 clause 7.7.1 2) 3)		
TSS reference	ISUP-SIP/Basic call/ Receipt of the Releas	e message (REL)	1		
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message <b>before</b> an early dialogue with the message 100 Trying has been established  The SUT shall hold the REL message until a <b>100 Trying</b> response has been received.  The SUT shall send a CANCEL The received REL is encapsulated in the CANCEL				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC	SUT	SIP-I		
	IAM →	<b>→</b>	INVITE(IAM)		
	REL →				
	RLC ←				
		+	100 Trying		
		<b>→</b>	CANCEL(REL)		
		+	200 OK CANCEL		
		+	487 Request terminated		
		<b>→</b>	ACK		

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

TP307006	SIP reference: RFC 3261				ISUP reference: 112.5 clause 7.7.1 3)
TSS reference	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established  • The SUT shall send a CANCEL or BYE request. The received REL is encapsulated in the BYE or CANCEL				
SIP parameter					
values					
ISUP parameter values					
Comments	ISUP/BICC		SU	Т	SIP-I
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)
				+	SIP_MESSAGE_VA
	REL	<b>→</b>			
	RLC	+			
	CASE A				
				<b>→</b>	CANCEL(REL)
				+	200 OK CANCEL
				+	487 Request terminated
				<b>→</b>	ACK
	CASE B				
				<b>→</b>	BYE(REL)
				+	200 OK BYE
				+	487 Request terminated
				<b>→</b>	ACK

Table 8

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

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

TP308001	SIP reference: RFC 3261		Q	ISUP reference: .1912.5 clause 7.7.2				
TSS	ISUP-SIP /Basic call/ Sending of the Release message (REL)/							
reference	_							
SIP								
selection								
criteria								
ISUP								
selection								
criteria								
Test		Ensure that the SUT after receiving the IAM sends out an INVITE message and on receipt of a						
purpose	BYE message in the confirmed dial	ogue						
	<ul> <li>sends a REL message constru</li> </ul>	cted from t	he encapsulated	REL in the received BYE				
SIP								
parameter								
values								
ISUP	REL; Cause value "Normal call clea	aring"						
parameter								
values								
Comments	ISUP/BICC	SU		SIP-I				
	IAM	→	<b>→</b>	INVITE(IAM)				
	ACM	<del>(</del>	+	180 Ringing(ACM)				
	ANM	+	<del>(</del>	200 OK INVITE(ANM)				
			<b>→</b>	ACK				
		Conversa	tion					
	REL	<b>←</b>	<b>(</b>	BYE(REL)				
	RLC	<b>→</b>	<b>→</b>	200 OK BYE(RLC)				

TP308002	SIP reference: RFC	3261		=	SUP reference: 912.5 clause 7.7.6		
TSS reference	ISUP-SIP /Basic call/ Sending	of the	Release mes				
SIP selection criteria		<u> </u>					
ISUP selection criteria							
Test purpose	a Failure message (4xx, 5xx, 6	Ensure that the SUT after receiving the IAM sends out an INVITE message. On receipt of a Failure message (4xx, 5xx, 6xx) defined as SIP_Failure_VA  • sends a REL message constructed from the encapsulated REL.					
SIP parameter values							
ISUP parameter values	REL; cause value: CV_ISUP	REL; cause value: CV_ISUP					
Comments	ISUP/BICC		SUT		SIP-I		
	IAM	<b>↑</b>		<b>→</b>	INVITE(IAM)		
				+	100 Trying		
	REL	+		+	SIP_Failure_VA(REL)		
	RLC	1		<b>→</b>	ACK		

Table 9

	Values for test purpose TP308003					
VA	←REL (Cause Value) CV_ ISUP	←4XX/5XX/6XX SIP message SIP_Failure_VA				
VA_01	127 Interworking	400 Bad Request				
VA_02	127 Interworking	402 Payment Required				
VA_03	127 Interworking	403 Forbidden				
VA_04	1 Unallocated number	404 Not Found				
VA_05	127 Interworking	405 Method Not Allowed				
VA_06	127 Interworking	406 Not Acceptable				
VA_07	127 Interworking	408 Request Timeout				
VA_08	22 Number changed (without diagnostic)	410 Gone				
VA_9	127 Interworking	423 Interval Too Brief				
VA_10	20 Subscriber absent	480 Temporarily Unavailable				
VA_11	127 Interworking	481 Call/Transaction does not exist				
VA_12	127 Interworking	482 Loop Detected				
VA_13	127 Interworking	483 Too many hops				
VA_14	127 Interworking	485 Ambiguous				
VA_15	17 User busy	486 Busy Here				
VA_16	127 Interworking	488 Not acceptable here				
VA_17	127 Interworking	493 Undecipherable				
VA_18	127 Interworking	500 Server Internal error				
VA_19	127 Interworking	501 Not implemented				
VA_20	127 Interworking	502 Bad Gateway				
VA_21	127 Interworking	504 Server timeout				
VA_22	17 User busy	600 Busy Everywhere				
VA_23	21 Call rejected	603 Decline				
VA_24	1 Unallocated number	604 Does not exist anywhere				
VA_25	127 Interworking	606 Not acceptable				

TP308003	SIP reference: RF	C 3261				SUP reference: 912.5 clause 7.7.6	
TSS reference	ISUP-SIP /Basic call/ Sendin	g of the	Release r	nessage (	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 <b>487 Request terminated</b> • no action is taken on the ISUP if a CANCEL request was previously sent before an answer to an INVITE was received.						
SIP parameter values							
ISUP parameter values							
Comments	ISUP/BICC		SU <sup>*</sup>	Т		SIP-I	
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)	
					+	100 Trying	
	REL	<b>→</b>			<b>→</b>	CANCEL(REL)	
	RLC						
					+	487 Request Terminated	
					<b>→</b>	ACK	

TP308004	SIP reference: RFC	3261		=	SUP reference: 912.5 clause 7.7.6			
TSS reference	ISUP-SIP /Basic call/ Sending	of the	Release mess	sage (REL)	)/			
SIP selection criteria								
ISUP selection criteria								
Test purpose	message defined as <b>SIP MES</b> 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		SUT		SIP-I			
	IAM	AM → INVITE(IAM)						
	XXX							
	REL	4		+	SIP_Failure_VA(REL)			
	RLC	1		<b>→</b>	ACK			

#### Table 10

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

Table 11

	Values for test purposes TP308004						
VA	←REL (Cause Value) CV_ ISUP	←4XX/5XX/6XX SIP message SIP_Failure_VA					
VA_01	127 Interworking	400 Bad Request					
VA_02	127 Interworking	402 Payment Required					
VA_03	127 Interworking	403 Forbidden					
VA_04	1 Unallocated number	404 Not Found					
VA_05	127 Interworking	405 Method Not Allowed					
VA_06	127 Interworking	406 Not Acceptable					
VA_07	127 Interworking	408 Request Timeout					
VA_08	22 Number changed (without diagnostic)	410 Gone					
VA_09	127 Interworking	423 Interval Too Brief					
VA_10	20 Subscriber absent	480 Temporarily Unavailable					
VA_11	127 Interworking	481 Call/Transaction does not exist					
VA_12	127 Interworking	482 Loop Detected					
VA_13	127 Interworking	483 Too many hops					
VA_14	127 Interworking	485 Ambiguous					
VA_15	17 User busy	486 Busy Here					
VA_16	127 Interworking	488 Not acceptable here					
VA_17	127 Interworking	493 Undecipherable					
VA_18	127 Interworking	500 Server Internal error					
VA_19	127 Interworking	501 Not implemented					
VA_20	127 Interworking	502 Bad Gateway					
VA_21	127 Interworking	504 Server timeout					
VA_22	17 User busy	600 Busy Everywhere					
VA_23	21 Call rejected	603 Decline					
VA_24	1 Unallocated number	604 Does not exist anywhere					
VA_25	127 Interworking	606 Not acceptable					

TP308005	SIP reference: RFC	3261				SUP reference: 912.5 clause 7.7.6		
TSS reference	ISUP-SIP /Basic call/ Sending	of the	Release r	nessage (F	REL)	/		
SIP selection criteria	NOT PICS 4/10							
ISUP selection criteria								
Test purpose	Ensure that the SUT after receiving the IAM sends out an INVITE message a 180 ringing message has been received on receipt of a Failure message (4xx, 5xx, 6xx) defined as SIP_Failure_VA  • sends a REL message constructed from the encapsulated REL.							
SIP parameter values				•				
ISUP parameter values	REL; cause value: CV_ISUP							
Comments	ISUP/BICC		SU <sup>*</sup>	Γ		SIP-I		
	IAM	IAM → INVITE(IAM)						
	ACM	ACM ← 180 Ringing						
	REL	+			<del>(</del>	SIP_Failure_VA(REL)		
	RLC	<b>→</b>			<b>→</b>	ACK		

Table 12

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

TP30806	SIP reference: RFC 3261				SUP reference: 912.5 clause 7.7.6		
TSS reference	ISUP-SIP /Basic call/ Sending	of the	Release messag	je (REL)	)/		
SIP selection criteria	NOT PICS 4/21						
ISUP selection criteria							
Test purpose	Ensure that the SUT after receiving the IAM sends out an INVITE message. On receipt of a response message (3xx) defined as SIP_Response_VA, the SUT  sends a REL message with the Cause value CV_ ISUP						
SIP parameter values							
ISUP parameter values	REL; cause value: CV_ISUP	REL; cause value: CV_ISUP					
Comments	ISUP/BICC		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
				+	100 Trying		
	REL	+		+	SIP_Response_VA		
	RLC	<b>→</b>		→	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" (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 Q.850.

NOTE 2: Due to the fact that the Cause Indications parameter does not include the definition text as defined in table1/Q.850 this is based on provisioning in the O-IWU.

### 5.2.2.9 Autonomous release at O-IWU

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

TP309001	SIP reference: RFC 32	261		ISUP reference:		
			Q.1912.5 c	lause 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	ISUP/BICC SUT SIP-I				
	IAM =	<b>)</b>				
	RSC ÷	•				
	RLC	<del>-</del>				

TP309002	SIP reference: RFC			Q.1912.5	SUP reference: clause 7.7.1, 7.7.4, 7.7.5
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt RSC message before a SIP MESSAGE_VA response message has been received  The SUT shall hold the RSC message until a SIP response has been received  The SUT shall send a CANCEL or BYE request. The RSC is not encapsulated				
SIP parameter values	CANCEL or BYE: A REL is er	ncapsula	ated with o	cause 31	
ISUP parameter values					
Comments	ISUP/BICC		SUT	ŗ	SIP-I
	IAM	<b>→</b>		→	INVITE(IAM)
	RSC	<b>→</b>			
	RLC	+			
				+	SIP_MESSAGE_VA
	CASE A				
				→	CANCEL
				+	200 OK CANCEL
				+	487 Request terminated
				→	ACK
	CASE B	•			
				→	BYE(REL#31)
				+	200 OK BYE(RLC)
				+	487 Request terminated
				→	ACK

Table 15

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

TP309003	SIP reference: RFC 3261	ISUP reference: Q.1912.5 clause 7.7.1, 7.7.4, 7.7.5					
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt RSC message before a 200 OK response message has been received  On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent The RSC is not encapsulated						
SIP parameter values	BYE: A REL is encapsulated with cause 31						
ISUP parameter values							
Comments	ISUP/BICC	SU	T	SIP-I			
	IAM         →         INVITE(IAM)           RSC         →         INVITE(IAM)           RLC         ←         INVITE(IAM)						
	200 014 11 11 (175 (2011)						
	→         BYE(REL#31)           ←         200 OK BYE(RLC)						

TP309005	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause 7.7.1, 7.7.4, 7.7.5		
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message with the complete called party number, sending a BYE message on receipt RSC message after a 200 OK response message has been received  The SUT shall send a BYE request The RSC is not encapsulated					
SIP parameter values	BYE: A REL is encapsulated with cause 31					
ISUP parameter values						
Comments	ISUP/BICC IAM ACM ANM	<b>+</b> + + + + + + + + + + + + + + + + + +	SU	÷ ÷ ÷ ÷ ÷ ÷	SIP-I INVITE(IAM) 180 Ringing(ACM) 200 OK INVITE(ANM) ACK  BYE(REL#31)	
	RLC ← 200 OK BYE(RLC					

TP309006	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause 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 after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established The SUT shall send a CANCEL or BYE request The RSC is not encapsulated.					
SIP parameter values	CANCEL or BYE: A REL is encapsulated with cause 31					
ISUP parameter values						
Comments	ISUP/BICC		SU	T	SIP-I	
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)	
				<del>-</del>	SIP_MESSAGE_VA	
	RSC	→				
	RLC	+				
	CASE A					
				→	CANCEL	
				<b>←</b>	200 OK CANCEL	
				<b>←</b>	487 Request terminated	
				→	ACK	
	CASE B					
				<b>→</b>	BYE(REL#31)	
				<b>←</b>	200 OK BYE(RLC)	
				<b>←</b>	487 Request terminated	
	→ ACK					

Table 16

	Values for test purpose; TP309006					
VA	A 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	P reference: RFC 3261		ISUP reference:			
			Q.1912.5 c	lause 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 GRS message no action is required on the SIP side other than to terminate local procedures if any are in progress						
SIP parameter values							
ISUP parameter values							
Comments	ISUP/BICC	SU	T	SIP-I			
	IAM	<b>→</b>					
	GRS	<b>→</b>					
	GRA	+					

TP309008	SIP reference: RFC	3261		-	SUP reference: clause 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	Т	SIP-I	
	IAM	<b>→</b>		→	INVITE(IAM)	
	GRS	<b>→</b>				
	GRA	<b>←</b>				
				<del>(</del>	SIP_MESSAGE_VA	
	CASE A					
				<b>→</b>	CANCEL	
				<del>(</del>	200 OK CANCEL	
				<del>(</del>	487 Request terminated	
				<b>→</b>	ACK	
	CASE B					
				<b>→</b>	BYE(REL#31)	
				<b>+</b>	200 OK BYE(RLC)	
				+	487 Request terminated	
	→ ACK					

Table 17

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

TP309009	SIP reference: RFC 32	61	_	SUP reference:			
			Q.1912.5 c	lause 7.7.1 3), 7.7.4, 7.7.5			
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt GRS message before a 200 OK response message has been received  The SUT shall hold the GRS message until a response has been received. A CANCEL is sent The GRS is not encapsulated  On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent						
SIP parameter values							
ISUP parameter values							
Comments	ISUP/BICC	SU	Т	SIP-I			
	IAM -	•	→	INVITE(IAM)			
			+	100 Trying			
	GRS +						
	GRA ← CANCEL						
			+	200 OK INVITE(CON)			
			→	ACK			
			+	200 OK CANCEL			
			<b>→</b>	BYE(REL#31)			
			+	200 OK BYE(RLC)			

TP309011	SIP reference: RFC	3261			ISUP reference:		
	Q.1912.5 clause 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	g mes	sage (CGI	3) with the in	dication hardware failure		
	oriented						
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Ensure that the SUT after rece						
	sending a INVITE message w						
		•		•	se message has been received		
	<ul> <li>The SUT shall send a BY</li> </ul>	E requ	est The G	RS is not en	capsulated		
SIP parameter							
values							
ISUP parameter							
values							
Comments	ISUP/BICC		SU	Т	SIP-I		
	IAM	<b>↑</b>		7	INVITE(IAM)		
	ACM	<b>+</b>		•	180 Ringing(ACM)		
	ANM ← 200 OK INVITE(ANM)						
	→ ACK						
	GRS	<b>→</b>		7	BYE(REL#31)		
	GRA	+	•	•	200 OK BYE(RLC)		

TP309012	SIP reference: RFC	3261		Q.1912.	ISUP reference: 5 clause 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 <b>after</b> an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established  The SUT shall send a CANCEL or BYE request The GRS is not encapsulated					
SIP parameter values					·	
ISUP parameter values						
Comments	ISUP/BICC		SU	T	SIP-I	
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)	
				+	SIP_MESSAGE_VA	
	GRS	<b>→</b>				
	GRA	+				
	CASE A					
				→	CANCEL	
				+	200 OK CANCEL	
				+	487 Request terminated	
				→	ACK	
	CASE B					
				→	BYE(REL#31)	
				+	200 OK BYE(RLC)	
				+	487 Request terminated	
				→	ACK	

Table 18

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

TP309013	SIP reference: RFC 3261		ISUP reference: Q.1912.5 clause 7.7.1, 7.7.4, 7.7.5			
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Ensure that the SUT after receiving more than one IAM's sending an INVITE message for each call association on receipt of a GRS message were the Range Parameter value is bigger than "1"  the SUT shall send a BYE requests for each call association The GRS is not encapsulated					
SIP parameter values	BYE1 contains the CS BYE2 contains the CS					
ISUP parameter values						
Comments	ISUP/BICC		SUT		SIP-I	
	IAM	→		→	INVITE1(IAM)	
	ACM	+		<b>←</b>	180 Ringing(ACM)	
	ANM	←		<b>←</b>	200 OK INVITE(ANM)	
				<b>→</b>	ACK	
	IAM	<b>→</b>		<b>→</b>	INVITE2(IAM)	
	ACM	+		+	180 Ringing(ACM)	
	ANM	+		+	200 OK INVITE(ANM)	
				→	ACK	
	GRS	<b>→</b>				
	GRA	<b>←</b>				
				→	BYE1(REL#31)	
				<del>-</del>	200 OK BYE(RLC)	
				<b>→</b>	BYE2(REL#31)	
				+	200 OK BYE(RLC)	

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

TP3090014	SIP reference: RFC 326	51		P reference: nuse 7.7.1, 1) , 7.7.4				
TSS reference	•	Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message 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 CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented"  no action is required on the SIP side other than to terminate local procedures if any are in progress							
SIP parameter values								
ISUP parameter values	CGB(hardware failure oriented)							
Comments	ISUP/BICC SUT SIP-I							
	IAM →							
	CGBA <b>←</b>							

TP309015	SIP reference: RFC 3261			-	SUP reference:	
				Q.191	2.5 clause 7.7.1, 7.7.4	
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" before a SIP MESSAGE_VA response message has been received  The SUT shall hold the CGB message until a SIP 200 OK response has been received  The SUT shall send a CANCEL request The CGB is not encapsulated					
SIP parameter values						
ISUP parameter values	CGB(hardware failure orie	ented)				
Comments	ISUP/BICC		SU	Т	SIP-I	
	IAM	<b>→</b>		→	INVITE(IAM)	
	CGB	<b>→</b>				
	CGBA	+				
				+	SIP_MESSAGE_VA	
	CASE A					
				<b>→</b>	CANCEL	
				+	200 OK CANCEL	
				+	487 Request terminated	
				→	ACK	
	CASE B					
				→	BYE(REL#31)	
				+	200 OK BYE(RLC)	
				+	487 Request terminated	
				<b>→</b>	ACK	

Table 19

	Values for test purpose TP309014						
VA	VA SIP MESSAGE_VA						
VA_1	100 Trying						
VA_2	180 Ringing						
VA_3	183 Session Progress						

TP3090016	SIP reference: RFC 3261			-	SUP reference: 5 clause 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" <b>before</b> a 200 OK response message has been received  On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent The CGB is not encapsulated						
SIP parameter values		•					
ISUP parameter values	CGB(hardware failure	oriented)					
Comments	ISUP/BICC		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
				+	100 Trying		
	CGB	<b>→</b>			CANOFI		
	CGBA	+		<b>→</b>	CANCEL		
				<b>←</b>	200 OK INVITE(CON) ACK		
				+	200 OK CANCEL		
				→ <del>-</del>	BYE(REL#31)		
				<del>-</del>	200 OK BYE(RLC)		

TP309017	SIP reference: RFC 3261 ISUP reference: Q.1912.5 clause 7.7.1, 7.7.4						
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" after a 200 OK response message has been received  The SUT shall send a BYE request The CGB is not encapsulated						
SIP parameter values		•					
ISUP parameter values	CGB(hardware failure oriented)						
Comments	ISUP/BICC	SU		SIP-I			
	,,	<b>→</b>	<b>→</b>	INVITE(IAM)			
	ACM         ←         180 Ringing(ACM)           ANM         ←         200 OK INVITE(ANM)						
	→ ACK						
	000	_		D)(E(DEL #04)			
	002	<del>}</del>	<b>→</b>	BYE(REL#31)			
	CGBA	<del>(</del>	<del>(</del>	200 OK BYE(RLC)			

TP309018	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause 7.7.1, 7.7.4		
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" <b>after</b> an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established  • The SUT shall send a CANCEL or BYE request The CGB is not encapsulated					
SIP parameter values				•	·	
ISUP parameter values	CGB(hardware failure oriented	d)				
Comments	ISUP/BICC		SU	Т	SIP-I	
	IAM	→		→	INVITE(IAM)	
				+	SIP_MESSAGE_VA	
	CGB	→				
	CGBA	<b>←</b>				
	CASE A					
				<b>→</b>	CANCEL	
				+	200 OK CANCEL	
	← 487 Request termi					
				<b>→</b>	ACK	
	CASE B					
				<b>→</b>	BYE(REL#31)	
				+	200 OK BYE(RLC)	
				<b>←</b>	487 Request terminated	
				→	ACK	

Table 20

	Values for test purpose TP309114; TP309018					
VA	VA SIP MESSAGE_VA					
VA_1	180 Ringing					
VA_2	183 Session Progress					

TP309019	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause 7.7.1, 7.7.4, 7.7.5			
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Ensure that the SUT after receiving more than one IAM's sending an INVITE message for each call association on receipt of a CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" were the Range and Status Parameter value is bigger than "1"  the SUT shall send a BYE requests for each call association The CGB is not encapsulated						
SIP parameter values	BYE1 contains the CSe BYE2 contains the CSe						
ISUP parameter values	CGB(hardware failure	oriented)					
Comments	ISUP/BICC		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE1(IAM)		
	ACM	+		<b>←</b>	180 Ringing(ACM)		
	ANM	<del>-</del>		<b>←</b>	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
	IAM	<b>→</b>		<b>→</b>	INVITE2(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
	CGB	<b>→</b>					
	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 reference: RFC 3261				ISUP reference: Q.1912.5 clause A.1.1.3					
TSS reference	ISUP-SIP/ ISU	ISUP-SIP/ ISUP Messages for special consideration / Confusion message								
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	contains an un party number a Ensure that wh Confusion mes of a Confusion Progress is ser Ensure ISUP n Session Progres	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 F	rogress with en	capsulated CF	N						
ISUP parameter values	CFN									
	ISUP				SIP-I					
	IAM	→		<b>→</b>	INVITE(IAM with unknown parameter)					
	CFN	+		<b>←</b>	183 Session Progress(CFN)					
	ACM	ACM         ←         180 Ringing(ACM)           ANM         ←         200 OK INVITE(ANM)								
	ANM									
				<b>→</b>	ACK					
		C	ommunication							
	REL	<b>→</b>		<b>→</b>	BYE(REL)					
	RLC	+		<b>←</b>	200 OK BYE(RLC)					

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

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

#### 5.2.2.12 Receipt of a Blocking message

TP312001	SIP reference: RFC 32	61	Q.	ISUP reference: 1912.5 clause A.1.1.3.1						
TSS reference	ISUP-SIP/ ISUP Messages for sp	pecial conside	eration / Rec	eipt 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 is not encapsulated within SIP messages									
SIP parameter										
values										
ISUP parameter										
values										
Comments	ISUP/BICC	SU	JT	SIP-I						
	BLO •	<b>→</b>								
	BLA	<del>(</del>								
	UBL •	<b>→</b>								
	UBA ⋅	<del>(</del>								

TP312002	SIP reference: RFC 3261		Q.1	ISUP reference: 912.5 clause A.1.1.3.1					
TSS reference	ISUP-SIP/ ISUP Messages for spec	ial conside	ration / Rece	eipt of a Blocking message					
SIP selection criteria									
ISUP selection criteria									
Test purpose	Ensure that the blocking from both ends; removal of blocking from one end can be correctly initiated Ensure the BLO messages is not encapsulated within SIP messages								
SIP parameter				•					
values									
ISUP parameter									
values									
Comments	ISUP/BICC	SU	Т	SIP-I					
	BLO →								
	BLA 🗲								
	BLO ←								
	BLA →								
	UBL →								
	UBA <b>←</b>								

TP312003	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause A.1.1.3.1				
TSS reference	ISUP-SIP/ ISUP Messages for	special	conside	ration / Re	eceipt 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 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				'	J			
ISUP parameter values								
Comments	ISUP		SU	T	SIP-I			
	CGB	<b>→</b>						
	CGBA ←							
	CGU	<b>→</b>						
	CGUA	+						

TP312004	SIP reference: RF	C 3261	ISUP reference: Q.1912.5 clause A.1.1.3.1					
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Receipt of a Blocking message							
SIP selection criteria	-							
ISUP selection criteria								
Test purpose	Ensure that the SUT on receipt of a CGB, which is received encapsulated within SIP messages, discards the ISUP information.							
SIP parameter values								
ISUP parameter values								
Comments	ISUP	SUT	SIP-I					
			← INFO(CBG)					

TP312005	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause A.1.1.3.1 Q.784 clause 1.3.2.4		
TSS reference	ISUP-SIP/ ISUP Messa	ages for special	conside	eration / F	Receip	t of a Blocking message
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Ensure that a received	IAM will unbloc	k a rem	otely blod	cked c	ircuit.
SIP parameter						
values						
ISUP parameter						
values						
Comments	ISUP		SL	JT		SIP-I
	BLO	→				
	BLA	+				
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)
	ACM	+			+	180 Ringing(ACM)
	ANM	+			+	200 OK INVITE(ANM)
					<b>→</b>	ACK
	REL	<b>→</b>			<b>→</b>	BYE(REL)
	RLC	+			+	200 OK BYE(RLC)

#### 5.2.2.13 Receipt of a user part test message

TP313001	SIP reference: RF	C 3261	= -	SUP reference: 2.5 clause A.1.1.3.1			
			Q.7	'84 clause 1.3.2.4			
TSS reference	ISUP-SIP/ ISUP Messages f	or special conside	eration / Receip	t of a user part test message			
SIP selection criteria							
ISUP selection	PICS 4/22						
criteria							
Test purpose	Ensure that on receipt of a user part test 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 SUT SIP-I						
	UPT	<b>→</b>					
	UPA	<del>-</del>					

TP313002	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause A.1.1.3.1			
TSS reference	ISUP-SIP/ ISUP Messages	for specia	l consideration	on / Receipt	of a user part test message		
SIP selection criteria							
ISUP selection	PICS 4/22						
criteria							
Test purpose	Ensure that the SUT is able	to send a	user part te	st message			
SIP parameter							
values							
ISUP parameter							
values							
Comments	ISUP		SUT		SIP-I		
	UPT <b>←</b>						
	UPA	<b>→</b>					

TP313003	SIP reference: RFC 3	3261	ISUP reference:			
			Q.1912	.5 clause A.1.1.3.1		
TSS reference	ISUP-SIP/ ISUP Messages for	special conside	eration / Receipt	of a user part test message		
SIP selection						
criteria						
ISUP selection	PICS 4/22					
criteria						
Test purpose	T4 Waiting to receive a resp	onse to a user	part test messa	ige		
	Ensure that the SUT is able to	restart the avai	lability test proce	dure after expiry of timer T4		
SIP parameter						
values						
ISUP parameter						
values						
Comments	ISUP	SL	JT	SIP-I		
	UPT 🗲					
	T4 expiry					
	UPT	+				
	UPA	<b>→</b>				

#### 5.2.2.14 Segmentation

TP314001	SIP reference	e: RFC 3261	ISUP reference: Q.1912.5 clause A.1.1.3.1							
TSS reference	ISUP-SIP/ ISUP Messa	ISUP-SIP/ ISUP Messages for special consideration / Receipt of a user part test message								
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	Ensure that a call can l direction.	Ensure that a call can be successfully completed if segmentation applies in forward direction.								
SIP parameter	INVITE - encapsulated IAM: Forward call indicator absent or set to "no additional									
values	information will be sent"									
	No action takes place on the SIP side									
ISUP parameter	IAM: optional forward of	IAM: optional forward call indicator: additional information will be sent in a segmentation								
values	message				-					
	SGM: optional parame	ters								
Comments	ISUP		SUT		SIP-I					
	IAM	<b>→</b>								
	SGM	<b>→</b>		<b>→</b>	INVITE(IAM)					
	ACM	+		+	180 Ringing(ACM)					
	ANM ← 200 OK INVITE(ANM)  → ACK									
			Convers	ation						
	REL	<b>→</b>		<b>→</b>	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: RF	C 326	1		-	SUP reference:		
TCC vofeveres	IOLID OID IOLID/OO/OLID				Q.T	912.5 clause 7.1.3		
TSS reference	ISUP-SIP-ISUP/SS/CLIP							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the SUT can successfully transmit a call having a calling party number with							
		the screening indicator set to "network provided" and the presentation restricted indicator						
	set to "presentation allowed"	"						
SIP parameter								
values								
ISUP parameter	IAM;	IAM;						
values	Calling party number para	meter						
	Address signals = PIXIT1							
	Numbering plan indicator = '	'001'B						
	Nature of address indicator	= '0000	0011'B					
	Screening indicator = '11'B							
	presentation restricted indica	ator =	presentation	n allowed,	'00'B			
Comments	ISUP		SU	Γ		SIP-I		
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)		
	ACM	<b>←</b>			+	180 Ringing(ACM)		
	ANM	+			+	200 OK INVITE(ANM)		
			Convers	ation				
	REL	<b>→</b>			<b>→</b>	BYE(REL)		
	RLC	+			+	200 OK BYE(RLC)		

TP401002	SIP reference: RF	C 326		-	SUP reference: 912.5 clause 7.1.3					
TSS reference	ISUP-SIP-ISUP/SS/CLIP									
SIP selection										
criteria										
ISUP selection criteria										
Test purpose	Ensure that the SUT can successfully transmit a call having a calling party number with the screening indicator set to "network provided" and an access transport parameter containing the calling sub-address									
SIP parameter										
values										
ISUP parameter values	IAM;  Calling party number parameter  Address signals = PIXIT1  Numbering plan indicator = '001'B  Nature of address indicator = '0000011'B  Screening indicator = '11'B  presentation restricted indicator = presentation allowed, '00'B									
Comments	Access transport parameter	iriciuui	SUT	IIIIOIIII	SIP-I					
	IAM	<b>→</b>	001	<b>→</b>	INVITE(IAM)					
	ACM	+		Ť	180 Ringing(ACM)					
	ANM	+		+	200 OK INVITE(ANM)					
			Conversation							
	REL	<b>↑</b>		<b>→</b>	BYE(REL)					
	RLC	+		+	200 OK BYE(RLC)					

TP401003	SIP reference: RF	C 326′	I	-	SUP reference: 912.5 clause 7.1.3				
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection criteria									
ISUP selection criteria									
Test purpose	Ensure that the SUT can successfully transmit a call having the <b>calling party number</b> with the screening indicator set to "user provided, verified and passed" and the presentation restricted indicator set to "presentation allowed"								
SIP parameter									
values									
ISUP parameter	IAM;	ĪAM;							
values	Calling party number parameter Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '01'B presentation restricted indicator = presentation allowed, '00'B								
Comments	ISUP		SUT		SIP-I				
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM	+		<b>←</b>	180 Ringing(ACM)				
	ANM	+		←	200 OK INVITE(ANM)				
			Conversat	ion					
	REL	1		<b>→</b>	BYE(REL)				
	RLC	+	_	+	200 OK BYE(RLC)				

TP401004	SIP reference: RF	C 326	1			SUP reference: 912.5 clause 7.1.3					
TSS reference	ISUP-SIP-ISUP/SS/CLIP										
SIP selection											
criteria											
ISUP selection											
criteria											
Test purpose	Ensure that the SUT can successfully transmit a call having a <b>calling party number</b> with the screening indicator set to "user provided, verified and passed" and an <b>access transport</b> parameter containing the <b>calling sub-address</b>										
SIP parameter											
values											
ISUP parameter	IAM;										
values	Calling party number para	meter									
	Address signals = PIXIT1										
	Numbering plan indicator =	'001'B									
	Nature of address indicator	= '0000'	0011'B								
	Screening indicator = '01'B										
	Presentation restricted indic										
	Access transport parameter	includ		dress info	orma						
Comments	ISUP		SUT			SIP-I					
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)					
	ACM	+			<b>←</b>	180 Ringing(ACM)					
	ANM ← 200 OK INVITE(ANM)										
			Conversati	ion							
	REL	<b>^</b>			<b>→</b>	BYE(REL)					
	RLC	<b>+</b>			<b>←</b>	200 OK BYE(RLC)					

TP401005	SIP reference: RF	C 3261	1			SUP reference: 912.5 clause 7.1.3			
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Ensure that the SUT can successfully transmit a call having a default <b>calling party number</b> with the screening indicator set to "network provided" and a <b>generic number</b> 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 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								
	Generic number paramete	r							
	Address signals = PIXIT2	•							
	Numbering plan indicator = '	'001'B							
	Nature of address indicator		0011'B						
	Screening indicator = '00'B	0000							
	Presentation restricted indic	ator = ı	presentation	allowed.	'00'B				
Comments	ISUP		SUT			SIP-I			
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)			
	ACM	+			<del>-</del>	180 Ringing(ACM)			
	ANM	+			<del>-</del>	200 OK INVITE(ANM)			
			Conversa	tion		` ′			
	REL	<b>→</b>			<b>→</b>	BYE(REL)			
	RLC	+			<del>-</del>	200 OK BYE(RLC)			

TP401006	SIP reference: RF0	C 3261				SUP reference:					
					Q.19	912.5 clause 7.1.3					
TSS reference	ISUP-SIP-ISUP/SS/CLIP	ISUP-SIP-ISUP/SS/CLIP									
SIP selection											
criteria											
ISUP selection											
criteria											
Test purpose	Ensure that the SUT can successfully transmit a call having a default calling party number with the screening indicator set to "network provided", a generic number containing the additional calling party number with the screening indicator set to "user provided, not verified" and an access transport parameter containing the calling sub-address										
SIP parameter											
values											
ISUP parameter	1004	IANA.									
values	IAM;										
	Calling party number parameter										
	Address signals = PIXIT1  Numbering plan indicator = '001'B  Nature of address indicator = '0000011'B  Screening indicator = '11'B										
	Generic number parameter	•									
	Address signals = PIXIT2  Numbering plan indicator = '001'B  Nature of address indicator = '0000011'B  Screening indicator = '00'B  Access transport parameter including the subaddress information										
Comments	ISUP		SUT			SIP-I					
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)					
	ACM	+			+	180 Ringing(ACM)					
	ANM	+			+	200 OK INVITE(ANM)					
			Conversa	ation							
	REL	<b>→</b>			<b></b>	BYE(REL)					
	RLC	+			+	200 OK BYE(RLC)					

SIP selection criteria ISUP selection criteria Test purpose SIP parameter values	ISUP-SIP-ISUP/SS/CLIP PICS 6/8 Ensure that the calling party address presentation restricted			in case of t			
SIP selection criteria ISUP selection criteria Test purpose SIP parameter values	PICS 6/8 Ensure that the calling party						
criteria ISUP selection criteria Test purpose SIP parameter values	Ensure that the calling party						
ISUP selection criteria Test purpose SIP parameter values	Ensure that the calling party						
criteria Test purpose  SIP parameter values	Ensure that the calling party						
SIP parameter values							
SIP parameter values							
SIP parameter values	address presentation restricte	ed indi	cator is set to	"presentati	on allowed" (see note).		
values							
ISUP parameter I							
	IAM;						
values	No calling party number pa	ramet	er				
Comments	SIP-I		SUT		ISUP		
Ī	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM		
, 1	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
, [	Conversation						
Ī	BYE(REL)	<b>→</b>		<b>→</b>	REL		
	200 OK BYE(RLC)	+		+	RLC		
	ral agreement prohibits the tra			• • •			

TP401008	SIP reference: F	RFC 3261			SUP reference:				
				Q.1	912.5 clause 7.1.3				
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection									
criteria									
ISUP selection criteria	PICS 6/7	PICS 6/7							
Test purpose		Ensure that the additional calling party number in the <b>generic number</b> is discarded in case of bilateral agreements, if the address presentation restricted indicator is set to presentation allowed"							
SIP parameter									
values									
ISUP parameter	IAM;								
values	No calling party number	paramete	r						
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	<b>←</b>		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
			Conversatio	n					
	BYE(REL) → REL								
	200 OK BYE(RLC)	+		+	RLC				
	eral agreement prohibits the ess presentation restricted in				nber in any case. The test with ted" is a CLIR test.				

TP401009	SIP reference: F	RFC 326	1	ISUP reference: Q.1912.5 clause 7.1.3				
TSS reference	ISUP-SIP-ISUP/SS/CLIP							
SIP selection criteria								
ISUP selection criteria	PICS 6/6							
Test purpose	Ensure that the <b>calling party number</b> 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)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+	ANM					
	Conversation							
	BYE(REL)	<b>→</b>		→	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP401010	SIP reference: RFC 3261				ISUP reference: 1912.5 clause 7.1.3			
TSS reference	ISUP-SIP-ISUP/SS/CLIP	ISUP-SIP-ISUP/SS/CLIP						
SIP selection criteria								
ISUP selection criteria								
Test purpose	Ensure that the calling party number in the sent IAM is generated from the calling party number in the encapsulated IAM							
SIP parameter		·						
values								
ISUP parameter values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
			Conversation	•				
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP401011	SIP reference: RFC 3261		IS	UP reference	e:				
			Q.	1912.5 clau	se 7.1.3				
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection criteria									
ISUP selection criteria									
Test purpose	Ensure that the additional calling party number in the sent IAM is generated from the additional calling party number in the encapsulated IAM								
SIP parameter	<u> </u>		•						
values									
ISUP parameter values									
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
			Conversati	sation					
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP401012	SIP reference: R		ISUP reference: Q.1912.5 clause 7.1.3					
TSS reference	ISUP-SIP-ISUP/SS/CLIP		•					
SIP selection								
criteria								
ISUP selection criteria								
Test purpose		Ensure that if the <b>calling party number</b> is not sent, then an additional calling party number in a <b>generic number</b> will be omitted.						
SIP parameter	INVITE: No calling party n	INVITE: No calling party number included in the encapsulated IAM, additional calling party						
values	number included.							
ISUP parameter	IAM;							
values	No calling party number	parame	ter					
	No generic number para	meter						
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
			Conversation	n				
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP401013	SIP reference: RI	FC 3261	I	ı	SUP reference: 3.5/Q.731					
TSS reference	ISUP-SIP-ISUP/SS/CLIP									
SIP selection										
criteria										
ISUP selection	PICS 1/7									
criteria										
Test purpose	Convert the Calling party Ensure that the SUT can consetting the nature of address address presentation restricts.	onvert these indicates	ne <b>calling party nur</b> ator to "international	<b>nber</b> i numb	nto an international number, er" and can pass on the					
SIP parameter values										
ISUP parameter	IAM;									
values	Calling party number para	ameter								
	Address signals = PIXIT1									
	Numbering plan indicator =									
	Nature of address indicator	= '0000'	)100'B							
	Screening indicator = '11'B									
	Presentation restricted indi	cator =p		<u>, '00'B</u>						
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM					
	180 Ringing(ACM)	+		+	ACM					
	200 OK INVITE(ANM)	+		+	ANM					
			Conversation	1						
	BYE(REL)	<b>→</b>		<b>→</b>	REL					
	200 OK BYE(RLC)	<b>←</b>		+	RLC					

TP401014	SIP reference: RF	C 3261	1		IS	SUP reference: 3.5/Q.731					
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  Ensure that the SUT can convert the additional calling party number in the <b>generic</b> number into an international number, if the numbering plan indicator is "ISDN Telephony", setting the nature of address indicator to "international number" and can pass on the address presentation restricted indicator and the screening indicator transparently										
SIP parameter						1 ,					
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 indicator Generic number paramete Address signals = PIXIT2 Numbering plan indicator = Nature of address indicator Screening indicator = '00'B Presentation restricted indicator	'001'B = '0000 ator =p r '001'B = '0000	oresentation al 0100'B oresentation al	·	'00'B						
Comments	SIP-I		SUT			ISUP					
	INVITE(IAM)	<b>→</b>				IAM					
	180 Ringing(ACM)	<b>←</b>				ACM					
	200 OK INVITE(ANM)	+			+	ANM					
			Conversati	on							
	BYE(REL)	<b>→</b>				REL					
	200 OK BYE(RLC)	+			<del>-</del>	RLC					

TP401015	SIP reference: RFC 3261				ISUP reference: 3.5/Q.731					
TSS reference	ISUP-SIP-ISUP/SS/CLIP	ISUP-SIP-ISUP/SS/CLIP								
SIP selection										
criteria										
ISUP selection	PICS 1/7 AND NOT PICS	1/9								
criteria										
Test purpose	Discarding an incomplete	• .	•							
	Ensure that the calling par				eived with the calling party					
	number incomplete indica	tor set to	"incomplete	e" (see note).						
SIP parameter										
values										
ISUP parameter	IAM:									
values	No calling party number	parame	ter							
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM					
	180 Ringing(ACM)	+		+	ACM					
	200 OK INVITE(ANM)	+		+	ANM					
			Convers	ation						
	BYE(REL)	<b>→</b>		<b>→</b>	REL					
	200 OK BYE(RLC)	200 OK BYE(RLC) ← RLC								
NOTE: This test	case is only applicable with	an ITU i	mplementat	ion.						

TP401016	SIP reference: RF	C 326	1	Į:	SUP reference:						
T00 (	3.5/Q.731										
TSS reference	ISUP-SIP-ISUP/SS/CLIP										
SIP selection											
criteria											
ISUP selection	PICS 1/8										
criteria											
Test purpose	Converting the calling par										
	Ensure that the country cod										
	removed if it is the network's	s own c	ountry code. Th	ne nature of	f address indicator shall be						
		numbe	r". The address	presentation	on restricted indicator shall be						
	transferred transparently.										
SIP parameter	INVITE: encapsulated IAM										
values	Calling party number		meter								
	Address signals = PI										
	Numbering plan indic										
	Nature of address in		= '0000011'B								
	Screening indicator =										
	Presentation restricte	ed indic	cator = presenta	tion allowe	d, '00'B						
ISUP parameter	IAM										
values	Calling party number param	eter									
	Address signals = PIXIT1	100410									
	Numbering plan indicator =		1400ID								
	Nature of address indicator	= '0000	)100 <sup>-</sup> B								
	Screening indicator = '11'B	-4									
Commonto	Presentation restricted indic	ator =		owed, 00 B							
Comments	SIP-I		SUT	<b>→</b>	ISUP						
	IAM	<b>→</b>			INVITE(IAM)						
	ACM	<b>+</b>		<del>-</del>	180 Ringing(ACM)						
	ANM	+	0	<b>+</b>	200 OK INVITE(ANM)						
	551		Conversation		DVE (DEL)						
	REL	<b>→</b>		<b>→</b>	BYE(REL)						
	RLC	<b>←</b>		+	200 OK BYE(RLC)						

TP401017	SIP reference: RF	C 326	1		ISUP reference: 3.5/Q.731						
TSS reference:	ISUP-SIP-ISUP/SS/CLIP										
SIP selection	1001 011 1001/00/0211										
criteria											
ISUP selection	PICS 1/8										
criteria	1										
Test purpose	Ensure that the country code "additional calling party num removed if it is the network's	Converting the additional calling party number to national format, if necessary Ensure that the country code in the address signals of the <b>generic number</b> coded as an "additional calling party number", if the numbering plan indicator is "ISDN Telephony" is removed if it is the network's own country code. The nature of address indicator shall be set to "national (significant) number". The address presentation restricted indicator shall be									
SIP parameter	INVITE: encapsulated IAM										
values	Generic number par		r								
	Address signals = Pl										
	Numbering plan indic										
	Nature of address inc		= '0000011'I	3							
	Screening indicator =										
10115	Presentation restricte	ed indic	cator = prese	ntation allow	red, '00'B						
ISUP parameter	IAM;										
values	Calling party number para Address signals = PIXIT1	meter									
	Numbering plan indicator = '	םי 1001									
	Nature of address indicator :		0011'R								
	Screening indicator = '11'B	_ 0000	DOTTE								
	Presentation restricted indic	ator =	presentation	allowed. '00'	'B						
	Generic number paramete			,							
	Address signals = PIXIT2										
	Numbering plan indicator = '										
	Nature of address indicator	= '0000	0011'B								
	Screening indicator = '00'B										
	Presentation restricted indic	ator =		allowed, '00'							
Comments	SIP-I		SUT		ISUP						
	IAM	<u>→</u>		<b>→</b>	INVITE(IAM)						
	ACM	<u>+</u>		<del>-</del>	180 Ringing(ACM)						
	ANM	+		· · ·	200 OK INVITE(ANM)						
	DEL		Conversa		DVE(DEL)						
	REL	<u>→</u>		<b>→</b>	BYE(REL)						
	RLC	+		←	200 OK BYE(RLC)						

TP401018	SIP reference: RFC 3261				ISUP reference: 3.5/Q.731			
TSS reference	ISUP-SIP-ISUP/SS/CLIP							
SIP selection criteria								
ISUP selection criteria	PICS 1/7							
Test purpose	Ensure that a prefix is add	Adding a prefix to an international calling party number 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)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	,	Conversation						
	BYE(REL)	BYE(REL) → REL						
	200 OK BYE(RLC)							
NOTE: The cod	ing "unknown" is a national o	option (@)		•	•			

TP401019	SIP reference: RI	C 326	1		IS	SUP reference:
						3.5/Q.731
TSS reference	ISUP-SIP-ISUP/SS/CLIP					
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Handling of address pres Ensure that the screening in presentation restricted indicavailable".(see note).	ndicato	r shall be se	t to "netwo	rk pr	
SIP parameter values						
ISUP parameter	IAM;					
values	Calling party number para	ameter				
	Address signals = PIXIT1					
	Numbering plan indicator =	' *'B				
	Nature of address indicator	= '*'B				
	Screening indicator = '11'B					
	Presentation restricted indic	cator =a	address not	available, '	10'B	
Comments	SIP-I		SUT	Γ		ISUP
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM
	180 Ringing(ACM)	+		1	<b>←</b>	ACM
	200 OK INVITE(ANM)	+		١.	<b>←</b>	ANM
	,		Convers	ation		
	BYE(REL)	<b>→</b>			<b>→</b>	REL
	200 OK BYE(RLC)	+		•	<b>←</b>	RLC
NOTE: The codi	ng "address not available" is	a natior	nal option (@	②).		

TP401020	SIP reference: RF	C 326	1		SUP reference: 912.5 clause 7.1.3		
TSS reference	ISUP-SIP-ISUP/SS/CLIP		•				
SIP selection criteria							
ISUP selection criteria							
Test purpose	Ensure that when the SUT I parameter and the Generic Sends an INVITE message header field" set to "anonyn	Numbe withou	er are not applicate t the "P-Asserted	ole Identity h			
SIP parameter values	INVITE: No P-Asserted Idea	ntity, Fr	om Header: anon	ymous@	anonymous.inv		
ISUP parameter values	IAM; no Calling party number	er and	no Additional calli	ng party	number present		
Comments	ISUP		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	ANM ← 200 OK INVITE(ANM)						
			Conversation				
	REL	<b>→</b>		<b>→</b>	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

TP401021	SIP reference: RF	C 326	1	-	SUP reference: 912.5 clause 7.1.3
TSS reference:	ISUP-SIP-ISUP/SS/CLIP				
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose  SIP parameter	parameter is <b>not applicable</b> presentation restriction para Address Indicator is set to N Sends an INVITE message field" where the user portion number and the country coc"+"CC+NCD+SN and no "Pi	e and the ameter NoAS_\ without of the deciracy I	he Generic Nu is set to "prese /ALUE t the "P-Asserte addr-spec is set to the country Header field"	mber is ap ntation allow ed-Identity het to value of where the	pereby Calling Party Number plicable whereby the address wed" and the Nature of neader field", a "From header of the additional calling party MGCF is located in the format adder contains the value of the
values	additional calling party num		71 Tivacy ficade	1, 1 10111 1100	duct contains the value of the
ISUP parameter	IAM; no Calling party number		ent, Additional	calling party	/ number present
values		•		· · ·	•
Comments	ISUP		SUT		SIP-I
	IAM	<b>→</b>		→	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
			Conversation	n	
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP401022	SIP reference: RF	C 326	1		ISUP reference: 1912.5 clause 7.1.3					
TSS reference	ISUP-SIP-ISUP/SS/CLIP									
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 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+NCD+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+NCD+SN  without "Privacy Header field" or "id" is not included									
SIP parameter	INVITE: P-Asserted-Identity				ımber, Privacy=id, From					
values	header derived from the add									
ISUP parameter values	IAM; Calling party number is	s prese	ent and no Add	litional callin	g party number is present					
Comments	ISUP		SUT		SIP-I					
	IAM	1		<b>→</b>	INVITE(IAM)					
	ACM	+		+	180 Ringing(ACM)					
	ANM	+		+	200 OK INVITE(ANM)					
			Conversati	on						
	REL	<b>→</b>		→	BYE(REL)					
	RLC	+		+	200 OK BYE(RLC)					

TP401023	SIP reference: RF	C 3261	I			SUP reference: 912.5 clause 7.1.3					
TSS reference	ISUP-SIP-ISUP/SS/CLIP				Q. I	912.3 Clause 7.1.3					
SIP selection	ISOF-SIF-ISOF/SS/CLIF										
criteria											
ISUP selection											
criteria Test purpose	E (1 ( ) ( ) ( ) ( ) ( )					Outility Dead Novel and					
	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+NCD+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+NCD+SN  and without "Privacy Header field" or "id" is not included										
SIP parameter	INVITE: P-Asserted-Identity										
values	From header derived from the	he addi	tional callin	g party nu	ımber						
ISUP parameter	IAM; Calling party number a	ınd Add	ditional calli	ng party n	umbe	r are present					
values			T								
Comments	ISUP		SU	Γ		SIP-I					
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)					
	ACM	+			+	180 Ringing(ACM)					
	ANM	<b>+</b>			+	200 OK INVITE(ANM)					
			Convers	ation	•						
	REL	<b>→</b>			<b>→</b>	BYE(REL)					
	RLC	+			+	200 OK BYE(RLC)					

Values for test purpose TP401024							
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					

TP401024	SIP reference: RF	C 326	1	=	SUP reference: 912.5 clause 7.1.3				
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection									
criteria									
ISUP selection	PICS 1/7								
criteria									
Test purpose	Ensure when no calling party number is included in the encapsulated IAM or the calling party number in the in the encapsulated IAM is not identical to the P-Asserted-Identity, 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 the fo	rmat "+"Co	C+NDC+SN, Privacy value				
values	"id" is not present				•				
ISUP parameter	IAM message with the Calli								
values	Address signals = nu			P-Asserte	ed-Identity				
	Screening indicator =								
	Number Incomplete								
	Numbering plan indic								
	Address Presentation			Presentat	ion allowed				
0	NoAS: "international	numbe			licup				
Comments	SIP-I		SUT		ISUP				
		INVITE(IAM) → IAM							
	180 Ringing(ACM) ← ACM								
	200 OK INVITE(ANM)	+	<u> </u>	<b>←</b>	ANM				
			Conversation						
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	<b>←</b>		+	RLC				

TP401025	SIP reference: RF	C 326	1	•	ISUP reference: 1912.5 clause 7.1.3				
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection criteria									
ISUP selection criteria	NOT PICS 1/7								
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, 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 the fo	rmat "+"C	C+NDC+SN, Privacy value				
values	"id" is not present				•				
ISUP parameter	IAM message with the Calli	ng par	ty number para	meter co	ded				
values	Address signals = nu Screening indicator = Number Incomplete I Numbering plan indic Address Presentation NoAS: "national (sign	= netwo Indicate cator = n Resti	ork provided or = PIXIT ISDN numbering icted Indicator =	g plan	·				
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>^</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
			Conversation	1					
	BYE(REL)	<b>^</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP401026	SIP reference: RF	C 326			SUP reference: 912.5 clause 7.1.3				
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection									
criteria									
ISUP selection	PICS 1/7								
criteria									
Test purpose	Ensure when no additional calling party number is included in the encapsulated IAM or the additional calling party number in the in the encapsulated IAM is not identical to the From header field, 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 the for	mat "+"Co	C+NDC+SN, Privacy value				
values	"id" is not present								
ISUP parameter	IAM message with the Addi								
values	Address signals = nu				der				
	Screening indicator =			fied"					
	Number Incomplete								
	Numbering plan indic								
	Address Presentation			resentat	ion allowed				
0	NoAS: "international	numbe		-	lioup				
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM) ←								
	200 OK INVITE(ANM)	+		<b>←</b>	ANM				
			Conversation						
	BYE(REL)	<b>→</b>		→	REL				
	200 OK BYE(RLC)	<b>←</b>		<b>←</b>	RLC				

TP401027	SIP reference: RF	C 326	1		SUP reference: 912.5 clause 7.1.3				
TSS reference:	ISUP-SIP-ISUP/SS/CLIP								
SIP selection									
criteria									
ISUP selection	NOT PICS 1/7								
criteria									
Test purpose	Ensure when no additional calling party number is included in the encapsulated IAM or the additional calling party number in the in the encapsulated IAM is not identical to the From header field, 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 the	format "+"C	C+NDC+SN, Privacy value				
values	"id" is not present								
ISUP parameter	IAM message with the Addi								
values	Address signals = nu				ader				
	Screening indicator =			erified"					
	Number Incomplete I								
	Numbering plan indic								
	Address Presentation NoAS: "national (sign			= Presenta	tion allowed				
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>↑</b>		<b>→</b>	IAM				
	180 Ringing(ACM) ← ACM								
	ANM								
			Conversation	on					
	BYE(REL)	<b>↑</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				

# 5.3.2 Calling Line Identification Restriction (CLIR)

TP402001	SIP reference: RF	C 3261	1		I	SUP reference:
					0.7	Q.1912.5
TSS reference:	ISUP-SIP-ISUP/SS/CLIR				Q.7	31 clause 4.5.2.1.1
SIP selection	150P-51P-150P/55/CLIR					
criteria						
ISUP selection criteria						
Test purpose						calling party number with
	the screening indicator set t			d" and th	e add	ress presentation restricted
	indicator set to "presentation	n restric	cted"			
SIP parameter						
values						
ISUP parameter	IAM;					
values	Calling party number para	meter				
	Screening indicator = '11'B					
	Address presentation restric	cted pai	rameter = '0	)1'B		
	Generic number paramete					
	Access transport paramet	er is n	ot including	the suba	ddres	s information
Comments	ISUP		SU	Γ		SIP-I
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)
	ACM	+			+	180 Ringing(ACM)
	ANM	+			+	200 OK INVITE(ANM)
			Convers	ation		
	REL	<b>→</b>			<b>→</b>	BYE(REL)
	RLC	+			+	200 OK BYE(RLC)

TP402002	SIP reference: RF	C 326	1		SUP reference: Q.1912.5
				Q.7	31 clause 4.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	screening indicator set to "n	ss tran etwork	sparently a call hav provided", the addr	ing a <b>c</b> ess pr	calling sub-address calling party number with the esentation restricted indicator ameter containing the calling
SIP parameter values					
ISUP parameter values	IAM; Calling party number para Screening indicator = '11'B Address presentation restric Generic number paramete Access transport paramete	ted pa	resent	format	ion
Comments	ISUP		SUT	T	SIP-I
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
			Conversation		
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	<b>←</b>		<b>←</b>	200 OK BYE(RLC)

TP402003	SIP reference: RF	C 326	1	I	SUP reference: Q.1912.5
				Q.7	Q.1912.5 31 clause 4.5.2.1.1
TSS reference:	ISUP-SIP-ISUP/SS/CLIR				
SIP selection					
criteria					
ISUP selection criteria					
Test purpose	Restricted calling party nu Ensure that the SUT can pa screening indicator set to "u presentation restricted indic	ss tran ser pro	sparently a call hav vided, verified and	ing the	calling party number with the I" and the address
SIP parameter			•		
values					
ISUP parameter	IAM				
values	Calling party number para	meter			
	Address signals = PIXIT1				
	Numbering plan indicator =	'001'B			
	Nature of address indicator	= '0000'	)011'B		
	Screening indicator = '01'B				
	Address presentation restrict	ted pa		•	
Comments	ISUP		SUT		SIP-I
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		<b>←</b>	200 OK INVITE(ANM)
			Conversation		
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	<b>←</b>		<b>←</b>	200 OK BYE(RLC)

TP402004	SIP reference: RF	C 326	1	].	SUP reference: Q.1912.5			
				Q.73	31 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	meter						
	Address signals = PIXIT1							
	Numbering plan indicator =							
	Nature of address indicator	= '0000	)011'B					
	Screening indicator = '01'B							
	Address presentation restric							
0	Access transport paramet	er inci		ntormat				
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		<del>-</del>	180 Ringing(ACM)			
	ANM	+	0	7	200 OK INVITE(ANM)			
	DEL		Conversation		DVE(DEL)			
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	<b>←</b>		<b>←</b>	200 OK BYE(RLC)			

TP402005	SIP reference: RF	C 326	I	I	SUP reference:			
				Q.7	Q.1912.5 31 clause 4.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/CLIR		l .					
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Restricted calling party number (user provided, not verified)  Ensure that the SUT can pass transparently a call having a default calling party number with the screening indicator set to "network provided" and a generic number containing the additional calling party number with the screening indicator set to "user provided, not verified", both having the address presentation restricted indicator set to "presentation restricted"							
SIP parameter								
values								
ISUP parameter	IAM;							
values	Calling party number parameter  Address signals = PIXIT1  Numbering plan indicator = '001'B  Nature of address indicator = '0000011'B  Screening indicator = '11'B  Address presentation restricted parameter = '01'B  Generic number parameter  Address signals = PIXIT2  Numbering plan indicator = '001'B  Nature of address indicator = '0000011'B  Screening indicator = '00'B  Address presentation restricted parameter = '01'B							
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+	0	<b>←</b>	200 OK INVITE(ANM)			
	DEL		Conversation		DVE(DEL)			
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		<b>←</b>	200 OK BYE(RLC)			

SUP-SIP-ISUP/SS/CLIR   ISUP-SIP-ISUP/SS/CLIR   ISUP-SIP-ISUP/SS/CLIR	TP402006	SIP reference: RF	C 326	I	Į	SUP reference: Q.1912.5			
TSS reference SIP selection criteria ISUP selection criteria  Restricted calling party number (user provided, not verified) with calling sub-address Ensure that the SUT can pass transparently a call having a default calling party 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  ISUP parameter values  IAM; Calling party number parameter Address signals = PIXIT1 Numbering plan indicator = '0000011'B Screening indicator = '11'B Address presentation restricted parameter = '01'B Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '0000011'B Screening indicator = '0000011'B Screening indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '001'B Address presentation restricted parameter = '01'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments  ISUP SUT SIP-I IAM → NINVITE(IAM) ACM ← 180 Ringing(ACM) ANM ← NINVITE(IAMM)  Conversation  REL → BYE(REL)					Q.7				
Criteria   ISUP selection criteria   Suppose   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	TSS reference	ISUP-SIP-ISUP/SS/CLIR		L					
Suparameter values   IAM;   Calling party number parameter   Address   Pixit   Numbering plan indicator = '001'B   Nature of address   pixit   Numbering plan indicator = '01'B   Nature of address   Pixit   Numbering plan indicator = '000'B   Nature of address indicator = '000'B   Nature of address presentation restricted parameter   Nature of address presentation restricted parameter   Signals = Pixit   Numbering plan indicator = '000'B   Nature of address indicator = '000'B   Nature of address indicator = '001'B   Nature of address indicator = '0000011'B   Screening indicator = '000'B   Nature of address indicator = '000B   Address presentation restricted parameter = '01'B   Nature of address parameter   Na	SIP selection								
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  IAM; Calling party number parameter Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '00000011'B Screening indicator = '11'B Address presentation restricted parameter = '01'B Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '001'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments  ISUP  SUT  IAM  ACM  ACM  CONVERSATION  SIP-I  IAM  PINVITE(IAM)  ACM  ANM  CONVERSATION  CONVERSATION  BYE(REL)	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  IAM; Calling party number parameter Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '11'B Address presentation restricted parameter = '01'B Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '000'B Nature of address indicator = '000'B Nature of address indicator = '000'B Address presentation restricted parameter = '01'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments  ISUP SUT IAM ACM ACM ACM ACM ACM ANM BCM ACM BCM ANM BCM ACM ANM BCM ACM BCM ANM BCM ACM BCM BCM BCM BCM BCM BCM BCM BCM BCM B	ISUP selection								
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  IAM;  Calling party number parameter Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '101'B Address presentation restricted parameter = '01'B Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments  ISUP  SUT  SIP-I  IAM  ACM  FINVITE(IAM)  ACM  FINVITE(IAM)  Conversation  REL  BYE(REL)									
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  IAMI;  Calling party number parameter Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '11'B Address presentation restricted parameter = '01'B Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '000'B Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments  ISUP SUT ISIP-I IAM ACM - INVITE(IAM) ACM - 180 Ringing(ACM) ANM - 200 OK INVITE(ANM)  Conversation  REL - BYE(REL)	Test purpose		ımber	(user provided, r	not verifi	ed) with calling			
Values   ISUP parameter   IAM;   Calling party number parameter   Address signals = PIXIT1   Numbering plan indicator = '001'B   Nature of address indicator = '00000011'B   Screening indicator = '11'B   Address presentation restricted parameter = '01'B   Generic number parameter   Address signals = PIXIT2   Numbering plan indicator = '001'B   Nature of address indicator = '001'B   Nature of address indicator = '00'B   Address presentation restricted parameter = '01'B   Access transport parameter including subaddress information		with the screening indicator set to "network provided", a <b>generic number</b> 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"							
ISUP parameter values  IAM; Calling party number parameter Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '11'B Address presentation restricted parameter = '01'B Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments  ISUP SUT IAM → INVITE(IAM) ACM ← 180 Ringing(ACM) ANM ← 200 OK INVITE(ANM)  Conversation  REL → BYE(REL)	SIP parameter								
Calling party number parameter         Address signals = PIXIT1         Numbering plan indicator = '001'B         Nature of address indicator = '0000011'B         Screening indicator = '11'B         Address presentation restricted parameter = '01'B         Generic number parameter         Address signals = PIXIT2         Numbering plan indicator = '001'B         Nature of address indicator = '0000011'B         Screening indicator = '00'B         Address presentation restricted parameter = '01'B         Access transport parameter including subaddress information         Comments         ISUP       SUT       SIP-I         IAM       →       →       INVITE(IAM)         ACM       ←       (4.180 Ringing(ACM)         ANM       ←       (4.180 Ringing(ACM)         ANM       ←       (5.00 OK INVITE(ANM)         Conversation       →       BYE(REL)									
Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '11'B Address presentation restricted parameter = '01'B Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments  ISUP SUT SIP-I IAM ACM CM ANM CM ANM CONVERSATION BYE(REL)  BYE(REL)		IAM;							
Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '11'B Address presentation restricted parameter = '01'B Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments  ISUP SUT IAM SIP-I IAM ACM CM CONVERSATION ANM CONVERSATION BYE(REL)  SYPE(REL)	values		meter						
Nature of address indicator = '0000011'B Screening indicator = '11'B Address presentation restricted parameter = '01'B Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments  ISUP SUT IAM ACM CM CONVERSATION ANM CONVERSATION ANM CONVERSATION BYE(REL)  SIP-I  IAN SIP									
Screening indicator = '11'B Address presentation restricted parameter = '01'B Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '00000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments  ISUP SUT IAM  SIP-I IAM  ACM  ACM  ANM  Conversation  REL  BYE(REL)		Ŭ .							
Address presentation restricted parameter = '01'B  Generic number parameter  Address signals = PIXIT2  Numbering plan indicator = '001'B  Nature of address indicator = '0000011'B  Screening indicator = '00'B  Address presentation restricted parameter = '01'B  Access transport parameter including subaddress information  Comments  ISUP  IAM  SUT  SIP-I  IAM  ACM  COM  ANM  CONVERSATION  SIP-I  IAM  ACM  ANM  CONVERSATION  BYE(REL)			= '0000	)011'B					
Generic number parameter  Address signals = PIXIT2  Numbering plan indicator = '001'B  Nature of address indicator = '0000011'B  Screening indicator = '00'B  Address presentation restricted parameter = '01'B  Access transport parameter including subaddress information  Comments  ISUP  IAM  → SUT  INVITE(IAM)  ACM  ← 180 Ringing(ACM)  ANM  ← 200 OK INVITE(ANM)  Conversation  REL  → BYE(REL)									
Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments    SUP				rameter = '01'B					
Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments    SUP			r						
Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments    SUP									
Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments    SUP				204415					
Address presentation restricted parameter = '01'B           Access transport parameter including subaddress information           Comments         ISUP         SUT         SIP-I           IAM         →         INVITE(IAM)           ACM         ←         180 Ringing(ACM)           ANM         ←         ←         200 OK INVITE(ANM)           Conversation         REL         →         BYE(REL)			= 0000	DOTTE					
Access transport parameter including subaddress information           Comments         ISUP         SUT         SIP-I           IAM         →         INVITE(IAM)           ACM         ←         180 Ringing(ACM)           ANM         ←         ←         200 OK INVITE(ANM)           Conversation         REL         →         BYE(REL)			+ad aa	romotor '01'D					
ISUP         SUT         SIP-I           IAM         →         INVITE(IAM)           ACM         ←         ←         180 Ringing(ACM)           ANM         ←         ←         200 OK INVITE(ANM)           Conversation         REL         →         BYE(REL)					informati	ion			
IAM         →         INVITE(IAM)           ACM         ←         180 Ringing(ACM)           ANM         ←         200 OK INVITE(ANM)           Conversation           REL         →         BYE(REL)	Comments		CI IIICI		Intomati				
ACM         ←         180 Ringing(ACM)           ANM         ←         200 OK INVITE(ANM)           Conversation           REL         →         BYE(REL)	Comments		-	301	-	-			
ANM ← 200 OK INVITE(ANM)  Conversation  REL → BYE(REL)						\ /			
Conversation  REL → BYE(REL)									
REL → BYE(REL)		, 11 1111		Conversation		200 0.( 1147112( / 11410)			
		REL	<b>→</b>	23	→	BYE(REL)			
I IRIC   ←     ←  200 OK BYF(RIC)		RLC	<del>-</del>		<del>-</del>	200 OK BYE(RLC)			

TP402007	SIP reference: I	RFC 3261			ISUP reference: Q.1912.5 '31 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 numl	per if the prese	ntation i	s restricted		
	Ensure that the <b>calling pa</b> address presentation rest				of bilateral agreements, if the on restricted"		
SIP parameter	•						
values							
ISUP parameter	IAM;						
values	No Calling party numbe	r paramet	er				
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	Conversation						
	BYE(REL)	→		<b>→</b>	REL		
		+		+			

TP402008	SIP reference: R	FC 326	1	ISUP reference: Q.1912.5				
				Q.7	31 clause 4.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/CLIR							
SIP selection								
criteria								
ISUP selection	PICS 6/4 AND PICS 6/5							
criteria								
Test purpose	Discarding the additiona							
					number is discarded in case			
	of bilateral agreements, if t	the addr	ess presentat	ion restricted	l 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		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		→	IAM			
	180 Ringing(ACM)	<b>←</b>		+	ACM			
	200 OK INVITE(ANM)	<del>-</del>		<b>←</b>	ANM			
			Conversat	ion				
	BYE(REL)	<b>→</b>		→	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP402009	SIP reference: F	RFC 326	1	_	SUP reference: Q.1912.5 31 clause 4.5.2.1.1
TSS reference:	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the calling pain the ISUP IAM	rty numb	er contained in the	e encaps	ulated IAM is unchanged sent
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Conversation		
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	+		+	RLC

TP402010	SIP reference: RFC 3261			ISUP reference:				
						Q.1912.5		
				(	Q.7:	31 clause 4.5.2.1.1		
TSS reference:	ISUP-SIP-ISUP/SS/CLIR							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the additional		arty numbe	r contained	in t	he encapsulated IAM is		
	unchanged sent in the ISI	JP IAM						
SIP parameter								
values								
ISUP parameter								
values								
Comments	SIP-I		SU	Γ		ISUP		
	INVITE(IAM)	<b>→</b>		-	<b>→</b>	IAM		
	180 Ringing(ACM)	+		•	<del>(</del>	ACM		
	200 OK INVITE(ANM)	+		•	<del>(</del>	ANM		
	,		Convers	ation				
	BYE(REL)	<b>→</b>		-	<del>}</del>	REL		
	200 OK BYE(RLC)	+		•	<del>(</del>	RLC		

TP402011	SIP reference: RF	C 3261	1		-	SUP reference: 912.5 clause 7.1.3			
TSS reference	ISUP-SIP-ISUP/SS/CLIR								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation restricted and the Generic Number is 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+NCD+SN  a "From header field" where the user portion of the addr-spec is set to value of the additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NCD+SN  and with "Privacy Header field" set to "id"								
SIP parameter values	INVITE: P-Asserted-Identity	, From	header field	d, Privacy	"id"				
ISUP parameter values	IAM: Calling party number. I	No add	litional callin	ng party nu	umbei	r			
Comments	ISUP		SU	Γ		SIP-I			
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)			
	ACM	+			+	180 Ringing(ACM)			
	ANM	+			+	200 OK INVITE(ANM)			
			Convers	ation					
	REL	<b>→</b>			<b>→</b>	BYE(REL)			
	RLC	<b>+</b>			+	200 OK BYE(RLC)			

TP402012	SIP reference: RF	C 326′	I	-	SUP reference: 912.5 clause 7.1.3				
TSS reference	ISUP-SIP-ISUP/SS/CLIR								
SIP selection criteria									
ISUP selection criteria									
Test purpose	Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation restricted and the Generic Number is applicable  Sends an INVITE message with the  "P-Asserted-Identity header field", where the user portion of the addr-spec is set to value of the calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NCD+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+NCD+SN  and with "Privacy Header field" is set to "id"								
SIP parameter values	INVITE: P-Asserted-Identity,			ivacy "id"					
ISUP parameter values	IAM: Calling party number. a	dditior	nal calling party	number					
Comments	ISUP		SUT		SIP-I				
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
			Conversatio	n					
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

	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: RF	C 326	1		-	SUP reference:					
					Q.1	912.5 clause 7.1.3					
TSS reference	ISUP-SIP-ISUP/SS/CLIR										
SIP selection											
criteria											
ISUP selection	PICS 1/7										
criteria											
Test purpose	Ensure when no calling party number is included in the encapsulated IAM or the calling party number in the in the encapsulated IAM is not identical to the P-Asserted-Identity, Privacy value "id" received  Send an IAM the calling party number is derived from SIP P-Asserted-Identity. The Address Presentation Restricted Indicator is set to Presentation restricted										
SIP parameter	INVITE: P-Asserted identity	user p	ortion is in t	he forma	t "+"C	C+NDC+SN, Privacy value					
values	"id" is present	•				•					
ISUP parameter	IAM message with the Calli	ng par	ty number	paramet	er cod	ded					
values	Address signals = nu Screening indicator = Number Incomplete Numbering plan indic Address Presentation NoAS: "international	= netwo Indicato cator = n Restr	ork provided or = PIXIT ISDN numb ricted Indica er"	l pering pla ator = Pre	n	tion restricted					
Comments	SIP-I		SU	Γ		ISUP					
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM					
	180 Ringing(ACM)	+			+	ACM					
	200 OK INVITE(ANM)	+			+	ANM					
			Convers	ation							
	BYE(REL)	<b>→</b>			<b>→</b>	REL					
	200 OK BYE(RLC)	+			+	RLC					

TP402014	SIP reference: RF	C 326	1		-	SUP reference: 912.5 clause 7.1.3				
TSS reference:	ISUP-SIP-ISUP/SS/CLIR									
SIP selection										
criteria										
ISUP selection	NOT PICS 1/7									
criteria										
Test purpose	Ensure when no calling party number is included in the encapsulated IAM or the calling party number in the in the encapsulated IAM is not identical to the P-Asserted-Identity, Privacy value "id" received  Send an IAM the calling party number is derived from SIP P-Asserted-Identity. The Address Presentation Restricted Indicator is set to Presentation restricted									
SIP parameter	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value									
values	"id" is present					•				
ISUP parameter	IAM message with the Calli									
values	Address signals = nu Screening indicator = Number Incomplete I Numbering plan indic Address Presentation NoAS: "national (sign	= netwo Indicato cator = n Restr	ork provided or = PIXIT ISDN numb ricted Indicat	ering pla	n	·				
Comments	SIP-I	illicarit	SUT			ISUP				
Comments	INVITE(IAM)	<b>→</b>	301		<b>→</b>	IAM				
	180 Ringing(ACM)	+			<del>′</del>	ACM				
	200 OK INVITE(ANM)	+			+	ANM				
		•	Conversa	ation						
	BYE(REL)	<b>→</b>		-	<b>→</b>	REL				
	200 OK BYE(RLC)	+			+	RLC				

TP402015	SIP reference: RF	C 3261			I	SUP reference:			
					Q.1	912.5 clause 7.1.3			
TSS reference	ISUP-SIP-ISUP/SS/CLIR								
SIP selection									
criteria									
ISUP selection	PICS 1/7								
criteria									
Test purpose	Ensure when no additional calling party number is included in the encapsulated IAM or the additional calling party number in the in the encapsulated IAM is not identical to the From header field, Privacy value "id" received Send an IAM the additional calling party number is derived from From header field. The Address Presentation Restricted Indicator is set to Presentation restricted								
SIP parameter	INVITE: P-Asserted identity	user po	rtion is in t	he format	"+"C	C+NDC+SN, Privacy value			
values	"id" is present	•				•			
ISUP parameter	IAM message with the Add	itional C	Calling par	ty numbe	r pa	rameter coded			
values	Address signals = nu	ımber de	erived from	SIP Fron	n hea	der			
	Screening indicator =			ot verified'	'				
	Number Incomplete								
	Numbering plan indic								
	Address Presentation	n Restri	cted Indica	tor = Pres	entat	tion restricted			
	NoAS: "international	number							
Comments	SIP-I		SUT			ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	180 Ringing(ACM)	<b>←</b>			+	ACM			
	200 OK INVITE(ANM)	<b>←</b>			+	ANM			
			Convers	ation					
	BYE(REL)	<b>→</b>			<b>→</b>	REL			
	200 OK BYE(RLC)	+			+	RLC			

TP402016	SIP reference: RF	C 326	1		-	SUP reference: 912.5 clause 7.1.3					
TSS reference	ISUP-SIP-ISUP/SS/CLIR										
SIP selection											
criteria											
ISUP selection	NOT PICS 1/7										
criteria											
Test purpose	Ensure when no additional calling party number is included in the encapsulated IAM or the additional calling party number in the in the encapsulated IAM is not identical to the From header field, Privacy value "id" received  Send an IAM the additional calling party number is derived from From header field. The Address Presentation Restricted Indicator is set to Presentation restricted										
SIP parameter	INVITE: P-Asserted identity	user p	ortion is in the	e format	"+"C	C+NDC+SN, Privacy value					
values	"id" is present	-				•					
ISUP parameter	IAM message with the Addi	tional	Calling party	/ numbe	r pa	rameter coded					
values	Address signals = nu Screening indicator =	User	provided, not			ader					
	Number Incomplete I										
	Numbering plan indic										
	Address Presentation NoAS: "national (sign			or = Pres	enta	tion restricted					
Comments	SIP-I		SUT			ISUP					
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM					
	180 Ringing(ACM)	+			+	ACM					
	200 OK INVITE(ANM)	+			+	ANM					
			Conversat	ion							
	BYE(REL)	<b>→</b>			<b>→</b>	REL					
	200 OK BYE(RLC)	+			+	RLC					

# 5.3.3 Connected line identification presentation (COLP)

TP403001	SIP reference: RF	C 326	1		-	SUP reference: Q.1912.5 31 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 c the optional forward call in			fully a cal	l reque	esting the COLP service in
SIP parameter values						
ISUP parameter	IAM;					
values	optional forward call indic	ators	Connected	line ident	ity req	uest indicator = requested
Comments	SIP-I		SUT			ISUP
	INVITE(IAM)	<b>→</b>			1	IAM
	180 Ringing(ACM)	+			4	ACM
	200 OK INVITE(ANM)	+			+	ANM
			Convers	ation		
	BYE(REL)	<b>→</b>			1	REL
	200 OK BYE(RLC)	+			+	RLC

TP403002	SIP reference: RFC	326′	1		ISUP reference: Q.1912.5				
				Q.	731 clause 5.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/COLP								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Connected number (user pr								
		twork creen	provided", a ing indicator	a <b>generic nu</b> set to "user	<b>mber</b> containing the additional provided, not verified" and an				
SIP parameter									
values									
ISUP parameter	IAM;								
values	optional forward call indical Connected line identity reques a)  ANM;  Connected number parameted Address presentation restricted Nature of address indicator = '0 Screening indicator = '11'B Address signals = PIXIT  Additional connected number Address presentation restricted Nature of address indicator = '0 Screening indicator = '10'B Address signals = PIXIT and an access transport par b)	st ind ter ed pa '0000 01'B eer pr ed pa '0000 01'B	rameter = '0 0011'B esent rameter = '0 0011'B	0'B 0'B	ted sub-address.				
	CON; Connected number paramer Address presentation restricte Nature of address indicator = '0 Screening indicator = '11'B Address signals = PIXIT Additional connected numb Address presentation restricte Nature of address indicator = '0 Screening indicator = '0'Screening indicator = '0'Screening indicator = '0'B Address signals = PIXIT and an access transport par	ed pa '0000 01'B <b>er</b> pr ed pa '0000 01'B	0011'B esent rameter = '0 0011'B	0'B	ted sub-address				
Comments	SIP-I		SUT		ISUP				
		<b>→</b>		<b>→</b>	IAM				
	CASE À		•	•					
		<del>(</del>		+	ACM				
		<del>`</del>		+	ANM				
	CASE B		I	1	P ** ****				
		<del>(</del>	1	+	CON				
	200 01(1141112(0014)		Conversa						
	BYE(REL)	<b>→</b>	COLIVEIS	- <del>- →</del>	REL				
		<u> </u>		+					
	200 OK BYE(RLC)		I		RLC				

TP403003	SIP reference: RFC	3261		ISUP reference: Q.1912.5
TSS reference	ISUP-SIP-ISUP/SS/COLP			Q.731 clause 5.5.2.1.1
SIP selection	130F-31F-130F/33/COLF			
criteria				
ISUP selection				
criteria				
Test purpose		ransparently a work provided creening indica	default <b>conne</b> ", a <b>generic n</b> e ator set to "use	ected number with the umber containing the additional r provided, not verified" without
SIP parameter values		<u> 3</u>		
ISUP parameter	IAM;			
values	optional forward call indicat	ors		
	Connected line identity reques		auested	
	a)	ot maioator. Te	quodicu	
	ÄNM;			
	Connected number paramet			
	Address presentation restricte		= '00'B	
	Nature of address indicator = Numbering plan indicator = '00'			
	Screening indicator = '11'B	υιь		
	Address signals = PIXIT			
	Additional connected numb	<b>er</b> present		
	Address presentation restricte		= '00'B	
	Nature of address indicator =			
	Numbering plan indicator = '00' Screening indicator = '00'B	01'B		
	Address signals = PIXIT			
	b)			
	CON;			
	Connected number paramet	ter		
	Address presentation restricte		= '00'B	
	Nature of address indicator =			
	Numbering plan indicator = '00	01'B		
	Screening indicator = '11'B Address signals = PIXIT			
	Additional connected numb	<b>er</b> nresent		
	Address presentation restricte		= '00'B	
	Nature of address indicator =	'0000011'B		
	Numbering plan indicator = '00	01'B		
	Screening indicator = '00'B			
	Address signals = PIXIT			
Comments	SIP-I		UT	ISUP
		<b>→</b>	-3	IAM
	CASE A	_	1	
	5 5 7	<del>(</del>	•	_
	200 OK INVITE(ANM)  CASE B	<b>←</b>	•	ANM
		<del>(</del>	•	CON
	ZOU OK INVITE(CON)		rsation	CON
	BYE(REL)	→ Conve		REL
		<del>´</del>	•	I .

TP403004	SIP reference: RF	C 326			I	SUP reference:					
						Q.1912.5					
					Q.7	31 clause 5.5.2.1.1					
TSS reference	ISUP-SIP-ISUP/SS/COLP										
SIP selection											
criteria											
ISUP selection	PICS 1/7										
criteria	Converting the connected number to national format if necessary										
Test purpose	Converting the connected number to national format, if necessary  Ensure that the country code in the address signals of the connected number is remo										
	if it is the network's own cou										
	"national (significant) number										
	screening indicator shall be					noted indicator and the					
SIP parameter	200 OK: encapsulated ANM			a. 0y							
values	Connected number										
	Address presentation			eter = '00'	'В						
	Nature of address inc										
	Numbering plan indic	cator =	'001'B								
	Screening indicator =		_SI								
	Address signals = PI	XIT									
ISUP parameter	IAM;										
values	optional forward call indic	ators									
	Connected line identity requ	est ind	icator: requ	ested							
	a)		•								
	ANM;										
	Connected number param										
	Address presentation restrict			00'B							
	Nature of address indicator		)100'B								
	Numbering plan indicator =										
	Screening indicator = ISUP_										
	Address signals = CC+PIXI	I									
	b) CON;										
	Connected number param	eter									
	Address presentation restrict		rameter = '0	00'B							
	Nature of address indicator			-							
	Numbering plan indicator =	'001'B									
	Screening indicator = ISUP_	_SI									
	Address signals = CC+PIXI										
_	Generic number paramete	r not p			1						
Comments	SIP-I		SUT		<u> </u>	ISUP					
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM					
	CASE A	-			-	14014					
	180 Ringing(ACM)	+			<del>(</del>	ACM					
	200 OK INVITE(ANM)	+			<b>←</b>	ANM					
	CASE B	-			· /	loon					
	200 OK INVITE(CON)	+	Comme	-4i	<b>←</b>	CON					
	DVE(DEL)		Convers	ation		DEI					
	BYE(REL)	<b>→</b>			<b>→</b>	REL					
	200 OK BYE(RLC)	+			<b>←</b>	RLC					

TP403005	SIP reference: RFC 3	261			I	SUP reference:		
					0.7	Q.1912.5 31 clause 5.5.2.1.1		
TSS reference	ISUP-SIP-ISUP/SS/COLP				Ψ.,	01 Clau3C 0.0.2.1.1		
SIP selection	1001 011 1001 700/0021							
criteria								
ISUP selection	PICS 1/7							
criteria								
Test purpose	Converting the additional con							
	Ensure that the country code in							
	"additional connected number",							
	removed if it is the network's ow set to "national (significant) num							
	screening indicator shall be tran				illalic	in restricted indicator and the		
SIP parameter	200 OK: encapsulated ANM or 0			archiny				
values	additional connected number							
	Address presentation restricted		ameter = '0	0'B				
	Nature of address indicator = '00							
	Numbering plan indicator = '001	'Β						
	Screening indicator = '01'B							
	Address signals = PIXIT							
ISUP parameter	IAM;							
values	optional forward call indicator	rs						
	Connected line identity request	indic	cator: requ	ested				
	a)							
	ANM;							
	Connected number parameter additional connected number		sent					
	Address presentation restricted		motor - 'O	Λ'R				
	Nature of address indicator = '00			OD				
	Numbering plan indicator = '001		.002					
	Screening indicator = '01'B							
	Address signals = CC+PIXIT							
	b)							
	CON; Connected number parameter	r nro	sont					
	additional connected number		:SCIIL					
	Address presentation restricted		ameter = '0	0'B				
	Nature of address indicator = '00							
	Numbering plan indicator = '001							
	Screening indicator = '01'B							
	Address signals = CC+PIXIT				1	I		
Comments	SIP-I		SUT	-		ISUP		
	INVITE(IAM) →	,			<b>→</b>	IAM		
	CASE A				4	ACM		
	180 Ringing(ACM) ← 200 OK INVITE(ANM) ←				+	ACM ANM		
	CASE B							
	200 OK INVITE(CON) ←	. T			+	CON		
	200 010 1100112 (0010)	<u> </u>	Conversa	ation	•			
	BYE(REL) →	,	30.110.0		<b>→</b>	REL		
	200 OK BYE(RLC) ←				<del>-</del>	RLC		
	ZUU UK BYE(KLC)				_	KLU		

TP403006	SIP reference: RFC	3261		ISUP reference: Q.1912.5 31 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 interna							
	Ensure that a prefix is added to			the nature of address				
	indicator is set to "u							
SIP parameter	200 OK INVITE with encapsul		ON					
values	Connected number p							
	Address presentation r							
	Nature of address indic		1'B					
	Numbering plan indica							
	Screening indicator = '							
	Address signals = PIXI	T						
ISUP parameter	ANM/CON:							
values	Connected number paramet							
	Address presentation restricte		00'B					
	Nature of address indicator =							
	Numbering plan indicator = '00	)1'B						
	Screening indicator = '11'B	<del>-</del>						
0	Address signals = Prefix+PIXI		_	loup				
Comments	SIP-I	SU		ISUP				
		<b>→</b>	→	INVITE(IAM)				
	CASE A	<del>-</del> 1		1,000				
		<del>-</del>	<b>+</b>	180 Ringing(ACM)				
	7 (1 414)	<del>(</del>	←	200 OK INVITE(ANM)				
	CASE B	1	r					
	CON	<b>←</b>	<b>+</b>	200 OK INVITE(CON)				
		Convers						
	REL → BYE(REL)							
	ILLO	<del>-</del>	<b>+</b>	200 OK BYE(RLC)				
NOTE: The cod	ing "unknown" is a national optic	n (@)						

TP403007	SIP reference: RFC	3261			SUP reference: Q.1912.5 31 clause 5.5.2.1.1					
TSS reference	ISUP-SIP-ISUP/SS/COLP		-							
SIP selection criteria										
ISUP selection criteria	PICS 1/8 AND PICS 7/3									
Test purpose	Discarding the connected in Ensure that the connected in address presentation restricts	<mark>umber</mark> is d	iscarded in ca	ase of b	ilateral agreements, if the					
SIP parameter values	address presentation restricted indicator is set to "presentation allowed" (see note).  200 OK INVITE with encapsulated ANM or CON  Connected number parameter  Address presentation restricted parameter = '00'B  Nature of address indicator = '0000011'B  Numbering plan indicator = '001'B  Screening indicator = '11'B  Address signals = PIXIT									
ISUP parameter values	IAM optional forward call indica Connected line identity reque a) ANM No Connected number para b) CON; No Connected number para	st indicator	: requested							
Comments	ISUP		SUT		SIP-I					
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)					
	CASE A									
	ACM	+		+	180 Ringing(ACM)					
	ANM	<del>(</del>		+	200 OK INVITE(ANM)					
	CASE B									
	CON	+		+	200 OK INVITE(CON)					
		Co	nversation		, ,					
	REL	<b>→</b>		<b>→</b>	BYE(REL)					
	RLC	+		+	200 OK BYE(RLC)					
	teral agreement prohibits the tracess presentation restricted indic				er in any case. The test with					

TP403008	SIP reference: RFC	326	I			SUP reference: Q.1912.5 31 clause 5.5.2.1.1				
TSS reference:	ISUP-SIP-ISUP/SS/COLP									
SIP selection										
criteria										
ISUP selection	PICS 1/8 AND PICS 7/4									
criteria										
Test purpose	Discarding the additional con Ensure that the additional con of bilateral agreements, if the	nect	ed number i	n the <b>gen</b> e	eric r	number is discarded in case				
	"presentation allowed" see (no	ote).	•							
SIP parameter	200 OK INVITE with encapsul									
values	Additional Connected									
	Address presentation r				3					
	Nature of address indic			'В						
	Numbering plan indicate		.001.R							
	Screening indicator = '0 Address signals = PIXI									
ISUP parameter		<u> </u>								
values	IAM;									
Value	optional forward call indicators									
	Connected line identity reques	st ind	icator: requ	ested						
	a)									
	ANM;									
	No Connected number param No Additional connected numl b) CON; No Connected number param No Additional connected numl	ber p eter								
Comments	ISUP		SUT	T		SIP-I				
		<b>→</b>			<b>→</b>	INVITE(IAM)				
	CASE A			•						
	ACM	<del>(</del>			+	180 Ringing(ACM)				
	ANM	<del>(</del>			<b>←</b>	200 OK INVITE(ANM)				
	CASE B									
	CON	<del>(</del>			+	200 OK INVITE(CON)				
			Convers	ation						
		<b>→</b>			<b>→</b>	BYE(REL)				
	ILLO	<del>(</del>			+	200 OK BYE(RLC)				
	teral agreement prohibits the tra n any case.	nsfei	ral of the ad	dditional c	onne	cted number in the generic				
TIGHT DOT 1	, 0000.									

TP403009	SIP reference: RF	C 3261	I		ı	SUP reference:				
						Q.1912.5				
<del></del>					Q.7	31 clause 5.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/COLP									
SIP selection										
criteria	PICS 1/8									
ISUP selection criteria	PICS 1/8									
Test purpose	Converting the connected	numh	or to intorn	ational f	orma	•				
rest purpose	Ensure that the exchange ca									
						nal number" and can pass on				
	the address presentation res									
SIP parameter	200 OK INVITE with encaps					ig marcator transparently				
values	Connected number									
	Address presentation			ter = '00'	В					
	Nature of address ind									
	Numbering plan indic	ator =	'001'B							
	Screening indicator =									
	Address signals = CC	+PIXI	Т							
ISUP parameter	IAM;									
values	optional forward call indica									
	Connected line identity reque	est ind	icator: requ	ested						
	a) <b>ANM</b>									
	Connected number parame	otor								
	Address presentation restrict		rameter – '0	n'B						
	Nature of address indicator =	•		OB						
	Numbering plan indicator = '0		7100 B							
	Screening indicator = '11'B									
	Address signals = PIXIT									
	Presentation restricted indica	ator = '	00'B							
	additional connected numl	<b>ber</b> pre	esent							
	b)									
	CON;									
	Connected number parame									
	Address presentation restrict			0'B						
	Nature of address indicator =		)100 <sup>-</sup> B							
	Numbering plan indicator = '0 Screening indicator = '11'B	UUID								
	Address signals = PIXIT									
	Presentation restricted indica	ator = '	00'B							
	additional connected numl									
Comments	SIP-I	P'\	SUT	•		ISUP				
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM				
	CASE A		1		1	1				
	180 Ringing(ACM)	+			+	ACM				
	200 OK INVITE(ANM)	+			<b>←</b>	ANM				
	CASE B									
	200 OK INVITE(CON)	+			<b>←</b>	CON				
			Convers	ation						
	BYE(REL)	<b>→</b>			<b>→</b>	REL				
	200 OK BYE(RLC)	+			+	RLC				

TP403010	SIP reference: I	RFC 326	ISUP reference:									
				0 -	Q.1912.5							
TCC reference		,		Q.	731 clause 5.5.2.1.1							
TSS reference SIP selection	ISUP-SIP-ISUP/SS/COLF	1001 -011 -1001 /00/00LF										
criteria												
ISUP selection												
criteria												
Test purpose	Handling unrequested C	.OI										
rest purpose			sfully set up	if the SUT red	ceives an unsolicited COL							
SIP parameter	200 OK INVITE with enca	psulated	ANM or CO	N								
values	Connected numb											
	Address presentat											
	Nature of address	indicator	= '0000011'	В								
	Numbering plan in		'001'B									
	Screening indicato											
	Address signals =	PIXIT										
ISUP parameter	IAM;											
values	optional forward call inc											
	Connected line identity request indicator: not requested											
	a)											
	ANM;											
	Connected number parameter											
	Address presentation rest			0.B								
	Nature of address indicate		0011B									
	Numbering plan indicator											
	Screening indicator = '11'	Б										
	Address signals = PIXIT additional connected nu	ımbar nr	ocont									
		iiiibei pie	eseni									
	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	Address signals - PIXIT										
	additional connected nu	ı <b>mber</b> pre	esent									
Comments	SIP-I		SUT		ISUP							
	INVITE(IAM)	<b>→</b>		→	IAM							
	CASE A		1		•							
	180 Ringing(ACM)	<b>←</b>		+	ACM							
	200 OK INVITE(ANM)	+		+	ANM							
	CASE B	1		<u>i</u> _	<b>'</b>							
	200 OK INVITE(CON)	<b>←</b>		<b>+</b>	CON							
	15 21111112(3311)		Conversa									
	BYE(REL)	<b>→</b>		<b>→</b>	REL							
	200 OK BYE(RLC)	<del>,</del>	1	<del>-</del>	RLC							
			ı		1 = 0							

TP403012	SIP reference: RF	C 3261			ISUP reference:					
					Q.1912.5					
				Q.7	731 clause 5.5.2.1.1					
TSS reference	ISUP-SIP-ISUP/SS/CLIR									
SIP selection										
criteria	DIGG 4/7									
ISUP selection	PICS 1/7									
criteria	F tht ANIM CON			- 000 OK INIV	TE is sent on the IOUD side					
Test purpose	without changing. The conn				ITE is sent on the ISUP side					
	connected sub address is in			ichanged. The	e ATF contained the					
SIP parameter	200 OK INVITE: encapsulat			cludod						
values	200 OK INVITE. encapsulat	eu Aivii	A OI CON III	iciuueu						
ISUP parameter	a)									
values	ANM;									
raiacc	Connected number param	eter								
	Address presentation restrict		ameter = '0	0'B						
	Nature of address indicator									
	Numbering plan indicator =	'001'B								
	Screening indicator = '11'B									
	Address signals = PIXIT									
	and an access transport pa	aramete	er containin	g the connect	ed sub-address.					
	b)									
	CON;									
	Connected number param									
	Address presentation restrict			0'B						
	Nature of address indicator		011'B							
	Numbering plan indicator = 144/D	001B								
	Screening indicator = '11'B Address signals = PIXIT									
	and an access transport pa	aramete	ar containin	a the connect	ad sub-addrass					
Comments	ISUP	aramet	SUT		SIP-I					
Commonto	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)					
	CASE A			-	111111111111111111111111111111111111111					
	ACM	+		+	180 Ringing(ACM)					
	ANM	+		+	200 OK INVITE(ANM)					
	CASE B									
i e	CON	<b>←</b>		←	200 OK INVITE(CON)					
	CON	+	Conversa	_	200 OK INVITE(CON)					
	CON	<b>←</b>	Convers	_	200 OK INVITE(CON) BYE(REL)					

TP403013	SIP reference: RFC	3261			I	SUP reference:
					Q.7	Q.1912.5 31 clause 5.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/CLIR		1			
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Ensure that an ANM or CON	encap	osulated in a 2	200 OK I	NVI	TE is sent on the ISUP side
	without changing. The conne			nanged.	The	ATP contained the
	connected sub address is inc					
SIP parameter	200 OK INVITE: encapsulate	d ANI	M or CON incl	uded		
values						
ISUP parameter	a)					
values	ANM;					
	Connected number parame			_		
	Address presentation restrict			В		
	Nature of address indicator =		0011'B			
	Numbering plan indicator = '0	101'B				
	Screening indicator = '11'B					
	Address signals = PIXIT					
	Additional connected number Address presentation restricts			D		
	Nature of address indicator =			Б		
	Numbering plan indicator = 'C		DULLE			
	Screening indicator = '00'B	010				
	Address signals = PIXIT					
	and an access transport pa	ramet	er containing	the conn	ecte	d sub-address
	b)	amot	or cornairing			a das addition.
	CON;					
	Connected number parame	ter				
	Address presentation restrict		rameter = '00'	В		
	Nature of address indicator =	•				
	Numbering plan indicator = '0	01'B				
	Screening indicator = '11'B					
	Address signals = PIXIT					
	Additional connected number	<b>oer</b> pr	esent			
	Address presentation restrict			В		
	Nature of address indicator =		0011'B			
	Numbering plan indicator = 'C	001'B				
	Screening indicator = '00'B					
	Address signals = PIXIT					
0	and an access transport pa	ramet		ine conn	ecte	
Comments	ISUP		SUT			SIP-I
	IAM CASE A	<b>→</b>			<b>→</b>	INVITE(IAM)
	CASE A	<b>←</b>		1	_	190 Dinging (ACM)
	ACM	<del>-</del>			<del>(</del>	180 Ringing(ACM)
	ANM	_			<del>(</del>	200 OK INVITE(ANM)
	CASE B			1		200 OK INIVITE (CONI)
	CON	<del>-</del>	Camara 11		<del>(</del>	200 OK INVITE(CON)
	DEL		Conversat		_	DVE(DEL)
	REL	<b>→</b>			<u>→</u>	BYE(REL)
	RLC	+			<del>(</del>	200 OK BYE(RLC)

## 5.3.4 Connected Line Identification Restriction (COLR)

TP404001	SIP reference: RF	C 326			SUP reference:
					Q.1912.5
				Q.7	31 clause 6.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/COLR				
SIP selection					
criteria					
ISUP selection criteria					
	Passing on information re	loting	to COLP		
Test purpose	Ensure that the SUT shall pa			rmation	related to the COLR
	supplementary service in the				
	number	o addire	oo procentation i	ootiiotoa	
SIP parameter					
values					
ISUP parameter	IAM;				
values	optional forward call indic	ators			
	Connected line identity requ	est ind	icator: requested		
	a)				
	ANM;				
	Connected number param		. 10.41.5		
	Address presentation restric				
	Nature of address indicator		0011 <sup>1</sup> B		
	Numbering plan indicator = 'Screening indicator = '11'B	0016			
	Address signals = PIXIT				
	Additional connected num	<b>ber</b> pr	esent		
	Address presentation restrict				
	Nature of address indicator				
	Numbering plan indicator =	'001'B			
	Screening indicator = '00'B				
	Address signals = PIXIT				
	b)				
	CON;				
	Connected number param		1041 D		
	Address presentation restrict				
	Nature of address indicator Numbering plan indicator =		1011 B		
	Screening indicator = '11'B	0016			
	Address signals = PIXIT				
	Additional connected num	<b>nber</b> pr	esent		
	Address presentation restrict				
	Nature of address indicator				
	Numbering plan indicator =	100415			
	Screening indicator = '00'B				
_	Address signals = PIXIT			1	T
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		→	IAM
	CASE A	-		1 -	Ta 014
	180 Ringing(ACM)	<b>+</b>		<del>-</del>	ACM
	200 OK INVITE(ANM)	+		+	ANM
	CASE B				Tooki
	200 OK INVITE(CON)	+	Company = -1!	←	CON
	DVE(DEL)		Conversation		DEL
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	+		<b>←</b>	RLC

TP404002	SIP reference: RFC 32		ISUP reference:					
			Q.1912.5					
TSS reference			Q.7	'31 clause 6.5.2.1.1				
SIP selection	ISUP-SIP-ISUP/SS/COLR							
criteria								
ISUP selection								
criteria								
Test purpose	Passing on information relatin	a to COLR						
	Ensure that the SUT shall pass transparently all information related to the COLR							
	supplementary service in the add							
	number and the additional conn							
SIP parameter								
values								
ISUP parameter	IAM;							
values	optional forward call indicator							
	Connected line identity request in	ndicator: requ	uested					
	a)							
	ANM; Connected number parameter							
	Address presentation restricted p		01' B					
	Nature of address indicator = '00		OT B					
	Numbering plan indicator = '001'							
	Screening indicator = '11'B	_						
	Address signals = PIXIT							
	Additional connected number present							
	Address presentation restricted parameter = '01' B							
	Nature of address indicator = '00							
	Numbering plan indicator = '001'	В						
	Screening indicator = '00'B							
	Address signals = PIXIT							
	b) CON;							
	Connected number parameter							
	Address presentation restricted p		01' B					
	Nature of address indicator = '00		01 5					
	Numbering plan indicator = '001'							
	Screening indicator = '11'B							
	Address signals = PIXIT							
	Additional connected number present							
	Address presentation restricted parameter = '01' B							
	Nature of address indicator = '0000011'B							
	Numbering plan indicator = '001'	В						
	Screening indicator = '00'B							
Commonto	Address signals = PIXIT	611	<del>-</del> 1	leup				
Comments	SIP-I INVITE(IAM) →	SU	<u> </u>	ISUP IAM				
	INVITE(IAM) →		7	IMIVI				
	180 Ringing(ACM)		+	ACM				
	200 OK INVITE(ANM)		<del>-</del>	ANM				
	CASE B		~	\( \text{LIMINI} \)				
	200 OK INVITE(CON)		+	CON				
	200 OK INVITE(CON)	Convers		OON				
	BYE(REL) →	Convers	<u>→</u>	REL				
	200 OK BYE(RLC)		+	RLC				
	1200 OK BIL(KLO)			1,750				

TP404003	SIP reference: RFC 3261 ISUP reference:								
			0.7	Q.1912.5					
TCC reference			Q.7	31 clause 6.5.2.1.1					
TSS reference SIP selection	ISUP-SIP-ISUP/SS/COLR								
criteria									
ISUP selection									
criteria									
Test purpose	Restricted connected number (user provided, verified and passed) with connecte sub-address								
	Ensure that the SUT can pass	transparently a	connected nu	ımber with the screening					
	indicator set to "user provided,	verified and pa	assed" and with	the address presentation					
	restricted indicator set to "pres								
	Additionally, an access transp	ort parameter	containing the	connected sub-address					
SIP parameter	shall also be provided								
values									
ISUP parameter	IAM;								
values	optional forward call indicate	ors							
	Connected line identity reques		uested						
	a)	•							
	ANM;								
	Connected number parameter		0.41.5						
	Address presentation restricted Nature of address indicator = '		01' B						
	Numbering plan indicator = '00								
	Screening indicator = '01'B	71 0							
	Address signals = PIXIT								
	access transport parameter co	ntaining the co	nnected sub-ad	ldress					
	b)	_							
	CON;								
	Connected number parameter		041.0						
	Address presentation restricted parameter = '01' B								
	Nature of address indicator = '0000011'B Numbering plan indicator = '001'B								
	Screening indicator = '01'B	71 6							
	Address signals = PIXIT								
	access transport parameter co	ntaining the co	nnected sub-ad	ldress					
Comments	SIP-I	SU	Т	ISUP					
		<b>→</b>	→	IAM					
	CASE A								
	10011119	<del>(</del>	<del>-</del>	ACM					
	=00 011 11111 = (7 11 1111)	<b>-</b>	+	ANM					
	CASE B	·		loon					
	200 OK INVITE(CON)	Conver	←	CON					
	BVE(DEL)	Convers	sation -	DEI					
	<u> </u>	<b>→</b>   ←	<del> </del>	REL RLC					
	200 ON BTE(NEO)	<u> </u>		INLO					

TP404004	SIP reference: RFC 3261 ISUP reference:								
						Q.1912.5			
T00 (	10115 015 10115 (00 (00) 5				Q.7	31 clause 6.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/COLR								
SIP selection									
criteria	DIOC 7/4								
ISUP selection	PICS 7/1								
criteria	Discouding the convected		:f th		!	and winter of			
Test purpose		Discarding the connected number if the presentation is restricted Ensure that the connected number is discarded in case of bilateral agreements, if the							
	address presentation restrict								
SIP parameter	200 INVITE: encapsulated A			to presen	lall	on restricted			
values	No Connected number			dad					
ISUP parameter	IAM;	er para	ineter inclu	ueu					
values	optional forward call indica	atore							
values	Connected line identity requi		icator: requ	petad					
	a)	est ind	icator. requ	colcu					
	ANM;								
	Connected number parame	eter							
	Address presentation restrict		rameter = 'C	1'B					
		Nature of address indicator = '0000011'B							
	Numbering plan indicator = '0								
	Screening indicator = '11'B								
	Address signals = PIXIT								
	b)								
	CON;								
	Connected number parame								
		Address presentation restricted parameter = '01'B							
	Nature of address indicator =		0011'B						
	Numbering plan indicator = '001'B								
		Screening indicator = '11'B							
	Address signals = PIXIT								
0	OID I		0117	-		LOUB			
Comments	SIP-I		SUT			ISUP			
	INVITE(IAM)	<b>→</b>			<u>→</u>	IAM			
	CASE A		I			In one			
	180 Ringing(ACM)	<u>+</u>			<u> </u>	ACM			
	200 OK INVITE(ANM)	<b>←</b>			<b>←</b>	ANM			
	CASE B		ı	<u> </u>		loon			
	200 OK INVITE(CON)	<del>-</del>			<del>(</del>	CON			
			Convers						
	BYE(REL)	<u>→</u>			<u>→</u>	REL			
	200 OK BYE(RLC)	+			<del>(</del>	RLC			

TP404005	SIP reference: R		ISUP reference:							
			Q.1912.5							
				Q.7	731 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 presentation is restricted			_						
					number is discarded in case					
	of bilateral agreements, if the	he addr	ess presenta	ation restricted	d indicator is set to					
	"presentation restricted"									
SIP parameter	200 INVITE: encapsulated	ANM o	r CON							
values	No Additional Conne	ected n	umber paran	neter included						
ISUP parameter	IAM;									
values	optional forward call indi									
	Connected line identity req	uest ind	dicator: reque	ested						
	a)									
	ÁNM;									
	Connected number parameter present									
	Additional Connected number parameter									
	Address presentation restricted parameter = '01'B									
	Nature of address indicator									
	Numbering plan indicator =									
	Screening indicator = '11'B									
	Address signals = PIXIT									
	b)									
	CON;									
	Connected number parameter present									
	Additional Connected number parameter									
	Address presentation restricted parameter = '01'B									
	Nature of address indicator = '0000011'B									
	Numbering plan indicator = '001'B Screening indicator = '11'B									
	Address signals = PIXIT									
	Address signals = 1 IXII									
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	<b>→</b>		→	IAM					
	CASE A	1 -	1							
	180 Ringing(ACM)	<b>+</b>		+	ACM					
	200 OK INVITE(ANM)	+		+	ANM					
	CASE B		1		Is at any					
	200 OK INVITE(CON)	+	1	<b>+</b>	CON					
	200 OK HAVITE(CON)	_	Conversa		OON					
	DVE(DEL)		Universa		DEI					
	BYE(REL) 200 OK BYE(RLC)	<b>→</b>	-	→ ←	REL RLC					
	ZUU UN DIE(KLU)	_	1	7	NLO					

TP404007	SIP reference: RF	C 326	1		I	SUP reference: Q.1912.5		
					Q.7	31 clause 6.5.2.1.1		
TSS reference	ISUP-SIP-ISUP/SS/COLR							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that an ANM or CON	V enca	psulated in a	200 Ok	(INVI	TE is sent on the ISUP side		
	without changing. The conn							
	connected sub address is in	cluded	l	_				
SIP parameter	200 OK INVITE: encapsulat	ed ANI	M or CON in	cluded				
values	·							
ISUP parameter	ANM;							
values	Connected number param	eter						
	Address presentation restric		rameter = '0	1'B				
	Nature of address indicator	= '0000	0011'B					
	Numbering plan indicator =	'001'B						
	Screening indicator = '11'B							
	Address signals = PIXIT							
	Additional connected num							
	Address presentation restrict			1'B				
	Nature of address indicator		0011'B					
	Numbering plan indicator =	'001'B						
	Screening indicator = '00'B							
	Address signals = PIXIT							
	and an access transport p	aramet	er containing	the co	nnecte	d sub-address.		
	b)							
	CON;							
	Connected number param			u.D				
	Address presentation restrict	•		IВ				
	Nature of address indicator		J011B					
	Numbering plan indicator =	0016						
	Screening indicator = '11'B Address signals = PIXIT							
		hor nr	econt					
	Additional connected number present Address presentation restricted parameter = '01'B							
	Nature of address indicator			טו				
	Numbering plan indicator =							
	Screening indicator = '00'B	5510						
	Address signals = PIXIT							
	and an access transport p	aramet	er containing	the co	nnecte	d sub-address.		
Comments			SUT			SIP-I		
	IAM	<b>→</b>	1		<b>→</b>	INVITE(IAM)		
	CASE A	1	1		1	, ,		
	ACM	<b>←</b>			+	180 Ringing(ACM)		
	ANM	+			+	200 OK INVITE(ANM)		
	CASE B	<u> </u>	1					
	CON	+			+	200 OK INVITE(CON)		
			Conversa	tion		200 010 1100112 (0010)		
	REL	<b>→</b>	Jonverse		<b>→</b>	BYE(REL)		
	RLC	+			+	200 OK BYE(RLC)		
	INLO		1		_	ZOO ON DIL(NLO)		

## 5.3.5 Terminal Portability (TP)

TP405001	SIP reference: RFC 3261			ISUP reference: Q.1912.5					
				Q.	.733clause 4.5.2.1				
TSS reference:	ISUP-SIP-ISUP/SS/TP	1							
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Ensure that SUT inform requested by the calling	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	INFO: Content-Type: a	pplication/IS	UP; SUS and R	ES enca	psulated in the MIME body				
values									
ISUP parameter									
values									
Comments	ISUP		SUT		SIP-I				
	IAM	→		→	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	<b>←</b>		<b>←</b>	200 OK INVITE(ANM)				
			Conversation						
	SUS	<b>→</b>		<b>→</b>	INFO(SUS)				
				<b>←</b>	200 OK INFO				
	RES	<b>→</b>		<b>→</b>	INFO(RES)				
				+	200 OK INFO				
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	<b>+</b>		+	200 OK BYE(RLC)				

TP405002	SIP referen	ce: RFC 3261			ISUP reference: Q.1912.5 .733clause 4.5.2.1				
TSS reference:	ISUP-SIP-ISUP/SS/T	P							
SIP selection criteria									
ISUP selection criteria									
Test purpose	Ensure that SUT info requested by the call	Terminal portability, requested by the called party Ensure that SUT informs the calling party that a suspend and a resume have been requested by the called party upon receipt of user initiated SUS and RES messages							
SIP parameter	INFO: Content-Type:	application/IS	SUP ; SUS a	and RES enca	psulated in the MIME body				
values									
ISUP parameter values									
Comments	ISUP		SUT		SIP-I				
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM	1		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
			Conversa	ation					
	SUS	+		+	INFO(SUS)				
				→	200 OK INFO				
	RES	+		+	INFO(RES)				
				<b>→</b>	200 OK INFO				
	DE:				D)(E(DEL)				
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	<b>←</b>		<b>←</b>	200 OK BYE(RLC)				

TP405003	SIP reference: R	FC 3261		ISUP reference: Q.1912.5 Q.733clause 4.5.2.1				
TSS reference	ISUP-SIP-ISUP/SS/TP							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Ensure that the call is relea	Terminal portability, requested by local served user, no Resume after Suspend Ensure that the call is released with cause #102 (recovery on timer expiry) by the SUT if timer T2 expires because the local served user does not resume the call						
SIP parameter values	INFO: Content-Type: applic BYE : Content-Type: applic							
ISUP parameter values								
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
			Conversation	1				
	SUS	<b>→</b>		<b>→</b>	INFO(SUS)			
				+	200 OK INFO			
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		<b>←</b>	200 OK BYE(RLC)			

TP405004	SIP reference: RF	C 3261	I	ISUP reference: Q.1912.5 Q.733clause 4.5.2.1				
TSS reference	ISUP-SIP-ISUP/SS/TP		<u>'</u>					
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	Terminal portability, relea	se sus	pended call					
	Ensure that a suspended ca	all can b	e released, if the	remote i	user releases the call			
SIP parameter	INFO: Content-Type: applic							
values	BYE : Content-Type: application	ation/IS	UP; REL encaps	sulated in	the MIME body			
ISUP parameter								
values								
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
		Conversation						
	SUS → INFO(SUS)							
					200 OK INFO			
	REL	+	_	+	BYE(REL)			
	RLC	<b>→</b>		<b>→</b>	200 OK BYE(RLC)			

# 5.3.6 SUB-addressing (SUB)

TP406001	SIP reference: RFC 3261		Q.19	-	se: 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 the access transport parameter  Ensure that the SUT can include the called sub-address in the access transport parameter in the encapsulated IAM						
SIP parameter values	INVITE: Content-Type: app	olication	ISUP ; IAM enca	psulated	in the MIME body		
ISUP parameter values							
Comments	ISUP		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
			Conversation	•	, , ,		
	REL	<b>→</b>		→	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

TP406002	SIP reference: RFC 3261				SUP reference: Q.1912.5 31 clause 8.5.2.1.1/		
TSS reference	ISUP-SIP-ISUP/SS/SUB						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Receiving the called sub-address in the access transport parameter Ensure that the SUT can include the called sub-address in the access transport parameter in the ISUP IAM						
SIP parameter values							
ISUP parameter values							
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM) ← ← ANM						
			Conversation				
	BYE(REL)	<b>→</b>		<b>→</b>	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP406003	SIP reference: R	FC 326	1		SUP reference: Q.1912.5 31 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-address in the access transport parameter  Ensure that the SUT can include the called sub-address in the access transport parameter in the encapsulated IAM							
SIP parameter values	INVITE: Content-Type: app	olication	/ISUP ; IAM encap	sulated	in the MIME body			
ISUP parameter values								
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
			Conversation		` '			
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP406004	SIP reference: R	RFC 326	1		I:	SUP reference:
						Q.1912.5
				C	2.73	31 clause 8.5.2.1.1/
TSS reference	ISUP-SIP-ISUP/SS/SUB					
SIP selection criteria						
ISUP selection						
criteria						
Test purpose	Receiving the calling sul Ensure that the SUT can in parameter in the ISUP IAM	nclude th			•	-
SIP parameter values						
ISUP parameter values						
Comments	SIP-I		SU	Ī		ISUP
	INVITE(IAM)	<b>→</b>		-	<del>}</del>	IAM
	180 Ringing(ACM)	+		•	<b>-</b>	ACM
	200 OK INVITE(ANM)	+		•	<del>-</del>	ANM
			Convers	ation		
	BYE(REL)	<b>→</b>		-	<del>}</del>	REL
	200 OK BYE(RLC)	+		•	<del>-</del>	RLC

# 5.3.7 Malicious Call Identification (MCID)

TP407001	SIP referen	ce: RFC 3261		ISUP reference: Q.1912.5 Q.731.7 clause 7.5.2.1.1	
TSS reference:	ISUP-SIP-ISUP/SS/N	//CID	•		
SIP selection criteria					
ISUP selection criteria					
Test purpose	set to "MCID request	can successful and pass on a	lly pass on an <b>I</b> l an <b>IRS</b> with <b>MC</b>	OR having the MCID request indica ID response indicator set to "MCID SUP to SIP-I interworking	
SIP parameter values	body			SUP; IDR encapsulated in the MIME ulated in the MIME	
ISUP parameter values					
Comments	ISUP		SUT	SIP-I	
	IAM	<b>→</b>		→ INVITE(IAM)	
	IDR	+		← 183 Session Progress(IDR	.)
	IRS	<b>→</b>		→ INFO(IRS)	
				← 200 OK INFO	
	ACM	+		← 180 Ringing(ACM)	
	ANM	+		← 200 OK INVITE(ANM)	
			Conversation		
	REL	<b>→</b>		→ BYE(REL)	
	RLC	+		← 200 OK BYE(RLC)	

TP407002	SIP reference: RFC	3261		ISUP reference: Q.1912.5				
				Q.73	31.7 clause 7.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/MCID							
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	Successful MCID request I-N							
	Ensure that the SUT can succe							
	set to "MCID request" and pas included" and the calling party							
SIP parameter	183 Session Progress: Conten							
values	body	7.			•			
	INFO: Content-Type: application	on/ISUP	; IRS encapsulat	ed in	the MIME body			
ISUP parameter values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	183 Session Progress(IDR)	+		+	IDR			
	INFO(IRS)	<b>→</b>		<b>→</b>	IRS			
	200 OK INFO	+						
	180 Ringing(ACM) ← ACM							
	200 OK INVITE(ANM)	+		+	ANM			
			Conversation					
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP407003	SIP reference: RFC 3261			ISUP reference: Q.1912.5 Q.731.7 clause 7.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/MCID							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Successful MCID request - at							
	Ensure that the SUT will accep							
	been received. The SUT should							
	"MCID request" and pass on a							
	included" and the calling party							
SIP parameter	INFO: Content-Type: application	n/ISUP	; IDR er	ncapsulate	ed in	the MIME body		
values	INFO: Content-Type: application	INFO: Content-Type: application/ISUP; IRS encapsulated in the MIME body						
ISUP parameter values	IRS containing the calling party	/ numbe	er paran	neter				
Comments	SIP-I		S	UT		ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	CASE A	1				-		
	180 Ringing(ACM)	+			+	ACM		
	183 Session Progress(IDR)	+			+	IDR		
	INFO(IRS)	<b>→</b>			<b>→</b>	IRS		
	200 OK INFO	+						
	200 OK INVITE(ANM)	+			+	ANM		
	CASE B		<u>I</u>			) ii divi		
	183 Session Progress(ACM)	+			+	ACM(early)		
	183 Session Progress(IDR)	+			+	IDR		
	INFO(IRS)	→			<b>→</b>	IRS		
	200 OK INFO	+						
	180 Ringing(CPG)	+			+	CPG(alerting)		
	200 OK INVITE(ANM)							
	Conversation							
	BYE(REL)	<b>→</b>	00		<b>→</b>	REL		
	200 OK BYE(RLC)	<del>-</del>			<del>′</del>	RLC		
NOTE: This situa	ation may occur e.g. if the call ha	_	forward	ad bafara				

TP407004	SIP reference: RF	C 326	1	G	ISUP reference: Q.1912.5 ).731.7 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 seems that the SUT rejects indicator set to "MCID not in the second s	a MCI	D request b	y sending a	a IRS with the MCID response
SIP parameter	183 Session Progress: Con	tent-Ty	pe: applicat	ion/ISUP; I	DR encapsulated in the MIME
values	body	•			·
	INFO: Content-Type: applic	ation/IS	SUP; IRS er	capsulated	I in the MIME body
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	<b>→</b>		→	INVITE(IAM)
	IDR	+		+	183 Session Progress(IDR)
	IRS	<b>→</b>		→	INFO(IRS)
				+	200 OK INFO
	ACM	<b>←</b>		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
			Conversa	tion	
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP407005	SIP reference: RFC	3261			-	SUP reference: Q.1912.5
T00 (					Q.73	1.7 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 sup Ensure that the SUT rejects a indicator set to "MCID not incl	MCID r	equest b	y sendii	ng a <b>IR</b>	
SIP parameter values	183 Session Progress: Contenbody INFO: Content-Type: application	t-Type:	applicat	ion/ISU	P; IDR	encapsulated in the MIME
ISUP parameter values				•		
Comments	SIP-I		SI	JT		ISUP
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM
	183 Session Progress(IDR)	+			+	IDR
	INFO(IRS)	<b>→</b>			<b>→</b>	IRS
	200 OK INFO	+				
	180 Ringing(ACM)	+			+	ACM
	200 OK INVITE(ANM)	+			+	ANM
	, ,		Conver	sation	•	
	BYE(REL)	<b>→</b>			<b>→</b>	REL
	200 OK BYE(RLC)	+			+	RLC

TP407006	SIP reference: RFC 3	2261			-	SUP reference:				
17407000	SIF reference. NFC 3	201		Q.1912.5						
					0.73	1.7 clause 7.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/MCID	40.000.000								
SIP selection										
criteria										
ISUP selection	PICS 1/7									
criteria										
Test purpose	MCID information passed and									
	Ensure that a received <b>IDR</b> is to									
	subsequent IRS being transfer									
	the address signals of the calli			r is add	ed an	d the nature of address				
	indicator is set to "international		-							
	<ul> <li>The IDR request is transfe</li> </ul>									
						calling party number coded				
	as an "international numbe									
SIP parameter	183 Session Progress: Content	t-Type:	applicatio	n/ISUP	; IDR	encapsulated in the MIME				
values	body									
10115	INFO: Content-Type: application	n/ISUF	; IRS enca	apsulate	ed in t	he MIME body				
ISUP parameter										
values	OID I		0117			IOUR				
Comments	SIP-I		SUT			ISUP				
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM				
	183 Session Progress(IDR)	<b>+</b>			<u>+</u>	IDR				
	INFO(IRS)	<b>→</b>			<b>→</b>	IRS				
	200 OK INFO	<b>+</b>			-	A ON 4				
	180 Ringing(ACM)	+			<b>+</b>	ACM				
	200 OK INVITE(ANM)	_	Canyara	nti o m	7	ANM				
	DVE(DEL)		Convers	ation		DEL				
	BYE(REL)	<b>→</b>			<b>→</b>	REL				
	200 OK BYE(RLC)	<b>←</b>			+	RLC				

TP407007	SIP reference: RF	C 326	1		ISUP reference:
					Q.1912.5
				Q	.731.7 clause 7.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/MCID				
SIP selection					
criteria					
ISUP selection criteria					
Test purpose	set to "MCID request" by se	ccessfunding a	ully reply to a an <b>IRS</b> with <b>N</b> er and a calli	n <b>IDR</b> hav I <b>CID resp</b>	MGCF ing the MCID request indicator onse indicator set to "MCID dress in the access transport
SIP parameter				n/ISUP; II	OR encapsulated in the MIME
values	body INFO: Content-Type: applic	ation/IS	· SUP: IRS end	apsulated	in the MIME body
ISUP parameter values					•
Comments	ISUP		SUT		SIP-I
	IAM	<b>→</b>		→	INVITE(IAM)
	IDR	<b>←</b>		+	183 Session Progress(IDR)
	IRS	<b>→</b>		→	INFO(IRS)
				+	200 OK INFO
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
			Conversati	on	
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP407008	SIP reference: RFC	3261			ISUP reference: Q.1912.5 31.7 clause 7.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/MCID				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Successful MCID request wing Ensure that the SUT can successe to "MCID request" by send included", the calling party nuparameter. SIP-I to ISUP intensical successions.	essfully ing an <b>ımber</b>	reply to an IDF IRS with MCID and a calling su	R having respons	the MCID request indicator se indicator set to "MCID
SIP parameter values	183 Session Progress: Content body INFO: Content-Type: application	• •			•
ISUP parameter values			·		
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	183 Session Progress(IDR)	+		+	IDR
	INFO(IRS)	<b>→</b>		<b>→</b>	IRS
	200 OK INFO	+			
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Conversation	1	
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	+		+	RLC

TP407009	SIP reference: RI	FC 326	1	Q.	ISUP reference: Q.1912.5 731.7 clause 7.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/MCID				
SIP selection criteria					
ISUP selection criteria					
Test purpose		ntinued	(user is alerted)		S is received within timer T39 ator set to "MCID requested".
SIP parameter values	183 Session Progress: Con	itent-Ty	pe: application/IS	SUP; ID	R encapsulated in the MIME
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)
	IDR	+		+	183 Session Progress(IDR)
				T39 e	xpiry
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
			Conversation	•	
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP407010	SIP reference: RFC	3261		ISUP reference: Q.1912.5 Q.731.7 clause 7.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/MCID		•					
SIP selection criteria								
ISUP selection criteria								
Test purpose	Ensure that call setup is contin	MCID timer (T39) expiry O-MGCF Ensure that call setup is continued (user is alerted) if no IRS is received within timer T39 expiry, after having sent the IDR with MCID request indicator set to "MCID requested".  SIPJ to ISUP interworking						
SIP parameter values	183 Session Progress: Contenbody INFO: Content-Type: application				•			
ISUP parameter values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	183 Session Progress(IDR)	+		+	IDR			
		T39 expiry						
	180 Ringing(ACM)	<b>←</b>		+	ACM			
	200 OK INVITE(ANM)	<b>←</b>		+	ANM			
			Conversation		•			
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	<b>←</b>		+	RLC			

## 5.3.8 Call hold (HOLD)

TP408001	SIP reference: RFC 3261				ISUP reference: Q.1912.5
				Q.733 cla	use 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	messages having the even	s that a	call is place tor set to "	ed on hold and progress". O-N	
SIP parameter values	INVITE: Content-Type: app	olication	ISUP; CPG	encapsulated	I in the MIME body
ISUP parameter values					
Comments	ISUP		SU	Т	SIP-I
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
			Convers	ation	
	CPG(progress, hold)	<b>→</b>		<b>→</b>	INVITE(CPG, sendonly)
				<b>←</b>	200 OK INVITE
				→	ACK
	CPG(progress, retrieve)	<b>→</b>		<b>→</b>	INVITE(CPG, sendrecv)
				+	200 OK INVITE
				<b>→</b>	ACK
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	+		<b>+</b>	200 OK BYE(RLC)

TP408002	SIP reference: RFC 3261				ISUP reference: Q.1912.5 use 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 local 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; CPG	encapsulated	in the MIME body			
ISUP parameter values								
Comments	SIP-I		SU	Γ	ISUP			
	INVITE(IAM)	<b>→</b>		→	IAM			
	180 Ringing(ACM)	+		<b>←</b>	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
			Convers	ation				
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(progress, hold)			
	200 OK INVITE	+						
	ACK	<b>→</b>						
	INVITE(CPG, sendrecv)	<b>→</b>		<b>→</b>	CPG(progress, retrieve)			
	200 OK INVITE	+						
	ACK	<b>→</b>						
	BYE(REL)	<b>→</b>		→	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP408003	SIP reference: RFC 3261			ISUP reference: Q.1912.5 Q.733 clause 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, re					
	Ensure that the notification messages having the <b>ever</b>				retrieved are sent with <b>CPG</b> IGCF interworking	
SIP parameter values	INVITE: Content-Type: app	olication	/ISUP; CPG e	encapsulated	in the MIME body	
ISUP parameter values						
Comments	ISUP		SUT		SIP-I	
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)	
	ACM	+		<del>-</del>	180 Ringing(ACM)	
	ANM	+		+	200 OK INVITE(ANM)	
			Conversa	tion		
	CPG(progress, hold)	+		<del>-</del>	INVITE(CPG, sendonly)	
				→	200 OK INVITE	
				+	ACK	
	CPG(progress, retrieve)	+		+	INVITE(CPG, sendrecv)	
				→	200 OK INVITE	
				+	ACK	
	REL	<b>→</b>		<b>→</b>	BYE(REL)	
	RLC	+		+	200 OK BYE(RLC)	

TP408004	SIP reference: RFC 3261			ISUP reference: Q.1912.5 Q.733 clause 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	Ensure that the notification messages having the <b>even</b>	Call hold after answer, requested by the remote 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; CPG encap	sulated	in the MIME body				
ISUP parameter values									
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		<b>←</b>	ACM				
	200 OK INVITE(ANM)	<b>←</b>		+	ANM				
			Conversation						
	INVITE(CPG, sendonly)	<b>←</b>		<b>←</b>	CPG(progress, hold)				
	200 OK INVITE	→							
	ACK	+							
	INVITE(CPG, sendrecv)	+		+	CPG(progress, retrieve)				
	200 OK INVITE	<b>→</b>							
	ACK	+							
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	<b>←</b>		<b>+</b>	RLC				

TP408005	SIP reference: RFC 3261			ISUP reference: Q.1912.5 Q.733 clause 2.2.1; 2.5.2.1.1.1; 2.5.2.1.1.2		
TSS reference				Q.733 clause	2.2.1; 2.5.2.1.1.1; 2.5.2.1.1.2	
	ISUP-SIP-ISUP/SS/HOLD					
SIP selection						
criteria	DIOC 0/4					
ISUP selection criteria	PICS 8/1					
Test purpose	Call hold after alerting, re					
	Ensure that when a outgoin notifications are sent with	ng call is C <b>PG</b> me	s placed on ssages. O-	hold and retrie	eved after alerting the rking	
SIP parameter	INVITE: Content-Type: app					
values					<u> </u>	
ISUP parameter						
values						
Comments	ISUP		SU'	Т	SIP-I	
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)	
	ACM	+		+	180 Ringing(ACM)	
	CPG(progress, hold)	<b>→</b>		<b>→</b>	INVITE(CPG, sendonly)	
				+	200 OK INVITE	
				<b>→</b>	ACK	
	CPG(progress, retrieve)	<b>→</b>		<b>→</b>	INVITE(CPG, sendrecv)	
				+	200 OK INVITE	
				<b>→</b>	ACK	
	ANM	+		<b>+</b>	200 OK INVITE(ANM)	
			Convers	sation		
	REL	<b>→</b>		<b>→</b>	BYE(REL)	
	RLC	+		+	200 OK BYE(RLC)	

TP408006	SIP reference: RFC 3261	Q.191	ISUP reference: Q.1912.5 Q.733 clause 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 a outgoir notifications are sent with	Call hold after alerting, requested by the local user  Ensure that when a outgoing call is placed on hold and retrieved after alerting the notifications are sent with CPG messages. I-MGCF interworking						
SIP parameter	INVITE: Content-Type: app	olication	/ISUP; CPG enca	psulated	in the MIME body			
values					·			
ISUP parameter values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		→	IAM			
	180 Ringing(ACM)	+		+	ACM			
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(progress, hold)			
	200 OK INVITE	<b>←</b>						
	ACK	<b>→</b>						
	INVITE(CPG, sendrecv)	<b>→</b>		<b>→</b>	CPG(progress, retrieve)			
	200 OK INVITE	<b>←</b>						
	ACK	<b>→</b>						
	200 OK INVITE(ANM)	+		+	ANM			
			Conversation					
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP408007	SIP reference: RFC 3261			ISUP reference: Q.1912.5 Q.764 clause 2.3		
TSS reference	ISUP-SIP-ISUP/SS/HOL	D				
SIP selection						
criteria						
ISUP selection criteria						
Test purpose	Call hold after answer,					
			can be released	by the us	er who activated the Call hold	
	service. O-MGCF interwo					
SIP parameter values	INVITE: Content-Type: a	pplication/	ISUP; CPG end	apsulated	in the MIME body	
ISUP parameter						
values						
Comments	ISUP		SUT		SIP-I	
	IAM	<b>→</b>		→	INVITE(IAM)	
	ACM	+		+	180 Ringing(ACM)	
	ANM	+		+	200 OK INVITE(ANM)	
			Conversation	n		
	CPG(progress, hold)	<b>→</b>		<b>→</b>	INVITE(CPG, sendonly)	
				+	200 OK INVITE	
				<b>→</b>	ACK	
	REL	<b>→</b>		<b>→</b>	BYE(REL)	
	RLC	<b>←</b>		<b>←</b>	200 OK BYE(RLC)	

TP408008	SIP reference: RFC 3261			ISUP reference: Q.1912.5			
T00 (					Q.764 clause 2.3		
TSS reference	ISUP-SIP-ISUP/SS/HOLD						
SIP selection							
criteria							
ISUP selection criteria							
Test purpose	Call hold after answer, re	elease o	f the call by the	local se	rved user		
	Ensure that a call in the he	ld state	can be released	by the us	er who activated the Call hold		
	service. I-MGCF interworki	ing					
SIP parameter	INVITE: Content-Type: app	olication	ISUP; CPG enca	psulated	in the MIME body		
values					-		
ISUP parameter							
values							
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
			Conversation	•			
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(progress, hold)		
	200 OK INVITE	+					
	ACK	<b>→</b>					
	BYE(REL)	<b>→</b>		<b>→</b>	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP408009	SIP reference:	RFC 3261			ISUP reference: Q.1912.5 Q.764 clause 2.3				
TSS reference	ISUP-SIP-ISUP/SS/HOLI	D	L						
SIP selection criteria									
ISUP selection criteria									
Test purpose	Ensure that a call in the h	Call hold after answer, release of the call by the non-served user Ensure that a call in the held state can be released by the user who did not activate the Call hold service. O-MGCF interworking							
SIP parameter values		INVITE: Content-Type: application/ISUP; CPG encapsulated in the MIME body							
ISUP parameter values									
Comments	ISUP		SUT		SIP-I				
	IAM	<b>→</b>		→	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
			Conversa	tion					
	CPG(progress, hold)	+		+	INVITE(CPG, sendonly)				
				→	200 OK INVITE				
				+	ACK				
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	+	•	+	200 OK BYE(RLC)				

TP408010	SIP reference: R	FC 3261		ISUP reference: Q.1912.5			
				(	Q.764 clause 2.3		
TSS reference	ISUP-SIP-ISUP/SS/HOLD						
SIP selection							
criteria							
ISUP selection criteria							
Test purpose	Call hold after answer, re						
				by the us	ser who did not activate the		
	Call hold service. I-MGCF						
SIP parameter	INVITE: Content-Type: app	olication/	'ISUP; CPG enca	psulated	I in the MIME body		
values							
ISUP parameter							
values		1		1			
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		→	IAM		
	180 Ringing(ACM)	+		<b>←</b>	ACM		
	200 OK INVITE(ANM)	+		<b>←</b>	ANM		
			Conversation				
	INVITE(CPG, sendonly)	+		<b>←</b>	CPG(progress, hold)		
	200 OK INVITE	<b>→</b>					
	ACK	+					
	BYE(REL)	<b>→</b>		<b>→</b>	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP408011	SIP reference:	RFC 3261			ISUP reference: Q.1912.5 Q.764 clause 2.3			
TSS reference	ISUP-SIP-ISUP/SS/HOLI	D	l .					
SIP selection criteria								
ISUP selection criteria								
Test purpose	Ensure that a held call ca	Call hold after alerting, release of the call by the local served user  Ensure that a held call can be released by the user who activated the Call hold service without retrieving the call. O-MGCF interworking						
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	→		<b>→</b>	INVITE(IAM)			
	ACM	<b>←</b>		+	180 Ringing(ACM)			
			Ringing					
	CPG(progress, hold)	<b>→</b>		<b>→</b>	INVITE(CPG, sendonly)			
				+	200 OK INVITE			
				<b>→</b>	ACK			
	REL	→		→	BYE(REL)			
	RLC	←		<b>←</b>	200 OK BYE(RLC)			

TP408012	SIP reference: RFC 3261 ISUP reference:					
					Q.1912.5	
				C	Q.764 clause 2.3	
TSS reference	ISUP-SIP-ISUP/SS/HOLD					
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Call hold after alerting, re					
	Ensure that a held call can			r who activ	ated the Call hold service	
	without retrieving the call. I					
SIP parameter	INVITE: Content-Type: app	lication/IS	SUP; CPG end	apsulated	in the MIME body	
values						
ISUP parameter						
values						
Comments	SIP-I		SUT		ISUP	
	INVITE(IAM)	<b>→</b>		→	IAM	
	180 Ringing(ACM)	+		+	ACM	
			Ringing			
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(progress, hold)	
	200 OK INVITE	+				
	ACK →					
	BYE(REL)	<b>→</b>		<b>→</b>	REL	
	200 OK BYE(RLC)	+	-	+	RLC	

# 5.3.9 Call Waiting (CW)

TP409001	SIP reference: RFC 3261				ISUP reference: Q.1912.5 '33 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 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 values	180 Ringing: Content-Type	e: applica	ation/ISUP; /	ACM encapsu	llated in the MIME body			
ISUP parameter values	ACM: Generic notification i	indicator	"Call is a w	aiting call"				
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		→	INVITE(IAM)			
	ACM(waiting)	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
			Conversa	ition				
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP409002	SIP reference: RFC 3261			ISUP reference:					
					Q.1912.5				
				Q.7	33 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 ACM								
	Ensure that a call can be s	Ensure that a call can be successfully established if the <b>ACM</b> indicates that this call is a							
	waiting call. I-MGCF interv	waiting call. I-MGCF interworking							
SIP parameter	180 Ringing: Content-Type	180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body							
values									
ISUP parameter	ACM: Generic notification indicator "Call is a waiting call"								
values									
Comments	SIP-I		SUT ISUP						
	INVITE(IAM)	<b>→</b>		→	IAM				
	180 Ringing(ACM)	+		+	ACM(waiting)				
	200 OK INVITE(ANM)	+		+	ANM				
	Conversation								
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP409003	SIP reference: RFC 3261				SUP reference: Q.1912.5 33 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 in CPG Ensure that a call can be successfully established if the CPG indicates that this call is a waiting call. O-MGCF interworking						
SIP parameter values	180 Ringing: Content-Type: application/ISUP; CPG encapsulated in the MIME body						
ISUP parameter values	CPG: Generic notification indicator "Call is a waiting call"						
Comments	ISUP		SUT	9	SIP-I		
	IAM	<b>→</b>		<b>→</b>	NVITE(IAM)		
	ACM	+		← 1	83 Session Progress(ACM)		
	CPG(waiting)	<del>-</del>		← 1	80 Ringing(CPG)		
	ANM	<del>-</del>			200 OK INVITE(ANM)		
	Conversation						
	REL	→		<b>→</b> E	BYE(REL)		
	RLC	+		← 2	00 OK BYE(RLC)		

TP409004	SIP reference: RFC 3261			ISUP reference: Q.1912.5				
				Q.7	33 clause 1.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/CW		<b> </b>					
SIP selection criteria								
ISUP selection criteria								
Test purpose	Call waiting indication in CPO	Call waiting indication in CPG						
	Ensure that a call can be succe	Ensure that a call can be successfully established if the CPG indicates that this call is a						
		waiting call. I-MGCF interworking						
SIP parameter	180 Ringing: Content-Type: application/ISUP; CPG encapsulated in the MIME body							
values								
ISUP parameter values	CPG: Generic notification indicator "Call is a waiting call"							
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	183 Session Progress ACM)	+		+	ACM			
	180 Ringing(CPG)	+		+	CPG(waiting)			
	200 OK INVITE(ANM)	+		+	ANM			
	Conversation							
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP409005	SIP reference:	: RFC 3261 ISUP reference: Q.1912.5						
					Q.733 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 <b>REL</b> with cause #21 (call rejected) if a busy user rejects							
	the waiting call. O-MGCF interworking							
SIP parameter	180 Ringing: Content-Type: application/ISUP; ACM or CPG encapsulated in the MIME							
values	body							
	480 Temporarily unavailable: Content-Type: application/ISUP; REL encapsulated in the							
	MIME body							
ISUP parameter	ACM or CPG: Generic n	otificat	ion indicator "Ca	ll is a v	vaiting call"			
values	REL: Cause #21 (call re	jected)						
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM(waiting)	+		+	180 Ringing(ACM)			
	REL(#21)	+		+	480 Temporarily Unavailable(REL)			
	RLC	<b>→</b>		<b>→</b>	ACK			

TP409006	SIP reference: RFC 3261	ISUP reference: Q.1912.5 Q.733 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. I-MGCF interworking									
SIP parameter values	180 Ringing: Content-Type: application/ISUP; ACM or CPG encapsulated in the MIME body 480 Temporarily unavailable: Content-Type: application/ISUP; REL encapsulated in the MIME body									
ISUP parameter values	ACM or CPG: Generic notification indicator "Call is a waiting call" REL: Cause #21 (call rejected)									
Comments	SIP-I SUT ISUP									
	INVITE(IAM) → IAM									
	180 Ringing(ACM)	<b>←</b>		+	ACM(waiting)					
	480 Temporarily Unavailable(REL)	+		+	REL(#21)					
	ACK → RLC									

TP409007	SIP reference: RFC 3261			ISUP reference: Q.1912.5 Q.733 clause 1.5.2.1.1					
TSS reference	ISUP-SIP-ISUP/SS/C	CW							
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Ensure that the SUT	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. O-MGCF interworking							
SIP parameter values	180 Ringing: Content body	180 Ringing: Content-Type: application/ISUP; ACM or CPG encapsulated in the MIME body 480 Temporarily unavailable: Content-Type: application/ISUP; REL encapsulated in the							
ISUP parameter		ACM or CPG: Generic notification indicator "Call is a waiting call"							
values	REL: Cause #19 (no								
Comments	ISUP		SUT		SIP-I				
	IAM	<b>→</b>		→	INVITE(IAM)				
	ACM(waiting)	+		+	180 Ringing(ACM)				
	T9 expiry								
	CASE A								
	REL(#19)	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				
				+	487 Request Terminated				
				<b>→</b>	ACK				
	CASE B								
	REL(#19)	<b>→</b>		<b>→</b>	CANCEL				
	RLC	+		+	200 OK CANCEL				
			_	+	487 Request Terminated				
				<b>→</b>	ACK				

TP409008	SIP reference: RF	C 326	1		-	SUP reference: Q.1912.5 33 clause 1.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/CW								
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Call waiting ignored (expiry of call waiting supervision timer)								
	Ensure that the SUT pass on a <b>REL</b> 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								
	480 Temporarily unavailable	e: Cont	ent-Type: ap	plication/l	SUP	; REL encapsulated in the			
	MIME body								
ISUP parameter	ACM or CPG: Generic notification indicator "Call is a waiting call"								
values	REL: Cause #19 (no answe	r from	user, user al	erted)					
Comments	SIP-I		SUT			ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	180 Ringing(ACM)	+			<del>(</del>	ACM(waiting)			
	T9 expiry								
	BYE(REL) → REL(#19)								
	200 OK BYE(RLC)	+			<del>(</del>	RLC			
	487 Request Terminated	+							
	ACK	<b>→</b>							

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

TP410001	SIP reference: R	FC 326	1		ISUP reference:				
				Q.19	912.5, Q.732 clause 2.5				
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	"Call is diverting" indicat								
					on occurs. The <b>ACM</b> contains				
			r set to "call	is diverting",	the call diversion information				
	and the <b>redirection numb</b>								
	The Redirection reason is								
	CPG (alerting) is coded as	if it has	been mapp	ed from ACN	М.				
	O-MCGF interworking								
SIP parameter		ntent-Ty	pe: applicat	ion/ISUP; A0	CM encapsulated in the MIME				
values	body								
	180 Ringing: Content-Type				sulated in the MIME body				
ISUP parameter	ACM: BCI Called party sta		cator "No in	dication"					
values	Generic notification								
	Call diversion inform								
	Redirection number								
	CPG: Event indicator=alert	ting		1	loin i				
Comments	ISUP		SUT		SIP-I				
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM(no indication)	<del>-</del>		+	183 Session Progress(ACM)				
	CPG(alerting)	<del>-</del>		<b>←</b>	180 Ringing(CPG)				
	ANM	<b>←</b>		· · ·	200 OK INVITE(ANM)				
			Conversa						
	REL	→		<b>→</b>	BYE(REL)				
	RLC	<b>←</b>		←	200 OK BYE(RLC)				

TP410002	SIP reference: RFC 3	3261			-	SUP reference: 2.5, Q.732 clause 2.5			
TSS reference	ISUP-SIP-ISUP/SS/Call Divers	ion							
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	"Call is diverting" indication received in ACM  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 ACM.  I-MCGF interworking								
SIP parameter	183 Session Progress: Content	t-Type:	applicat	ion/ISUF	; ACM	I encapsulated in the MIME			
values	body								
	180 Ringing: Content-Type: ap					ated in the MIME body			
ISUP parameter values	Generic notification	ACM: BCI Called party status indicator "No indication" Generic notification Call diversion information Redirection number							
Comments	SIP-I		SI	JT		ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	183 Session Progress(ACM)	+			+	ACM(no indication)			
	180 Ringing(CPG)	<b>←</b>			+	CPG(alerting)			
	200 OK INVITE(ANM)	+			+	ANM			
			Conver	sation	•				
	BYE(REL)	<b>→</b>			<b>→</b>	REL			
	200 OK BYE(RLC)	+			+	RLC			

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

TP410003	SIP reference: RF	C 326	1	Q.1912	ISUP reference: .5, Q.732 clause 2.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/Call Dive	ersion							
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	"Call diversion may occur" received in ACM  Verify that a call can be successfully established, if diversion may occur. The ACM indicates that "call diversion may occur" in the optional backward call indicators. The following CPG contains the generic notification indicator set to "call is diverting", the call diversion information and the redirection number, if diversion occurs. The CPG (progress) contains CV_redirection_reason in call diversion information and also Redirection number. The CPG (alerting) is coded as if it has been mapped from ACM, with RnNbRes parameter (optional).  O-MCGF interworking								
SIP parameter	180 Ringing: Content-Type:								
values		ent-Ty	pe: application	n/ISUP; CP	PG encapsulated in the MIME				
IOLID	body				e 11 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1				
ISUP parameter values	"Call diversion may o								
Comments	ISUP		SUT		SIP-I				
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM(free)	+		+	180 Ringing(ACM)				
	CPG	+		+	183 Session Progress(CPG)				
	CPG(alerting)	+		+	183 Session Progress(CPG)				
	ANM	+		+	200 OK INVITE(ANM)				
			Conversati	on					
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP410004	SIP reference: RFC 3261		Q.19	_	SUP reference: , Q.732 clause 2.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	"Call diversion may occur" received in ACM Verify that a call can be successfully established, if diversion may occur. The ACM indicates that "call diversion may occur" in the optional backward call indicators. The following CPG contains the generic notification indicator set to "call is diverting", the call diversion information and the redirection number, if diversion occurs. The CPG (progress) contains CV_redirection_reason in call diversion information and also Redirection number. The CPG (alerting) is coded as if it has been mapped from ACM, with RnNbRes parameter (optional). I-MCGF interworking							
SIP parameter	180 Ringing: Content-Type: application,	/ISUP;	ACM enc	apsul	ated in the MIME body			
values	183 Session Progress: Content-Type: a	pplicat	ion/ISUP;	CPG	encapsulated in the MIME			
	body							
ISUP parameter values	"Call diversion may occur"	CPG: Event information=progress, Call diversion information; Generic notification; Redirection number						
Comments	SIP-I	SI	JT		ISUP			
	INVITE(IAM) →			<b>→</b>	IAM			
	180 Ringing(ACM) ←			<b>←</b>	ACM(free)			
	183 Session Progress(CPG) ←			<b>←</b>	CPG			
	183 Session Progress(CPG) ←			+	CPG(alerting)			
	200 OK INVITE(ANM) ←			+	ANM			
		Conver	sation					
	BYE(REL) →			<b>→</b>	REL			
	200 OK BYE(RLC) ←			+	RLC			

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

TP410005	SIP reference: R	SIP reference: RFC 3261 ISUP reference:								
				Q	2.1912.5, Q.732 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				!					
				version	information are received, as if					
	multiple forwardings have of The CV_redirection_reason									
	The Redirection number re O-MCGF interworking	Suiction	parameter	is passed	u on.					
SIP parameter		tont Tv	no: applicat	ion/ISLID	; ACM encapsulated in the MIME					
values	body	петп-ту	ре. арріісаі	1011/130F	, ACIVI ericapsulated in the Milvic					
values		tent-Tv	ne: annlicat	ion/ISLIP	c; CPG encapsulated in the MIME					
	body	iterit i y	po. applicat	1011/1001	, or o cheapsulated in the Minne					
		· applica	ation/ISUP	CPG end	capsulated in the MIME body					
ISUP parameter	ACM: BCI Called party sta									
values	Generic notification									
	Call diversion inform	nation R	edirection re	eason ur	nconditional					
	Redirection number									
	CPG1: Event information=r	orogress	6							
	Generic notification	J								
	Call diversion inform	nation R	edirection re	eason C\	V_redirection_reason					
	Redirection number									
	Redirection number									
	CPG2: Event information=a	alerting,		number						
Comments	ISUP		SUT		SIP-I					
	IAM	<b>→</b>			→ INVITE(IAM)					
	ACM(no indication)	+			← 183 Session Progress(ACM)					
	CPG1	+			← 183 Session Progress(CPG)					
	CPG2(alerting) ← 180 Ringing(CPG)									
	ANM	+	<u> </u>		← 200 OK INVITE(ANM)					
		<u> </u>	Conversa							
	REL	<b>→</b>			→ BYE(REL)					
	RLC	<b>←</b>			← 200 OK BYE(RLC)					

TP410006	SIP reference: RFC 3	SUP reference:						
				Q.	1912	.5, Q.732 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 contain		e call dive	rsion in	forn	nation are received, as if		
	multiple forwardings have occur							
	The CV_redirection_reason is							
	The Redirection number restrict	ion pai	rameter is	passed	on.			
010	I-MCGF interworking	_						
SIP parameter	183 Session Progress: Content-	· I ype:	application	n/ISUP;	ACIV	I encapsulated in the MIME		
values	body	_		// CL ID	000	La li al Markar		
	183 Session Progress: Content-	· i ype:	application	1/ISUP;	CPG	encapsulated in the MIIME		
	body		-/ICLID: CI	20		atadia tha NANAT bady		
ISUP parameter	180 Ringing: Content-Type: app ACM: BCI Called party status in				apsui	ated in the MilME body		
values	Generic notification	nuicaic	or ino maio	Jalion				
values	Call diversion information	n Padii	raction rea	eon uno	ondi	tional		
	Redirection number	ritean	rection rea	SOIT UITO	oriai	lional		
	CPG: Event information=progre	SS						
	Generic notification	00						
	Call diversion information	n Redii	rection rea	son CV	red	irection reason		
	Redirection number			-		_		
	Redirection number restr	riction						
	CPG: Event information=alerting	g, Redi	irection nu	mber re	strict	ion		
Comments	SIP-I		SUT			ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	183 Session Progress(ACM)	<b>←</b>			<b>←</b>	ACM(no indication)		
	183 Session Progress(CPG)	<b>←</b>			+	CPG1		
	180 Ringing(CPG)	+			<b>←</b>	CPG2(alerting)		
	200 OK INVITE(ANM)	+			<b>←</b>	ANM		
			Conversa	tion				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	<b>←</b>			<b>←</b>	RLC		

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

TP410007	SIP reference: RF	C 326	1	0.10		SUP reference: , Q.732 clause 2.5.2.2.1				
TSS reference	ISUP-SIP-ISUP/SS/Call Div	ersion		Q.13	112.5	, Q.732 Clause 2.3.2.2.1				
SIP selection	ISST SIT ISST /SS/SQII BIVOISION									
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 restriction parameter (in ACM /CPG /ANM /CON)									
SIP parameter	O-MCGF interworking INVITE: Content-Type: appl	ication	ISUP; IAM e	encapsula	ted ir	the MIME body				
values	200 OK INVITE: Content-Ty									
ISUP parameter	IAM: Redirecting number, C			er, Redire	ection	information				
values	ANM: Redirection address r	estriction		1						
Comments	ISUP		SUT			SIP-I				
	IAM	<b>→</b>				INVITE(IAM)				
	ACM	+			<del>-</del>	180 Ringing(ACM)				
	ANM ← 200 OK INVITE(ANM)									
			Conversa	ition						
	REL	<b>→</b>			<u>→</u>	BYE(REL)				
	RLC	+			<b>←</b>	200 OK BYE(RLC)				
NOTE: Altered in	Gateways.									

TP410008	SIP reference: RF	C 326	1	Q.191	-	SUP reference: i, Q.732 clause 2.5.2.2.1			
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Notification procedures for a diverting call - after the diverting exchange								
	Verify that the IUT can succ								
	diversion) all the diversion in								
	It has to be checked that the direction:	follow	ing signallir	ng information	on i	s passed on in the forward			
	redirecting num	ber (se	ee note)						
	original called n	umber	(see note)						
	redirection infor	matio	า						
	It has to be checked that the	follow	ing signallir	ng information	on i	s passed on in the backward			
	direction:								
	redirection num	ber res	<b>striction</b> pa	rameter (in	AC	M /CPG /ANM /CON)			
	I-MCGF interworking								
SIP parameter	INVITE: Content-Type: appl								
values	200 OK INVITE: Content-Ty	pe: ap	plication/ISI	JP; ANM er	ncap	osulated in the MIME body			
ISUP parameter	IAM: Redirecting number, O			per, Redired	ction	n information			
values	ANM: Redirection address r	estricti							
Comments	SIP-I		SU	Γ		ISUP			
	INVITE(IAM)	<b>→</b>		-	<b>→</b>	IAM			
	180 Ringing(ACM)	<b>←</b>		•	<del>-</del>	ACM			
	200 OK INVITE(ANM)								
			Convers	ation					
	BYE(REL)	<b>→</b>		-	<b>&gt;</b>	REL			
	200 OK BYE(RLC)	+		•	<del>(</del>	RLC			
NOTE: Altered in	n Gateways.		•	· ·					

TP410009	SIP reference: RF	C 326	1	<b>]</b> ,	SUP reference:			
				Q.7:	Q.1912.5 31 clause 3.5.2.4.1			
TSS reference	ISUP-SIP-ISUP/SS/Call Div	ersion			0.0000000000000000000000000000000000000			
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 values	INVITE: Content-Type: appl encapsulated in the MIME b		/ISUP; IAM co	ontaining an (	Original called number			
ISUP parameter values	IAM: No original called num	ber pre	esent					
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
			Conversat	ion				
	BYE(REL)	<b>→</b>		→	REL			
	200 OK BYE(RLC)	<b>←</b>		+	RLC			

TP410010	SIP reference: RF	C 326	1		IS	SUP reference:		
				_	73	Q.1912.5 31 clause 4.5.2.1.1		
TSS reference	ISUP-SIP-ISUP/SS/Call Div	ersion			<i>(.1</i> 3	71 Clause 4.3.2.1.1		
SIP selection		CISIOII						
criteria								
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						Original called number called		
values	number encapsulated in the	MIME	body			_		
ISUP parameter	IAM: Original called number	r "Interi	national num	ıber"				
values								
Comments	SIP-I		SUT	•		ISUP		
	INVITE(IAM)	<b>→</b>		-		IAM		
	180 Ringing(ACM)	+		€	-	ACM		
	200 OK INVITE(ANM) ← ANM							
			Conversa	ation				
	BYE(REL)	<b>→</b>		1	•	REL		
	200 OK BYE(RLC)	+		•	•	RLC		

TP410011	SIP reference: RI	-C 326 <sup>-</sup>	1			SUP reference: Q.1912.5 31 clause 4.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/Call Div	ISUP-SIP-ISUP/SS/Call Diversion								
SIP selection										
criteria										
ISUP selection	PICS 1/7									
criteria										
Test purpose	Original called number in									
						nipulates the original called				
	number according to the pr									
	Discarding the original									
	The PTC will send an IAM v									
SIP parameter				containin	g an C	Original called number called				
values	number encapsulated in the									
ISUP parameter	IAM: No original called num	ber pre	esent							
values										
Comments	SIP-I		SUT	'		ISUP				
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM				
	180 Ringing(ACM)	<b>←</b>			+	ACM				
	200 OK INVITE(ANM) ← ANM									
			Conversa	ation						
	BYE(REL)	<b>→</b>			<b>→</b>	REL				
	200 OK BYE(RLC)	+			+	RLC				

TP410012	SIP reference: RI	FC 326 <sup>-</sup>	1		SUP reference: Q.1912.5 31 clause 4.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/Call Div	version			
SIP selection criteria					
ISUP selection	PICS 1/8				
criteria					
Test purpose	Original called number in	the inc	coming internati	onal gate	eway
	Verify that the incoming into	ernation	al gateway chec	ks and ma	anipulates the original called
	number according to the pr	rocedur	es as defined for	CLIP.App	olicable tests:
	Converting the original	called	number to nation	nal forma	t, if necessary (own country
	code)				
SIP parameter	INVITE: Content-Type: app	lication	/ISUP; IAM conta	ining an	Original called number called
values	number encapsulated in the	e MIME	body	Ū	
ISUP parameter	IAM: Original called numbe	r "Natio	nal number"		
values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Conversation	•	
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	+		+	RLC

TP410013	SIP reference: R	1	ISUP reference: Q.1912.5 Q.731 clause 4.5.2.1.1							
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion									
SIP selection criteria										
ISUP selection criteria	PICS 10/2 AND 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 case of bilateral agreements									
SIP parameter values	INVITE: Content-Type: appencapsulated in the MIME		/ISUP; IAM	containin	g a Re	edirecting number				
ISUP parameter values	IAM: No Redirecting numb	er prese	ent							
Comments	SIP-I		SUT	•		ISUP				
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM				
	180 Ringing(ACM)	+			+	ACM				
	200 OK INVITE(ANM)	+			+	ANM				
			Convers	ation	•					
	BYE(REL)	<b>→</b>			<b>→</b>	REL				
	200 OK BYE(RLC)	+			+	RLC				

TP410014	SIP reference: RI	FC 326	1		SUP reference: Q.1912.5 31 clause 4.5.2.1.1		
TSS reference	ISUP-SIP-ISUP/SS/Call Div	version					
SIP selection criteria							
ISUP selection	PICS 1/7						
criteria							
SIP parameter values	Redirecting number in the Verify that the outgoing into number according to the publication because the PTC will send an IAM INVITE: Content-Type: appencapsulated in the MIME IAM: No Redirecting numbers	ernation rocedur ing nur with an older	al gateway checks es as defined for C nber, if the addres "address not avail! /ISUP; IAM contain	and ma CLIP s is mar able" Ro	anipulates the <b>redirecting</b> ked not available Nb		
values	IAW. NO Redirecting number	or prese	ii it				
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM) ← ANM						
	Conversation						
	BYE(REL)	<b>→</b>		<b>→</b>	REL		
	200 OK BYE(RLC)	<b>←</b>		+	RLC		

TP410015	SIP reference: R	FC 326	1		ISUP reference: 1912.5, Q.732 clause
				2.5.2	2.3Q.731clause 3.5.2.3
TSS reference:	ISUP-SIP-ISUP/SS/Call Di	iversion			
SIP selection criteria					
ISUP selection criteria	PICS 1/7				
Test purpose	number according to the p	ernation procedurating number of restrict	al gateway es as define <b>mber</b> to inte ion indicato	checks and ned for CLIP rnational forr r	nanipulates the redirecting  mat with transparent transferral
SIP parameter values	INVITE: Content-Type: app number" encapsulated in the			containing a	Redirecting number "National
ISUP parameter values	IAM: Redirecting number "	Internat	ional numbe	r"	
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Convers	ation	
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	+		+	RLC

TP410016	SIP reference: RI	C 326	I	Q.19	SUP reference: 912.5, Q.732 clause 3Q.731clause 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 Verify that the incoming into number according to the process Converting the redirect code) The PTC will send an IAM verification.	ernation rocedur ing nur	al gateway che es as defined for nber to national	ecks and ma or CLIP	
SIP parameter	INVITE: Content-Type: app				edirecting number
values	"International number" enca	apsulate	ed in the MIME	body	
ISUP parameter	IAM: Redirecting number "r	ational	number"		
values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Conversation	n	
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	+		+	RLC

TP410017	SIP reference: F	RFC 326 <sup>-</sup>	1	Q.1912.	SUP reference: 5, Q.732 clause 2.5.2.3, 731clause 3.5.2.3		
TSS reference	ISUP-SIP-ISUP/SS/Call D	iversion					
SIP selection							
criteria							
ISUP selection	PICS 1/8 AND 10/4						
criteria							
Test purpose	number according to the particle Adding a prefix to an in the PTC will send an IAM	iternation procedur nternation I with Rg	nal gateway che es as defined f nal <b>redirecting</b> Nb	ecks and moor CLIP	anipulates the <b>redirecting</b>		
SIP parameter values	INVITE: Content-Type: ap encapsulated in the MIME	•	/ISUP; IAM cor	itaining a R	edirecting number		
ISUP parameter values	IAM: Redirecting number						
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM) ← ANM						
			Conversation	on			
	BYE(REL)	<b>→</b>		<b>→</b>	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP410018	SIP reference: F	RFC 326	1		ISUP reference: 2.5, Q.732 clause 2.5.2.4, Q.731clause 3.5.2.4					
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion									
SIP selection criteria										
ISUP selection criteria	PICS 10/5 AND PICS 1/8	PICS 10/5 AND PICS 1/8								
Test purpose	Redirection number in the Verify that the incoming in number according to the properties Discarding the redirection removes the redirection.	ternation procedur tion nur	nal gateway es defined fondes nber in case	checks and or COLP of bilateral	manipulates the <b>redirection</b> agreements					
SIP parameter values	183 Session Progress: Conumber encapsulated in the	ontent-Ty ne MIME Type: ap	pe: applicati body plication/ISL	on/ISUP; A	CM containing a Redirection					
ISUP parameter values	ACM: Called party status= Generic notification Call diversion inform No Redirection num ANM: No Redirection num	=no indica า mation R mber	ation Redirection re	eason uncor	nditional					
Comments	ISUP		SUT		SIP-I					
	IAM	<b>→</b>		→	INVITE(IAM)					
	ACM(no indication)	+		+	183 Session Progress(ACM)					
	CPG ← 180 Ringing(CPG)									
	ANM									
			Conversa	ion						
	REL	<b>→</b>		<b>→</b>	BYE(REL)					
	RLC	+		+	200 OK BYE(RLC)					

TP410019	SIP reference: RF	C 326	1		ISUP reference: 2.5, Q.732 clause 2.5.2.3, 2.731clause 3.5.2.3					
TSS reference	ISUP-SIP-ISUP/SS/Call Div	ISUP-SIP-ISUP/SS/Call Diversion								
SIP selection criteria										
ISUP selection criteria	PICS 1/7									
Test purpose	Verify that the outgoing inte number according to the process Converting the redirecticode)  1. The PTC will provide the	Redirection number in the outgoing international gateway  Verify that the outgoing international gateway checks and manipulates the redirection  number according to the procedures defined for COLP  Converting the redirection number to national format, if necessary (own country code)  1. The PTC will provide the necessary stimulus  2. ACM with CDInf, GenNot = "call is diverting" and an international RnNb with own CC								
SIP parameter	183 Session Progress: Con	tent-Ty	pe: application/l	SUP; AC	M containing a Redirection					
values	number "International numb	er" enc	apsulated in the	MIME b	ody					
ISUP parameter	ACM: Called party status=n	o indica	ation							
values	Generic notification									
	Call diversion inform			n uncon	ditional					
	Redirection number	"Nation		1	T					
Comments	ISUP		SUT		SIP-I					
	IAM	<b>→</b>		→	INVITE(IAM)					
	ACM(no indication)	+		+	183 Session Progress(ACM)					
	CPG	+		+	180 Ringing(CPG)					
	ANM	<b>←</b>		<b>←</b>	200 OK INVITE(ANM)					
			Conversation							
	REL	<b>→</b>		<b>→</b>	BYE(REL)					
	RLC	+		+	200 OK BYE(RLC)					

TP410020	SIP reference: RI	FC 326	1	(		ISUP reference: 2.5, Q.732 clause 2.5.2.3, 0.731clause 3.5.2.3
TSS reference	ISUP-SIP-ISUP/SS/Call Div	version				
SIP selection criteria						
ISUP selection criteria	PICS 1/8					
Test purpose	Redirection number in the Verify that the incoming into number according to the process Converting the redirect	ernation rocedur	nal gateway es defined f	checks or COL	and n P	nanipulates the <b>redirection</b>
SIP parameter values		tent-Ty	pe: applicat	ion/ISL	JP; AC	M containing a Redirection
ISUP parameter values	ACM: Called party status=r Generic notification Call diversion inform Redirection number	no indica	ation edirection r	eason (		ditional
Comments	ISUP		SUT			SIP-I
	IAM ACM(no indication)	<b>→</b>			<b>→</b>	INVITE(IAM) 183 Session Progress(ACM)
	CPG	+			+	180 Ringing(CPG)
	ANM	+			+	200 OK INVITE(ANM)
			Conversa	tion		
	REL	<b>→</b>			<b>→</b>	BYE(REL)
	RLC	+			+	200 OK BYE(RLC)

TP410021	SIP reference: RF	C 326	1			ISUP reference:				
				C	Q.1912	.5, Q.731 clause 5.5.2.3.1				
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion									
SIP selection										
criteria										
ISUP selection	PICS 1/8 AND PICS 10/6									
criteria										
Test purpose	Redirection number in the									
	Verify that the outgoing inte					nanipulates the <b>redirection</b>				
	number according to the pr									
	Adding a prefix to an inte									
					ith CDI	nf, GenNot = "call is diverting"				
	and an international RnNb v									
SIP parameter	183 Session Progress: Con									
values	number "International numb			n the N	/IME b	ody				
ISUP parameter	ACM: Called party status=n	o indica	ation							
values	Generic notification									
	Call diversion inform				uncon	ditional				
	Redirection number	<u>Numbe</u>		Κ	•	<u></u>				
Comments	ISUP		SUT			SIP-I				
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)				
	ACM(no indication)	+			+	183 Session Progress(ACM)				
	CPG	+			+	180 Ringing(CPG)				
	ANM	<b>←</b>			+	200 OK INVITE(ANM)				
			Conversa	tion						
	REL	<b>^</b>			<b>^</b>	BYE(REL)				
	RLC	+			+	200 OK BYE(RLC)				

## 5.3.11 CONF

TP411001	SIP reference: RFC 3261			Q	reference: .1912.5 clause 1.6.15			
TSS reference	ISUP-SIP-ISUP/SS/CONF							
SIP selection criteria								
ISUP selection criteria								
Test purpose	To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message  1. Assist a call set up from SP A to SPD.  2. Check that the notification "conference established" is received in the CPG from conferee at SPC  3. Check the notification "other party added" in the CPG.  O-MGCF interworking							
SIP parameter values	INFO: Content-Type: application/ISI	JP; CPG	encapsulated	in the I	MIME body			
ISUP parameter	CPG: Generic notification: conferen	ce establi	shed					
values	CPG: Generic notification: other par	ty added						
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
			Conversatio	n				
	CPG(conference established)	→		→	INFO(CPG)			
				+	200 OK INFO			
	CPG(other party added)	<b>→</b>		<b>→</b>	INFO(CPG)			
				+	200 OK INFO			
	DEL				DVE(DEL)			
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	<b>←</b>		←	200 OK BYE(RLC)			

TP411002	SIP reference:	SIP reference: RFC 3261			ISUP reference: Q.1912.5 Q.734 clause 1.6.15			
TSS reference	ISUP-SIP-ISUP/SS/CON	IF						
SIP selection criteria								
ISUP selection criteria								
Test purpose	To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message  1. Assist a call set up from SP A to SPD  2. Check that the notification "conference established" is received in the CPG from conferee at SPC  3. Check the notification "other party added" in the CPG.  I-MGCF interworking							
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body							
ISUP parameter	CPG: Generic notification	n: conf	erence establ	lished				
values	CPG: Generic notification	n: othe	r party added					
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	<b>←</b>		<b>←</b>	ACM			
	200 OK INVITE(ANM)	<b>←</b>		<b>←</b>	ANM			
			Conversatio	n				
	INFO(CPG)	→		→	CPG(conference established)			
	200 OK INFO	+						
	INFO(CPG) → CPG(other party added)							
	200 OK INFO	+						
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP411003	SIP reference: RFC 3261			Q	reference: .1912.5 clause 1.6.15			
TSS reference	ISUP-SIP-ISUP/SS/CONF							
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message  1. Assist a call set up from SPA to SPD  2. Check that the notification "conference established" is received in the CPG from conferee at SPC  3. Check the notification "isolated" in the CPG  O-MGCF interworking							
SIP parameter values	INFO: Content-Type: application/ISUF	; CPG	encapsulated ir	the	MIME body			
ISUP parameter	CPG: Generic notification: conference	establi	shed					
values	CPG: Generic notification: isolated							
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
			Conversation					
	CPG(conference established)	→		<b>→</b>	INFO(CPG)			
				+	200 OK INFO			
	CPG(isolated)	<b>→</b>		→	INFO(CPG)			
				<b>←</b>	200 OK INFO			
	DEL	+		+	DVE(DEL)			
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	<b>←</b>		<b>←</b>	200 OK BYE(RLC)			

TP411004	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.734 clause 1.6.15			
TSS reference	ISUP-SIP-ISUP/SS/CON	lF.		•				
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message  1. Assist a call set up from SPA to SPD  2. Check that the notification "conference established" is received in the CPG from conferee at SPC  3. Check the notification "isolated" in the CPG I-MGCF interworking							
SIP parameter			n/ISUP: CPG	encaps	sulated in the MIME body			
values	, , , , , , , , , , , , , , , , , , , ,		, ,		,			
ISUP parameter	CPG: Generic notification	n: conf	erence establi	shed				
values	CPG: Generic notification	n: isola	ited					
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	<b>←</b>		<b>←</b>				
	200 OK INVITE(ANM)	<b>←</b>		<b>+</b>	ANM			
			Conversation					
	INFO(CPG)	→		→	CPG(conference established)			
	200 OK INFO	+						
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(isolated)			
	200 OK INFO	+						
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP411005	SIP reference: RFC 3261	ISUP reference: Q.1912.5 Q.734 clause 1.6.15						
TSS reference	ISUP-SIP-ISUP/SS/CONF							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message  1. Assist a call set up from SPA to SPD  2. Check that the notification "conference established" is received in the CPG from conferee at SPC  3. Check the notification "reattached" in the CPG  O-MGCF interworking							
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body							
ISUP parameter values	CPG: Generic notification: conferen CPG: Generic notification: isolated CPG: Generic notification: reattache		shed					
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
			Conversation					
	CPG(conference established)	<b>→</b>		<b>→</b>	INFO(CPG)			
				+	200 OK INFO			
	CPG(isolated)	<b>→</b>		<b>→</b>	INFO(CPG)			
				+	200 OK INFO			
	CPG(reattached)	<b>→</b>		<b>→</b>	INFO(CPG)			
				+	200 OK INFO			
	REL	→		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP411006	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.734 clause 1.6.15				
TSS reference	ISUP-SIP-ISUP/SS/CON	IF							
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message  1. Assist a call set up from SPA to SPD  2. Check that the notification "conference established" is received in the CPG from conferee at SPC  3. Check the notification "reattached" in the CPG  I-MGCF interworking								
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body								
ISUP parameter values	CPG: Generic notificatio CPG: Generic notificatio CPG: Generic notificatio	n: isola	ated	shed					
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
	, , , , , , , , , , , , , , , , , , ,		Conversation						
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(conference established)				
	200 OK INFO	+							
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(other party added)				
	200 OK INFO	+							
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(isolated)				
	200 OK INFO	+							
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		<b>←</b>	RLC				

TP411007	SIP reference: RFC 3261	ISUP reference: Q.1912.5 Q.734 clause 1.6.15						
TSS reference	ISUP-SIP-ISUP/SS/CONF							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message  1. Assist a call set up from SPA to SPD  2. Check the notification "other party disconnected" in the CPG  O-MGCF interworking							
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body							
ISUP parameter	CPG: Generic notification: conference		shed					
values	CPG: Generic notification: other part CPG: Generic notification: other part		nected					
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
			Conversation					
	CPG(conference established)	<b>→</b>		<b>^</b>	INFO(CPG)			
				+	200 OK INFO			
	CPG(other party added)	<b>→</b>		<b>→</b>	INFO(CPG)			
				+	200 OK INFO			
	CPG(other party disconnected)	→		<b>→</b>	INFO(CPG)			
				+	200 OK INFO			
	REL	→		<b>→</b>	BYE(REL)			
	RLC	<b>←</b>		<b>←</b>	200 OK BYE(RLC)			

TP411008	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.734 clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/CON	IF.			
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose		n succ	essfully transfer	/deliv	er the required notifications in/from the
	CPG message				
	1. Assist a call set up fro				
	2. Check the notification I-MGCF interworking	"other	party disconned	cted"	in the CPG
SIP parameter	INFO: Content-Type: app	olicatio	n/ISUP; CPG ei	ncaps	sulated in the MIME body
values					•
ISUP parameter	CPG: Generic notification	n: conf	erence establish	ned	
values	CPG: Generic notification	n: othe	r party added		
	CPG: Generic notification	n: othe		ected	
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Conversation		
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(conference established)
	200 OK INFO	+			
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(other party added)
	200 OK INFO	+			
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(other party disconnected)
	200 OK INFO	+			
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	+		+	RLC

TP411009	SIP reference: RFC 3261		Q	reference: .1912.5 :lause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/CONF	Q.1	34 C	siause 1.6.15
SIP selection	1001 011 1001 700/00111			
criteria				
ISUP selection criteria				
Test purpose	To verify that the IUT can successfully transf CPG message 1. Assist a call set up from SPA to SPD 2. Check that the notification "conference es conferee at SPC 3. Release the conference O-MGCF interworking	·		
SIP parameter values	INFO: Content-Type: application/ISUP; CPG	encapsulated in	the I	MIME body
ISUP parameter values	CPG: Generic notification: conference estab	ished		
Comments	ISUP	SUT		SIP-I
	IAM →		<b>→</b>	INVITE(IAM)
	ACM ←		+	180 Ringing(ACM)
	ANM <b>←</b>		<b>←</b>	200 OK INVITE(ANM)
		Conversation		
	CPG(conference established) →		<b>→</b>	INFO(CPG)
			+	200 OK INFO
	REL →		<b>→</b>	BYE(REL)
	RLC +		<b>←</b>	200 OK BYE(RLC)

TP411010	SIP reference:	RFC :	3261		ISUP reference: Q.1912.5 Q.734 clause 1.6.15			
TSS reference	ISUP-SIP-ISUP/SS/CON	F						
SIP selection criteria								
ISUP selection criteria								
Test purpose	To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message  1. Assist a call set up from SPA to SPD  2. Check that the notification "conference established" is received in the CPG from conferee at SPC  3. Release the conference I-MGCF interworking							
SIP parameter values	INFO: Content-Type: app	olicatio	n/ISUP; CPG	encaps	sulated in the MIME body			
ISUP parameter values	CPG: Generic notification	n: conf	erence establi	shed				
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	<b>←</b>		+	ACM			
	200 OK INVITE(ANM)	<b>←</b>		+	ANM			
			Conversation	า				
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(conference established)			
	200 OK INFO	+						
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

## 5.3.12 ECT

TP412001	SIP reference: RFC 3261			ı	SUP reference: Q.1912.5			
			Q	.732.7	7 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 additional calling party number in the call transfer number Verify that the IUT is able to store the additional calling party number in the generic number when the calling party number and the generic number have been received from the remote user. This information is sent by the IUT to the other remote user in the call transfer number in either the FAC or CPG when the call transfer is activated. O-MGCF interworking							
SIP parameter values	INFO: Content-Type: application/ISL	IP; FAC	encapsula	ted in	the MIME body			
ISUP parameter values	FAC: Generic notification=call transfe	er activ	e, Call trans	sfer nu	ımber (PIXIT)			
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
	Conversation							
	FAC(call transfer active, CTNb)	<b>→</b>		<b>→</b>	INFO(FAC)			
				+	200 OK INFO			
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	<b>←</b>		<b>←</b>	200 OK BYE(RLC)			

TP412002	SIP reference: R	FC 326	1		ISUP reference: Q.1912.5 Q.734 clause 1.6.15			
TSS reference	ISUP-SIP-ISUP/SS/ECT							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Capability of sending the additional calling party number in the call transfer number Verify that the IUT is able to store the additional calling party number in the generic number when the calling party number and the generic number have been received from the remote user. This information is sent by the IUT to the other remote user in the call transfer number in either the FAC or CPG when the call transfer is activated. I-MGCF interworking							
SIP parameter values	INFO: Content-Type: appli	cation/IS	SUP; FAC en	capsu	lated in the MIME body			
ISUP parameter values	FAC: Generic notification=	call tran	sfer active, C	all tra	nsfer number (PIXIT			
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
		(	Conversatio	n				
	INFO(FAC)	<b>→</b>		<b>→</b>	FAC(call transfer active, CTNb)			
	200 OK INFO	+						
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP412005	SIP reference: RFC 3261		Q.73	ISUP reference: Q.1912.5 2.7 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 storing and sending	the add	ditional conne	cted number in the call
	transfer number			
	Verify that the IUT is able to store the			
	when the <b>connected number</b> and t			
	remote user. This information is sen			
	transfer number in either the FAC	or <b>CPG</b>	when the call t	ransfer is activated.
	O-MGCF interworking			
SIP parameter	INFO: Content-Type: application/ISI			
values	INFO: Content-Type: application/ISI			in the MIME body
ISUP parameter	CPG: Event indicator=progress, Ger			
values	FAC: Generic notification=call transf	fer activ		
Comments	ISUP		SUT	SIP-I
	IAM	→	-	
	ACM	+	•	
	ANM	+	•	200 OK INVITE(ANM)
			onversation	
	CPG(hold)	→	-	
			•	200 OK INVITE(recvonly)
			-	ACK
	FAC(call transfer active, CTNb)	→	-	INFO(FAC)
			•	- 200 OK INFO
			-	INVITE(sendrecv)
			•	
			-	` ,
	REL	<b>→</b>	-	BYE(REL)
	RLC	+	•	. ,

TP412006	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.734 clause 1.6.15			
TSS reference	ISUP-SIP-ISUP/SS/CONF							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose		endin	g the additi	onal c	onnected number in the call			
	transfer number							
					ected number in the generic number			
					er have been received from the			
					other remote user in the call			
	transfer number in either th	e FAC	or <b>CPG</b> wh	en the	call transfer is activated.			
OID	I-MGCF interworking	. ,.			1 . 1:			
SIP parameter		INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body						
values	INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body							
ISUP parameter	CPG: Event indicator=progre							
values	FAC: Generic notification=ca	all trans		Call trai				
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		<b>←</b>	ACM			
	200 OK INVITE(ANM)	+		<b>←</b>	ANM			
			Conversation					
	INVITE(CPG, sendonly)	→		→	CPG(hold)			
	200 OK INVITE(recvonly)	+						
	ACK	<b>→</b>						
	INFO(FAC)	<b>→</b>		<b>→</b>	FAC(call transfer active, CTNb)			
	200 OK INFO	+						
	INVITE(sendrecv)	<b>→</b>						
	200 OK INVITE(sendrecv)	+						
	ACK	<b>→</b>						
	- 7.7							
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	<del>/</del>		<del>/</del>	RLC			
	1200 OK DIL(INLO)	•			INCO			

TP412009	SIP reference: RFC 3261		Į;	SUP reference: Q.1912.5	
			Q.	732.7	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 - initiati				
	Verify that the local exchange controlli	ng the	ECT can su	ucces	sfully initiate the loop
	prevention procedure by sending LOP			ition	indicator set to "request"
	and with <b>call transfer reference</b> for b	otn ca	IIIS.		
CID noromotor	O-MGCF interworking INFO: Content-Type: application/ISUF	· CDC	` anaonaulat	ما ام	the MIME hady
SIP parameter values	INFO: Content-Type: application/ISUF				
values	INFO: Content-Type: application/ISUF				
ISUP parameter	CPG: Event indicator=progress, Gene				the Milvie Body
values	LOP: request: Call transfer reference	110 110	incation=noi	u	
Value	LOP: response: Call transfer reference	ż			
	FAC: Generic notification=call transfer		e. Call transf	er nu	mber(PIXIT)
Comments	ISUP		SUT		SIP-I
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
		С	onversation	า	, ,
	CPG(hold)	<b>→</b>		<b>→</b>	INVITE(CPG, sendonly)
				<b>←</b>	200 OK INVITE(recvonly)
				<b>→</b>	ACK
	LOP(request)	<b>→</b>		<b>→</b>	INFO(LOP)
				+	200 OK INFO
	LOP(response)	+		+	INFO(LOP)
				<b>→</b>	200 OK INFO
	FAC(call transfer active, CTNb)	<b>→</b>		<b>→</b>	INFO(FAC)
				+	200 OK INFO
				<b>→</b>	INVITE(sendrecv)
				+	200 OK INVITE(sendrecv)
				<b>→</b>	ACK
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC(RLC)	<b>←</b>		<b>←</b>	200 OK BYE

TP412010	SIP reference: RFC 3261			ISUP reference: Q.1912.5 Q.734 clause 1.6.15					
TSS reference	ISUP-SIP-ISUP/SS/ECT								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Loop prevention procedure								
					successfully initiate the loop				
					ention indicator set to "request"				
	and with call transfer refere	nce fo	or both calls	i.					
	I-MGCF interworking								
SIP parameter	INFO: Content-Type: applica	tion/IS	SUP; CPG e	encapsu	lated in the MIME body				
values	INFO: Content-Type: applica								
	INFO: Content-Type: applica								
ISUP parameter	CPG: Event indicator=progre			cation=r	nold				
values	LOP: request: Call transfer re								
	LOP: response: Call transfer reference FAC: Generic notification=call transfer active, Call transfer number(PIXIT)								
Commonto	SIP-I	ıı trans		Call trar	ISUP				
Comments		<b>→</b>	SUT						
	INVITE(IAM)	<del>  7</del>		<b>→</b>	IAM				
	180 Ringing(ACM)	<del>-</del>		<del>-</del>	ACM				
	200 OK INVITE(ANM)		 Conversati	•	ANM				
	INIVITE (ODO dauly)	<b>→</b>	Conversati		ODO(11-1)				
	INVITE(CPG, sendonly)	<del>  7</del>		→	CPG(hold)				
	200 OK INVITE(recvonly)	<b>→</b>							
	ACK	<del>  7</del>							
	INEC(LOD)	<b>→</b>			I OD(========+)				
	INFO(LOP)	<del>  7</del>		→	LOP(request)				
	200 OK INFO								
	INEO(LOD)	+		+	I OD(respects)				
	INFO(LOP)	<b>→</b>			LOP(response)				
	200 OK INFO	<del>                                     </del>							
	INEO/EAC)	<del></del>		_	EAC(call transfer active CTNIh)				
	INFO(FAC) 200 OK INFO	<b>→</b>		<b>→</b>	FAC(call transfer active, CTNb)				
	ZOU OK IINFO	+							
	INVITE(sendrecv)	<b>→</b>							
	200 OK INVITE(sendrecv)	<del></del>							
	ACK	<b>→</b>							
	ACK	+ <b>-</b> -							
	BYE(REL)	<b>→</b>			REL				
		<del>  7</del>		→ ←					
	200 OK BYE(RLC)		l	7	RLC				

TP412011	SIP reference: RFC 3261			-	SUP reference: Q.1912.5			
T00 (	NOUR OID IOUR/OO/FOT		Q.7	32.7	' 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 local exchange controlling the ECT rejects the call transfer if no LOP is received within T <sub>ECT</sub> expiry.  O-MGCF interworking							
SIP parameter values	INFO: 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, General LOP: request: Call transfer reference				·			
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	<b>4</b>		<b>←</b>	180 Ringing(ACM)			
	ANM	<b>4</b>		<b>←</b>	200 OK INVITE(ANM)			
		O	onversation					
	CPG(hold)	1		<b>→</b>	INVITE(CPG, sendonly)			
				<b>←</b>	200 OK INVITE(recvonly)			
				<b>→</b>	ACK			
	LOP(request)	<b>→</b>		<b>→</b>	INFO(LOP)			
				<b>←</b>	200 OK INFO			
	REL	<b>→</b>		<u>→</u>	BYE(REL)			
	RLC	<b>←</b>		<b>←</b>	200 OK BYE(RLC)			

TP412012	SIP reference: RF	C 326	1	ISUP reference: Q.1912.5 Q.734 clause 1.6.15		
TSS reference	ISUP-SIP-ISUP/SS/CONF					
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Loop prevention procedure					
			ontrolling the E	CT i	rejects the call transfer if no LOP is	
	received within T <sub>ECT</sub> expiry.					
	I-MGCF interworking					
SIP parameter	INFO: Content-Type: applica					
values	INFO: Content-Type: applica	ation/IS	SUP; LOP enca	ıpsu	lated in the MIME body	
ISUP parameter	CPG: Event indicator=progre			on=l	hold	
values	LOP: request: Call transfer re	<u>eferen</u>				
Comments	SIP-I		SUT		ISUP	
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM	
	180 Ringing(ACM)	+		+	ACM	
	200 OK INVITE(ANM)	+		<b>←</b>	ANM	
			Conversation			
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(hold)	
	200 OK INVITE(recvonly)	+				
	ACK	→				
	INFO(LOP)	<b>→</b>		<b>→</b>	LOP(request)	
	200 OK INFO	+				
	BYE(REL)	<b>→</b>		<b>→</b>	REL	
	200 OK BYE(RLC)	<b>←</b>		<b>←</b>	RLC	

TP412013	SIP reference: RFC 3261		Q.7		SUP reference: Q.1912.5 ' clause 7.5.2.1.1.1 a)				
TSS reference	ISUP-SIP-ISUP/SS/ECT		•		,				
SIP selection criteria									
ISUP selection criteria									
Test purpose									
SIP parameter	INFO: Content-Type: application/ISL	JP; CPG	encapsulate	ed in	the MIME body				
values	INFO: Content-Type: application/ISUINFO: Content-Type: application/ISU	INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body INFO: Content-Type: application/ISUP; LOP encapsulated in the MIME body							
ISUP parameter values	LOP: request: Call transfer reference FAC: Generic notification=call transf	CPG: Event indicator=progress, Generic notification=hold LOP: request: Call transfer reference FAC: Generic notification=call transfer active, Call transfer number(PIXIT)							
Comments	ISUP		SUT		SIP-I				
	IAM	→		<b>→</b>	INVITE(IAM)				
	ACM	<b>←</b>		<b>←</b>	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
	Conversation								
	CPG(hold)	<b>→</b>		<b>→</b>	INVITE(CPG, sendonly)				
				<b>←</b>	200 OK INVITE(recvonly)				
				<b>→</b>	ACK				
	LOP(request)	<b>→</b>		<b>→</b>	INFO(LOP)				
	zor (roquest)			<del>-</del>	200 OK INFO				
	FAC(call transfer active, CTNb)	<b>→</b>		<b>→</b>	INFO(FAC)				
				<del>-</del>	200 OK INFO				
				<b>→</b>	INVITE(sendrecv)				
				+	200 OK INVITE(sendrecv)				
				<b>→</b>	ACK				
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	<del>-</del>		<del>7</del>	200 OK BYE(RLC)				
	INLO			_	ZUU UN BIE(NLU)				

TP412014	SIP reference: RF	SIP reference: RFC 3261			ISUP reference: Q.1912.5 Q.734 clause 1.6.15				
TSS reference	ISUP-SIP-ISUP/SS/ECT	ISUP-SIP-ISUP/SS/ECT							
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Verify that the local exchang	Loop prevention procedure - successful on timer expiry  Verify that the local exchange controlling the ECT completes the call transfer if no LOP is received within T <sub>ECT</sub> expiry.							
SIP parameter		ation/IS	UP: CPG enc	ansu	lated in the MIME body				
values	INFO: Content-Type: applica INFO: Content-Type: applica	INFO: Content-Type: application/ISUP; CPG 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			ion=ł	nold				
values	LOP: request: Call transfer r								
_	FAC: Generic notification=ca	all trans		ll trar					
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		<b>←</b>	ANM				
			Conversation						
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(hold)				
	200 OK INVITE(recvonly)	+							
	ACK	<b>→</b>							
	INFO(LOP)	<b>→</b>		<b>→</b>	LOP(request)				
	200 OK INFO	+							
	INFO(FAC)	<b>→</b>		<b>→</b>	FAC(call transfer active, CTNb)				
	200 OK INFO	+							
	INVITE(sendrecv)	<b>→</b>							
	200 OK INVITE(sendrecv)	+							
	ACK	<b>→</b>							
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		<b>←</b>	RLC				

TP412015	SIP reference: RFC 3261	0.7	ISUP reference: Q.1912.5 Q.732.7 clause 7.5.2.1.1.1 a)					
TSS reference	ISUP-SIP-ISUP/SS/ECT		α.,	02.7	Olduse 7.5.2.1.1.1 uj			
SIP selection	1001 011 1001 7007201							
criteria								
ISUP selection								
criteria								
Test purpose	Facility message with generic noti	ficatio	n sent to the	ren	note user			
	Verify that the local exchange contro							
	by sending FAC with the generic no							
	alerting" and the service activation							
SIP parameter	INFO: Content-Type: application/ISU							
values	INFO: Content-Type: application/ISU				the MIME body			
ISUP parameter	CPG: Event indicator=progress, Gen	eric no	tification=hole	d				
values	FAC: Generic notification=call transfe	er activ		er nu				
Comments	ISUP		SUT		SIP-I			
	IAM	→		<b>→</b>	INVITE(IAM)			
	ACM	<b>←</b>		+	180 Ringing(ACM)			
	ANM	<b>←</b>		+	200 OK INVITE(ANM)			
			onversation	1				
	CPG(hold)	→		<b>→</b>	INVITE(CPG, sendonly)			
				+	200 OK INVITE(recvonly)			
				<b>→</b>	ACK			
	FAC(call transfer active, CTNb)	→		<b>→</b>	INFO(FAC)			
				+	200 OK INFO			
				<b>→</b>	INVITE(sendrecv)			
				+	200 OK INVITE(sendrecv)			
				<b>→</b>	ACK			
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP412016	SIP reference: RFC 3261			ISUP reference: Q.1912.5 Q.734 clause 1.6.15					
TSS reference	ISUP-SIP-ISUP/SS/ECT	ISUP-SIP-ISUP/SS/ECT							
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Facility message with gene								
					successfully initiate a call transfer				
					call transfer, active" or "call transfer,				
					"call transfer". O-MGCF interworking				
SIP parameter	INFO: Content-Type: applicat								
values	INFO: Content-Type: applicat								
ISUP parameter	CPG: Event indicator=progre								
values	FAC: Generic notification=cal	l tran		Call trar					
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	<b>←</b>		<b>←</b>	ACM				
	200 OK INVITE(ANM)	<b>←</b>		←	ANM				
			Conversation						
	INVITE(CPG, sendonly)	→		→	CPG(hold)				
	200 OK INVITE(recvonly)	<b>←</b>							
	ACK	<b>→</b>							
	INFO(FAC)	→		→	FAC(call transfer active, CTNb)				
	200 OK INFO	<b>←</b>							
	INVITE(sendrecv)	<b>→</b>							
	200 OK INVITE(sendrecv)	+							
	ACK	<b>→</b>							
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	<b>←</b>		+	RLC				

TP412017	SIP reference: RFC 3261			•	SUP reference: Q.1912.5		
			Q	.732.	7 clause 7.5.2.1.1.1 a)		
TSS reference	ISUP-SIP-ISUP/SS/ECT						
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Call progress message with gener						
	Verify that the local exchange (controlling the ECT) can successfully initiate a call transfer by sending <b>CPG</b> with the <b>generic notification</b> set to "call transfer, active" and the <b>service activation</b> parameter set to "call transfer".  O-MGCF interworking						
SIP parameter values	INFO: Content-Type: application/ISL	JP; CPC	encapsula	ated in	the MIME body		
ISUP parameter values	CPG: Generic notification=call transf	er activ	e, Call trans	sfer n	umber (PIXIT)		
Comments	ISUP		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	CPG(call transfer active, CTNb)	<b>→</b>		<b>→</b>	INFO(CPG)		
				+	200 OK INFO		
	ANM	<b>←</b>		+	200 OK INVITE(ANM)		
		C	onversatio	n			
	REL	<b>→</b>		<b>→</b>	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

TP412018	SIP reference: RFC 3261			ISUP reference: Q.1912.5 Q.734 clause 1.6.15				
TSS reference	ISUP-SIP-ISUP/SS/ECT	ISUP-SIP-ISUP/SS/ECT						
SIP selection criteria								
ISUP selection criteria								
Test purpose	Call progress message with generic notification sent to the remote user  Verify that the local exchange (controlling the ECT) can successfully initiate a call transfer by sending CPG with the generic notification set to "call transfer, active" and the service activation parameter set to "call transfer".  I-MGCF interworking							
SIP parameter values	INFO: Content-Type: applica	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body						
ISUP parameter values	CPG: Event indicator=progre CPG: Generic notification=c							
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	,	(	Conversation					
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(hold)			
	200 OK INVITE(recvonly)	+			,			
	ACK	<b>→</b>						
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(call transfer active, CTNb)			
	200 OK INFO	+						
	INVITE(sendrecv)	<b>→</b>						
	200 OK INVITE(sendrecv) ACK	<b>←</b>						
	ACK	7						
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	<b>←</b>		<b>←</b>	RLC			

TP412019	SIP reference: RFC 3261			ISUP reference: Q.1912.5 Q.734 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 FAC or CPG before							
	sending it to the next exchange, if its indicator is set to "presentation restricted" and the is no bilateral agreement to transfer the number							
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC(CTNb=restricted) encapsulated in the MIME body							
ISUP parameter	CPG: Event indicator=progre							
values	FAC: Generic notification=call transfer active, no Call transfer number(PIXIT)							
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		<b>←</b>	ANM			
			Conversation	on				
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(hold)			
	200 OK INVITE(recvonly)	<b>←</b>						
	ACK	<b>→</b>						
	INFO(FAC)	<b>→</b>		<b>→</b>	FAC(call transfer active)			
	200 OK INFO	+						
	INVITE(sendrecv)	<b>→</b>						
	200 OK INVITE(sendrecv)	+						
	ACK	<b>→</b>						
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		<b>←</b>	RLC			

TP412020	SIP reference: RFC 3261			ISUP reference: Q.1912.5 Q.734 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 to international format. The nature of address indicator shall be set to "international number".							
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC(CTNb=national) encapsulated in the MIME body							
ISUP parameter values	CPG: Event indicator=progress, Generic notification=hold FAC: Generic notification=call transfer active, Call transfer number=international(PIXIT)							
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	Conversation							
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(hold)			
	200 OK INVITE(recvonly)	+						
	ACK	<b>→</b>						
	INFO(FAC)	<b>→</b>		<b>→</b>	FAC(call transfer active, CTNb)			
	200 OK INFO	+						
	INVITE(sendrecv)	<b>→</b>						
	200 OK INVITE(sendrecv)	+						
	ACK	<b>→</b>						
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP412021	SIP reference: RFC 3261	Q.7	ISUP reference: Q.1912.5 Q.732.7 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						
	<b>number</b> if it is the network's own country code. The nature of address indicator shall be set to "national (significant) number"						
SIP parameter	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body						
values	INFO: Content-Type: application/ISUP; FAC(CTNb=international) encapsulated in the MIME body						
ISUP parameter	CPG: Event indicator=progress, Generic notification=hold						
values	FAC: Generic notification=call trans	fer active	e, Call transfe	er nu	umber=national(PIXIT)		
Comments	ISUP		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
	Conversation						
	CPG(hold)	<b>→</b>		<b>→</b>	INVITE(CPG, sendonly)		
				<b>←</b>	200 OK INVITE(recvonly)		
				<b>→</b>	ACK		
	FAC(call transfer active, CTNb)	<b>→</b>		<b>→</b>	INFO(FAC)		
				+	200 OK INFO		
				<b>→</b>	INVITE(sendrecv)		
				<b>←</b>	200 OK INVITE(sendrecv)		
				<b>→</b>	ACK		
	REL	<b>→</b>		<b>→</b>	BYE(REL)		
	RLC	+		<b>←</b>	200 OK BYE(RLC)		

TP412022	SIP reference: RFC 3261	ISUP reference: Q.1912.5 Q.732.7 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 receive and re-send the sub-address in the access transport parameter in the FAC message in either direction after activating the call transfer service. These are the calling sub-address for incoming calls and the connected sub-address for outgoing calls.  O-MGCF interworking						
SIP parameter values	INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body						
ISUP parameter values	FAC: Generic notification=call transfer active, Call transfer number(PIXIT) FAC: ATP contained the connected sub address						
Comments	ISUP		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
		C	Conversation				
	FAC(call transfer active, CTNb)	<b>→</b>		<b>→</b>	INFO(FAC)		
				+	200 OK INFO		
	FAC(ATP=SUB)	+		+	INFO(FAC)		
				<b>→</b>	200 OK INFO		
	FAC(ATP=SUB)	<b>→</b>		<b>→</b>	INFO(FAC)		
				+	200 OK INFO		
	REL	<b>→</b>		<b>→</b>	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

SIP selection criteria ISUP selection criteria Test purpose t t SIP parameter	sub-address for outgoing calls.  O-MGCF interworking  INFO: Content-Type: application  CPG: Event indicator=progress  FAC: Generic notification=call to	call on/IS	essage in eithing sub-addre  GUP; FAC enceneric notificat	er dir ss for apsul	rection after activating the call r incoming calls and the connected lated in the MIME body				
criteria ISUP selection criteria Test purpose t t SIP parameter	Verify that if the IUT is able to retransport parameter in the FA transfer service. These are the sub-address for outgoing calls. O-MGCF interworking INFO: Content-Type: application CPG: Event indicator=progress FAC: Generic notification=call to	call on/IS	essage in eithing sub-addre  GUP; FAC enceneric notificat	er dir ss for apsul	rection after activating the call r incoming calls and the connected lated in the MIME body				
ISUP selection criteria Test purpose t t SIP parameter	Verify that if the IUT is able to retransport parameter in the FA transfer service. These are the sub-address for outgoing calls. O-MGCF interworking INFO: Content-Type: application CPG: Event indicator=progress FAC: Generic notification=call to	call on/IS	essage in eithing sub-addre  GUP; FAC enceneric notificat	er dir ss for apsul	rection after activating the call r incoming calls and the connected lated in the MIME body				
criteria Test purpose t t SIP parameter	Verify that if the IUT is able to retransport parameter in the FA transfer service. These are the sub-address for outgoing calls. O-MGCF interworking INFO: Content-Type: application CPG: Event indicator=progress FAC: Generic notification=call to	call on/IS	essage in eithing sub-addre  GUP; FAC enceneric notificat	er dir ss for apsul	rection after activating the call r incoming calls and the connected lated in the MIME body				
Test purpose E t t S SIP parameter	Verify that if the IUT is able to retransport parameter in the FA transfer service. These are the sub-address for outgoing calls. O-MGCF interworking INFO: Content-Type: application CPG: Event indicator=progress FAC: Generic notification=call to	call on/IS	essage in eithing sub-addre  GUP; FAC enceneric notificat	er dir ss for apsul	rection after activating the call r incoming calls and the connected lated in the MIME body				
t t s (	Verify that if the IUT is able to retransport parameter in the FA transfer service. These are the sub-address for outgoing calls. O-MGCF interworking INFO: Content-Type: application CPG: Event indicator=progress FAC: Generic notification=call to	call on/IS	essage in eithing sub-addre  GUP; FAC enceneric notificat	er dir ss for apsul	rection after activating the call r incoming calls and the connected lated in the MIME body				
SIP parameter	NFO: Content-Type: application  CPG: Event indicator=progress FAC: Generic notification=call to	s, Ge trans	eneric notificat	ion=l	nold				
	CPG: Event indicator=progress FAC: Generic notification=call t	s, Ge trans	eneric notificat	ion=l	nold				
values	FAC: Generic notification=call t	trans							
	FAC: Generic notification=call t	trans							
				ıll trar	ister number(PIXII)				
	FAC: ATP contained the connected sub address								
	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
_	Conversation								
l li	INVITE(sendonly,CPG hold)	<b>←</b>		+	CPG(hold)				
	200 OK INVITE(recvonly)	<b>→</b>							
	ACK	<b>←</b>							
 	INITO/FAC)	<b>→</b>		→	FAC(call transfer active, CTNb)				
	INFO(FAC) 200 OK INFO	<b>→</b>		7	FAC(call transfer active, CTNb)				
	200 OK INFO	_							
	INFO(FAC)	+		+	FAC(ATP=SUB)				
	200 OK INFO	<u>`</u>		+	17.0(7.11 =000)				
	200 010 1141 0	<del>-</del>		+					
	INFO(FAC)	<b>→</b>		<b>→</b>	FAC(ATP=SUB)				
	200 OK INFO	<del>′</del>		Ť	1.7.0, =000,				
	Conversation								
	BYE(REL)	<b>→</b>		→	REL				
	200 OK BYE(RLC)	+		+	RLC				

## 5.3.13 3PTY

TP413001	SIP reference: RFC 3261			_	SUP reference: Q.1912.5				
				Q.734	4.2 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 (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 sh 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 values	INFO: Content-Type: application/ISUP; C	CPG	encapsulat	ed in	the MIME body				
ISUP parameter	CPG: Event indicator=progress, Generic	not	fication=ho	ld					
values	CPG: Event indicator=progress, Generic	not	fication=co	nferer	nce established				
Comments	ISUP		SUT		SIP-I				
	IAM -	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM	-		+	180 Ringing(ACM)				
	ANM	<b>(</b>		+	200 OK INVITE(ANM)				
		С	onversatio	n					
	CPG(hold)	<b>→</b>		<b>→</b>	INVITE(CPG, sendonly)				
				+	200 OK INVITE(recvonly)				
				<b>→</b>	ACK				
	CPG(conference established)	<b>→</b>		<b>→</b>	INVITE(CPG, sendrecv)				
				+	200 OK INVITE(sendrecv)				
				<b>→</b>	ACK				
		С	onversatio	n					
	REL -	<b>&gt;</b>		<b>→</b>	BYE(REL)				
	RLC •	<b>(</b>		+	200 OK BYE(RLC)				

TP413002	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.734.2 clause 2.4; 2.2.1				
TSS reference	ISUP-SIP-ISUP/SS/3PTY								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	(remote held user) to a three accordingly The IUT should send a CPG	with two active calls is located, can successfully join this call ee-way conversation, and notify the implied remote party <b>PG</b> message with the <b>generic notification indicator</b> set to both implied parties. The <b>event indicator</b> in the <b>CPG</b> should							
SIP parameter	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body								
values	7		,	•	,				
ISUP parameter		G: Event indicator=progress, Generic notification=hold							
values	CPG: Event indicator=progre	ess, Ge		cation=					
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		<b>←</b>	ACM				
	200 OK INVITE(ANM)	<b>←</b>		+	ANM				
			onversati						
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(hold)				
	200 OK INVITE(recvonly)	+							
	ACK	<b>→</b>							
	INVITE(CPG, sendrecv)	<b>→</b>		<b>→</b>	CPG(conference established)				
	200 OK INVITE(sendrecv)	<del>-</del>							
	ACK	<b>→</b>							
		0	onversati	on					
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP413003	SIP reference: RFC 3261			_	SUP reference: Q.1912.5 4.2 clause 2.4; 2.2.1				
TSS reference	ISUP-SIP-ISUP/SS/3PTY								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Served user initiates 3PTY  Verify that the served user with two active calls is located, can successfully join this call (remote active user) to a three-way conversation, and notify the implied remote party accordingly  The IUT should send a CPG message with the generic notification indicator set to "conference established" to both implied parties. The event indicator in the CPG should be set to "progress"  1. Setup a call to user B  2. establish a conference  O-MGCF interworking								
SIP parameter values	INFO: Content-Type: application/ISUP;	CPG	encapsulat	ed in	the MIME body				
ISUP parameter values	CPG: Event indicator=progress, Generic	c not	ification=cor	nferer	nce established				
Comments	ISUP		SUT		SIP-I				
	IAM	<b>\</b>		<b>→</b>	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	<b>←</b>		<b>←</b>	200 OK INVITE(ANM)				
	Conversation								
	CPG(conference established)	<b>→</b>		<b>→</b>	INFO(CPG)				
				<b>←</b>	200 OK INFO				
		С	onversatio	n					
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	<b>←</b>		<b>←</b>	200 OK BYE(RLC)				

TP413004	SIP reference: R	RFC 3261	I		ISUP reference: Q.1912.5 Q.734.2 clause 2.4; 2.2.1			
TSS reference	ISUP-SIP-ISUP/SS/3PTY							
SIP selection criteria								
ISUP selection criteria								
Test purpose	(remote active user) to a thaccordingly The IUT should send a CP	ser with two active calls is located, can successfully join this call a three-way conversation, and notify the implied remote party  CPG message with the generic notification indicator set to "to both implied parties. The event indicator in the CPG should						
SIP parameter values	INFO: Content-Type: appli	cation/IS	SUP; CPG enca	apsı	lated in the MIME body			
ISUP parameter values	CPG: Event indicator=proc	gress, Ge	eneric notificati	ion=	conference established			
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		1	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	<b>←</b>		+	ANM			
		(	Conversation					
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(conference established)			
	200 OK INFO	<b>←</b>						
		(	Conversation					
	BYE(REL)	<b>→</b>		1	REL			
	200 OK BYE(RLC)	<b>←</b>		<b>←</b>	RLC			

C.1912.5	TP413005	SIP reference: RFC 3261			ı	SUP reference:					
SSIP selection criteria				Q	.734.						
SIP selection criteria  Test purpose  Served user creates a private communication with a remote user  Verify that the IUT (controlling the conference) on a 3PTY call can successfully create private communication with the active-held user. The appropriate notification is sent in CPG messages to the user  O-MGCF interworking  SIP parameter values  SIP parameter values  CPG 1, 4: Event indicator=progress, Generic notification=hold  CPG 5: Event indicator=progress, Generic notification=conference established  CPG 5: Event indicator=progress, Generic notification=conference disconnected  CPG 3: Event indicator=progress, Generic notification=conference disconnected  SIP   SUP   SUP   SIP-1  IAM       SUP   SIP-1  IAM           SUT   SIP-1  IAM                   SUT   SIP-1  IAM	TSS reference	ISUP-SIP-ISUP/SS/3PTY									
Suparameter values   Served user creates a private communication with a remote user	SIP selection										
Served user creates a private communication with a remote user	criteria										
Test purpose    Served user creates a private communication with a remote user	ISUP selection										
Verify that the IUT (controlling the conference) on a 3PTY call can successfully create private communication with the active-held user. The appropriate notification is sent in CPG messages to the user	criteria										
SUP parameter values   CPG 1, 4: Event indicator=progress, Generic notification=hold   CPG 5: Event indicator=progress, Generic notification=retrieve   CPG 2: Event indicator=progress, Generic notification=conference established   CPG 3: Event indicator=progress, Generic notification=conference disconnected   SUP	Test purpose	Verify that the IUT (controlling the corprivate communication with the active CPG messages to the user	nferenc	e) on a 3PT	Y cal	I can successfully create					
values         CPG 5: Event indicator=progress, Generic notification=retrieve         CPG 2: Event indicator=progress, Generic notification=conference established           CPG 3: Event indicator=progress, Generic notification=conference disconnected           Comments         ISUP         SUT         SIP-I           IAM         →         →         INVITE(IAM)           ACM         ←         ←         180 Ringing(ACM)           ANM         ←         ←         200 OK INVITE(ANM)           CPG 1(hold)         →         INVITE(CPG, sendonly)           ←         200 OK INVITE(recvonly)           →         ACK           CPG 2(conference established)         →         INVITE(CPG, sendrecv)           ←         200 OK INVITE(sendrecv)           ←         200 OK INVITE(sendrecv)           ←         200 OK INVITE(cPG, sendonly)           ←         200 OK INVITE(cPG, sendrecv)           ←         200 OK INVITE(sendrecv)           →         ACK           CPG 5(retrieve)         →         →           CPG 6(conference established)         →         INFO(CPG)           ←         200 OK INVITE(sendrecv)           →         ACK           Conversation         ←         200 OK INFO <td></td> <td>INFO: Content-Type: application/ISUF</td> <td>P; CPG</td> <td>encapsulate</td> <td>ed in</td> <td>the MIME body</td>		INFO: Content-Type: application/ISUF	P; CPG	encapsulate	ed in	the MIME body					
values         CPG 5: Event indicator=progress, Generic notification=retrieve         CPG 2: Event indicator=progress, Generic notification=conference established           CPG 3: Event indicator=progress, Generic notification=conference disconnected           Comments         ISUP         SUT         SIP-I           IAM         →         →         INVITE(IAM)           ACM         ←         ←         180 Ringing(ACM)           ANM         ←         ←         200 OK INVITE(ANM)           CPG 1(hold)         →         INVITE(CPG, sendonly)           ←         200 OK INVITE(recvonly)           →         ACK           CPG 2(conference established)         →         INVITE(CPG, sendrecv)           ←         200 OK INVITE(sendrecv)           ←         200 OK INVITE(sendrecv)           ←         200 OK INVITE(cPG, sendonly)           ←         200 OK INVITE(cPG, sendrecv)           ←         200 OK INVITE(sendrecv)           →         ACK           CPG 5(retrieve)         →         →           CPG 6(conference established)         →         INFO(CPG)           ←         200 OK INVITE(sendrecv)           →         ACK           Conversation         ←         200 OK INFO <td>ISUP parameter</td> <td>CPG 1, 4: Event indicator=progress, 0</td> <td>Generi</td> <td>c notification</td> <td>=hold</td> <td>t</td>	ISUP parameter	CPG 1, 4: Event indicator=progress, 0	Generi	c notification	=hold	t					
CPG 2: Event indicator=progress, Generic notification=conference established CPG 3: Event indicator=progress, Generic indication=conference disconnected Comments   SUP											
SUP   IAM   →   INVITE(IAM)   IAW   ACM   ←   ←   ←   ←   ←   ←   ←   ←   ←		CPG 2: Event indicator=progress, Ge	neric n	otification=c	onfer	ence established					
IAM			neric n	otification=c	onfei	ence disconnected					
ACM ANM ←	Comments	ISUP		SUT		SIP-I					
ANM  Conversation  CPG 1(hold)  NVITE(CPG, sendonly)  ACK  CPG 2(conference established)  ACK  CPG 3(conference disconnected)  CPG 3(conference disconnected)  CPG 4(hold)  CPG 4(hold)  CPG 5(retrieve)  CPG 5(retrieve)  CPG 6(conference established)  CONVERSATION  CONVERSATION  CONVERSATION  CONVERSATION  CONVERSATION  CONVERSATION  CONVERSATION  BYE(REL)		IAM			<b>→</b>						
Conversation         Conversation           CPG 1(hold)         →         INVITE(CPG, sendonly)           ←         200 OK INVITE(recvonly)         →           ACK         →         INVITE(CPG, sendrecv)           ←         200 OK INVITE(sendrecv)           →         →         ACK           CPG 3(conference disconnected)         →         INFO(CPG)           ←         200 OK INFO           CPG 4(hold)         →         INVITE(CPG, sendonly)           ←         200 OK INVITE(recvonly)           →         ACK           CPG 5(retrieve)         →         INVITE(CPG, sendrecv)           ←         200 OK INVITE(sendrecv)           →         ACK           Conversation         CONVERSATION           REL         →         BYE(REL)		ACM	+		+	180 Ringing(ACM)					
CPG 1(hold)         →         INVITE(CPG, sendonly)           ←         200 OK INVITE(recvonly)           →         ACK           CPG 2(conference established)         →         INVITE(CPG, sendrecv)           ←         200 OK INVITE(sendrecv)           →         ACK           CPG 3(conference disconnected)         →         INFO(CPG)           ←         200 OK INFO           CPG 4(hold)         →         INVITE(CPG, sendonly)           ←         200 OK INVITE(recvonly)           →         ACK           CPG 5(retrieve)         →         INVITE(CPG, sendrecv)           →         ACK           Conversation         Conversation           CPG 6(conference established)         →         INFO(CPG)           ←         200 OK INFO           Conversation         BYE(REL)		ANM	+		+	200 OK INVITE(ANM)					
CPG 2(conference established)         →         INVITE(CPG, sendrecv)           ←         200 OK INVITE(sendrecv)           ←         200 OK INVITE(sendrecv)           →         ACK           CPG 3(conference disconnected)         →         INFO(CPG)           ←         200 OK INFO           CPG 4(hold)         →         INVITE(CPG, sendonly)           ←         200 OK INVITE(recvonly)           →         ACK           CPG 5(retrieve)         →         INVITE(CPG, sendrecv)           ←         200 OK INVITE(sendrecv)           →         ACK           Conversation           CPG 6(conference established)         →         INFO(CPG)           ←         200 OK INFO											
CPG 2(conference established)  → INVITE(CPG, sendrecv)  ← 200 OK INVITE(sendrecv)  → ACK  CPG 3(conference disconnected)  ← 200 OK INFO  CPG 4(hold)  → INVITE(CPG, sendonly)  ← 200 OK INVITE(recvonly)  → ACK  CPG 5(retrieve)  → INVITE(CPG, sendonly)  ← 200 OK INVITE(recvonly)  → ACK  CONVERSATION  CPG 6(conference established)  → INFO(CPG)  ← 200 OK INVITE(sendrecv)  → ACK  Conversation  CPG 6(conference established)  → INFO(CPG)  ← 200 OK INFO  BYE(REL)		CPG 1(hold)	<b>→</b>		<b>→</b>	INVITE(CPG, sendonly)					
CPG 2(conference established)    INVITE(CPG, sendrecv)					+	200 OK INVITE(recvonly)					
←         200 OK INVITE(sendrecv)           →         ACK           CPG 3(conference disconnected)         →         INFO(CPG)           ←         200 OK INFO           CPG 4(hold)         →         INVITE(CPG, sendonly)           ←         200 OK INVITE(recvonly)           →         ACK           CPG 5(retrieve)         →         INVITE(CPG, sendrecv)           ←         200 OK INVITE(sendrecv)           →         ACK           Conversation           CPG 6(conference established)         →         INFO(CPG)           ←         200 OK INFO           Conversation         →         BYE(REL)					<b>→</b>	ACK					
CPG 3(conference disconnected)         →         INFO(CPG)           CPG 4(hold)         →         INVITE(CPG, sendonly)           CPG 5(retrieve)         →         INVITE(CPG, sendrecv)           CPG 5(retrieve)         →         INVITE(CPG, sendrecv)           COnversation         COnversation           CPG 6(conference established)         →         INFO(CPG)           Conversation         ←         200 OK INFO		CPG 2(conference established)	<b>→</b>								
CPG 3(conference disconnected)         →         INFO(CPG)           ←         200 OK INFO           CPG 4(hold)         →         INVITE(CPG, sendonly)           ←         200 OK INVITE(recvonly)           ACK         →         INVITE(CPG, sendrecv)           ←         200 OK INVITE(sendrecv)           →         ACK           Conversation         CPG 6(conference established)         →         INFO(CPG)           ←         200 OK INFO											
CPG 4(hold)         →         INVITE(CPG, sendonly)           ←         200 OK INVITE(recvonly)           →         ACK           CPG 5(retrieve)         →         INVITE(CPG, sendrecv)           ←         200 OK INVITE(sendrecv)           →         ACK           Conversation         CPG 6(conference established)         →         INFO(CPG)           ←         200 OK INFO           Conversation         FREL         →         BYE(REL)					<b>→</b>	ACK					
CPG 4(hold)         →         INVITE(CPG, sendonly)           ←         200 OK INVITE(recvonly)           →         ACK           CPG 5(retrieve)         →         INVITE(CPG, sendrecv)           ←         200 OK INVITE(sendrecv)           →         ACK           Conversation         CPG 6(conference established)         →         INFO(CPG)           ←         200 OK INFO           Conversation         FREL         →         BYE(REL)		CPG 3(conference disconnected)			_	INFO(CPG)					
CPG 4(hold)         →         INVITE(CPG, sendonly)           ←         200 OK INVITE(recvonly)           ACK         →         INVITE(CPG, sendrecv)           ←         200 OK INVITE(sendrecv)           ←         200 OK INVITE(sendrecv)           ACK         Conversation           CPG 6(conference established)         →         INFO(CPG)           ←         200 OK INFO           Conversation         FREL         →         BYE(REL)		Of O o(contenence discontracted)									
←         200 OK INVITE(recvonly)           →         ACK           CPG 5(retrieve)         →         INVITE(CPG, sendrecv)           ←         200 OK INVITE(sendrecv)           →         ACK           Conversation         INFO(CPG)           ←         200 OK INFO    Conversation  REL  P BYE(REL)						200 61(1141 6					
←         200 OK INVITE(recvonly)           →         ACK           CPG 5(retrieve)         →         INVITE(CPG, sendrecv)           ←         200 OK INVITE(sendrecv)           →         ACK           Conversation         INFO(CPG)           ←         200 OK INFO    Conversation  REL  P BYE(REL)		CPG 4(hold)	<b>→</b>		<b>→</b>	INVITE(CPG_sendonly)					
CPG 5(retrieve)         →         INVITE(CPG, sendrecv)           ←         200 OK INVITE(sendrecv)           →         ACK           Conversation         INFO(CPG)           ←         200 OK INFO		0. 0 I(II0I0)	+-								
CPG 5(retrieve)  → INVITE(CPG, sendrecv)  ← 200 OK INVITE(sendrecv)  → ACK  Conversation  CPG 6(conference established)  → INFO(CPG)  ← 200 OK INFO  Conversation  FEL  → BYE(REL)											
Conversation  CPG 6(conference established)  Conversation  CPG 6(conference established)  Conversation  Conversation  Conversation  Conversation  REL  BYE(REL)					<b>–</b>	,					
Conversation  CPG 6(conference established)  Conversation  CPG 6(conference established)  Conversation  Conversation  Conversation  Conversation  REL  BYE(REL)		CPG 5(retrieve)	<b>→</b>		<b>→</b>	INVITE(CPG, sendrecv)					
CPG 6(conference established)  CPG 6(conference established)  CONVERSATION  CONVERSATION  CONVERSATION  REL  BYE(REL)			†-								
Conversation  CPG 6(conference established) → INFO(CPG)  ← 200 OK INFO  Conversation  REL → BYE(REL)					_						
CPG 6(conference established) → INFO(CPG) ← 200 OK INFO  Conversation  REL → BYE(REL)			0	onversation							
Conversation           REL         → BYE(REL)		CPG 6(conference established)	_			INFO(CPG)					
Conversation  REL → BYE(REL)		5. 5 5(551116161166 651421161164)	†-								
REL → BYE(REL)					<b> </b>						
REL → BYE(REL)			0	onversation	<u>'</u>						
		REL	_			BYE(REL)					
I IRIC I ← I I ← I200 OK BYF(RLC)		RLC	<del>-</del>		<del>-</del>	200 OK BYE(RLC)					

TP413006	SIP reference: RF0	C 326		ISUP reference:						
				Q.1912.5						
	Q.734.2 clause 2.5.2.1.1.3									
TSS reference	ISUP-SIP-ISUP/SS/3PTY									
SIP selection										
criteria										
ISUP selection										
criteria	<u> </u>									
Test purpose	Served user creates a priva									
					PTY call can successfully create					
		ne ac	tive-neid use	er. The	appropriate notification is sent in					
	CPG messages to the user I-MGCF interworking									
CID managed an		4:/	CLID: CDC a		lated in the NAINAE books					
SIP parameter	INFO: Content-Type: applica	ition/i	SUP; CPG e	encapsu	liated in the MilNE body					
values	CDC: Frant in disease, magnet		·	4:	h al d					
ISUP parameter values	CPG: Event indicator=progre									
values	CPG: Event indicator=progre									
	CPG: Event indicator=progre CPG: Event indicator=progre									
Comments	SIP-I	<del>3</del> 55, G	SUT	Callon=	ISUP					
Comments		<b>→</b>	301	<b>→</b>	IAM					
	INVITE(IAM)	+		+						
	180 Ringing(ACM)	<del>-</del>	-	<del></del>	ACM					
	200 OK INVITE(ANM)		0		ANM					
	Conversation									
	INVITE(CPG, sendonly)	<b>→</b>	-	→	CPG(hold)					
	200 OK INVITE(recvonly)	<b>←</b>								
	ACK	<b>→</b>								
	W 11 (175 (000 )	+_			000( ( ( ( ( ( ( ( ( ( ( ( ( ( ( ( ( (					
	INVITE(CPG, sendrecv)	<b>→</b>		→	CPG(conference established)					
	200 OK INVITE(sendrecv)	+								
	ACK	<b>→</b>								
	INFO(CPG)	<b>→</b>		→	CPG(conference disconnected)					
	200 OK INFO	+								
	NN//TE/ODO : : :	+_			0004 10					
	INVITE(CPG, sendonly)	<b>→</b>	1	→	CPG(hold)					
	200 OK INVITE(recvonly)	<b>←</b>	1							
	ACK	<b>→</b>	1							
	NN (ITE (ODO)	+		-						
	INVITE(CPG, sendrecv)	<b>→</b>		→	CPG(retrieve)					
	200 OK INVITE(sendrecv)	+	1							
	ACK	<b>→</b>	<u> </u>							
		<u> </u>	Conversation							
	INFO(CPG)	<b>→</b>	1	→	CPG(conference established)					
	200 OK INFO	+	1							
			Conversation							
	BYE(REL)	<b>→</b>		<b>→</b>	REL					
	200 OK BYE(RLC)	+		<b>←</b>	RLC					

TP413007	SIP reference: RFC 3261		ISUP refer Q.1912.5 Q.734.2 cl		e: e 2.5.2.1.1.3 a
TSS reference	ISUP-SIP-ISUP/SS/3PTY				
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Verify that the IUT (controlling the corprivate communication with the active messages to the user O-MGCF interworking	ferenc	e) on a 3PTY	call	can successfully create
SIP parameter values	INFO: Content-Type: application/ISUI	P; CPG	encapsulate	d in	the MIME body
ISUP parameter	CPG: Event indicator=progress, Gene				
values	CPG: Event indicator=progress, Gene	eric not	ification=conf	erer	nce disconnected
Comments	ISUP		SUT		SIP-I
	IAM	→		<b>→</b>	INVITE(IAM)
	ACM	+		<b>←</b>	180 Ringing(ACM)
	ANM	+		<b>←</b>	200 OK INVITE(ANM)
			onversation		
	CPG(conference established)	→		<b>→</b>	INFO(CPG)
				<b>←</b>	200 OK INFO
	CPG(conference disconnected)	<b>→</b>		<b>→</b>	INFO(CPG)
				<b>←</b>	200 OK INFO
		_	onversation		
	CPG(conference established)	→		<b>→</b>	INFO(CPG)
				<b>←</b>	200 OK INFO
			onversation		
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	+		<del>(</del>	200 OK BYE(RLC)

TP413008	SIP reference: R	FC 326	1		ISUP reference: Q.1912.5 Q.734.2 clause 2.5.2.1.1.3 a
TSS reference	ISUP-SIP-ISUP/SS/3PTY				
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose		ling the	conference) on	a 31	n a remote user PTY call can successfully create appropriate notification is sent in CPG
SIP parameter values	INFO: Content-Type: applic		,		,
ISUP parameter	CPG: Event indicator=prog	gress, G	eneric notificati	on=	conference established
values	CPG: Event indicator=prog	gress, G	eneric notificati	on=	conference disconnected
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		<b>→</b>	IAM
	180 Ringing(ACM)	<b>←</b>		+	ACM
	200 OK INVITE(ANM)	+		<b>←</b>	ANM
			Conversation		
	INFO(CPG)	→		<b>→</b>	CPG(conference established)
	200 OK INFO	+			
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(conference disconnected)
	200 OK INFO	+	<u> </u>		
	111111111111111111111111111111111111111		Conversation		
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(conference established)
	200 OK INFO	+			
			Conversation		
	BYE(REL)	<b>→</b>		<u>→</u>	REL
	200 OK BYE(RLC)	<b>←</b>		<b>←</b>	RLC

TP413009	SIP reference: RFC 3261		G	_	SUP reference: Q.1912.5 2 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 remo Verify that the IUT (controlling the cor the active-held user and retain and no messages The IUT should send to the appropria notification indicator. The event inc O-MGCF interworking	ference otify the	e) on a 3PT other user ote users <b>C</b> l	Y cal appro	I can successfully disconnect opriately using CPG essages with a generic
SIP parameter	INFO: Content-Type: application/ISUF	o CPG	encansulat	ed in	the MIME body
values	in a comon type application to con	, 0. 0	опоароша		and minimize south
ISUP parameter	CPG: Event indicator=progress, Gene	ric not	ification=co	nferer	nce established
values	CPG: Event indicator=progress, Gene				
Comments	ISUP		SUT		SIP-I
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)
	ACM	<b>←</b>		<b>←</b>	180 Ringing(ACM)
	ANM	<b>←</b>		<b>←</b>	200 OK INVITE(ANM)
		С	onversatio	n	
	CPG(conference established)	→		<b>→</b>	INFO(CPG)
				<b>←</b>	200 OK INFO
	CPG(conference disconnected)	<b>→</b>		→	INFO(CPG)
				<b>←</b>	200 OK INFO
		С	onversatio	n	
	REL	→		<b>→</b>	BYE(REL)
	RLC	<b>←</b>		<b>←</b>	200 OK BYE(RLC)

TP413010	SIP reference: RF0	326	1		ISUP reference: Q.1912.5 Q.734.2 clause 2.5.2.1.1.3 b					
TSS reference	ISUP-SIP-ISUP/SS/3PTY									
SIP selection criteria										
ISUP selection criteria										
Test purpose	the active-held user and retain messages The IUT should send to the a	g the in and	conference) on I notify the othe oriate remote us	a 3 er us sers	PTY call can successfully disconnect					
SIP parameter values	INFO: Content-Type: applica	tion/IS	SUP; CPG enca	apsu	lated in the MIME body					
ISUP parameter	CPG: Event indicator=progre	ss, G	eneric notificati	on=	conference established					
values	CPG: Event indicator=progre	ss, G	eneric notificati	on=	conference disconnected					
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	<b>→</b>		1	IAM					
	180 Ringing(ACM)	+		+	ACM					
	200 OK INVITE(ANM)	+		+	ANM					
			Conversation							
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(conference established)					
	200 OK INFO	+								
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(conference disconnected)					
	200 OK INFO	+								
			Conversation							
	BYE(REL)	<b>→</b>		<b>→</b>	REL					
	200 OK BYE(RLC)	+		+	RLC					

TP413011	SIP reference: RFC 3261			I	SUP reference:					
					Q.1912.5					
<i>(</i>			Q.	734.	2 clause 2.5.2.1.1.3 b					
TSS reference	ISUP-SIP-ISUP/SS/3PTY									
SIP selection										
criteria ISUP selection										
criteria Test purpose	Served user disconnects one remote user and retains the other									
	Verify that the IUT (controlling the country the active-idle user and retain and not not included in the appropriation indicator. The event in O-MGCF interworking	onference otify the iate remo	e) on a 3PTY other user ap ote users <b>CP</b> in the <b>CPG</b> s	cal opro <b>G</b> m	I can successfully disconnect priately using CPG messages essages with a generic ald be set to "progress"					
SIP parameter values	INFO: Content-Type: application/ISU	JP; CPG	encapsulate	d in	the MIME body					
ISUP parameter	CPG: Event indicator=progress, Ger	neric noti	fication=hold	1						
values	CPG: Event indicator=progress, Ger				nce established					
raidoo	CPG: Event indicator=progress, Ger									
Comments	ISUP		SUT		SIP-I					
	IAM	→		<b>→</b>	INVITE(IAM)					
	ACM	+		+	`					
	ANM	+		<b>←</b>	200 OK INVITE(ANM)					
	Conversation									
	CPG(hold)	<b>→</b>		<b>→</b>	INVITE(CPG, sendonly)					
				+	200 OK INVITE(recvonly)					
				<b>→</b>	ACK					
	CPG(conference established)	→		<b>→</b>	INVITE(CPG, sendrecv)					
				<b>←</b>	200 OK INVITE(sendrecv)					
				<b>→</b>	ACK					
	CPG(conference disconnected)	→		<u>→</u>	INFO(CPG)					
				+	200 OK INFO					
	CPG(hold)	→		<b>→</b>	INVITE(CPG, sendonly)					
	J. J. (	<del>  -</del>		<del></del>	200 OK INVITE(recvonly)					
				<u>→</u>	ACK					
	REL	→		<b>→</b>	BYE(REL)					
	RLC	+		<b>←</b>	200 OK BYE(RLC)					

TP413012	SIP reference: RF	C 3261			ISUP reference: Q.1912.5						
		Q.734.2 clause 2.5.2.1.1.3 b									
TSS reference	ISUP-SIP-ISUP/SS/3PTY										
SIP selection											
criteria											
ISUP selection											
criteria											
Test purpose	Served user disconnects one remote user and retains the other Verify that the IUT (controlling the conference) on a 3PTY call can successfully disconnect the active-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: Content-Type: applica	ation/IS	SUP: CPG e	ncapsu	lated in the MIME body						
values	3		·	•	,						
ISUP parameter	CPG: Event indicator=progre	ess, Ge	eneric notific	cation=l	hold						
values	CPG: Event indicator=progre										
	CPG: Event indicator=progre	ess, Ge	eneric notific	cation=	conference disconnected						
Comments	SIP-I		SUT		ISUP						
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM						
	180 Ringing(ACM)	+		+	ACM						
	200 OK INVITE(ANM)	+		+	ANM						
		Conversation									
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(hold)						
	200 OK INVITE(recvonly)	+									
	ACK	<b>→</b>									
	INVITE(CPG, sendrecv)	<b>→</b>		<b>→</b>	CPG(conference established)						
	200 OK INVITE(sendrecv)	+									
	ACK	<b>→</b>									
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(conference disconnected)						
	200 OK INFO	+									
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(hold)						
	200 OK INVITE(recvonly)	+									
	ACK	<b>→</b>									
	BYE(REL)	<b>→</b>		<b>→</b>	REL						
	200 OK BYE(RLC)	+		+	RLC						

## 5.3.14 User-to-user service

### 5.3.14.1 User-to-user service 1

TP414001	SIP refere	ence: RFC 3261			SUP reference: Q.1912.5 A.1.1 .5.2.3 and 4/Q.737		
TSS reference	ISUP-SIP-ISUP/SS	JUUS1					
SIP selection criteria							
ISUP selection criteria							
Test purpose		Ensure that the SUT can successfully transfer the User-to-user service 1 implicit request in the encapsulated IAM. O-MGCF interworking					
SIP parameter	INVITE: Content-T	pe: application/l	SUP; IAM cont	aining the	user-to-user information		
values	parameter encapsu	lated in the MIMI	E body				
ISUP parameter values	IAM: User-to-user i	nformation paran	neter				
Comments	ISUP		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
			Conversation	n			
	REL	→		<b>→</b>	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

TP414002	SIP reference: F	RFC 326	1		SUP reference: Q.1912.5 A.1.1
				1.1	.5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Ensure that the SUT can	successf	ully transfer the U	ser-to-us	er service 1 implicit request in
	the encapsulated IAM. I-M	IGCF int	erworking		
SIP parameter	INVITE: Content-Type: ap	plication	/ISUP; IAM contai	ning the	user-to-user information
values	parameter encapsulated in	n the MIN	ЛE body	Ū	
ISUP parameter	IAM: User-to-user informa	tion para	meter		
values		-			
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Conversation	•	
	BYE(REL)	→		<b>→</b>	REL
	200 OK BYE(RLC)	+		+	RLC

TP414003	SIP refere	ence: RFC 3261			ISUP reference: Q.1912.5 A.1.1
				1.1	.5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS	/UUS1			
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Ensure that the SU	T can successfull	y transfer the U	ser-to-us	ser service 1 explicit request
	not essential in the	encapsulated IAN	I. O-MGCF inte	rworking	
SIP parameter	INVITE: Content-Ty	pe: application/IS	SUP; IAM contain	ining the	user-to-user indicator
values	parameter encapsu	lated in the MIME	body		
ISUP parameter	IAM: User-to-user in	nformation param	eter, User-to-us	ser indica	ator = service 1 explicit
values	request				
Comments	ISUP		SUT		SIP-I
	IAM	→		<b>→</b>	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
			Conversation		
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP414004	SIP reference: R	FC 326	1		SUP reference: Q.1912.5 A.1.1 5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection					
criteria					
Test purpose	Ensure that the SUT can seessential in the encapsulat		•		er service 1 explicit request
SIP parameter	INVITE: Content-Type: app	olication	ISUP; IAM conta	ining the	user-to-user indicator
values	parameter encapsulated in	the MIN	/IE body	· ·	
ISUP parameter values	IAM: User-to-user informat	ion para	meter, User-to-u	ser indica	tor
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Conversation		
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	+		+	RLC

TP414005	SIP reference: RF	C 326	1		_	SUP reference: Q.1912.5 A.1.1 5.2.3 and 4./Q.737		
TSS reference	ISUP-SIP-ISUP/SS/UUS1							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the SUT can su in the encapsulated ACM. O-MGCF interworking							
SIP parameter	INVITE: Content-Type: appl	ication	/ISUP; IAM	containin	g the	user-to-user information		
values	parameter encapsulated in	the MIN	∕IE body					
	180 Ringing: Content-Type:	applica	ation/ISUP;	ACM con	itainin	g the user-to-user information		
	parameter encapsulated in	the MIN	ЛE body					
ISUP parameter	IAM: User-to-user information							
values	ACM: User-to-user informat	ion par	ameter					
Comments	ISUP		SUT	Γ		SIP-I		
	IAM	<b>→</b>			<b>↑</b>	INVITE(IAM)		
	ACM	+			+	180 Ringing(ACM)		
	ANM	+			+	200 OK INVITE(ANM)		
			Convers	ation				
	REL	<b>→</b>			<b>→</b>	BYE(REL)		
	RLC	+			+	200 OK BYE(RLC)		

TP414006	SIP reference: R	SIP reference: RFC 3261 ISUP reference Q.1912.5 A.1.1 1.1.5.2.3 and 4./Q.								
TSS reference	ISUP-SIP-ISUP/SS/UUS1									
SIP selection criteria										
ISUP selection criteria										
Test purpose	Ensure that the SUT can so in the encapsulated ACM. I-MGCF interworking									
SIP parameter values	INVITE: Content-Type: app parameter encapsulated in 180 Ringing: Content-Type parameter encapsulated in	the MIN	ME body ation/ISUP;			user-to-user information g the user-to-user information				
ISUP parameter values	IAM: User-to-user informati ACM: User-to-user informa									
Comments	SIP-I		SUT			ISUP				
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM				
	180 Ringing(ACM)	+			+	ACM				
	200 OK INVITE(ANM)									
			Convers	ation	•					
	BYE(REL)	<b>→</b>			<b>→</b>	REL				
	200 OK BYE(RLC)	+			+	RLC				

TP414007	SIP reference: RFC 3261 ISUP reference: Q.1912.5 A.1.1 1.1.5.2.3 and 4./Q.737								
TSS reference	ISUP-SIP-ISUP/SS/UUS1								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Ensure that the SUT can su in the encapsulated ACM. O-MGCF interworking								
SIP parameter	INVITE: Content-Type: appl	ication/	ISUP; IAM	containin	g the i	user-to-user information			
values	parameter encapsulated in	the MIN	∕IE body						
	180 Ringing: Content-Type:			ACM cor	ntainin	g the user-to-user indicator			
	parameter encapsulated in	the MIN	ЛE body						
ISUP parameter	IAM: User-to-user information								
values	ACM: User-to-user indicator	r set to	service 1 su	upported	respoi	nse			
Comments	ISUP		SUT	•		SIP-I			
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)			
	ACM	+			+	180 Ringing(ACM)			
	ANM								
			Convers	ation					
	REL	<b>→</b>			<b>→</b>	BYE(REL)			
	RLC	+			+	200 OK BYE(RLC)			

TP414008	SIP reference: RF	C 326	1	-	SUP reference: Q.1912.5 A.1.1 5.2.3 and 4./Q.737			
TSS reference	ISUP-SIP-ISUP/SS/UUS1							
SIP selection criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the SUT can su in the encapsulated ACM. I-MGCF interworking	:						
SIP parameter	INVITE: Content-Type: appl	ication	ISUP; IAM containi	ng the	user-to-user information			
values	parameter encapsulated in	the MIN	/IE body					
	180 Ringing: Content-Type: parameter encapsulated in			ontainin	g the user-to-user indicator			
ISUP parameter	IAM: User-to-user information	on para	meter, User-to-use	r indica	tor set to service 1 request			
values	ACM: User-to-user indicator	r set to	service 1 supported	d respo	nse			
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)							
			Conversation					
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP414009	SIP reference: RF	C 326	I	-	SUP reference: Q.1912.5 A.1.1 5.2.3 and 4./Q.737			
TSS reference	ISUP-SIP-ISUP/SS/UUS1							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose		Ensure that the SUT can successfully transfer the User-to-user service 1 discarded by the network in the encapsulated ACM O-MGCF interworking						
SIP parameter	INVITE: Content-Type: appl	ication	ISUP; IAM contain	ning the	user-to-user information			
values	parameter encapsulated in 180 Ringing: Content-Type: parameter encapsulated in 180 Ringing:	applica	ation/ISUP; ACM o	ontainin	ng the User-to-user indicator			
ISUP parameter	IAM: User-to-user information							
values	ACM: User-to-user indicator			network	response			
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	<del>(</del>		+	180 Ringing(ACM)			
	ANM	<del>(</del>		+	200 OK INVITE(ANM)			
			Conversation		, ,			
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP414010	SIP reference: RF	-C 326	1		_	SUP reference: Q.1912.5 A.1.1 5.2.3 and 4./Q.737				
TSS reference	ISUP-SIP-ISUP/SS/UUS1									
SIP selection criteria										
ISUP selection criteria										
Test purpose		Ensure that the SUT can successfully transfer the User-to-user service 1 discarded by the network in the encapsulated ACM								
SIP parameter values	INVITE: Content-Type: app parameter encapsulated in 180 Ringing: Content-Type: parameter encapsulated in	the MIN application	ME body ation/ISUP;		•	user-to-user information g the User-to-user indicator				
ISUP parameter values	IAM: User-to-user information ACM: User-to-user indicato	on para	meter	y the net	work i	response				
Comments	SIP-I		SUT			ISUP				
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM				
	180 Ringing(ACM)	+			+	ACM				
	200 OK INVITE(ANM)									
			Convers	ation	•					
	BYE(REL)	<b>→</b>		_	<b>→</b>	REL				
	200 OK BYE(RLC)	+			+	RLC				

#### 5.3.14.2 User-to-user service 2

TP414101	SIP reference: RI	FC 326 <sup>-</sup>	1		SUP reference: Q.1912.5 A.1.1 .5.2.3 and 4./Q.737			
TSS reference	ISUP-SIP-ISUP/SS/UUS2							
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	and User-to user information	Ensure that the SUT can successfully transfer the User-to-user service 2 explicit request and User-to user information in the encapsulated IAM. An additional User-to-user information is sent in a USR message encapsulated in an INFO request.						
SIP parameter	INVITE: Content-Type: app	lication	ISUP; IAM cont	aining the	user-to-user indicator and			
values	User-to-user information er							
	INFO: Content-Type: applic				Jser-to-user information			
	parameter encapsulated in			Ū				
ISUP parameter	IAM: User-to-user informati			user indica	ator			
values	USR: User-to-user informat	tion						
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
			Conversation	n				
	USR	<b>→</b>		<b>→</b>	INFO(USR)			
				+	200 OK INFO			
	USR	+		+	INFO(USR)			
				<b>→</b>	200 OK INFO			
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP414102	SIP reference: RFC 3261			ISUP reference: Q.1912.5 A.1.1 1.2.5.2.3 and 4./Q.737			
TSS reference	ISUP-SIP-ISUP/SS/UUS2	2					
SIP selection criteria							
ISUP selection criteria							
Test purpose		additional			ser service 2 explicit request in n is sent in a USR message		
SIP parameter	INVITE: Content-Type: ap		SUP; IAM conta	ining the	user-to-user indicator		
values	encapsulated in the MIME						
	INFO: Content-Type: app			ing the l	Jser-to-user information		
	parameter encapsulated i						
ISUP parameter	IAM: User-to-user informa		eter, User-to-us	ser indica	ator		
values	USR: User-to-user inform	ation					
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		<b>→</b>	IAM		
	180 Ringing(ACM)	<b>←</b>		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
			Conversation				
	INFO(USR)	→		<b>→</b>	USR		
	200 OK INFO	+					
	INFO(USR)	+	•	+	USR		
	200 OK INFO	<b>→</b>					
	BYE(REL)	<b>→</b>		<b>→</b>	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP414103	SIP reference: RF	C 326			SUP reference: Q.1912.5 A.1.1 .5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS2				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can su in the encapsulated ACM. I-MGCF interworking	ccessf	ully transfer the U	ser-to-us	er service 2 explicit response
SIP parameter values	INVITE: Content-Type: appl parameter encapsulated in 180 Ringing: Content-Type: parameter encapsulated in	the MIN applica	ME body ation/ISUP; ACM	Ū	user-to-user information
ISUP parameter values	IAM: User-to-user information ACM: User-to-user indicator	on para	meter, User-to-us		
Comments	ISUP		SUT		SIP-I
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
			Conversation		
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP414104	SIP reference: I	RFC 326	1		ISUP reference: Q.1912.5 A.1.1 .5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS2	2			
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can the encapsulated IAM. I-N		•	ser-to-us	ser service 2 explicit request in
SIP parameter values	INVITE: Content-Type: apparameter encapsulated i			ining the	user-to-user indicator
ISUP parameter values	IAM: User-to-user informa			ser indica	ator
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Conversation		
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	+		+	RLC

### 5.3.14.3 User-to-user service 3

TP414201	SIP reference: RFC 3261			ISUP reference: Q.1912.5 A.1.1 1.3.5.2.3 and 4./Q.737		
TSS reference	ISUP-SIP-ISUP/SS/UUS3					
SIP selection						
criteria						
ISUP selection criteria						
Test purpose	Ensure that the SLIT can su	ccassfi	ılly transfer t	ha Hsar-to-us	ser service 3 explicit request in	
rost parpose		tional l			s sent in several USR message	
SIP parameter	INVITE: Content-Type: appl		ISUP; IAM o	ontaining the	user-to-user indicator	
values	encapsulated in the MIME b INFO: Content-Type: application parameter encapsulated in the	ation/IS the MIN	/IE body	· ·		
ISUP parameter	IAM: User-to-user information	on para	meter, User	-to-user indica	ator	
values	USR: User-to-user informati	on				
Comments	ISUP		SUT		SIP-I	
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)	
	ACM	+		+	180 Ringing(ACM)	
	ANM	+	0	<b>(</b>	200 OK INVITE(ANM)	
	1100		Conversa		IN IEO (110P)	
	USR	<b>→</b>		<b>→</b>	INFO(USR)	
				+	200 OK INFO	
	USR	+		+	INFO(USR)	
	USK	_		→ <del>-</del>	200 OK INFO	
				7	200 OK INFO	
	USR	+		+	INFO(USR)	
	00.1			<b>→</b>	200 OK INFO	
	USR	<b>→</b>		<b>→</b>	INFO(USR)	
				+	200 OK INFO	
	REL	<b>↑</b>		<b>→</b>	BYE(REL)	
	RLC	+		+	200 OK BYE(RLC)	

TP414202	SIP reference: I	SIP reference: RFC 3261			ISUP reference: Q.1912.5 A.1.1 1.3.5.2.3 and 4./Q.737			
TSS reference	ISUP-SIP-ISUP/SS/UUS3	ISUP-SIP-ISUP/SS/UUS3						
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	Ensure that the SUT can	successf	ully transfer	the User-to-u	ser service 3 explicit	request in		
	the encapsulated IAM. Ac							
	encapsulated in an INFO					· ···occago		
	I-MGCF interworking							
SIP parameter	INVITE: Content-Type: ap	plication	/ISUP: IAM	containing the	e user-to-user indicat	or		
values	encapsulated in the MIME		,					
	INFO: Content-Type: app		SUP: USR c	ontaining the	User-to-user informa	tion		
	parameter encapsulated i			3				
ISUP parameter	IAM: User-to-user informa			r-to-user indic	ator			
values	USR: User-to-user inform		,					
Comments	SIP-I		SUT	Г	ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	,		Convers	ation				
	INFO(USR)	<b>→</b>		→	USR			
	200 OK INFO	+						
	INFO(USR)	+		+	USR			
	200 OK INFO	<b>→</b>						
	INFO(USR)	+		+	USR			
	200 OK INFO	<b>→</b>						
	INFO(USR)	<b>→</b>		<b>→</b>	USR			
	200 OK INFO	+						
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP414203	SIP reference: RFC 3261				ISUP reference: Q.1912.5 A.1.1 .5.2.3 and 4./Q.737			
TSS reference	ISUP-SIP-ISUP/SS/UUS3							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that an User-to-user request service 3 encapsulated in an INFO request during the confirmed state can successful proceeded. The User-to-user information is passed on in several encapsulated USR messages.  O-MGCF interworking							
SIP parameter	INFO: Content-Type: applic		SUP; FAR containir	ng the u	ser-to-user indicator			
values	encapsulated in the MIME INFO: Content-Type: applice encapsulated in the MIME INFO: Content-Type: application parameter encapsulated in	cation/IS body cation/IS the MIN	SUP; USR containi ME body	ng the l				
ISUP parameter	FAR: User-to-user indicator							
values	FAA: User-to-user indicator USR: User-to-user informat		e 3 response provid	ded				
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
	Conversation							
	FAR	<b>→</b>		<b>→</b>	INFO(FAR)			
				+	200 OK INFO			
	FAA	+		+	INFO(FAA)			
				<b>→</b>	200 OK INFO			
	USR	<b>→</b>		→	INFO(USR)			
				+	200 OK INFO			
	USR	+		+	INFO(USR)			
	0011			<b>→</b>	200 OK INFO			
	USR	+		+	INFO(USR)			
				<b>→</b>	200 OK INFO			
	USR	<b>→</b>		<b>→</b>	INFO(USR)			
				+	200 OK INFO			
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP414204	SIP reference: RFC 3261				ISUP reference: Q.1912.5 A.1.1 .5.2.3 and 4./Q.737					
TSS reference	ISUP-SIP-ISUP/SS/UUS3	ISUP-SIP-ISUP/SS/UUS3								
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	confirmed state can succe several encapsulated USF I-MGCF interworking	Ensure that an User-to-user request service 3 encapsulated in an INFO request during the confirmed state can successful proceeded. The User-to-user information is passed on in several encapsulated USR messages								
SIP parameter	INFO: Content-Type: appl		SUP; FAR co	ontaining the υ	ser-to-user indicator					
values	encapsulated in the MIME INFO: Content-Type: appl encapsulated in the MIME INFO: Content-Type: appl parameter encapsulated in	ication/IS body ication/IS n the MIN	SUP; USR co	ontaining the l						
ISUP parameter	FAR: User-to-user indicate	or service	3 request r	not essential						
values	FAA: User-to-user indicate	or service	3 response	provided						
	USR: User-to-user information	ation								
Comments	SIP-I		SUT	•	ISUP					
	INVITE(IAM)	<b>→</b>		→	IAM					
	180 Ringing(ACM)	+		+	ACM					
	200 OK INVITE(ANM)	+		+	ANM					
	`		Conversa	ation						
	INFO(FAR)	<b>→</b>		→	FAR					
	200 OK INFO	+		_	1					
	INFO(FAA)	+		+	FAA					
	200 OK INFO	<b>→</b>			1701					
	200 010 1141 0									
	INFO(USR)	<b>→</b>		<b>→</b>	USR					
	200 OK INFO	+			COIX					
	200 010 1101 0	+								
	INFO(USR)	+		+	USR					
	200 OK INFO	→ ·			USIX					
	200 OK INFO	7								
	INICO(LICE)				LICD					
	INFO(USR)	<b>←</b>		+	USR					
	200 OK INFO	7								
	INIEO/LICD)				LICD					
	INFO(USR)	<b>→</b>		<b>→</b>	USR					
	200 OK INFO	+								
	BYE(REL)	<b>→</b>		<b>→</b>	REL					
	200 OK BYE(RLC)	+		<b>←</b>	RLC					

TP414205	SIP reference: RF	C 3261	1		ISUP reference: Q.1912.5 A.1.1 1.3.5.2.3 and 4./Q.737		
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	ccessfu	ully transfer	the User	-to-us	er service 3 explicit response	
SIP parameter	INVITE: Content-Type: appl	ication/	/ISUP; IAM	containin	g the	user-to-user indicator	
values	parameter encapsulated in	the MIN	∕IE body				
	200 OK INVITE: Content-Ty				conta	ining the user-to-user	
	indicator parameter encaps	ulated i	n the MIME	body			
ISUP parameter	IAM: User-to-user indicator						
values	ANM: User-to-user indicator	set to	service 3 pi	rovided re	espons	se	
Comments	ISUP		SUT			SIP-I	
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)	
	ACM	+	-		+	180 Ringing(ACM)	
	ANM	+		•	+	200 OK INVITE(ANM)	
			Convers	ation			
	REL	<b>→</b>			<b>→</b>	BYE(REL)	
	RLC	+			+	200 OK BYE(RLC)	

TP414206	SIP reference: RF	C 326	I		SUP reference: Q.1912.5 A.1.1 5.2.3 and 4./Q.737						
TSS reference	ISUP-SIP-ISUP/SS/UUS3	ISUP-SIP-ISUP/SS/UUS3									
SIP selection criteria											
ISUP selection											
criteria											
Test purpose		ccessf	ully transfer the	User-to-us	er service 3 explicit response						
	in the encapsulated ANM										
	O-MGCF interworking										
SIP parameter	INVITE: Content-Type: appl			taining the	user-to-user indicator						
values	parameter encapsulated in										
	200 OK INVITE: Content-Ty				ining the user-to-user						
	indicator parameter encaps	ulated i	n the MIME boo	dy							
ISUP parameter	IAM: User-to-user indicator	set to s	ervice 3 reques	st							
values	ANM: User-to-user indicator	set to	service 3 provid	ded respon	se						
Comments	SIP-I		SUT		ISUP						
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM						
	180 Ringing(ACM)	+		+	ACM						
	200 OK INVITE(ANM) ← ANM										
		Conversation									
	BYE(REL)	<b>→</b>		<b>→</b>	REL						
	200 OK BYE(RLC)	+		+	RLC						

TP414207	SIP reference: RF	C 326	1		SUP reference: Q.1912.5 A.1.1 5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	confirmed state can success	sful pro	ceeded. The	e user to user	
SIP parameter	INFO: Content-Type: applic		SUP; FAR co	ontaining the u	ser-to-user indicator
values	encapsulated in the MIME b				
	INFO: Content-Type: applic		SUP; FRJ co	ntaining the u	ser-to-user indicator
	encapsulated in the MIME b				
ISUP parameter	FAR: User-to-user indicator				
values	FRJ: User-to-user indicator	service			
Comments	ISUP		SUT	•	SIP-I
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)
	ACM	<b>←</b>		<b>←</b>	180 Ringing(ACM)
	ANM	+		<del>-</del>	200 OK INVITE(ANM)
			Conversa	ation	
	FAR	<b>→</b>		→	INFO(FAR)
				<b>←</b>	200 OK INFO
	FRJ	+		+	INFO(FRJ)
				<b>→</b>	200 OK INFO
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP414208	SIP reference: R	RFC 326 <sup>-</sup>	1		ISUP reference: Q.1912.5 A.1.1 .5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that an User-to-us confirmed state can succe				in an INFO request during the request is rejected
SIP parameter	INFO: Content-Type: appl		SUP; FAR contain	ing the ι	ser-to-user indicator
values	encapsulated in the MIME INFO: Content-Type: appliencapsulated in the MIME	ication/IS	SUP; FRJ containi	ng the u	ser-to-user indicator
ISUP parameter	FAR: User-to-user indicate		e 3 request not es	sential	
values	FRJ: User-to-user indicate				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Conversation		
	INFO(FAR)	<b>→</b>		→	FAR
	200 OK INFO	+			
	INFO(FRJ)	+		+	FRJ
	200 OK INFO	<b>→</b>			
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	+		+	RLC

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

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

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

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

#### A.1.1.1.1 Sending of the SETUP Message

TP501001	SIP reference: RFC 3261				ISDN reference: Q.1912.5 clause 6.1.2 (i,1)			
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message							
SIP selection	NOT PICS 4/4 AND NOT PI	ICS 4/5						
criteria								
ISDN selection	NOT PICS 1/6							
criteria								
Test purpose  SIP parameter values	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer including a SDP for A-law (PCMA) and µ-law (PCMU)  the SUT shall delete µ-law (PCMU) from the media description that it will send back in the SDP answer  the SUT shall immediately send out the SETUP. The BC is constructed from the ISUP TMR or USI  SIP INVITE: Audio RTP/AVP 0 8  200 OK: Audio RTP/AVP 8							
ISDN parameter	SETUP BC: A-law	•						
values				_		1		
Comments	SIP-I		SU	T		ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
	180 Ringing(ACM)	+			+	ALERTING		
	200 OK INVITE(ANM)	+			<b>←</b>	CONNECT		
	ACK	→						
			Convers	sation				
	BYE(REL)	→			<b>→</b>	DISCONNECT		
	200 OK BYE(RLC)	<b>←</b>			+	RELEASE		
					<b>→</b>	RELEASE COMPLETE		

TP501002	SIP reference: RFC	3261				ISDN reference:		
						12.5 clause 6.1.2 (i,2b		
TSS reference:	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message							
SIP selection	PICS 4/4 AND PICS 4/5							
criteria								
ISDN selection criteria	NOT PICS 1/6							
Test purpose  SIP parameter	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer and 100rel extensions and preconditions extensions in the SIP Supported header: including a SDP for A-law (PCMA) and μ-law (PCMU)</li> <li>the SUT shall delete μ-law (PCMU), if present, from the media description that it will send back in the SDP answer</li> <li>the SETUP shall be deferred until all preconditions have been met. The BC is constructed from the ISUP TMR or USI</li> <li>The bearer path shall be connected in both directions when both of the following conditions are satisfied:</li> <li>The I-IWU determines (BCng the procedures defined in RFC 3312) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable).</li> <li>In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "notification not required", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312) that sufficient preconditions have been met for the session to proceed</li> <li>SIP INVITE: Audio RTP/AVP 0 8</li> </ul>							
values	200 OK: Audio RTP/AVI							
ISDN parameter	SETUP BC: A-law							
values								
Comments	SIP-I		SU	JT		ISDN		
	INVITE(IAM)	<b>→</b>						
	183 session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>			<b>→</b>	SETUP		
	200 OK UPDATE	+						
	180 Ringing(ACM)	+			+	ALERTING		
	200 OK INVITE(ANM) ← CONNECT							
	ACK	<b>→</b>						
			Conver	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT		
	200 OK BYE(RLC)	<del>-</del>			<b>←</b>	RELEASE		
					<b>→</b>	RELEASE COMPLETE		

TP501003	SIP reference: RFC 3261				0.40	ISDN reference:		
TCC voference	Q.1912.5 clause 6.1.2 (i,1)  SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message							
TSS reference			ne_SETU	P_messa	ige			
SIP selection	NOT PICS 4/4 AND NOT PICS	4/5						
criteria								
ISDN selection	NOT PICS 1/6							
criteria								
SIP parameter values	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer including a SDP for A-law (PCMA) and μ-law (PCMU)  • the SUT shall delete μ-law (PCMU) from the media description that it will send back in the SDP answer  • the SUT shall immediately send out the SETUP. The BC is constructed from the ISUP TMR or USI  SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 8							
ISDN parameter values	SETUP BC: A-law							
Comments	SIP-I		SU	T		ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
	180 Ringing(ACM)	+			+	ALERTING		
	200 OK INVITE(ANM)	+			+	CONNECT		
	ACK	<b>→</b>						
	Conversation							
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT		
	200 OK BYE(RLC)	+			+	RELEASE		
					<b>→</b>	RELEASE COMPLETE		

TP501004	SIP reference: RFC 3261			ISDN reference: Q.1912.5 clause 6.1.2 (i,2b)		
TSS reference	SIP-I-ISDN/Basic_call/ Sending	n of the	SETU		312.3 clad3c 0.1.2 (1,25)	
SIP selection	PICS 4/4 AND PICS 4/5	<u>g_0i tilo_</u>	<u>OL 101</u>	_mcssage		
criteria	1100 1/171112 1100 1/0					
ISDN selection	NOT PICS 1/6					
criteria						
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer and 100rel extensions and preconditions extensions in the SIP Require header including a SDP for A-law (PCMA) and μ-law (PCMU)</li> <li>the SUT shall delete μ-law (PCMU), if present, from the media description that it will send back in the SDP answer</li> <li>the SETUP shall be deferred until all preconditions have been met. The BC is constructed from the ISUP TMR or USI.</li> </ul>					
SIP parameter	SIP INVITE: Audio RTP/AVP 0.8					
values	200 OK: Audio RTP/AVP	8				
ISDN parameter	SETUP BC: A-law					
values						
Comments	SIP-I		SU	Т	ISDN	
	INVITE(IAM)	→				
	183 session Progress	+				
	PRACK	→				
	200 OK PRACK	+				
	UPDATE	→		<b>→</b>	SETUP	
	200 OK UPDATE	<del>(</del>				
	180 Ringing(ACM)	<del>(</del>		+	ALERTING	
	200 OK INVITE(ANM)	+		+	CONNECT	
	ACK	<b>→</b>				
		С	onvers	sation		
	BYE(REL)	<b>→</b>		<b>→</b>	DISCONNECT	
	200 OK BYE(RLC)	+		+	RELEASE	
				→	RELEASE COMPLETE	

TP501005	SIP reference: RFC 3261				0.40	ISDN reference:		
TSS reference	Q.1912.5 clause 6.1.2 (i,1)  SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message							
SIP selection	NOT PICS 4/4 AND NOT PICS		ne_SETU	P_messa	ige			
criteria	NOT PICS 4/4 AND NOT PICS	4/5						
ISDN selection	PICS 1/6							
criteria	PICS 1/6							
	Ensure that if the SUT upon re-	ooint (	of the first	INI\/ITE \	with	oufficient digits, with an SDB		
Test purpose	offer including a SDP for A-lav							
						scription that it will send back in		
	the SDP answer	/ (PCI	viA) IIOIII i	ne media	a ues	scription that it will send back in		
		لممما	out the C	CTUD T	ha D	C is sometimented from the ICLID		
	TMR or USI	sena	out the S	ETUP. I	пе ь	C is constructed from the ISUP		
SIP parameter	SIP INVITE: Audio RTP/AVP	8 0						
values	200 OK: Audio RTP/AVP	8						
ISDN parameter	SETUP BC: A-law							
values								
Comments	SIP-I		SU	Т		ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
	180 Ringing(ACM)	+			+	ALERTING		
	200 OK INVITE(ANM)	+			+	CONNECT		
	ACK	<b>→</b>						
	Conversation							
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT		
	200 OK BYE(RLC)	+			<del>(</del>	RELEASE		
					<b>→</b>	RELEASE COMPLETE		

TP501006	SIP reference: RFC 3261			ISDN reference: Q.1912.5 clause 6.1.2 (i,2b)			
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message						
SIP selection	PICS 4/4 AND PICS 4/5						
criteria							
ISDN selection	PICS 1/6						
criteria			(1) (1)	1 N // TE ://	(C) +		
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer and 100rel extensions and preconditions extensions in the SIP Supported header: including a SDP for A-law (PCMA) and μ-law (PCMU)</li> <li>the SUT shall delete A-law (PCMA), if present, from the media description that it will send back in the SDP answer</li> <li>the SETUP shall be deferred until all preconditions have been met. The BC is constructed from the ISUP TMR or USI</li> </ul>						
SIP parameter	SIP INVITE: Audio RTP/AVP 0.8						
values	200 OK: Audio RTP/AVP	8					
ISDN parameter	SETUP BC: A-law						
values							
Comments	SIP-I		SU	Т	ISDN		
	INVITE(IAM)	<b>→</b>					
	183 session Progress	<b>←</b>					
	PRACK	<b>→</b>					
	200 OK PRACK	+					
	UPDATE	<b>→</b>		<b>→</b>	SETUP		
	200 OK UPDATE	+					
	180 Ringing(ACM)	<b>←</b>		+	ALERTING		
	200 OK INVITE(ANM)	<b>←</b>		+	CONNECT		
	ACK	<b>→</b>					
			Convers	sation			
	BYE(REL)	<b>→</b>		<b>→</b>	DISCONNECT		
	200 OK BYE(RLC)	<b>←</b>		+	RELEASE		
				<b>→</b>	RELEASE COMPLETE		

TP501007	SIP reference: RFC	3261			O 10	ISDN reference: 912.5 clause 6.1.2 (i,1)	
TSS reference	SIP-I-ISDN/Basic_call/ Sending	a of t	ne SETU			712.3 Clause 0.1.2 (1,1)	
SIP selection criteria	NOT PICS 4/4 AND NOT PICS				3-		
ISDN selection criteria	PICS 1/6	PICS 1/6					
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer including a SDP for A-law (PCMA) and μ-law (PCMU)</li> <li>the SUT shall delete A-law (PCMA) from the media description that it will send back in the SDP answer</li> <li>the SUT shall immediately send out the SETUP. The BC is constructed from the ISUP TMR or USI</li> </ul>						
SIP parameter values	SIP INVITE: Audio RTP/AVP 200 OK: Audio RTP/AVP						
ISDN parameter values	SETUP BC: A-law						
Comments	SIP-I		SU	Т		ISDN	
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP	
	180 Ringing(ACM)	+			+	ALERTING	
	200 OK INVITE(ANM)	<b>←</b>			<b>←</b>	CONNECT	
	ACK	<b>→</b>					
	Conversation						
	BYE(REL)	<b>→</b>		_	<b>→</b>	DISCONNECT	
	200 OK BYE(RLC)	+			<b>←</b>	RELEASE	
					<b>→</b>	RELEASE COMPLETE	

TP501008	SIP reference: RF0	C 3261				ISDN reference:	
				G	0.19	12.5 clause 6.1.2 (i,2b)	
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message						
SIP selection	PICS 4/4 AND PICS 4/5						
criteria							
ISDN selection	PICS 1/6						
criteria							
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer and 100rel extensions and preconditions extensions in the SIP Require header including a SDP for A-law (PCMA) and μ-law (PCMU)</li> <li>the SUT shall delete A-law (PCMA), if present, from the media description that it will send back in the SDP answer</li> <li>the SETUP shall be deferred until all preconditions have been met. The BC is constructed from the ISUP TMR or USI. The BC is constructed from the ISUP TMR or USI</li> </ul>						
SIP parameter	INVITE: Audio RTP/AVP 0 8						
values	200 OK: Audio RTP/AVP 8						
ISDN parameter	SETUP: BC: A-law						
values							
Comments	SIP-I		SU	JT		ISDN	
	INVITE(IAM)	<b>→</b>					
	183 session Progress	+					
	PRACK	<b>→</b>					
	200 OK PRACK	+					
	UPDATE	→			<b>→</b>	SETUP	
	200 OK UPDATE	+					
	180 Ringing(ACM)	<b>←</b>			<b>+</b>	ALERTING	
	200 OK INVITE(ANM)	+			<b>←</b>	CONNECT	
	ACK	→					
			Conver	sation			
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT	
	200 OK BYE(RLC)	+			<b>+</b>	RELEASE	
					<b>→</b>	RELEASE COMPLETE	

TP501009	SIP reference: RI	FC 3261			ISDN reference:			
	EN 383 001 clause 6.1.3.5.2.2							
TSS reference	SIP-I-ISDN/Basic_call/ Sen	ding_of the	_SETUP_	_message				
SIP selection	NOT PICS 4/4 AND NOT P	ICS 4/5 AN	ID PICS 1	/9				
criteria								
ISDN selection								
criteria								
Test purpose	A-Law, <b>then independent</b> Sends a SETUP message	Ensure that the SUT on receipt of an INVITE message with a SDP offer for µ-Law and A-Law, then independent from the received order of preference Sends a SETUP message						
	the G.711 a-law codec sha	I be returne	ed in the S	DP answei	r as preferred codec			
SIP parameter	Offer: m=audio 4711 R	TP/AVP 0 8	3					
values	Answer: m=audio 4712 R	TP/AVP 8 (	)					
ISDN parameter values								
Comments	SIP-I		SUT		ISDN			
	INVITE(IAM)	→		→	SETUP			
	180 Ringing(ACM)	<b>←</b>		+	ALERTING			
	200 OK INVITE(ANM)	+		+	CONNECT			
	ACK	<b>→</b>						
	Conversation							
	BYE(REL)	<b>→</b>		<b>→</b>	DISCONNECT			
	200 OK BYE(RLC)	+		+	RELEASE			
				<b>→</b>	RELEASE COMPLETE			

TP501010	SIP reference: RF	C 3261			ISDN reference:		
					33 001 clause 6.1.3.5.2.2		
TSS reference	SIP-I-ISDN/Basic_call/ Send	ding_of the	_SETU	P_message			
SIP selection	PICS 4/4 AND PICS 4/5 AN	D PICS 1/	9				
criteria							
ISDN selection							
criteria							
Test purpose	Ensure that the SUT on rece						
					in the SIP Supported header:,		
	then independent from the						
	<ul> <li>the SETUP shall be def</li> </ul>						
	• the G.711 a-law codec	shall be re	turned i	n the SDP ar	nswer as preferred codec		
SIP parameter	Offer: m=audio 4711 R						
values	Answer: m=audio 4712 R	TP/AVP 8	0				
ISDN parameter							
values							
Comments	SIP-I		SU	Т	ISDN		
	INVITE(IAM)	<b>→</b>					
	183 session Progress	<b>←</b>					
	PRACK	<b>→</b>					
	200 OK PRACK	+					
	UPDATE	<b>→</b>		<b>→</b>	SETUP		
	200 OK UPDATE	+					
	180 Ringing(ACM)	+		+	ALERTING		
	200 OK INVITE(ANM)	+		+	CONNECT		
	ACK	<b>→</b>					
		(	Convers	sation			
	BYE(REL)	<b>→</b>		→	DISCONNECT		
	200 OK BYE(RLC)	+		+	RELEASE		
				<b>→</b>	RELEASE COMPLETE		

TP501011	SIP reference: R	FC 3261			ISDN reference:		
					33 001 clause 6.1.3.5.2.2		
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message						
SIP selection	PICS 4/4 AND PICS 4/5 AND	ND PICS 1/	9				
criteria							
ISDN selection							
criteria							
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for µ-Law and						
					in the SIP Require header,		
	then independent from th						
	<ul> <li>the SETUP shall be de</li> </ul>	eferred unti	l all prec	onditions hav	ve been met		
	<ul> <li>the G.711 a-law coded</li> </ul>	shall be re	turned i	n the SDP ar	nswer as preferred codec		
SIP parameter	Offer: m=audio 4711 R	TP/AVP 0	8				
values	Answer: m=audio 4712 R	TP/AVP 8	0				
ISDN parameter							
values							
Comments	SIP-I		SU	T	ISDN		
	INVITE(IAM)	→					
	183 session Progress	+					
	PRACK	→					
	200 OK PRACK	<b>←</b>					
	UPDATE	→		<b>→</b>	SETUP		
	200 OK UPDATE	+					
	180 Ringing(ACM)	<b>←</b>		<b>←</b>	ALERTING		
	200 OK INVITE(ANM)	+		+	CONNECT		
	ACK	<b>→</b>					
			Convers	sation			
	BYE(REL)	<b>→</b>		→	DISCONNECT		
	200 OK BYE(RLC)	+		+	RELEASE		
	, ,			<b>→</b>	RELEASE COMPLETE		

TP501012	SIP reference: RFC 3261 ISDN reference:							
TSS reference	Q.1912.5 clause 6.1.2 (i,1)  SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message							
SIP selection	NOT PICS 4/4 AND NOT PI					T DICC 4/40		
criteria	NOT PICS 4/4 AND NOT PI	JS 4/5 F	AND PICS	1/9 ANL	טאו כ	11 PICS 4/19		
ISDN selection								
criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams and based on operator policy then if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected							
SIP parameter values	Offer: m=audio 4711 RT m= video 4712 RT  Answer: m=audio 4711 RT m=video 0 RTP/A	ΓΡ/AVP 8	31					
ISDN parameter								
values Comments	SIP-I		SU	T .	1	ISDN		
Comments		<b>→</b>	30	<u> </u>	<b>→</b>	SETUP		
	INVITE(IAM)	+			<del>7</del>	ALERTING		
	180 Ringing(ACM)	+			<del>-</del>			
	200 OK INVITE(ANM)	<b>∀</b>			~	CONNECT		
	ACK	7	<b>^</b>					
	D)(E(DEL)	+	Conver	sation		DIOCONINIECT		
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT		
	200 OK BYE(RLC)	+			+	RELEASE		
					<b>→</b>	RELEASE COMPLETE		

TP501013	SIP reference: RF	C 3261		Q.1	ISDN reference: 912.5 clause 6.1.2 (i,1)				
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message								
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19								
criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams, 100rel extensions and preconditions extensions in the SIP Supported header: and based on operator policy then if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected								
SIP parameter	Offer: m=audio 4711 R	TP/AVP 8							
values	m= video 4712 R	TP/AVP 3	1						
	Answer: m=audio 4711 R m=video 0 RTP//	-							
ISDN parameter									
values									
Comments	SIP-I		SU	IT	ISDN				
	INVITE(IAM)	→							
	183 session Progress	←							
	PRACK	→							
	200 OK PRACK	<b>←</b>							
	UPDATE	<b>→</b>		→	SETUP				
	200 OK UPDATE	+							
	180 Ringing(ACM)	+		+	ALERTING				
	200 OK INVITE(ANM)	+		+	CONNECT				
	ACK	<b>→</b>							
			Conver	sation					
	BYE(REL)	<b>→</b>		<b>→</b>	DISCONNECT				
	200 OK BYE(RLC)	+		+					
	,			→	RELEASE COMPLETE				

TP501014	SIP reference: RFC 3261			ISDN reference:				
					Q.1912.5 clause 6.1.2 (i,1)			
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message							
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19							
criteria								
ISDN selection								
criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams, 100rel extensions and preconditions extensions in the SIP Require header and based on operator policy then if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected							
SIP parameter	Offer: m=audio 4711 R	TP/AVP 8						
values	m= video 4712 R	TP/AVP 31						
ISDN parameter values	Answer: m=audio 4711 R' m=video 0 RTP//	-						
Comments	SIP-I		SUT		ISDN			
	INVITE(IAM)	<b>→</b>						
	183 session Progress	+						
	PRACK	→						
	200 OK PRACK	+						
	UPDATE	→		→	SETUP			
	200 OK UPDATE	+						
	180 Ringing(ACM)	<b>←</b>		<b>←</b>	ALERTING			
	200 OK INVITE(ANM)	+		<b>←</b>	CONNECT			
	ACK	<b>→</b>						
			Conversation					
	BYE(REL)	<b>→</b>		<b>→</b>	DISCONNECT			
	200 OK BYE(RLC)	+		<b>←</b>	RELEASE			
				→	RELEASE COMPLETE			

TP501015	SIP reference: RFC 3261		ISDN reference: Q.1912.5 clause 6.1.2 (i,1)			
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message					
SIP selection	NOT PICS 4/4 AND NOT PICS 4/5 AND PICS 1/9 AND PICS 4/19					
criteria						
ISDN selection						
criteria						
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams and based on operator policy then					
	the call is refused with a 415 Unsupported media type response					
SIP parameter	Offer: m=audio 4711 RTP/AVP 8					
values	m= video 4712 RTP/AVP 31					
ISDN parameter						
values						
Comments	SIP-I		SUT	ISDN		
	INVITE(IAM)	<b>→</b>				
	415 Unsupported media type	+				
	ACK	<b>→</b>				

TP501016	SIP reference: RFC 3261			ISDN reference:		
			Q.19	912.5 clause 6.1.2 (i,1)		
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message					
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND PICS 4/19					
criteria						
ISDN selection						
criteria						
Test purpose	Ensure that the SUT on receipt of an					
	one media streams, 100rel extensions and preconditions extensions in the SIP Supported					
	header: and based on operator policy then					
	the call is refused with a 415 Unsupported media type response					
SIP parameter	Offer: m=audio 4711 RTP/AVP 8					
values	m= video 4712 RTP/AVP 31					
ISDN parameter						
values						
Comments	SIP-I		SUT	ISDN		
	INVITE(IAM)	<b>→</b>				
	415 Unsupported media type	+				
	ACK	<b>→</b>				

TP501017	SIP reference: RFC 3261		0.19	ISDN reference: 912.5 clause 6.1.2 (i,1)		
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message					
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND PICS 4/19					
ISDN selection criteria						
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams, 100rel extensions and preconditions extensions in the SIP Require header and based on operator policy then the call is refused with a 415 Unsupported media type response					
SIP parameter	Offer: m=audio 4711 RTP/AVP 8					
values	m= video 4712 RTP/AVP 31					
ISDN parameter						
values						
Comments	SIP-I		SUT	ISDN		
	INVITE(IAM)	<b>→</b>				
	415 Unsupported media type	+				
	ACK	<b>→</b>				

TP501018	SIP reference: RFC 3	261				ISDN reference:		
	•					.1912.5 clause 6.1.3		
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message							
SIP selection	PICS 1/2							
criteria								
ISDN selection	PICS 1/9							
criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message containing an encapsulated IAM with the media description defined with the "a = " "b =" and "m=" lines set to a_b_m_LINE_VALUE sends the SETUP message with the Bearer Capability (BC) constructed from the USI parameter in the encapsulated IAM or, if absent, constructed from the TMR of the encapsulated IAM according the ISUP rules							
SIP parameter values								
ISDN parameter	SETUP; BC Coding standard:	CCIT	T standa	rdized cod	ding			
values	Information transfer capability: Constructed from the USI or from the TMR							
	transfer mode: circuit mode							
	information transfer rate: 64 kbits/s							
Comments	SIP-I		SU	T		ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
	180 Ringing(ACM)	+			<b>←</b>	ALERTING		
	200 OK INVITE(ANM)	+			<b>←</b>	CONNECT		
	ACK	<b>→</b>						
	Conversation							
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT		
	200 OK BYE(RLC)	<del>-</del>			<del>(</del>	RELEASE		
	→ RELEASE COMPLETE							

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

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

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

TP501021	SIP reference: RF	C 3261				ISDN reference:				
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message									
SIP selection										
criteria										
ISDN selection										
criteria										
Test purpose	Ensure that the SUT on receipt of an INVITE message with a Called party number +CC NDC SN where CC is the country code of the network in which the next hop terminates, component of the Request-URI send a SETUP message Type of number: "Subscriber number", remove "+CC NDC" and use the remaining digits to fill the Address signals contained in the user info.  Numbering plan Indicator ISDN (Telephony) numbering plan									
SIP parameter values										
ISDN parameter values	SETUP : Called party numb	er								
Comments	SIP-I		SU	T		ISDN				
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
	180 Ringing(ACM)	<b>←</b>			+	ALERTING				
	200 OK INVITE(ANM)	+			+	CONNECT				
	ACK	→								
	Conversation									
	BYE(REL)	→			<b>→</b>	DISCONNECT				
	200 OK BYE(RLC)	+			+	RELEASE				
					<b>→</b>	RELEASE COMPLETE				

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

## A.1.1.1.2 Sending of the INFO

TP502001	SIP reference: RFC		ISDN reference:						
TSS reference	SIP-I-ISDN/Basic_call/Sending	of IN	NFO_mes	sage					
SIP selection	PICS 3/4								
criteria									
ISDN selection	PICS 3/8								
criteria									
Test purpose	previous INVITE which whereby the number of digits already accepted sends a INFO and pass it  The INFO shall contain in	Ensure that the SUT receives an INVITE with the INFOe Call-ID and From tag as a previous INVITE which whereby the number of digits in the Request-URI is <b>greater</b> than the number of digits already accumulated for the call sends a INFO and pass it to outgoing ISDN procedures  The INFO shall contain in its Subsequent Number parameter only the additional digits received in this Request-URI compared with the digits already accumulated for the call							
SIP parameter					<u> </u>	,,			
values									
ISDN parameter									
values									
Comments	SIP-I		SU	IT		ISDN			
	INVITE(IAM)	→			<b>→</b>	SETUP			
	INVITE(SAM)	<b>→</b>			<b>→</b>	INFO			
	INVITE(SAM)	<b>→</b>			<b>→</b>	INFO			
	180 Ringing(ACM)	+			<b>←</b>	ALERTING			
	200 OK INVITE(ANM)	+			<b>←</b>	CONNECT			
	ACK	<b>→</b>							
			Convers	sation					
	BYE(REL)	→			<b>→</b>	DISCONNECT			
	200 OK BYE(RLC)	+			<b>←</b>	RELEASE			
					<b>→</b>	RELEASE COMPLETE			

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

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

TP503001	SIP reference: RFC 3261			ISDN reference: Q.1912.5 clause 6.5 1)						
TSS reference	SIP-I-ISDN/Basic_call/Re	ceipt_of ALE	RTING_	_CALL-PF	ROC	C_PROGRESS_message				
SIP selection										
criteria										
ISDN selection criteria	PICS 3/8 AND PICS 1/6	PICS 3/8 AND PICS 1/6								
Test purpose	<ul> <li>Ensure that the SUT in call state N25, on receipt the ALERTING message</li> <li>the 180 Ringing SIP response is sent. Ensure that the in-band information can be transmitted to the calling user</li> </ul>									
SIP parameter										
values										
ISDN parameter values										
Comments	SIP-I		SU	T		ISDN				
	INVITE(IAM)	→			<b>→</b>	SETUP				
	,				<b>+</b>	SETUP ACK				
	180 Ringing(ACM)	+			<b>+</b>	ALERTING				
	In	Inband Info								
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT				
	200 OK BYE(RLC)	+			<b>←</b>	RELEASE				
					<b>→</b>	RELEASE COMPLETE				

TP503002	SIP reference: RFC	3261			Q.	ISDN reference: 1912.5 clause 6.5 1)				
TSS reference	SIP-I-ISDN/Basic_call/Receipt	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message								
SIP selection criteria						-				
ISDN selection criteria										
Test purpose	<ul> <li>Ensure that the SUT in call state N6, on receipt the ALERTING message,</li> <li>a 180 Ringing SIP response is sent. Ensure that the in-band information can be transmitted to the calling user</li> </ul>									
SIP parameter										
values										
ISDN parameter values										
Comments	SIP-I		SU	T		ISDN				
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
	180 Ringing(ACM)	+			<b>←</b>	ALERTING				
	Inband	Inband Info								
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT				
	200 OK BYE(RLC)	<b>←</b>			<del>-</del>	RELEASE				
					<b>→</b>	RELEASE COMPLETE				

TP503003	SIP reference: RFC 3261				Q.1	ISDN reference: 1912.5 clause 6.5 2)					
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message										
SIP selection											
criteria											
ISDN selection											
criteria											
Test purpose	<ul> <li>Ensure that the SUT in call state N9, on receipt the ALERTING message,</li> <li>a 180 Ringing SIP response is sent. Ensure that the in-band information can be transmitted to the calling user</li> </ul>										
SIP parameter values											
ISDN parameter values											
Comments	SIP-I		SU	ΙT		ISDN					
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP					
	183 Session Progress(ACM)	<b>←</b>			<del>(</del>	CALL PROCEEDING					
	180 Ringing(CPG)	+			<del>(</del>	ALERTING					
	Inband	Info		<u>'</u>							
	BYE(REL) → DISCONNECT										
	200 OK BYE(RLC)	+			<del>(</del>	RELEASE					
					<b>→</b>	RELEASE COMPLETE					

TP503004	SIP reference: RFC 3261			ISDN reference:							
					Q.1	912.5 clause 6.5 2)					
TSS reference	SIP-I-ISDN/Basic_call/Receipt	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message									
SIP selection											
criteria											
ISDN selection	PICS 3/8 AND PICS 1/6										
criteria											
Test purpose	Ensure that the SUT in call state N25, on receipt of the CALL PROCEEDING message										
	a 183 Session Progress with an encapsulated ACM is sent to the previous entity										
SIP parameter	183 Session Progress encapsu	183 Session Progress encapsulated ACM: BCi Called party status = no indication									
values											
ISDN parameter	CALL PROCEEDING										
values											
Comments	SIP-I		SU	IT		ISDN					
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP					
	183 Session Progress(ACM)	+			<b>←</b>	CALL PROCEEDING					
	CANCEL(REL)	<b>→</b>			<b>→</b>	RELEASE					
	200 OK CANCEL(RLC)	+			+	RELEASE COMPLETE					

TP503005	SIP reference: RFC 3261				ISDN reference:						
						1912.5 clause 6.5 2)					
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message										
SIP selection											
criteria											
ISDN selection											
criteria											
Test purpose	Ensure that the SUT in call state N6, on receipt of the CALL PROCEEDING message										
	containing a progress indicator set to PI_VALUE,										
	<ul> <li>a 183 Session Progress w</li> </ul>	ith an	encapsul	ated ACN	/l is	sent to the previous entity					
SIP parameter	183 Session Progress encapsu										
values	with Progress indicator			•	•						
ISDN parameter	CALL PROCEEDING										
values											
Comments	SIP-I		SU	T		ISDN					
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP					
	183 Session Progress(ACM)	+			<b>←</b>	CALL PROCEEDING(PI)					
	CANCEL(REL)	<b>→</b>			<b>→</b>	RELEASE					
	200 OK CANCEL(RLC)	+			<b>←</b>	RELEASE COMPLETE					

TP503006	SIP reference: RFC 3261				0 1	ISDN reference: 1912.5 clause 6.5 2)		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message							
SIP selection		_0.7.						
criteria								
ISDN selection								
criteria								
Test purpose	Ensure that the SUT in call state N9, on receipt of the PROGRESS message containing a <b>progress indicator</b> set to PI_VALUE a 183 Session Progress with an encapsulated ACM is sent to the previous entity							
SIP parameter values	183 Session Progress with encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with Progress indicator							
ISDN parameter values	CALL PROCEEDING							
Comments	SIP-I		SU	ΙΤ		ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
	183 Session Progress(ACM)	+			<b>←</b>	CALL PROCEEDING		
	183 Session Progress(CPG)	+			+	PROGRESS(PI)		
				•				
	CANCEL(REL)	<b>→</b>			<b>→</b>	RELEASE		
	200 OK CANCEL(RLC)	<b>←</b>			<b>←</b>	RELEASE COMPLETE		

TP503007	SIP reference: RFC	3261			Q.1	ISDN reference: 912.5 clause 6.5 2)			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message								
SIP selection criteria						<u> </u>			
ISDN selection criteria	PICS 1/6								
Test purpose		Ensure that the SUT in call state N9, on receipt of the ALERTING message containing a progress indicator set to PI_VALUE, the 180 Ringing SIP response is sent							
SIP parameter	180 Ringing encapsulated AC			rty status	=su	bscriber free, ATP with			
values	Progress indicator			•		·			
ISDN parameter values	ALERTING(PI)								
Comments	SIP-I		SU	IT		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	180 Ringing(ACM)	+			<b>←</b>	ALERTING(PI)			
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT			
	200 OK BYE(RLC)	+			<del>(</del>	RELEASE			
					<b>→</b>	RELEASE COMPLETE			

TP503008	SIP reference: RF	C 3261				ISDN reference:			
						1912.5 clause 6.5 2)			
TSS reference	SIP-I-ISDN/Basic_call/Recei	ipt_of ALE	RTING_	_CALL-PR	OC	C_PROGRESS_message			
SIP selection criteria									
ISDN selection criteria	PICS 1/6	PICS 1/6							
Test purpose	progress indicator set to P	Ensure that the SUT in call state N25, on receipt of a ALERTING message containing the progress indicator set to PI_VALUE the 180 Ringing SIP response is sent							
SIP parameter	180 Ringing encapsulated A	180 Ringing encapsulated ACM: BCi called party status=subscriber free, ATP with							
values	Progress indicator		•	•					
ISDN parameter	ALERTING(PI)								
values	, ,								
Comments	SIP-I		SU	IT		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	INVITE(SAM)	<b>→</b>			<del>)</del>	INFO			
	400 Dinging (ACM)					ALEDTING(DI)			
	180 Ringing(ACM) ← ALERTING(PI)								
	DVE(DEL)			1.	_	DICCONNECT			
	BYE(REL)	<b>→</b>			<u> </u>	DISCONNECT			
	200 OK BYE(RLC)	<b>←</b>		•	<del>(</del>	RELEASE			
				•	<b>→</b>	RELEASE COMPLETE			

TP503009	SIP reference: RFC 3261				ISDN reference: Q.1912.5 clause 6.5 2)				
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message								
SIP selection criteria									
ISDN selection criteria	PICS 1/6								
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message sends out a SETUP message in state N6, where the end of address signalling is determined by analysis of the called party number to indicate that <b>a sufficient number of digits has been received</b> to route the call to the called party, on receipt of a CALL PROCEEDING a 183 Session Progress with an encapsulated ACM is sent to the previous entity								
SIP parameter values									
ISDN parameter values									
Comments	SIP-I		SL	IT		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+		•	+	CALL PROCEEDING			
	CANCEL(REL)	<b>→</b>			<b>→</b>	RELEASE			
	200 OK CANCEL(RLC)	+			<del>(</del>	RELEASE COMPLETE			

TP503010	SIP reference: RFC	SIP reference: RFC 3261			ISDN reference: Q.1912.5 clause 6.5 2)			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message							
SIP selection			_					
criteria								
ISDN selection criteria								
Test purpose	Ensure that the SUT in call state N25, on receipt of a PROGRESS message containing no progress indicator,  a 183 Session Progress with an encapsulated ACM is sent to the previous entity							
SIP parameter values		183 Session Progress encapsulated ACM: BCi Called party status = no indication						
ISDN parameter values	CALL PROCEEDING							
Comments	SIP-I	;	SUT		ISDN			
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP			
	183 Session Progress(ACM)	+		+	CALL PROCEEDING			
				+	PROGRESS			
	CANCEL(REL)	<b>→</b>		→	RELEASE			
	200 OK CANCEL(RLC)	<del>-</del>		<b>←</b>	RELEASE COMPLETE			

TP503011	SIP reference: R	FC 3261				ISDN reference:				
					Q	1.1912.5 clause 6.6				
TSS reference	SIP-I-ISDN/Basic_call/Rec	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message								
SIP selection										
criteria										
ISDN selection	PICS 1/6	PICS 1/6								
criteria										
Test purpose	Ensure that the SUT in the	Idle state or	receip	t of a INVI	ΤE	message sends out a SETUP				
	message, receives an ALE	RTING mes	sage, h	aving sent	ta ′	180 Ringing message, on				
	receipt of a PROGRESS m	nessage								
	<ul> <li>the PROGRESS is no</li> </ul>	t interworked								
SIP parameter										
values										
ISDN parameter values										
Comments	SIP-I		SU	Т		ISDN				
	INVITE(IAM)	<b>→</b>		•	<b>→</b>	SETUP				
	180 Ringing(ACM)	+		•	<del>(</del>	ALERTING				
	<b>←</b> PROGRESS									
	BYE(REL)	<b>→</b>		•	<b>→</b>	DISCONNECT				
	200 OK BYE(RLC)	+		•	<del>(</del>	RELEASE				
				•	<b>→</b>	RELEASE COMPLETE				

TP503012	SIP reference: RFC 3261				_	ISDN reference: 0.1912.5 clause 6.6			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message								
SIP selection	Oil -1-10DN/Dasic_call/Necelpt	_01 /\L	LIVIIIVO_	_OALL-I	INOC	NOONEOO_message			
criteria									
ISDN selection	PICS 1/6								
criteria	1100 170	1 100 1/0							
Test purpose		9							
SIP parameter values	183 Session Progress: Encaps	ulated	I ACM, ca	alled part	y sta	tus indicator=no indication			
ISDN parameter values									
Comments	SIP-I		SU	JT		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROCEEDING			
					+	PROGRESS			
	CANCEL(REL)	<b>→</b>			<b>→</b>	RELEASE			
	200 OK CANCEL(RLC)	+			+	RELEASE COMPLETE			

TP503013	SIP reference: RFC 3	3261		ISDN reference: Q.1912.5 clause 6.6					
TSS reference	SIP-I-ISDN/Basic call/Receipt	SIP-I-ISDN/Basic call/Receipt of ALERTING CALL-PROC PROGRESS message							
SIP selection									
criteria									
ISDN selection	PICS 1/6								
criteria									
Test purpose	message, receives a CALL PR	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message, receives an ALERTING sends a 180 Ringing with encapsulated CPG Alerting							
SIP parameter	183 Session Progress with end					status indicator=no indication			
values	180 Ringing encapsulated GPC	3: Eve	nt indicat	or=Alertin	ng				
ISDN parameter values									
Comments	SIP-I		SU	IT		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			4	CALL PROCEEDING			
	180 Ringing(CPG)	+			<b>+</b>	ALERTING			
	Inband	Info							
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT			
	200 OK BYE(RLC)	+			<b>4</b>	RELEASE			
					<b>→</b>	RELEASE COMPLETE			

TP503014	SIP reference: RFC	3261			ISDN reference	ce:			
					Q.1912.5 clause				
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message								
SIP selection									
criteria									
ISDN selection	PICS 1/6								
criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message sends out a SETUP message, receives a CALL PROCEEDING message, receives a PROGRESS message								
	with a progress indicator who								
	receipt of a ALERTING Messa		Jiogress	s description	i value is set to F	I_VALUE, UII			
	<ul> <li>sent a 180 Ringing messa</li> </ul>								
SIP parameter	183 Session Progress with end		ed ACM	1· called na	rty status indicato	r-no indication			
values	183 Session Progress with end								
	Progress indicator	Japoula		ovom man	5a.o 10g.000, 7	***************************************			
	180 Ringing encapsulated GP	G: Even	t indicat	or=Alerting					
ISDN parameter									
values									
Comments	SIP-I		SU	IT	ISDN				
	INVITE(IAM)	→		] -	SETUP				
	183 Session Progress(ACM)	+		•	CALL PROCE	EDING			
	183 Session Progress(CPG)			•	PROGRESS(F	기)			
	180 Ringing(CPG)	<b>←</b>		•	<ul><li>ALERTING</li></ul>				
	Inband Info								
	BYE(REL)	→		=		<u> </u>			
	200 OK BYE(RLC)	+		•	· · · · · · · · · · · · · · · · · · ·				
				]-	RELEASE CO	MPLETE			

TP503015	SIP reference: RFC	3261			_	ISDN reference: 1.1912.5 clause 6.6			
T00 (			-DTINIO	0411 004					
TSS reference	SIP-I-ISDN/Basic_call/Receipt	of AL	=RTING_	CALL-PRO	UC	_PROGRESS_message			
SIP selection criteria									
ISDN selection criteria	PICS 1/6								
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message sends out a SETUP message, on receipt of a ALERTING Message, receives a PROGRESS message with a <b>progress indicator</b> where the progress description value is set to PI_VALUE  • sent a 183 Session Progress message containing a encapsulated CPG								
SIP parameter	180 Ringing encapsulated ACM	Л: BCi	called pa	rty status=	sul	oscriber free			
values	183 Session Progress with end	apsula	ted CPG	: event ind	lica	tor=Progress, ATP with			
	Progress indicator	•				•			
ISDN parameter values									
Comments	SIP-I		SU	T		ISDN			
	INVITE(IAM)	<b>→</b>		-	<del>}</del>	SETUP			
	180 Ringing(ACM)	+		•	<del>(</del>	ALERTING			
	183 Session Progress(CPG)	+		•	<del>(</del>	PROGRESS(PI)			
	Inband Info								
	BYE(REL)	BYE(REL) → DISCONNECT							
	200 OK BYE(RLC)	+		•	<b>(</b>	RELEASE			
				-	<del>&gt;</del>	RELEASE COMPLETE			

TP503016	SIP reference: RFC	3261			C	ISDN reference: Q.1912.5 clause 6.6				
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message									
SIP selection						-				
criteria										
ISDN selection	PICS 1/6									
criteria										
Test purpose	message, receives a CALL PR	Ensure that the SUT in the Idle state on receipt of a INVITE message sends out a SETUP message, receives a CALL PROCEEDING, receives a PROGRESS message with a progress indicator where the progress description value is set to PI_VALUE								
SIP parameter values		183 Session Progress with encapsulated ACM: called party status indicator=no indication 183 Session Progress with encapsulated CPG event indicator=Progress, ATP with								
ISDN parameter values										
Comments	SIP-I		SU	IT		ISDN				
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
	183 Session Progress(ACM)	+			+	CALL PROCEEDING				
	183 Session Progress(CPG)				+	PROGRESS(PI)				
	Inband	Info								
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT				
	200 OK BYE(RLC)	+			+	RELEASE				
				-	<b>→</b>	RELEASE COMPLETE				

## A.1.1.1.4 Receipt of the CONNECT Message

TP504001	SIP reference: RFC 32	261			ISDN reference:					
				(	Q.1912.5 clause 6.7					
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of CONNECT_message									
SIP selection										
criteria										
ISDN selection criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message sends out a SETUP message, receives an ALERTING message, on receipt of a CONNECT message  • sends a 200 OK INVITE to the previous entity  The bearer path shall be connected in both directions when the following condition is									
	satisfied:  • The BICC outgoing bearer set-up procedure, (Q.1902.4) is successfully completed In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "notification not required", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312) that sufficient preconditions have been met for the session to proceed									
SIP parameter values										
ISDN parameter										
values										
Comments	SIP-I		SUT		ISDN					
	INVITE(IAM)	<b>→</b>		→	SETUP					
	180 Ringing(ACM)	<del>(</del>		<b>←</b>	ALERTING					
	200 OK INVITE(ANM)	<del>-</del>		+	CONNECT					
	ACK	<b>→</b>								
			versation							
	BYE(REL)	<b>→</b>		<b>→</b>	DISCONNECT					
	200 OK BYE(RLC)	<del>(</del>		+	RELEASE					
				<b>→</b>	RELEASE COMPLETE					

TP504002	SIP reference: RFC 3261			ISDN reference: Q.1912.5 clause 6.7				
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of CONNECT_message							
SIP selection		•						
criteria								
ISDN selection								
criteria								
Test purpose	message, receives a ALER sends a 200 OK INVITE to 1	TING mes	sage, on	receipt of a	message sends out a SETUP CONNECT message			
SIP parameter								
values								
ISDN parameter values								
Comments	SIP-I		SU'	Τ	ISDN			
	INVITE(IAM)	<b>→</b>						
	183 session Progress	+						
	PRACK	→						
	200 OK PRACK	+						
	UPDATE	→		<b>→</b>	SETUP			
	200 OK UPDATE	+						
	180 Ringing(ACM)	+		+	ALERTING			
	200 OK INVITE(ANM)	+		+	CONNECT			
	ACK	→						
			Convers	ation				
	BYE(REL)	<b>→</b>		<b>→</b>	DISCONNECT			
	200 OK BYE(RLC)	+		<del>(</del>	RELEASE			
				<b>→</b>	RELEASE COMPLETE			

TP504003	SIP reference: RFC 32	61			ISDN reference: Q.1912.5 clause 6.7			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of CONNECT_message							
SIP selection criteria								
ISDN selection								
criteria								
Test purpose	ALERTING message, on receipt	SDP offer was not received in the initial INVITE. Ensure that the SUT, having received the ALERTING message, on receipt of an CONNECT message  sends a 200 OK INVITE to the UAC. The 200 OK INVITE shall include an SDP offer consistent with the BC used						
SIP parameter	200 OK SDP offer							
values	ACK SDP answer							
ISDN parameter values								
Comments	SIP-I		SI	JT		ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
	180 Ringing(ACM)	+			+	ALERTING		
	200 OK INVITE(ANM; SDP1)	+			+	CONNECT		
	ACK(SDP2)	<b>→</b>						
		Conversation						
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT		
	200 OK BYE(RLC)	<b>←</b>			<b>←</b>	RELEASE		
					<b>→</b>	RELEASE COMPLETE		

TP504004	SIP reference: RFC	3261				ISDN reference:		
				(	<b>Q.1</b>	912.5 clause 6.4, 6.7		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of CONNECT_message							
SIP selection criteria								
ISDN selection criteria								
Test purpose	message	SDP offer was received in the initial INVITE. Ensure that the SUT, on receipt of an CON message  sends a 200 OK INVITE to the previous entity						
SIP parameter values	200 OK INVITE: encapsulated	200 OK INVITE: encapsulated CON						
ISDN parameter values								
Comments	SIP-I		SU	T		ISDN		
	INVITE(IAM)	<b>→</b>			<del>&gt;</del>	SETUP		
	200 OK INVITE(CON)	+		•	<del>(</del>	CONNECT		
	ACK	<b>→</b>						
	Conversation							
	BYE(REL)	<b>→</b>		•	<b>→</b>	DISCONNECT		
	200 OK BYE(RLC)	+		•	<del>(</del>	RELEASE		
				-	<b>→</b>	RELEASE COMPLETE		

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

TP505001	SIP reference: RFC	ISDN reference: 1912.5 clause 6.11.2						
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE							
SIP selection	NOT PICS 4/10							
criteria								
ISDN selection								
criteria								
Test purpose  SIP parameter values	CV_ISDN, location LOC_ISDN the SUT immediately re	an R : quest appro	ELEASE s the disc priate SIF	COMPLE connection	TE i	message with the Cause value		
ISDN parameter	REL_COMP: cause value: CV	_ISDI	N (PIXIT)					
values								
Comments	SIP-I		SL	JT		ISDN		
	INVITE(IAM)	→			<b>→</b>	SETUP		
	SIP_FAILURE_VA(REL)	<b>←</b>			+	RELEASE COMPLETE		
	ACK	<b>→</b>						

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

TP505003	SIP reference: RFC 3261 ISDN reference: Q.1912.5 clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	NOT PICS 4/10	NOT PICS 4/10							
ISDN selection criteria									
Test purpose  SIP parameter	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, on receipt of an DISCONNECT message with the Cause value CV_ISDN, location LOC_ISDN:  the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN REL_COMP is returned to the ISDN side  the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA  SIP Statue-Code: SIP FAILURE VA (PIXIT)								
values		(,							
ISDN parameter values	DISC; cause value: CV_ISDN (F	PIXIT)							
Comments	SIP-I	SU	JT	ISDN					
	INVITE(IAM)	<b>→</b>	→	SETUP					
	SIP_FAILURE_VA(REL)	<b>-</b>	+	DISCONNECT					
	ACK -	<b>→</b>	→	RELEASE					
		€ RELEASE COMPLETE							

TP505004	SIP reference: RFC 3		ISDN reference:					
	Q.1912.5 clause 6.11.2							
TSS reference	SIP-I-ISDN/Basic_call/Receipt_	of_D	ISC_or_R	ELEASE				
SIP selection	NOT PICS 4/10							
criteria								
ISDN selection criteria								
Test purpose  SIP parameter values	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a SETUP ACKNOWLEDGE message, and on receipt of a RELEASE COMPLETE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN  • the SUT immediately requests the disconnection of the internal bearer path.  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA  SIP Statue-Code: SIP_FAILURE_VA (PIXIT)							
ISDN parameter values	REL_COMP: cause value: CV	_ISDI	N (PIXIT)					
Comments	SIP-I		SU	ΙΤ		ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
					<del>(</del>	SETUP ACK		
	SIP_FAILURE_VA(REL)	<b>←</b>			<del>(</del>	RELEASE COMPLETE		
	ACK →							

TP505005	SIP reference: RFC 3261 ISDN reference: Q.1912.5 clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic call/Receipt of DISC or RELEASE								
SIP selection criteria	NOT PICS 4/10	NOT PICS 4/10							
ISDN selection criteria									
Test purpose  SIP parameter values	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a SETUP ACKNOWLEDGE message, and on receipt of a RELEASE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, a REL_COMP is returned to the ISDN side  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA  SIP Statue-Code: SIP_FAILURE_VA (PIXIT)								
ISDN parameter values	RELEASE; cause value: CV_IS	DN (PIXIT)							
Comments	SIP-I	SI	JT	ISDN					
	INVITE(IAM)	<b>→</b>	<b>→</b>	SETUP					
			+	SETUP ACK					
	SIP_FAILURE_VA(REL)	<b>←</b>	+	RELEASE					
	ACK → RELEASE COMPLETE								

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

TP505007	SIP reference: RFC 3261				0 -	ISDN reference: 1912.5 clause 6.11.2			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	NOT PICS 4/10								
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a SETUP ACKNOWLEDGE message, on receipt of a re-INVITE message sends an INFORMATION message and on receipt of a REL EASE COMPLETE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN  the SUT immediately requests the disconnection of the internal bearer path.  the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA								
SIP parameter values	SIP Statue-Code: SIP_FAILURE	_VA (PI	IXIT)						
ISDN parameter values	REL_COMP: cause value: CV	_ISDN	(PIXIT)						
Comments	SIP-I		SU	T		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
					<b>←</b>	SETUP ACK			
	INVITE(IAM)	<b>→</b>			<b>→</b>	INFO			
	SIP_FAILURE_VA(REL)	+			<b>←</b>	RELEASE COMPLETE			
	ACK	<b>→</b>							

TP505008	SIP reference: RFC 3261 ISDN reference: Q.1912.5 clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection	NOT PICS 4/10								
criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a SETUP ACKNOWLEDGE message, on receipt of a re-INVITE message sends an INFORMATION message and on receipt of a REL EASE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN  the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side  the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA								
SIP parameter	SIP Statue-Code: SIP_FAILURE	_VA (PI	XIT)						
values ISDN parameter values	RELEASE; cause value: CV_IS	SDN (P	IXIT)						
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
					<b>←</b>	SETUP ACK			
	INVITE(IAM)	<b>→</b>			<b>→</b>	INFO			
	SIP_FAILURE_VA(REL)	+			+	RELEASE			
	ACK	→			<b>→</b>	RELEASE COMPLETE			

TP505009	SIP reference: RFC 3	261		Q.1	ISDN reference: 1912.5 clause 6.11.2			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE							
SIP selection	NOT PICS 4/10							
criteria								
ISDN selection								
criteria								
Test purpose		ETUP ACKNO NFORMATION CV_ISDN, Io sts the discor for re-selection	WLEDGE Namessage cation LOC nnection of ton, an ISDN	me: and C_IS the I RI	ssage, on receipt of a d on receipt of a DISCONNECT SDN internal bearer path. When the ELEASE message is returned			
SIP parameter values	SIP Statue-Code: SIP_FAILURE_	VA (PIXIT						
ISDN parameter values	DISC: cause value: CV_ISDN (	PIXIT)						
Comments	SIP-I	SI	JT		ISDN			
	INVITE(IAM)	<b>→</b>	-	<del>}</del>	SETUP			
			•	<del>(</del>	SETUP ACK			
	INVITE(IAM)	<b>→</b>			INFO			
	SIP_FAILURE_VA(REL)	<b>←</b>	•	<del>(</del>	DISCONNECT			
	ACK	<b>→</b>	-	<b>→</b>	RELEASE			
			•	<del>-</del>	RELEASE COMPLETE			

Values for test purposes TP108001 - TP108009							
	←SIP Message SIP_FAILURE_VA	← REL Cause Indicators parameter CV_ISDN,					
VA_1	404 Not Found	Cause Value No. 1 ("unallocated (unassigned) number")					
VA_2	500 Server Internal Error	Cause Value No. 2 ("no route to network")					
VA_3	500 Server Internal Error	Cause Value No. 3 ("no route to destination")					
VA_4	500 Server Internal Error	Cause Value No. 4 ("Send special information tone")					
VA_5	404 Not Found	Cause Value No. 5 ("Misdialled trunk prefix")					
VA_6	500 Server Internal Error	Cause Value No. 8 ("Pre-emption")					
VA_7	500 Server Internal Error	Cause Value No. 9 ("Pre-emption-circuit reserved for reuse")					
VA_8	486 Busy Here	Cause Value No. 17 ("user busy")					
VA_9	480 Temporarily unavailable	Cause Value No. 18 ("no user responding")					
VA_10	480 Temporarily unavailable	Cause Value No. 19 ("no answer from the user")					
VA_11	480 Temporarily unavailable	Cause Value No. 20 ("subscriber absent")					
VA_12	480 Temporarily unavailable	Cause Value No. 21 ("all rejected")					
VA_13	410 Gone	Cause Value No. 22 ("number changed")					
VA_14	480 Temporarily unavailable	Cause Value No. 25 ("Exchange routing error")					
VA_15	502 Bad Gateway	Cause Value No. 27 ("destination out of order")					
VA_16	484 Address Incomplete	Cause Value No. 28 ("invalid number format (address incomplete")					
VA_17	500 Server Internal Error	Cause Value No. 29 ("facility rejected")					
VA_18	480 Temporarily unavailable	Cause Value No. 31 ("normal unspecified") (Class default)					
VA_19	486 Busy here if Diagnostics indicator includes the (CCBS indicator = CCBS possible)	Cause Value in the Class 010 (No circuit/channel available, Cause Value No. 34)					
\/A 20	else 480 Temporarily unavailable	Causa Valua in the Class 040 (resource unavailable					
VA_20	500 Server Internal Error	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38-47) (47 is class default)					
VA_21	500 Server Internal Error	Cause Value No. 50 ("requested facility not subscribed")					
VA_21 VA_22	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_27 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)					

TP505010	SIP reference: RFC 3	3261			Q.	ISDN reference: 1912.5 clause 6.11.2				
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection criteria	NOT PICS 4/10									
ISDN selection criteria										
Test purpose  SIP parameter	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, and receives a CALL PROCEEDING message, on receipt of a RELEASE COMPLETE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN  • the SUT immediately requests the disconnection of the internal bearer path  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA  183 Session Progress encapsulated ACM: BCi Called party status = no indication									
values	SIP Statue-Code: SIP_FAILURE				,					
ISDN parameter values	REL_COMP: cause value: CV	_ISDN	l (PIXIT)							
Comments	SIP-I		SU	Т		ISDN				
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
	183 Session Progress(ACM)	+			+	CALL PROC				
	SIP_FAILURE_VA(REL)	+			+	RELEASE COMPLETE				
	ACK	<b>→</b>								

TP505011	SIP reference: RFC 3261 ISDN reference: Q.1912.5 clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	NOT PICS 4/10								
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, and receives a CALL PROCEEDING message, on receipt of a RELEASE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA								
SIP parameter	183 Session Progress encapsu			i Called p	party	status = no indication			
values	SIP Statue-Code: SIP_FAILURE								
ISDN parameter values	RELEASE; cause value: CV_I	SDN	(PIXIT)						
Comments	SIP-I		SU	T		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			<del>(</del>	CALL PROC			
	SIP_FAILURE_VA(REL)	+			<del>-</del>	RELEASE			
	ACK	<b>→</b>			<b>→</b>	RELEASE COMPLETE			

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

TP505013	SIP reference: RFC 3261 ISDN reference: Q.1912.5 clause 6.11.2									
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection	NOT PICS 4/10									
criteria										
ISDN selection criteria										
Test purpose  SIP parameter values	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message followed by a PROGRESS message with a <b>progress indicator</b> PI_VALUE, on receipt of a RELEASE COMPLETE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN  the SUT immediately requests the disconnection of the internal bearer path the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA  183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with									
Values	Progress indicator SIP Statue-Code: SIP_FAILUR	•		LVOILIII	aloa	101-110g1035,7111 Willi				
ISDN parameter values	REL_COMP: cause value: CV									
Comments	SIP-I		SU	Т		ISDN				
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
	183 Session Progress(ACM)	+			+	CALL PROC				
	183 Session Progress(CPG)	+			<b>←</b>	PROGRESS(PI)				
	SIP_FAILURE_VA(REL)	<b>←</b>			<b>←</b>	RELEASE COMPLETE				
	ACK	<b>→</b>								

TP505014	SIP reference: RFC 3261 ISDN reference:								
	Q.1912.5 clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection	NOT PICS 4/10								
criteria									
ISDN selection									
criteria									
Test purpose	Ensure that the SUT in the Idle message, receives a CALL PRO	OCEE	DING me	essage fo	llow	ed by a PROGRESS message			
	with a <b>progress indicator</b> PI_\			eipt of a F	RELE	EASE message with the <b>Cause</b>			
	value CV_ISDN, location LOC	_							
						internal bearer path. When the			
			e-selectio	n, an ISD	NR	ELEASE COMPLETE message			
	is returned to the ISDN sid	-							
OID 1	the SUT shall send the app								
SIP parameter	183 Session Progress encapsu								
values	183 Session Progress with enc	apsul	ated CPG	Event in	dica	tor= Progress, ATP with			
	Progress indicator	\/A /F	NVIT\						
ICDM noromotor	SIP Statue-Code: SIP_FAILURE								
ISDN parameter values	RELEASE; cause value: CV_IS	אוטכ (	PIXII)						
Comments	SIP-I		SU	T		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			<b>←</b>	CALL PROC			
	183 Session Progress(CPG)	+			<b>←</b>	PROGRESS(PI)			
	SIP_FAILURE_VA(REL)	+			<b>←</b>	RELEASE			
	ACK	<b>→</b>			<b>→</b>	RELEASE COMPLETE			

TP505015	SIP reference: RFC	3261			Q.	ISDN reference: 1912.5 clause 6.11.2				
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection	NOT PICS 4/10									
criteria										
ISDN selection criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message followed by a PROGRESS message with a progress indicator PI_VALUE, on receipt of a DISCONNECT message with the Cause value CV_ISDN, location LOC_ISDN  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA									
SIP parameter	183 Session Progress encapsu									
values	183 Session Progress with end Progress indicator	apsui	ated CPG	e Event in	aica	tor= Progress, ATP with				
	SIP Statue-Code: SIP_FAILURE	: \/\ (	PIXIT)							
ISDN parameter values	DISC; cause value: CV_ISDN									
Comments	SIP-I		SU	IT		ISDN				
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
	183 Session Progress(ACM)	<b>←</b>			<b>←</b>	CALL PROC				
	183 Session Progress(CPG)	+			<b>←</b>	PROGRESS(PI)				
	SIP_FAILURE_VA(REL)	+			<b>←</b>	DISCONNECT				
	ACK	→			<b>→</b>	RELEASE				
					<b>←</b>	RELEASE COMPLETE				

Table 21

	Values for test	purpose TP1080010- TP1080015
←SIP Message SIP_FAILURE_VA		← REL  Cause Indicators parameter  CV_ISDN,
VA_1	486 Busy Here Cause Value No. 17	Cause Value No. 17 ("user busy")
VA_2	480 Temporarily unavailable	Cause Value No. 18 ("No user responding")
VA_3	480 Temporarily unavailable	Cause Value No. 21 ("all rejected")
VA_4	410 Gone	Cause Value No. 22 ("number changed")
VA_5	502 Bad Gateway	Cause Value No. 27 ("destination out of order")
VA_6	484 Address Incomplete	Cause Value No. 28 ("invalid number format (address incomplete")
VA_7	480 Temporarily unavailable	Cause Value No. 31 ("normal unspecified") (Class default)
VA_8	500 Server Internal Error	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38-47) (47 is class default)
VA_9	500 Server Internal Error	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_10	500 Server Internal Error	Cause Value No. 88 ("incompatible destination")
VA_11	500 Server Internal Error	Cause Value No. 111 ("protocol error, unspecified") (Class default)

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

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

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

TP505019	SIP reference: RFC 3	3261	ISDN reference: Q.1912.5 clause 6.11.2						
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	NOT PICS 4/10								
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message followed by a ALERTING message, having sent a 180 Ringing message on receipt of an ISDN RELEASE COMPLETE message  the SUT immediately requests the disconnection of the internal bearer path the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA								
SIP parameter values	183 Session Progress encapsu SIP Statue-Code: SIP_FAILURE			i Called p	arty	status = no indication			
ISDN parameter values	REL_COMP: cause value: CV	_ISDN	I (PIXIT)						
Comments	SIP-I		SU	TI		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	<b>←</b>			<b>↓</b>	CALL PROC			
	180 Ringing(CPG)	+			4	ALERTING			
	SIP_FAILURE_VA(REL)	+			+	RELEASE COMPLETE			
	ACK	<b>→</b>							

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

TP505021	SIP reference: RFC 3261 ISDN reference: Q.1912.5 clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	NOT PICS 4/10								
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message followed by a ALERTING message, having sent a 180 Ringing message on receipt of an ISDN DISCONNECT message  the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE message is returned to the ISDN side  the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA								
SIP parameter	183 Session Progress encapsu	lated	ACM: BC	i Called p	arty	status = no indication			
values	SIP Statue-Code: SIP_FAILURE	_VA (	PIXIT)						
ISDN parameter values	DISC; cause value: CV_ISDN	(PIXI	Τ)						
Comments	SIP-I		SU	T		ISDN			
	INVITE(IAM)	<b>→</b>			1	SETUP			
	183 Session Progress(ACM)	+			<b>↓</b>	CALL PROC			
	180 Ringing(CPG)	+			₩	ALERTING			
	SIP_FAILURE_VA(REL)	+			<b>+</b>	DISCONNECT			
	ACK	<b>→</b>			1	RELEASE			
					+	RELEASE COMPLETE			

TP505022	SIP reference: RFC 3	3261			0	ISDN reference: 1912.5 clause 6.11.2		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_	of D	ISC or R	ELEASE		1312.3 014430 0.11.2		
SIP selection criteria	NOT PICS 4/10							
ISDN selection criteria								
Test purpose  SIP parameter values	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message, receives a PROGRESS message with a <b>progress indicator</b> where the progress description value is set to PI_VALUE, on receipt of a ALERTING Message, having sent a 180 Ringing message on receipt of an RELEASE COMPLETE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN  • the SUT immediately requests the disconnection of the internal bearer path  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA  183 Session Progress encapsulated ACM: BCi Called party status = no indication  183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with							
	Progress indicator SIP Statue-Code: SIP_FAILUR	F VA	(PIXIT)					
ISDN parameter values	REL_COMP: cause value: CV							
Comments	SIP-I		SU	ΙΤ		ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
	183 Session Progress(ACM)	+			<b>←</b>	CALL PROC		
	183 Session Progress(CPG)	+			<b>←</b>	PROGRESS(PI)		
	180 Ringing(CPG)	+			<b>←</b>	ALERTING		
	SIP_FAILURE_VA(REL)	<b>←</b>			<b>←</b>	RELEASE COMPLETE		
	ACK	<b>~</b>						

TP505023	SIP reference: RFC 3	3261			Q.	ISDN reference: 1912.5 clause 6.11.2		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE							
SIP selection criteria	NOT PICS 4/10							
ISDN selection								
criteria								
SIP parameter values	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message, receives a PROGRESS message with a <b>progress indicator</b> where the progress description value is set to PI_VALUE, on receipt of a ALERTING Message, having sent a 180 Ringing message on receipt of an RELEASE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA  183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with							
	Progress indicator	/^	(DI)(IT)					
ICDN norometer	SIP Statue-Code: SIP_FAILUR							
ISDN parameter values	RELEASE; cause value: CV_I	אוטפ (	riall)					
Comments	SIP-I		SU	JT		ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
	183 Session Progress(ACM)	+			+	CALL PROC		
	183 Session Progress(CPG)	+		· ·	+	PROGRESS(PI)		
	180 Ringing(CPG)	+			+	ALERTING		
	SIP_FAILURE_VA(REL)	+			+	RELEASE		
	ACK	<b>→</b>	·	<u> </u>	<b>→</b>	RELEASE COMPLETE		

		3261				ISDN reference:		
						1912.5 clause 6.11.2		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE							
SIP selection	NOT PICS 4/10							
criteria								
ISDN selection								
criteria								
Test purpose						E message sends out a SETUP		
	message, receives a CALL PRi with a progress indicator whe							
	receipt of a ALERTING Messag							
	DISCONNECT message with the							
						internal bearer path. When the		
						ELEASE COMPLETE message		
	is returned to the ISDN sid	е				· ·		
	<ul> <li>the SUT shall send the app</li> </ul>	oropri	ate SIP st	atus defir	ned a	as SIP_FAILURE_VA		
SIP parameter	183 Session Progress encapsu							
values	183 Session Progress with enc	apsul	ated CPG	Event in	dica	tor= Progress, ATP with		
	Progress indicator							
	SIP Statue-Code: SIP_FAILURE							
ISDN parameter	DISC; cause value: CV_ISDN	(PIXI	T)					
values				_		1		
Comments	SIP-I		SU	T		ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
	183 Session Progress(ACM)	+			+	CALL PROC		
	183 Session Progress(CPG)	+			+	PROGRESS(PI)		
	180 Ringing(CPG)	+			<b>←</b>	ALERTING		
	SIP_FAILURE_VA(REL)	+			<b>←</b>	DISCONNECT		
	ACK	<b>→</b>			<b>→</b>	RELEASE		
					<b>←</b>	RELEASE COMPLETE		

TP505025	SIP reference: RFC 3	261			Q.	ISDN reference: 1912.5 clause 6.11.2			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	NOT PICS 4/10								
ISDN selection criteria									
Test purpose  SIP parameter values	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message, on receipt of a ALERTING Message, having sent a 180 Ringing message, receives a PROGRESS message with a <b>progress indicator</b> where the progress description value is set to PI_VALUE, on receipt of an RELEASE COMPLETE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN  • the SUT immediately requests the disconnection of the internal bearer path  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA  183 Session Progress encapsulated ACM: BCi Called party status = no indication  183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with								
	Progress indicator SIP Statue-Code: SIP_FAILURE	_VA (	PIXIT)						
ISDN parameter values	REL_COMP: cause value: CV								
Comments	SIP-I		SU	IT		ISDN			
	INVITE(IAM)	<b>^</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROC			
	180 Ringing(CPG)	+			+	ALERTING			
	183 Session Progress(CPG)	+			+	PROGRESS(PI)			
	SIP_FAILURE_VA(REL)	+		•	+	RELEASE COMPLETE			
	ACK	<b>→</b>							

TP505026	SIP reference: RFC 3	3261			Q.	ISDN reference: 1912.5 clause 6.11.2			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection	NOT PICS 4/10								
criteria									
ISDN selection									
criteria									
Test purpose  SIP parameter	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message, on receipt of a ALERTING message, having sent a 180 Ringing message, receives a PROGRESS message with a progress indicator where the progress description value is set to PI_VALUE, on receipt of an RELEASE message with the Cause value CV_ISDN, location LOC_ISDN  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA  183 Session Progress encapsulated ACM: BCi Called party status = no indication								
values	183 Session Progress with enc								
values	Progress indicator	арзаі	alca or c	LVOIR	uica	tor= 1 rogress, 7(1) with			
	SIP Statue-Code: SIP_FAILURE	VA (	PIXIT)						
ISDN parameter values	RELEASE; cause value: CV_I		(PIXIT)						
Comments	SIP-I		SU	IT		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			<b>←</b>	CALL PROC			
	180 Ringing(CPG)	+			<b>←</b>	ALERTING			
	183 Session Progress(CPG)	+			<b>←</b>	PROGRESS(PI)			
	SIP_FAILURE_VA(REL)	+			<b>←</b>	RELEASE			
	ACK	<b>→</b>			<b>→</b>	RELEASE COMPLETE			

TP505027	SIP reference: RFC	3261			Q.	ISDN reference: 1912.5 clause 6.11.2			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection	NOT PICS 4/10								
criteria									
ISDN selection criteria									
SIP parameter values	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message, on receipt of a ALERTING Message, having sent a 180 Ringing message, receives a PROGRESS message with a <b>progress indicator</b> where the progress description value is set to PI_VALUE, on receipt of an DISCONNECT message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA  183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with								
	Progress indicator SIP Statue-Code: SIP_FAILURE	: \/A /DIYIT	Γ\						
ISDN parameter values	DISC; cause value: CV_ISDN		' /						
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROC			
	180 Ringing(CPG)	+			+	ALERTING			
	183 Session Progress(CPG)	+			<b>←</b>	PROGRESS(PI)			
	SIP_FAILURE_VA(REL)	+			+	DISCONNECT			
	ACK	<b>→</b>			<b>→</b>	RELEASE			
					+	RELEASE COMPLETE			

Table 22

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

TP505028	SIP reference: RFC 3261 ISDN reference: Q.1912.5 clause 6.1								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	NOT PICS 4/10								
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a CALL PROCEEDING message, receives a CONNECT message, a 200 OK message is sent, on receipt of a RELEASE COMPLETE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN  the SUT immediately requests the disconnection of the internal bearer path the SUT shall send a BYE message								
SIP parameter values	183 Session Progress encapsu	ulated	ACM: BC	i Called p	arty	status = no indication			
ISDN parameter values	REL_COMP: cause value: CV	_ISD	N (PIXIT)						
Comments	SIP-I		SU	IT		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROC			
	200 OK INVITE(ANM)	+			<b>←</b>	CONNECT			
			Commun	ication					
	BYE(REL)	+			<b>←</b>	RELEASE COMPLETE			
	200 OK BYE(RLC)	<b>→</b>							

TP505029	SIP reference: RFC	3261			Q.	ISDN reference: 1912.5 clause 6.11.2		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE							
SIP selection	NOT PICS 4/10							
criteria								
ISDN selection								
criteria								
Test purpose  SIP parameter	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a CALL PROCEEDING message, receives a CONNECT message, a 200 OK message is sent, on receipt of a RELEASE message with the Cause value CV_ISDN, location LOC_ISDN  the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side  the SUT shall send a BYE message  183 Session Progress encapsulated ACM: BCi Called party status = no indication							
values	Too Coolein Fogress chapes	aiatoa 7	tom. Bo	, Galloa I	Juity	otatao – no maleation		
ISDN parameter values	RELEASE; cause value: CV_I	ISDN (F	PIXIT)					
Comments	SIP-I		SU	Т		ISDN		
	INVITE(IAM)	→			<b>→</b>	SETUP		
	183 Session Progress(ACM)	+			+	CALL PROC		
	200 OK INVITE(ANM)	+			<b>←</b>	CONNECT		
		(	Commun	ication				
	BYE(REL)	+			+	RELEASE		
	200 OK BYE(RLC)	→			<b>→</b>	RELEASE COMPLETE		

TP505030	SIP reference: RFC		Q.	ISDN reference: 1912.5 clause 6.11.2				
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE							
SIP selection criteria	NOT PICS 4/10							
ISDN selection criteria								
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a CALL PROCEEDING message, receives a CONNECT message, a 200 OK message is sent, on receipt of a DISCONNECT message with the Cause value CV_ISDN, location LOC_ISDN  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE message is returned to the ISDN side  • the SUT shall send a BYE message							
SIP parameter values	183 Session Progress encapsu	ulated	ACM: BC	i Called	oarty	status = no indication		
ISDN parameter values	DISC; cause value: CV_ISDN	(PIXI	T)					
Comments	SIP-I		SU	Т		ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
	183 Session Progress(ACM)	+			<b>←</b>	CALL PROC		
	200 OK INVITE(ANM)	+			<b>←</b>	CONNECT		
			Commun	ication				
	BYE(REL)	+			<b>←</b>	DISCONNECT		
	200 OK BYE(RLC)	<b>→</b>			<b>→</b>	RELEASE		
					+	RELEASE COMPLETE		

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

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

TP505033	SIP reference: RFC 3261 Q.					ISDN reference: 1912.5 clause 6.11.2			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection	NOT PICS 4/10								
criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out an SETUP message, receives a CONNECT message, a 200 OK message is sent, on receipt of a DISCONNECT message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN  the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE message is returned to the ISDN side  the SUT shall send a BYE message								
SIP parameter values									
ISDN parameter values	DISC; cause value: CV_ISD	N (PIXI	T)						
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	200 OK INVITE(ANM) ← CONNECT								
	Communication								
	BYE(REL)	+		_	<b>←</b>	DISCONNECT			
	200 OK BYE(RLC)	<b>→</b>			<b>→</b>	RELEASE			
				·	<b>←</b>	RELEASE COMPLETE			

TP505034	SIP reference: RFC 3261	ISDN reference: Q.1912.5 clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection criteria	NOT PICS 4/10									
ISDN selection criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, via a broadcast data link, after time-out of <b>T303</b> :  • the SUT shall send a 480 Temporarily unavailable final response									
SIP parameter values	480 Temporarily unavailable: Encapsul	ated	REL with cause 1	8						
ISDN parameter values										
Comments	SIP-I		SUT		ISDN					
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP					
	→ SETUP									
		T303 expiry								
	480 Temporarily unavailable(REL)	+								
	ACK	<b>→</b>	•							

### Table 23

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

TP505035	SIP reference: RFC 320	61	Q	ISDN reference: .1912.5 clause 6.11.2			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of	_DISC_or_R	ELEASE				
SIP selection criteria	PICS 4/10						
ISDN selection criteria							
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, on receipt of an ISDN RELEASE COMPLETE, where the cause value defined as CV_ISDN  the SUT immediately requests the disconnection of the internal bearer path.  the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA  The ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field						
SIP parameter values	SIP Statue-Code: SIP_FAILURE_VA (PIXIT), Reason header value: CV_SIP (PIXIT)						
ISDN parameter values	REL_COMP: cause value: CV_ISDN (PIXIT)						
Comments	SIP-I	SU	Т	ISDN			
	INVITE(IAM)	<b>&gt;</b>	→	SETUP			
	SIP_FAILURÉ_VA(REL)	-	+	RELEASE COMPLETE			
	ACK -	<b>→</b>					

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

Table 24

	Values for test	purposes TP108036, TP108037
	←SIP Message SIP_FAILURE_VA CV_SIP	← REL Cause Indicators parameter CV_ISDN,
VA_1	404 Not Found Cause Value No. 1	Cause Value No. 1 ("unallocated (unassigned) number")
VA_2	500 Server Internal Error Cause Value No. 2	Cause Value No. 2 ("no route to network")
VA_3	500 Server Internal Error Cause Value No. 3	Cause Value No. 3 ("no route to destination")
VA_4	500 Server Internal Error Cause Value No. 4	Cause Value No. 4 ("Send special information tone")
VA_5	404 Not Found Cause Value No. 5	Cause Value No. 5 ("Misdialed trunk prefix")
VA_6	500 Server Internal Error Cause Value No. 8	Cause Value No. 8 ("Preemption")
VA_7	500 Server Internal Error Cause Value No. 9	Cause Value No. 9 ("Preemption-circuit reserved for reuse")
VA_8	486 Busy Here Cause Value No. 17	Cause Value No. 17 ("user busy")
VA_9	480 Temporarily unavailable Cause Value No. 18	Cause Value No. 18 ("no user responding")
VA_10	480 Temporarily unavailable Cause Value No. 19	Cause Value No. 19 ("no answer from the user")
VA_11	480 Temporarily unavailable Cause Value No. 20	Cause Value No. 20 ("subscriber absent")
VA_12	480 Temporarily unavailable Cause Value No. 21	Cause Value No. 21 ("all rejected")
VA_13	410 Gone Cause Value No. 22	Cause Value No. 22 ("number changed")
VA_14	480 Temporarily unavailable Cause Value No. 25	Cause Value No. 25 ("Exchange routing error")
/A_15	502 Bad Gateway Cause Value No. 27	Cause Value No. 27 ("destination out of order")
VA_16	484 Address Incomplete Cause Value No. 28	Cause Value No. 28 ("invalid number format (address incomplete")
VA_17	500 Server Internal Error	Cause Value No. 29 ("facility rejected")
VA_18	480 Temporarily unavailable Cause Value No. 31	Cause Value No. 31 ("normal unspecified") (Class default)

Values for test purposes TP108036, TP108037							
	←SIP Message	← REL					
	SIP_FAILURE_VA CV_SIP	Cause Indicators parameter CV_ISDN,					
VA_19	486 Busy here if Diagnostics indicator includes the (CCBS indicator = CCBS possible) else 480 Temporarily unavailable Cause Value No. 34	Cause Value in the Class 010 (resource unavailable, Cause Value No. 34)					
VA_20	500 Server Internal Error Cause Value No. 47	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38-47) (47 is class default)					
VA_21	500 Server Internal Error Cause Value No. 50	Cause Value No. 50 ("requested facility not subscribed")					
VA_22	500 Server Internal Error Cause Value No. 55	Cause Value No. 55 ("incoming calls barred within CUG")					
VA_23	500 Server Internal Error Cause Value No. 57	Cause Value No. 57 ("bearer capability not authorized")					
VA_24	500 Server Internal Error Cause Value No. 58	Cause Value No. 58 ("bearer capability not presently")					
VA_25	500 Server Internal Error Cause Value No. 63	Cause Value No. 63 ("service option not available, unspecified") (Class default)					
VA_26	500 Server Internal Error Cause Value No. 65 - 79	Cause Value in the Class 100 (service or option not implemented Cause Value No. 65 - 79) (79 is class default)					
VA_27	500 Server Internal Error Cause Value No. 87	Cause Value No. 87 ("user not member of CUG")					
VA_28	500 Server Internal Error Cause Value No. 88	Cause Value No. 88 ("incompatible destination")					
VA_29	500 Server Internal Error Cause Value No. 90	Cause Value No. 90 ("Non-existent CUG")					
VA_30	404 Not Found Cause Value No. 91	Cause Value No. 91 ("invalid transit network selection")					
VA_31	500 Server Internal Error Cause Value No. 95	Cause Value No. 95 ("invalid message") (Class default)					
VA_32	500 Server Internal Error Cause Value No. 97	Cause Value No. 97 ("Message type non-existent or not implemented")					
VA_33	500 Server Internal Error Cause Value No. 99	Cause Value No. 99 ("information element/parameter non- existent or not implemented")					
VA_34	480 Temporarily unavailable Cause Value No. 102	Cause Value No. 102 ("recovery on timer expiry")					
VA_35	500 Server Internal Error Cause Value No. 103	Cause Value No. 103 ("Parameter non-existent or not implemented, pass on")					
VA_36	500 Server Internal Error Cause Value No. 110	Cause Value No. 110 ("Message with unrecognized Parameter, discarded")					
VA_37	500 Server Internal Error Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)					
VA_38	480 Temporarily unavailable Cause Value No. 127	Cause Value No. 127 ("interworking unspecified") (Class default)					

TP505037	SIP reference: RFC 3	261			Q.	ISDN reference: 1912.5 clause 6.11.2		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_	of DI	SC or R	ELEASE				
SIP selection criteria	PICS 4/10							
ISDN selection criteria								
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, and receives a CALL PROCEEDING message, on receipt of a RELEASE COMPLETE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN  the SUT immediately requests the disconnection of the internal bearer path.  the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA  The ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field							
SIP parameter values	183 Session Progress encapsu							
ISDN parameter values	SIP Statue-Code: SIP_FAILURE_VA (PIXIT), Reason header value: CV_SIP (PIXIT)  REL_COMP: cause value: CV_ISDN (PIXIT)							
Comments	SIP-I		SU	T		ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
	183 Session Progress(ACM) ← CALL PROC							
	SIP_FAILURE_VA(REL)	+			<del>(</del>	RELEASE COMPLETE		
	ACK	<b>→</b>						

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

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

TP505040	SIP reference: RFC 3	261				ISDN reference:				
	Q.1912.5 clause 6.11.2									
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection	PICS 4/10									
criteria										
ISDN selection										
criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message followed by a PROGRESS message with a progress indicator PI_VALUE, on receipt of a RELEASE COMPLETE message with the Cause value CV_ISDN, location LOC_ISDN  the SUT immediately requests the disconnection of the internal bearer path the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA  The ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field									
SIP parameter	183 Session Progress encapsu									
values	183 Session Progress with enc	apsul	ated CPG	Event in	dica	tor= Progress, ATP with				
	Progress indicator				_					
1001	SIP Statue-Code: SIP_FAILURE			eason he	ade	r value: CV_SIP (PIXIT)				
ISDN parameter values	REL_COMP: cause value: CV	_ISDI	N (PIXII)							
Comments	SIP-I		SU	Т		ISDN				
	INVITE(IAM)	<b>^</b>			<b>→</b>	SETUP				
	183 Session Progress(ACM)	4		·	+	CALL PROC				
	183 Session Progress(CPG)	+			<del>(</del>	PROGRESS(PI)				
	SIP_FAILURE_VA(REL)	4		•	+	RELEASE COMPLETE				
	ACK	<b>→</b>		•						

TP505041	SIP reference: RFC 3	ISDN reference: 1912.5 clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection criteria	PICS 4/10									
ISDN selection criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message followed by a PROGRESS message with a progress indicator PI_VALUE, on receipt of a RELEASE message with the Cause value CV_ISDN, location LOC_ISDN  the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side  the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA  The ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field									
SIP parameter values	183 Session Progress encapsu 183 Session Progress with enca Progress indicator SIP Statue-Code: SIP_FAILURE	apsul	ated CPG	Event in	dica	tor= Progress, ATP with				
ISDN parameter values	RELEASE; cause value: CV_IS	SDN	(PIXIT)			· ,				
Comments	SIP-I		SU	IT		ISDN				
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
	183 Session Progress(ACM)	+			+	CALL PROC				
	183 Session Progress(CPG)	+			<b>←</b>	PROGRESS(PI)				
	SIP_FAILURE_VA(REL)	+			<b>←</b>	RELEASE				
	ACK	<b>→</b>			<b>→</b>	RELEASE COMPLETE				

TP505042	SIP reference: RFC 3	3261			Q.1	ISDN reference: 1912.5 clause 6.11.2				
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection criteria	PICS 4/10									
ISDN selection criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message followed by a PROGRESS message with a progress indicator PI_VALUE, on receipt of a DISCONNECT message with the Cause value CV_ISDN, location LOC_ISDN  the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE message is returned to the ISDN side  the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA  The ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field									
SIP parameter values	183 Session Progress encapsu 183 Session Progress with enc Progress indicator SIP Statue-Code: SIP_FAILURE	apsul	ated CPG	Event in	dicat	tor= Progress, ATP with				
ISDN parameter values	DISC; cause value: CV_ISDN					_ ,				
Comments	SIP-I		SU	JT		ISDN				
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
	183 Session Progress(ACM)	<b>←</b>			<b>←</b>	CALL PROC				
	183 Session Progress(CPG)	+			+	PROGRESS(PI)				
	SIP_FAILURE_VA(REL)	<b>←</b>			<b>←</b>	DISCONNECT				
	ACK	<b>→</b>				RELEASE				
					<b>←</b>	RELEASE COMPLETE				

Table 25

	Values for test purpose TP108038 - TP108043						
	←SIP Message SIP_FAILURE_VA CV_SIP	← REL Cause Indicators parameter CV_ISDN,					
VA_1	486 Busy Here Cause Value No. 17	Cause Value No. 17 ("user busy")					
VA_2	480 Temporarily unavailable Cause Value No. 18	Cause Value No. 18 ("No user responding")					
VA_3	480 Temporarily unavailable Cause Value No. 21	Cause Value No. 21 ("all rejected")					
VA_4	410 Gone Cause Value No. 22	Cause Value No. 22 ("number changed")					
VA_5	502 Bad Gateway Cause Value No. 27	Cause Value No. 27 ("destination out of order")					
VA_6	484 Address Incomplete Cause Value No. 28	Cause Value No. 28 ("invalid number format (address incomplete")					
VA_7	480 Temporarily unavailable Cause Value No. 31	Cause Value No. 31 ("normal unspecified") (Class default)					
VA_8	500 Server Internal Error Cause Value No. 47	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38-47) (47 is class default)					
VA_9	500 Server Internal Error Cause Value No. 63	Cause Value No. 63 ("service option not available, unspecified") (Class default)					
VA_10	500 Server Internal Error Cause Value No. 88	Cause Value No. 88 ("incompatible destination")					
VA_11	500 Server Internal Error Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)					

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

TP505044	SIP reference: RFC 3261 ISDN reference: Q.1912.5 clause 6.11.2									
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection criteria	PICS 4/10									
ISDN selection criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends a SETUP message, receives a SETUP ACKNOWLEDGE message followed by a ALERTING message, having sent a 180 Ringing message on receipt of an ISDN RELEASE message  the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RLC is returned to the ISDN side  the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA  The ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field									
SIP parameter values	SIP Statue-Code: SIP_FAILURE_	VA (F	PIXIT), Re	ason he	ade	r value: CV_SIP (PIXIT)				
ISDN parameter values	RELEASE; cause value: CV_IS	DN (	PIXIT)							
Comments	SIP-I		SU	T		ISDN				
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
				·	+	SETUP ACK				
	10011119(110111)	<b>←</b>			<b>←</b>	ALERTING				
	SIP_FAILURE_VA(REL)	<b>←</b>			+	RELEASE				
	ACK	<b>→</b>			<b>→</b>	RELEASE COMPLETE				

TP505045	SIP reference: RFC 3	3261			Q.	ISDN reference: 1912.5 clause 6.11.2				
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection	PICS 4/10									
criteria										
ISDN selection criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends a SETUP message, receives a SETUP ACKNOWLEDGE message followed by a ALERTING message, having sent a 180 Ringing message on receipt of an ISDN DISCONNECT message  the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE is returned to the ISDN side  the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA  The ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field									
SIP parameter	SIP Statue-Code: SIP_FAILURE	_VA (F	PIXIT), Re	eason he	ade	r value: CV_SIP (PIXIT)				
values										
ISDN parameter values	DISC: cause value: CV_ISDN	(PIXIT	<u> </u>							
Comments	SIP-I		SU	T		ISDN				
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
						SETUP ACK				
	180 Ringing(ACM) ← ALERTING									
	SIP_FAILURE_VA(REL)	+			+	DISCONNECT				
	ACK	→			<b>→</b>	RELEASE				
					+	RELEASE COMPLETE				

Table 26

	Values for test purpose	s TP108044 and TP108046
	←SIP Message SIP_FAILURE_VA CV_SIP	← REL Cause Indicators parameter CV_ISDN,
VA_1	486 Busy Here Cause Value No. 17	Cause Value No. 17 ("user busy")
VA_2	480 Temporarily unavailable Cause Value No. 18	Cause Value No. 18 ("No user responding")
VA_3	480 Temporarily unavailable Cause Value No. 21	Cause Value No. 21 ("all rejected")
VA_4	410 Gone Cause Value No. 22	Cause Value No. 22 ("number changed")
VA_5	502 Bad Gateway Cause Value No. 27	Cause Value No. 27 ("destination out of order")
VA_6	484 Address Incomplete Cause Value No. 28	Cause Value No. 28 ("invalid number format (address incomplete")
VA_7	480 Temporarily unavailable Cause Value No. 31	Cause Value No. 31 ("normal unspecified") (Class default)
VA_8	500 Server Internal Error Cause Value No. 47	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38-47) (47 is class default)
VA_9	500 Server Internal Error Cause Value No. 63	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_10	500 Server Internal Error Cause Value No. 88	Cause Value No. 88 ("incompatible destination")
VA_11	500 Server Internal Error Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)

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

TP505047	SIP reference: RFC	3261			Q.	ISDN reference: 1912.5 clause 6.11.2		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE							
SIP selection	PICS 4/10							
criteria								
ISDN selection criteria								
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message, receives a CONNECT message, a 200 OK message is sent, on receipt of a RELEASE message with the Cause value CV_ISDN, location LOC_ISDN  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RLC is returned to the ISDN side  • the SUT shall send a BYE message  • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field							
SIP parameter values	183 Session Progress encapsu SIP Statue-Code: SIP_FAILURE							
ISDN parameter	RELEASE: cause value: CV_I	SDN	(PIXIT)			, ,		
values	oin i	1				liony		
Comments	SIP-I	L_	SU	1	ļ.,	ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
	183 Session Progress(ACM)	<del>(</del>			<del>(</del>	CALL PROC		
	200 OK INVITE(ANM)	<b>←</b>			<b>←</b>	CONNECT		
			Commun	ication				
	BYE(REL)	+			<b>←</b>	RELEASE		
	200 OK BYE(RLC)	<b>→</b>			<b>→</b>	RELEASE COMPLETE		

TP505048	SIP reference: RFC 3	3261			Q.	ISDN reference: 1912.5 clause 6.11.2			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection	PICS 4/10								
criteria									
ISDN selection									
criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message, receives a CONNECT message, a 200 OK message is sent, on receipt of a DISCONNECT message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN  • the SUT immediately requests the disconnection of the internal bearer path When the bearer channel is available for re-selection, an ISDN RELEASE is returned to the ISDN side  • the SUT shall send a BYE message  • The ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field								
SIP parameter values	183 Session Progress encapsu								
	SIP Statue-Code: SIP_FAILURE DISC: cause value: CV_ISDN			eason ne	aue	r value. Cv_SIP (PIXII)			
ISDN parameter values	DISC. Cause value. CV_ISDIN	(FIXI	1)						
Comments	SIP-I		SU	ΙΤ		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			<b>←</b>	CALL PROC			
	200 OK INVITE(ANM)	+			+	CONNECT			
			Commun	ication					
	BYE(REL)	+			+	DISCONNECT			
	200 OK BYE(RLC)	<b>→</b>			1	RELEASE			
					<b>+</b>	RELEASE COMPLETE			

TP505049	SIP reference: RFC 3261 ISDN reference: Q.1912.5 clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection	PICS 4/10								
criteria									
ISDN selection									
criteria									
Test purpose  SIP parameter	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CONNECT message, a 200 OK message is sent, on receipt of a RELEASE COMPLETE message with the Cause value CV_ISDN, location LOC_ISDN  • the SUT immediately requests the disconnection of the internal bearer path. the SUT shall send a BYE message  • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field								
values	SIP Statue-Code: SIP_FAILURE	_ , , (,	17(11), 1(0	acon noud	or value. ov_o	(1.17(1.1)			
ISDN parameter values	REL_COMP: cause value: CV_	_ISDN	I (PIXIT)						
Comments	SIP-I		SU	Γ	ISDN				
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP				
	200 OK INVITE(ANM) ← CONNECT								
			Communi	cation					
	BYE(REL)	+		+	RELEASE CO	OMPLETE			
	200 OK BYE(RLC)	<b>→</b>				·			

TP505050	SIP reference: RFC 3261 ISDN reference: Q.1912.5 clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection	PICS 4/10								
criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CONNECT message, a 200 OK message is sent, on receipt of a RELEASE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RLC is returned to the ISDN side  • the SUT shall send a BYE message  • The ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field								
SIP parameter values	SIP Statue-Code: SIP_FAILUR	E_VA	(PIXIT), <b>R</b> e	ason head	de	r value: CV_SIP (PIXIT)			
ISDN parameter values	RELEASE: cause value: CV_	ISDN	(PIXIT)						
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	<b>→</b>		-	<del>}</del>	SETUP			
	200 OK INVITE(ANM)	+		•	<del>(</del>	CONNECT			
			Commun	ication					
	BYE(REL)	+		•	<b>(</b>	RELEASE			
	200 OK BYE(RLC)	<b>→</b>		-	<del>}</del>	RELEASE COMPLETE			

TP505051	SIP reference: RF	C 3261			Q.	ISDN reference: 1912.5 clause 6.11.2			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	PICS 4/10								
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CONNECT message, a 200 OK message is sent, on receipt of a DISCONNECT message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE is returned to the ISDN side  • the SUT shall send a BYE message  • The ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field								
SIP parameter values	SIP Statue-Code: SIP_FAILU	JRE_VA (	PIXIT), <b>R</b> e	eason h	eade	r value: CV_SIP (PIXIT)			
ISDN parameter values	DISC: cause value: CV_ISI	ON (PIXI	T)						
Comments	SIP-I		SU	IT		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	200 OK INVITE(ANM)	+			+	CONNECT			
			Commun	ication	,				
	BYE(REL)	+			+	DISCONNECT			
	200 OK BYE(RLC)	<b>→</b>			<b>→</b>	RELEASE			
					+	RELEASE COMPLETE			

Table 27

	Values for tes	st purposes TP108047 and TP108052
←SIP Message SIP_FAILURE_VA CV_SIP		← REL Cause Indicators parameter CV_ISDN,
VA_1	BYE Cause Value No. 16	Cause Value No. 16
VA_2	BYE Cause Value No. 31	Cause Value No. 31 ("normal unspecified") (Class default)
VA_3	BYE Cause Value No. 38	Cause Value No. 38 ("Network out of order")
VA_4	BYE Cause Value No. 41	Cause Value No. 41 ("Temporary failure ")
VA_5	BYE Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)

### A.1.1.1.6 Receipt of BYE / CANCEL messages

TP506001	SIP reference: RFC 3261			ISDN reference: Q.1912.5 clause 6.11.1				
TSS reference	SIP-I-ISDN/Basic_call/Rece	eipt_of_BYE	_or_CAI	NCEL				
SIP selection criteria								
ISDN selection criteria								
Test purpose		an ALERTI n ISDN DIS	NG and CONNEC	CONNECT OF With the	message. On receipt of SIP cause and location mapped			
SIP parameter values								
ISDN parameter values	DISC: Cause value and loc	ation mappe	ed from t	he encapsu	lated REL in the received BYE			
Comments	SIP-I		SUT		ISDN			
	INVITE(IAM)	<b>→</b>		→	SETUP			
	180 Ringing(ACM)	+		<del>(</del>	ALERTING			
	200 OK INVITE(ANM)	<del>(</del>		<del>(</del>	CONNECT			
	ACK	<b>→</b>						
	Conversation							
	BYE(REL)	<b>→</b>		→	DISCONNECT			
	200 OK BYE(RLC)	+		+	RELEASE			
				<b>→</b>	RELEASE COMPLETE			

TP506002	SIP reference: RFC 3261				Q.1	ISDN reference: 912.5 clause 6.11.1		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_	of_B	YE_or_CA	ANCEL				
SIP selection criteria								
ISDN selection criteria								
Test purpose	SETUP message, receives an the I-IWU shall send an ISDN [	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out a SETUP message, receives an ALERTING message, the SUT on receipt of SIP <b>CANCEL</b> , the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received CANCEL to the ISDN side						
SIP parameter values								
ISDN parameter values	DISC: Cause value and locatio CANCEL	n map	ped from	the enca	psul	ated REL in the received		
Comments	SIP-I		SU	T		ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
	180 Ringing(ACM)	<b>←</b>			<b>←</b>	ALERTING		
	CANCEL(REL)							
	200 OK CANCEL	+			+	RELEASE		
	487 Request Terminated	+			<b>→</b>	RELEASE COMPLETE		
	ACK	<b>→</b>						

TP506003	SIP reference: RFC 3261				Q.	ISDN reference: 1912.5 clause 6.11.1			
TSS reference	SIP-I-ISDN/Basic_call/Receip	t_of_B\	/E_or_C/	NCEL					
SIP selection criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idl SETUP message, the SUT on DISCONNECT with the cause received CANCEL to the ISDN	receip and lo	t of SIP <b>C</b>	ANCEL,	, the	I-IWU shall send an ISDN			
SIP parameter values									
ISDN parameter values	DISC: Cause value and location	on map	ped from	the enca	apsu	ated REL in the received			
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	<b>→</b>							
	100 Trying	+			<b>→</b>	SETUP			
	CANCEL(REL)								
	200 OK CANCEL	+			+	RELEASE			
	487 Request Terminated	+			<b>→</b>	RELEASE COMPLETE			
	ACK	<b>→</b>		•					

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

TP506005	SIP reference: RF0	3261		Q.	ISDN reference: 1912.5 clause 6.11.1				
TSS reference	SIP-I-ISDN/Basic_call/Recei	ot_of_BYE	or_CAN	ICEL					
SIP selection									
criteria									
ISDN selection									
criteria									
Test purpose	SETUP message, receives a INFO message on receipt of	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a SETUP ACKNOWLEDGE message, and on receipt of a INFO message on receipt of SIP <b>CANCEL</b> , the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received CANCEL to the ISDN side							
SIP parameter									
values									
ISDN parameter	DISC: Cause value and locat	tion mappe	ed from th	ne encapsu	lated REL in the received				
values	CANCEL								
Comments	SIP-I		SUT		ISDN				
	INVITE(IAM)	→		<b>→</b>	SETUP				
				←	SETUP ACK				
	INVITE(IAM)	<b>→</b>		<b>→</b>	INFO				
	CANCEL(REL)	<b>→</b>		<b>→</b>	DISCONNECT				
	200 OK CANCEL	+	-	+	RELEASE				
	487 Request Terminated	+		<b>→</b>	RELEASE COMPLETE				
	ACK	<b>→</b>							

TP506006	SIP reference: RFC 3261				Q.	ISDN reference: 1912.5 clause 6.11.1
TSS reference	SIP-I-ISDN/Basic_call/Receipt	of_B	YE_or_C/	NCEL		
SIP selection						
criteria						
ISDN selection						
criteria						
Test purpose	Ensure that the SUT in the Idle SETUP message, receives a C the I-IWU shall send an ISDN I encapsulated REL in the received.	CALL F	PROCEE! with the c	DING m ause ai	nessag nd loca	e, on receipt of SIP <b>CANCEL</b> , ation mapped from the
SIP parameter values	183 Session Progress encapsu	ılated	ACM: BC	i Calle	d party	status = no indication
ISDN parameter	DISC: Cause value and locatio	n map	ped from	the en	capsul	ated REL in the received
values	CANCEL					
Comments	SIP-I		SU	Т		ISDN
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP
	183 Session Progress(ACM)	+			+	CALL PROC
	CANCEL(REL)	<b>→</b>			<b>→</b>	DISCONNECT
	200 OK CANCEL	+			+	RELEASE
	487 Request Terminated	+			<b>→</b>	RELEASE COMPLETE
	ACK	<b>→</b>				

TP506007	SIP reference: RFC 3	3261	C	ISDN reference: 0.1912.5 clause 6.11.1					
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL								
SIP selection criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message followed by a PROGRESS message, on receipt of SIP <b>CANCEL</b> , the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received CANCEL to the ISDN side								
SIP parameter values	183 Session Progress encapsu 183 Session Progress with enc Progress indicator								
ISDN parameter values	DISC: Cause value and locatio CANCEL	n map	ped from	the encaps	ulated REL in the received				
Comments	SIP-I		SU	IT	ISDN				
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP				
	183 Session Progress(ACM)	<b>←</b>		+	· CALL PROC				
	183 Session Progress(CPG)	+		+	PROGRESS(PI)				
	CANCEL(REL)	<b>→</b>		<b>→</b>	DISCONNECT				
	200 OK CANCEL	+		+	RELEASE				
	487 Request Terminated	+		→	RELEASE COMPLETE				
	ACK	<b>→</b>							

TP506008	SIP reference: RF0	3261		Q.	ISDN reference: 1912.5 clause 6.11.1					
TSS reference	SIP-I-ISDN/Basic_call/Receip	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL								
SIP selection criteria										
ISDN selection criteria										
Test purpose	SETUP message, receives a ALERTING message, on rec	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a SETUP ACKNOWLEDGE message followed by a ALERTING message, on receipt of SIP <b>CANCEL</b> , the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received CANCEL to the								
SIP parameter values										
ISDN parameter values	DISC: Cause value and locat CANCEL	ion mapp	ed from the	encapsu	lated REL in the received					
Comments	SIP-I		SUT		ISDN					
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP					
				+	SETUP ACK					
	180 Ringing(ACM)	+		+	ALERTING					
	CANCEL(REL) → DISCONNECT									
	200 OK CANCEL	+		+	RELEASE					
	487 Request Terminated	+	•	<b>→</b>	RELEASE COMPLETE					
	ACK	<b>→</b>								

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

TP506010	SIP reference: RFC	3261			0	ISDN reference: 1912.5 clause 6.11.1		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_	of B	/F or C/	ANCEL	Q.	1912.3 Clause 0.11.1		
SIP selection criteria	011 110211/124010_0411/10001p1_	_0						
ISDN selection criteria								
Test purpose	SETUP message, receives a CPROGRESS message, on receives	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message, a ALERTING Message, a PROGRESS message, on receipt of SIP <b>CANCEL</b> , the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received CANCEL of the ISDN side						
SIP parameter values		183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with						
ISDN parameter values	DISC: Cause value and locatio CANCEL	n map	ped from	the encap	osul	ated REL in the received		
Comments	SIP-I		SU	IT		ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
	183 Session Progress(ACM)	<b>←</b>			<del>(</del>	CALL PROC		
	180 Ringing(CPG)	+			<del>(</del>	ALERTING		
	183 Session Progress(CPG)	<b>←</b>			<del>(</del>	PROGRESS(PI)		
	CANCEL(REL)	<b>→</b>			<b>→</b>	DISCONNECT		
	200 OK CANCEL	<b>←</b>			<del>(</del>	RELEASE		
	487 Request Terminated	<b>←</b>			<b>→</b>	RELEASE COMPLETE		
	ACK	<b>→</b>						

TP506011	SIP reference: RFC	3261		Q.	ISDN reference: 1912.5 clause 6.11.1	
TSS reference	SIP-I-ISDN/Basic_call/Receip	t_of_B\	E_or_C	ANCEL		
SIP selection criteria						
ISDN selection criteria						
Test purpose		n ALER <sup>*</sup> C with th	TING me ne cause	ssage, the	: Sl	message, sending out a JT on receipt of SIP <b>BYE</b> , the I- mapped from the encapsulated
SIP parameter values						
ISDN parameter values	DISC: Cause value and locati	on map	ped from	the encap	sul	ated REL in the received BYE
Comments	SIP-I		SU	Т		ISDN
	INVITE(IAM)	<b>→</b>		1	<b>→</b>	SETUP
	180 Ringing(ACM)	+			<del>(</del>	ALERTING
	BYE(REL)	<b>→</b>		1	<b>→</b>	DISCONNECT
	200 OK BYE(RLC)	+			<del>(</del>	RELEASE
	487 Request Terminated	+			<b>→</b>	RELEASE COMPLETE
	ACK	<b>→</b>				

TP506012	SIP reference: RFC	3261		ISDN reference: Q.1912.5 clause 6.11.1			
TSS reference	SIP-I-ISDN/Basic_call/Receipt	_of_B	YE_or_CAN(	CEL			
SIP selection criteria							
ISDN selection criteria							
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message, on receipt of SIP <b>BYE</b> , the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received BYE to the ISDN side						
SIP parameter values	183 Session Progress encapsu	ılated	ACM: BCi C	alled part	y status = no indication		
ISDN parameter values	DISC: Cause value and location	n map	ped from the	encapsu	llated REL in the received BYE		
Comments	SIP-I		SUT		ISDN		
	INVITE(IAM)	<b>→</b>		→	SETUP		
	183 Session Progress(ACM)	+		+	CALL PROC		
	BYE(REL)	<b>→</b>		→	DISCONNECT		
	200 OK BYE(RLC)	+		+	RELEASE		
	487 Request Terminated	+		→	RELEASE COMPLETE		
	ACK	<b>→</b>					

TP506013	SIP reference: RFC	3261	ISDN reference: Q.1912.5 clause 6.11.1						
TSS reference	SIP-I-ISDN/Basic_call/Receipt	of_B\	E_or_C/	ANCEL					
SIP selection									
criteria									
ISDN selection									
criteria									
Test purpose	SETUP message, receives a C message, on receipt of SIP BY	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message followed by a PROGRESS message, on receipt of SIP BYE, the I-IWU shall send an ISDN DISC with the cause and ocation mapped from the encapsulated REL in the received BYE to the ISDN side							
SIP parameter	183 Session Progress encapsu	183 Session Progress encapsulated ACM: BCi Called party status = no indication							
values	183 Session Progress with end Progress indicator	apsula	ated CPG	Event indi	cato	or= Progress, ATP with			
ISDN parameter values	DISC: Cause value and location	n map	ped from	the encaps	sula	ted REL in the received BYE			
Comments	SIP-I		SU	T	I	SDN			
	INVITE(IAM)	<b>→</b>		-	<b>→</b> 5	SETUP			
	183 Session Progress(ACM)	+		•	-	CALL PROC			
	183 Session Progress(CPG)	+		•	<b>-</b> F	PROGRESS(PI)			
	BYE(REL)								
	200 OK BYE(RLC)	+		•	<b>-</b> F	RELEASE			
	487 Request Terminated	+		-	<b>≯</b> F	RELEASE COMPLETE			
	ACK	<b>→</b>	•						

TP506014	SIP reference: RFC	3261			ISDN reference:
<i>(</i>					1912.5 clause 6.11.1
TSS reference	SIP-I-ISDN/Basic_call/Receip	ot_of_BYE_or	_CANCE	_	
SIP selection criteria					
ISDN selection criteria					
Test purpose		SETUP ACK eipt of SIP <b>BY</b>	NOWLED E, the I-I\	GE me NU sha	
SIP parameter values					
ISDN parameter values	DISC: Cause value and locat	ion mapped fr	om the ei	ncapsu	ated REL in the received BYE
Comments	SIP-I		SUT		ISDN
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP
	·			+	SETUP ACK
	180 Ringing(ACM)	<del>(</del>		+	ALERTING
	BYE(REL)	<b>→</b>		<b>→</b>	DISCONNECT
	200 OK BYE(RLC)	+		+	RELEASE
	487 Request Terminated	+		<b>→</b>	RELEASE COMPLETE
	ACK	<b>→</b>			

TP506015	SIP reference: RFC	3261		ISDN reference: Q.1912.5 clause 6.11.1					
TSS reference	SIP-I-ISDN/Basic_call/Receipt_	of_BYE	_or_CA	NCEL					
SIP selection criteria									
ISDN selection criteria									
Test purpose	SETUP message, receives a C Message, on receipt of SIP BY	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message, on receipt of a ALERTING Message, on receipt of SIP BYE, the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received BYE to the ISDN side							
SIP parameter values		183 Session Progress encapsulated ACM: BCi Called party status = no indication							
ISDN parameter values	DISC: Cause value and location	n mapp	ed from	the encap	osul	ated REL in the received BYE			
Comments	SIP-I		SU.	T		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			<del>(</del>	CALL PROC			
	180 Ringing(CPG)	+			<del>(</del>	ALERTING			
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT			
	200 OK BYE(RLC)	200 OK BYE(RLC) ← RELEASE							
	487 Request Terminated	+			<b>→</b>	RELEASE COMPLETE			
	ACK	<b>→</b>							

TP506016	SIP reference: RFC	3261		ISDN reference: Q.1912.5 clause 6.11.1						
TSS reference	SIP-I-ISDN/Basic_call/Receipt	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL								
SIP selection criteria										
ISDN selection criteria										
Test purpose	SETUP message, receives a C PROGRESS message, on rece	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message, a ALERTING Message, a PROGRESS message, on receipt of SIP BYE, the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received BYE to the ISDN side								
SIP parameter values		183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with								
ISDN parameter values	DISC: Cause value and location	n map	ped from	the enca	psul	ated REL in the received BYE				
Comments	SIP-I		SU	T		ISDN				
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
	183 Session Progress(ACM)	+			<b>+</b>	CALL PROC				
	180 Ringing(CPG)	+			<b>+</b>	ALERTING				
	183 Session Progress(CPG)	+			<b>←</b>	PROGRESS(PI)				
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT				
	200 OK BYE(RLC)	+			+	RELEASE				
	487 Request Terminated	+			<b>→</b>	RELEASE COMPLETE				
	ACK	<b>→</b>								

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

### A.1.1.2.1 Sending of the INVITE message

TP601001	SIP reference: R	FC 32	261	ISDN/ISDN reference: Q.1912.5 clause 7.1 1 a)					
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message								
SIP selection criteria									
ISDN selection criteria									
Test purpose		Ensure that the SUT in Idle state, on receipt of a SETUP message containing the complete called party number and the sending complete indication  Sends the INVITE message							
SIP parameter values									
ISDN parameter values	SETUP; Called party nur	nber:	with send con	nplete indi	cation				
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONNECT	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
			Conversation	on					
	DISCONNECT	<b>→</b>		→	BYE(REL)				
	RELEASE	+		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	<b>→</b>							

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

TP601003	SIP reference: R	FC 32	261	ISDN/ISDN reference: Q.1912.5 clause 7.1 1 c)					
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message								
SIP selection criteria									
ISDN selection criteria									
Test purpose	called party number who	ere the dicate I party	e end of addr that <b>a suffici</b>	ess signalli	JP message containing the complete ng is determined by analysis of the er of digits has been received to				
SIP parameter values									
ISDN parameter values	SETUP; Called party nu	mber:	sufficient nui	mber of dig	its to route to the called party				
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONNECT	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
			Conversat	tion					
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)				
	RELEASE	+		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	<b>→</b>							

TP601004	SIP reference: RFC 3261					ISDN/ISDN reference: Q.1912.5 clause 7.1 1 d)		
TSS reference	ISDN-SIP /Basic call/Send	lina o	f the INIVITE	macca		Q.1912.5 Clause 7.1 1 u)		
SIP selection	ISDN-SIF /Basic caii/Seric	all ig o	I the invite	messaç	je			
criteria								
ISDN selection criteria								
Test purpose	Ensure that the SUT in Idle state, on receipt of a SETUP message containing the complete called party number where the end of address signalling is determined by the expiration timer T302 after the receipt of the latest address message  sends the INVITE message							
SIP parameter								
values								
ISDN parameter								
values								
Comments	ISDN		SUT			SIP-I		
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)		
	ALERTING	+			<b>←</b>	180 Ringing(ACM)		
	CONNECT	+			<b>←</b>	200 OK INVITE(ANM)		
					<b>→</b>	ACK		
			Conversa	tion				
	DISCONNECT	<b>→</b>			<b>→</b>	BYE(REL)		
	RELEASE	+			+	200 OK BYE(RLC)		
	RELEASE COMPLETE	<b>→</b>						

TP601005	SIP reference: R	FC 326	51		ISDN/ISDN reference: Q.1912.5 clause 7.1.1					
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message									
	ISDN-SIP /Basic cail/Send	airig oi	the invite	message						
SIP selection criteria										
ISDN selection										
criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of an SETUP message, with the Bearer capability set to BC_VALUE  • sends an INVITE message, with the media description defined with the "a = " "b =" and "m=" lines set to a_b_m_LINE_VALUE  • the IAM is encapsulated unchanged in the INVITE									
SIP parameter values										
ISDN parameter values	INVITE: a_b_m_LINE_VA	LUE, IA	AM encapsu	lated in a	MIME-body					
Comments	ISDN		SUT		SIP-I					
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)					
	ALERTING	+		+	180 Ringing(ACM)					
	CONNECT	+		+	200 OK INVITE(ANM)					
				<b>→</b>	ACK					
			Conversat	ion						
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)					
	RELEASE	+		+	200 OK BYE(RLC)					
	RELEASE COMPLETE	<b>→</b>	•							

Table 28

				Values for t	est purpose	ΓP301005					
VA		IS	DN		SDP - a_b_m_LINE_VALUE						
		BC para	ameter	HLC		m= line		b= line	a= line		
	ВС	Information Transport Capability	User Information Layer 1 Protocol Indicator	High Layer Characteristics Identification	<media></media>	<transport></transport>	<fmt-list></fmt-list>	<modifier>:<bandwi dth-value=""></bandwi></modifier>	rtpmap: <dynamic-pt> <encoding name="">/<clock rate="">[/encoding parameters&gt;</clock></encoding></dynamic-pt>		
VA_01	"speech"	"Speech"	"G.711 μ-law"	Ignore	Audio	RTP/AVP	0 (and possibly 8) Note 1	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)		
VA_02	"speech"	"Speech"	"G.711 μ-law"	Ignore	Audio	RTP/AVP	Dynamic PT (and possibly a second Dynamic PT)	AS:64	rtpmap: <dynamic-pt> PCMU/8000 (and possibly rtpmap:<dynamic-pt> PCMA/8000)</dynamic-pt></dynamic-pt>		
VA_03	"speech"	"Speech"	"G.711 A-law"	Ignore	Audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000		
VA_04	"speech"	"Speech"	"G.711 A-law"	Ignore	Audio	RTP/AVP	Dynamic PT	AS:64	rtpmap: <dynamic-pt> PCMA/8000</dynamic-pt>		
VA_05	"3.1 KHz audio"	"3.1 KHz audio"	"G.711 μ-law"		Audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)		
VA_06	"3.1 KHz audio"	"3.1 KHz audio"	"G.711 A-law"		Audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000		
VA_07	"3.1 KHz audio"	"3.1 KHz audio"	"G.711 µ-law"	"Facsimile Group 2/3"	Image	udptl	t38	AS:64	Based on T.38.		
VA_08	"3.1 KHz audio"	"3.1 KHz audio"	"G.711 A-law"	"Facsimile Group 2/3"	Image	udptl	t38	AS:64	Based on T.38.		
VA_09	"3.1 KHz audio"	"3.1 KHz audio"	"G.711 µ-law"	"Facsimile Group 2/3"	Image	tcptl	t38	AS:64	Based on T.38.		
VA_10	"3.1 KHz audio"	"3.1 KHz audio"	"G.711 A-law"	"Facsimile Group 2/3"	Image	tcptl	t38	AS:64	Based on T.38.		
VA_11	"64 kbit/s unrestricted"	"Unrestricted digital inf. W/tone/ann."	N/A	Ignore	Audio	RTP/AVP	9	AS:64	RTPmap:9 G722/8000		
VA_12	"64 kbit/s unrestricted"	"Unrestricted digital information"	N/A	Ignore	Audio	RTP/AVP	Dynamic PT	AS:64	rtpmap: <dynamic-pt> CLEARMODE/8000</dynamic-pt>		

TP601006	SIP reference: RFC 3261					ISDN/ISDN reference: Q.1912.5 clause 7.1.2			
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message								
SIP selection criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter of the SETUP  to the addr-spec component of the To header field which shall include the "user=phone" URI parameter if the To header field contains a sip: URI								
SIP parameter	INVITE: To: sip:; user=				-				
values	•	•							
ISDN parameter values									
Comments	ISDN		SUT			SIP-I			
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)			
	ALERTING	+			<b>←</b>	180 Ringing(ACM)			
	CONNECT	+			<del>-</del>	200 OK INVITE(ANM)			
					<b>→</b>	ACK			
			Conversat	tion					
	DISCONNECT	<b>→</b>			<b>→</b>	BYE(REL)			
	RELEASE	+			<b>←</b>	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>							

TP601007	SIP reference: RFC 3261				ISDN/ISDN reference: Q.1912.5 clause 7.1.2					
TSS reference	ISDN-SIP /Basic call/Send	ISDN-SIP /Basic call/Sending of the INVITE message								
SIP selection										
criteria										
ISDN selection criteria										
Test purpose	Ensure that the SUT is ma Called Party Number para to the addr-spec com	mete	of the SETUP ar	d the a						
SIP parameter	INVITE: To:									
values										
ISDN parameter										
values										
Comments	ISDN		SUT		SIP-I					
	SETUP	<b>→</b>								
	INFO	<b>→</b>								
	INFO	<b>→</b>		<b>→</b>	INVITE(IAM)					
	ALERTING	<b>←</b>		<b>←</b>	180 Ringing(ACM)					
	CONNECT	+		+	200 OK INVITE(ANM)					
				<b>→</b>	ACK					
			Conversation							
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)					
	RELEASE	+		+	200 OK BYE(RLC)					
	RELEASE COMPLETE	<b>→</b>								

TP601008	SIP reference: R	FC 32	261		ISDN/ISDN reference: Q.1912.5 clause 7.1.2						
TSS reference	ISDN-SIP /Basic call/Send	ISDN-SIP /Basic call/Sending of the INVITE message									
SIP selection criteria											
ISDN selection criteria											
Test purpose	Called Party address infor to the addr-spec com	Ensure that the SUT is mapping in the Called Party Number parameter contained in the Called Party address information of the SETUP and followed INFO  to the addr-spec component of the <b>To header field</b> which shall include the "user=phone" URI parameter if the To header field contains a sip: URI									
SIP parameter values	INVITE: To: sip:; user=				·						
ISDN parameter											
values											
Comments	ISDN		SUT		SIP-I						
	SETUP	<b>→</b>									
	INFO	<b>→</b>									
	INFO	<b>→</b>		<b>→</b>	INVITE(IAM)						
	ALERTING	+		+	180 Ringing(ACM)						
	CONNECT	+		+	200 OK INVITE(ANM)						
				<b>→</b>	ACK						
			Conversation	n							
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)						
	RELEASE	+		+	200 OK BYE(RLC)						
	RELEASE COMPLETE	<b>→</b>									

TP601009	SIP reference: R	FC 32	261		ISDN/ISDN reference: Q.1912.5 clause 7.1.2					
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message									
SIP selection criteria										
ISDN selection criteria										
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, TON = "International number" of the SETUP  • to the addr-spec component of the To header field in the INVITE message  • The format of the To header field is "+CC+NDC+SN"  • the forward address information is derived from the user info component of the INVITE Request-URI									
SIP parameter values	INVITE: To:									
ISDN parameter values										
Comments	ISDN		SUT		SIP-I					
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)					
	ALERTING	+		+	180 Ringing(ACM)					
	CONNECT	<b>←</b>		+	200 OK INVITE(ANM)					
				<b>→</b>	ACK					
			Conversation	)						
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)					
	RELEASE	<b>←</b>		+	200 OK BYE(RLC)					
	RELEASE COMPLETE	<b>→</b>								

TP601010	SIP reference: R	261			ISDN/ISDN reference: Q.1912.5 clause 7.1.2				
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message								
SIP selection criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, TON = "National (significant) number" of the SETUP  to the addr-spec component of the To header field in the INVITE message  The format of the To header field is "+CC+NDC+SN"  the forward address information is derived from the user info component of the INVITE Request-URI								
SIP parameter values	INVITE: To:								
ISDN parameter values									
Comments	ISDN		SUT			SIP-I			
	SETUP	<b>→</b>		-	<b>&gt;</b>	INVITE(IAM)			
	ALERTING	+		•	-	180 Ringing(ACM)			
	CONNECT	+		•	-	200 OK INVITE(ANM)			
				-	•	ACK			
			Conversat	ion					
	DISCONNECT	<b>→</b>		-	<b>&gt;</b>	BYE(REL)			
	RELEASE	+		•	-	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>							

TP601011	SIP reference: R	FC 32	261		ISDN/ISDN reference: Q.1912.5 clause 7.1.2					
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message									
SIP selection										
criteria										
ISDN selection criteria										
Test purpose  SIP parameter	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, TON = "unknown" of the SETUP  to the addr-spec component of the To header field in the INVITE message.  The format of the To header field is "+CC+NDC+SN"  the forward address information is derived from the user info component of the INVITE Request-URI  INVITE: To:									
values										
ISDN parameter values										
Comments	ISDN		SUT		SIP-I					
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)					
	ALERTING	+		+	180 Ringing(ACM)					
	CONNECT	+		+	200 OK INVITE(ANM)					
				<b>→</b>	ACK					
			Conversatio	n						
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)					
	RELEASE	+		+	200 OK BYE(RLC)					
	RELEASE COMPLETE	<b>→</b>								

TP601012	SIP reference: R	FC 32	261	ISDN/ISDN reference: Q.1912.5 clause 7.1.2					
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message								
SIP selection									
criteria									
ISDN selection									
criteria									
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, TON = "International number" of the SETUP and the and the followed INFO  to the addr-spec component of the To header field.  The format of the To header field is "+CC+NDC+SN"  the forward address information is derived from the user info component of the INVITE Request-URI								
SIP parameter	INVITE: To:								
values									
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>							
	INFO	<b>→</b>							
	INFO	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONNECT	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
			Conversat	ion					
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)				
	RELEASE	+		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	<b>→</b>							

TP601013	SIP reference: R	FC 32	261		ISDN/ISDN reference: Q.1912.5 clause 7.1.2				
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message								
SIP selection									
criteria									
ISDN selection									
criteria									
Test purpose  SIP parameter	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, TON = "National (significant) number" of the SETUP and the followed INFO  to the addr-spec component of the To header field.  The format of the To header field is "+CC+NDC+SN"  the forward address information is derived from the user info component of the INVITE Request-URI								
values	INVITE: To:								
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>							
	INFO	<b>→</b>							
	INFO	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONNECT	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
			Conversati	on					
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)				
	RELEASE	+		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	<b>→</b>							

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

### A.1.1.2.2 Overlap sending

TP602001	SIP reference: R	FC 32	261	ISDN/ISDN reference: Q.1912.5 clause 7.2						
TSS reference	ISDN-SIP /Basic call/Ove	ISDN-SIP /Basic call/Overlap sending								
SIP selection	PICS 3/1									
criteria										
ISDN selection										
criteria										
Test purpose	Ensure if the SUT is supp INFOs received after the				ards the SIP network, subsequent gnored					
SIP parameter										
values										
ISDN parameter										
values										
Comments	ISDN		SUT		SIP-I					
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)					
	INFO	<b>→</b>								
	ALERTING	+		+	180 Ringing(ACM)					
	CONNECT	+		+	200 OK INVITE(ANM)					
				<b>→</b>	ACK					
			Conversation	on						
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)					
	RELEASE	+		+	200 OK BYE(RLC)					
	RELEASE COMPLETE	<b>→</b>			·					

TP602002	SIP reference: RI	FC 3261		ISDN/ISDN reference: Q.1912.5 clause 7.2.1						
TSS reference	ISDN-SIP /Basic call/Overlap sending									
SIP selection	PICS 3/2									
criteria										
ISDN selection										
criteria										
Test purpose	<ul> <li>Ensure that the SUT in Idle state, on receipt of an SETUP message sends a INVITE message. On receipt of a INFO from the ISDN access the SUT shall:</li> <li>1) Stop timer TOIW3 (if it is running)</li> <li>2) TOIW2 shall be restarted and the SUT shall invoke the following procedures: <ul> <li>a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call</li> <li>b) A new INVITE with the INFOe Call-ID and From header (including tag) as the previous INVITE is sent</li> <li>c) The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question</li> <li>d) All other contents of the new INVITE are interworked from the parameters of the original SETUP</li> </ul> </li> </ul>									
SIP parameter values										
ISDN parameter values										
Comments	ISDN	SUT		SIP-I						
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)						
	INFO	<b>→</b>	<b>→</b>	INVITE(IAM)						
	INFO	<b>→</b>	<b>→</b>	INVITE(IAM)						
	ALERTING	+	+	180 Ringing(ACM)						
	CONNECT	+	+	200 OK INVITE(ANM)						
		→ ACK								
		Conversa	ation							
	DISCONNECT	<b>→</b>	<b>→</b>	BYE(REL)						
	RELEASE	+	+	200 OK BYE(RLC)						
	RELEASE COMPLETE	<b>→</b>		, , ,						

TP602003	SIP reference: R	FC 32	261		ISDN/ISDN reference: Q.1912.5 clause 7.2.1		
TSS reference	ISDN-SIP /Basic call/Over	lap se	ending				
SIP selection criteria	PICS 3/2						
ISDN selection criteria							
Test purpose	The SUT in Idle state, on receipt of an SETUP message sends a INVITE message On receipt of a INFO from the ISDN access the SUT shall:  TOIW2 shall be restarted and the SUT shall invoke the following procedures: Ensure that if timer TOIW2 has expired, subsequent INFOs received after the SUT has sent the INVITE are ignored						
SIP parameter values							
ISDN parameter values							
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
	INFO	<b>→</b>		<b>→</b>	INVITE(IAM)		
			T <sub>oiw2</sub> expi	red			
	INFO	<b>→</b>					
	ALERTING	+		+	180 Ringing(ACM)		
	CONNECT	+		+	200 OK INVITE(ANM)		
				→	ACK		
			Conversat				
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)		
	RELEASE	+		+	200 OK BYE(RLC)		
	RELEASE COMPLETE	<b>→</b>					

TP602004	SIP reference: R	FC 32	261		ISDN/ISDN reference: Q.1912.5 clause 7.2.1			
TSS reference	ISDN-SIP /Basic call/Ove	rlap se	ending					
SIP selection criteria	PICS 3/1							
ISDN selection criteria								
Test purpose	BICC/ISDN the SUT shall sends an INVITE me received in the SETU shall invoke the follow Ensure that if timer T	received in the SETUP and the INFO TOIW1 and TIOW2 shall be started and the SUT shall invoke the following procedures:						
SIP parameter values								
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>						
	INFO	<b>→</b>		<b>→</b>	INVITE(IAM)			
			T <sub>oiw2</sub> expi	ired				
	INFO	<b>→</b>						
	ALERTING	+		<b>+</b>	180 Ringing(ACM)			
	CONNECT	+		+	200 OK INVITE(ANM)			
				→	ACK			
			Conversa	tion				
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)			
	RELEASE	+		+	200 OK BYE(RLC)			
	RELEASE COMPLETE	→						

	SDN-SIP /Basic call/Overl PICS 1/9 AND PICS 3/2				Q.1912.5 clause 7.2.1		
SIP selection P criteria		ap se	ending				
	105 1/9 AND PICS 3/2		<u> </u>				
ICDM coloction							
ISDIN SCIECTION							
criteria							
d S C S	Ensure that if the O-MGCF sends an INVITE request before the end of address signalling is determined, the O-MGCF shall an INVITE with incomplete address information reject with a SIP 404 or 484 error response On receipt of a INFO from the ISDN access, the O-MGCF shall: stop timer Ti/w3 (if it is running) send an INVITE request complying to the following:  The INVITE request shall use the SIP preconditions extension The INVITE request shall include all digits received so far for this call in the Request-URI						
-	restart Ti/w2						
SIP parameter							
values							
ISDN parameter values							
	SDN		SUT		SIP-I		
	SETUP	<b>→</b>	301	<b>→</b>	INVITE(IAM)		
	SE TOT			<del>/</del>	404/484		
<u> </u>				<b>→</b>	ACK		
					ACK		
 	NFO	<b>→</b>		<b>→</b>	INVITE(IAM)		
<u>                                   </u>	141 0			<del>/</del>	404/484		
				<b>→</b>	ACK		
<u> </u>					ACK		
1	NFO	<b>→</b>		<b>→</b>	INVITE(IAM)		
<del>"</del>	141 0			<del>′</del>	404/484		
				<b>→</b>	ACK		
					AOIC		
11	NFO	<b>→</b>		<b>→</b>	INVITE(IAM)		
<del>"</del>	141 0				INVITE(IAW)		
Δ	LERTING	+		+	180 Ringing(ACM)		
l	CONNECT	<del>`</del>		+	200 OK INVITE(ANM)		
	, O. 111EO 1	_		<b>→</b>	ACK		
			Conversat	_			
	DISCONNECT	<b>→</b>	20	<del>_</del>	BYE(REL)		
	RELEASE	<del>′</del>		<del>,</del>	200 OK BYE(RLC)		
	RELEASE COMPLETE	<u>`</u>					

TP602006	SIP reference: RFC	3261			DN/ISDN reference: .1912.5 clause 7.7.6						
TSS reference	ISDN-SIP /Basic call/Overlap sending										
SIP selection criteria	NOT PICS 3/2	NOT PICS 3/2									
ISDN selection criteria											
Test purpose	that the SUT before having message (4xx, 5xx, 6xx) def	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT before having received an backward message, on receipt of a Failure message (4xx, 5xx, 6xx) defined as SIP_Failure_VA  • sends a DISCONNECT or RELEASE message cause value 28									
SIP parameter values	SIP_Failure_VA: ISUP REL	encapsı	ılated in the M	IME bo	ody						
ISDN parameter values	DISCONNECT/RELEASE: 0	Cause va	alue constructe	ed from	the encapsulated REL						
Comments	ISDN		SUT		SIP-I						
	SETUP	→		<b>→</b>	INVITE(IAM)						
				+	484 Address Incomplete						
	CASE A			<b>→</b>	ACK						
	RELEASE	+									
	RELEASE COMPLETE	<b>→</b>									
	CASE B										
	DISCONNECT	+									
	RELEASE	<b>→</b>									
	RELEASE COMPLETE	+									

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

TP603001	SIP reference: R	FC 32	261	Q.1	ISDN/ISDN reference: 912.5 clause 7.1 1) a), 7.3.1					
TSS reference	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING									
SIP selection criteria	PICS 3/1									
ISDN selection criteria										
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the complete called party number and the sending complete indication  Sends the INVITE message to called user  Sends the CALL PROCEDING message  Sends the PROGRESS message, with the with progress description set to PI_VAL									
SIP parameter values	183 Session Progress with encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with Progress indicator									
ISDN parameter values	CALL PROCEEDING PROGRESS(PI value=PI	VAL)								
Comments	ISDN		SUT		SIP-I					
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)					
	CALL PROCEEDING	+		+	183 Session Progress(ACM)					
	PROGRESS(PI)	+		+	183 Session Progress(CPG)					
	ALERTING	+		<b>←</b>	180 Ringing(ACM					
	CONNECT	+		<b>←</b>	200 OK INVITE(ANM)					
	→ ACK									
			Conversa							
	DISCONNECT	<b>→</b>		→	BYE(REL)					
	RELEASE	+		+	200 OK BYE(RLC)					
	RELEASE COMPLETE	<b>→</b>								

TP603002	SIP reference: R	FC 32	261	0.1	ISDN/ISDN reference: 912.5 clause 7.1 1) b), 7.3.1			
TSS reference	ISDN-SIP /Basic call/Sen	dina d	of CALL PRO					
SIP selection	PICS 3/1	<u> </u>	_	_				
criteria								
ISDN selection criteria								
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the maximum number of digits used in the national numbering plan  Sends the INVITE message to called user  Sends the CALL PROCEDING message, with the with progress description value set to PI VAL							
SIP parameter values	183 Session Progress wit with PI	h enca	apsulated AC	M: BCi Call	led party status = no indication, ATP			
ISDN parameter values	CALL PROCEEDING(PI	value=	:PI_VAL)					
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		→	INVITE(IAM)			
	CALL PROCEEDING	<b>←</b>		+	183 Session Progress(ACM)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONNECT	+		+	200 OK INVITE(ANM)			
				→	ACK			
			Conversat					
	DISCONNECT	<b>→</b>		→	BYE(REL)			
	RELEASE	+		+	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>						

TP603003	SIP reference: R	FC 32	261		ISDN/ISDN reference:			
		Q.1912.5 clause 7.1, 7.3.1						
TSS reference	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING							
SIP selection criteria	PICS 3/2							
ISDN selection criteria								
Test purpose	Ensure that the SUT if overlap addressing is to be used toward the SIP network, on receipt of an SETUP message containing the minimum number of digits required for routing the call has been received (start timer TOIW2 and invoke the appropriate outgoing SIP signalling procedure)  Sends an INVITE message to the called user  after the expiration of T <sub>OIW2</sub> sends the CALL PROCEEDING message							
SIP parameter								
values								
ISDN parameter	CALL PROCEEDING							
values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>						
	INFO	<b>→</b>		<b>→</b>	INVITE(IAM)			
			T <sub>oiw2</sub> expired	t				
	CALL PROCEEDING	+						
	ALERTING	+		+	180 Ringing(ACM)			
	CONNECT	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
			Conversat	ion				
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)			
	RELEASE	+		+	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>						

TP603004	SIP reference: R	FC 32	261	Q.1	ISDN/ISDN reference: 912.5 clause 7.1 1) a), 7.3.1			
TSS reference	ISDN-SIP /Basic call/Send	ding_c	of_CALL PRO	DCEDING_/	ALERTING			
SIP selection criteria	PICS 3/1							
ISDN selection criteria								
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the complete called party number and the sending complete indication, on receipt of a 180 Ringing message  Sends the ALERTING message with the with the with progress description value PI VAL							
SIP parameter values	180 Ringing encapsulated Progress indicator PI_VAL		l: BCi Called	party status	s = subscriber free, ATP with			
ISDN parameter values	ALERTING: Progress indi	cator	value PI_VAI	L included				
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		→	INVITE(IAM)			
	CALL PROCEEDING	+						
	ALERTING	+		+	180 Ringing(ACM(PI))			
	CONNECT	+		+	200 OK INVITE(ANM)			
	→ ACK							
	Conversation							
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)			
	RELEASE	+		+	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>						

TP603005	SIP reference: F	• • •		ISDN/ISDN reference: Q.1912.5 clause 7.1 1 a)				
TSS reference	ISDN-SIP /Basic call/Sen	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING						
SIP selection	PICS 3/2							
criteria								
ISDN selection								
criteria								
Test purpose					IP message containing the complete			
	called party number who	ere the	e end of addres	ss signallir	ng is determined by the expiration			
	timer T <sub>OIW2</sub> after the rec	eipt o	f the latest add	ress mess	age on receipt of a 183 Session			
	Progress with encapsulat							
	<ul> <li>a PROGRESS is ser</li> </ul>							
SIP parameter	183 Session Progress en	183 Session Progress encapsulated ACM: BCi Called party status = no indication, ATP with						
values	Progress indicator							
ISDN parameter	PROGRESS							
values			1		1			
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>						
	INFO	→		→	INVITE(IAM)			
			T <sub>oiw2</sub> expire	ed				
	CALL PROCEEDING	+						
	PROGRESS	+		+	183 Session Progress(ACM(PI))			
	ALERTING	+		+	180 Ringing(CPG)			
	CONNECT	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
			Conversation	on				
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)			
	RELEASE	+		+	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>						

TP603006	SIP reference: R	FC 32	61	ISDN/ISDN reference: Q.1912.5 clause 7.1 1 a)				
TSS reference	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING							
SIP selection criteria	PICS 3/2							
ISDN selection criteria								
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the complete called party number where the end of address signalling is determined by the expiration timer T OIW2 after the receipt of the latest address message on receipt of a 183 Session Progress  • no information is sent backward							
SIP parameter values	183 Session Progress end ATP	capsul	ated ACM: B	Ci Called p	arty status = no indication, without			
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>						
	INFO	<b>→</b>		→	INVITE(IAM)			
			T <sub>oiw2</sub> expi	red				
				+	183 Session Progress(ACM)			
	ALERTING	+		+	180 Ringing(ACM			
	CONNECT	<b>←</b>		<b>←</b>	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
			Conversat	ion				
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)			
	RELEASE	+		+	200 OK BYE(RLC)			
	RELEASE COMPLETE	→						

TP603007	SIP reference: R		-0.		ISDN/ISDN reference: Q.1912.5 clause 7.1 1 a)				
TSS reference	ISDN-SIP /Basic call/Send	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING							
SIP selection criteria	PICS 3/2								
ISDN selection criteria									
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the complete called party number where the end of address signalling is determined by the expiration timer T <sub>OIW2</sub> after the receipt of the latest address message on receipt of a 183 Session Progress  • no information is sent backward								
SIP parameter values	183 Session Progress wit	hout e	encapsulated I	SUP mes	sage				
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>							
	INFO	<b>→</b>		<b>→</b>	INVITE(IAM)				
			T <sub>oiw2</sub> expir	ed					
				<b>←</b>	183 Session Progress				
	ALERTING	<b>←</b>		<b>←</b>	180 Ringing(ACM				
	CONNECT	+		+	200 OK INVITE(ANM)				
	→ ACK								
			Conversati	on					
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)				
	RELEASE	+		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	<b>→</b>			, ,				

# A.1.1.2.4 Sending of the CONNECT message

TP604001	SIP reference: RFC 3261					ISDN/ISDN reference: Q.1912.5 clause 7.5			
TSS reference	ISDN-SIP /Basic call/Sending_of_CONNECT								
SIP selection criteria		_							
ISDN selection criteria									
Test purpose	Ensure that the SUT having sent the ALERTING message, on receipt of a 200 OK INVITE for this call, it shall stop timer TOIW2 (if running)  Send CONNECT as determined by ISDN procedures  Stop any existing awaiting answer indication (e.g. ringing tone)								
SIP parameter values	200 OK INVITE: encapsulated ANM in the MIME body								
ISDN parameter values	CONNECT								
Comments	ISDN		SUT			SIP-I			
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)			
	ALERTING	+			+	180 Ringing(ACM			
	CONNECT	+			+	200 OK INVITE(ANM)			
					<b>→</b>	ACK			
			Conversa	tion					
	DISCONNECT	<b>→</b>			<b>→</b>	BYE(REL)			
	RELEASE	+			+	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>							

TP604002	SIP reference: RFC 3261				ISDN/ISDN reference: Q.1912.5 clause 7.5				
TSS reference	ISDN-SIP /Basic call/Sending_of_CONNECT								
SIP selection criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT does not having sent the ALERTING message, on receipt of a 200 OK INVITE for this call, it shall stop timer TOIW2 (if running)  Send CON as determined by ISDN procedures  Stop any existing awaiting answer indication (e.g. ringing tone)								
SIP parameter values	200 OK INVITE: encapsulated CON in the MIME body								
ISDN parameter values	CONNECT								
Comments	ISDN		SUT			SIP-I			
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)			
	CONNECT	+			+	200 OK INVITE(CON)			
					<b>→</b>	ACK			
			Conversat	tion					
	DISCONNECT	<b>→</b>			<b>→</b>	BYE(REL)			
	RELEASE	+			+	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>							

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

TP605001	SIP reference: R	FC 32	261		ISDN/ISDN reference: Q.1912.5 clause 7.7.1, 1)				
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT								
SIP selection criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT after receiving the SETUP and before an INVITE has been sent. On receipt of a RELEASE COMPLETE message  no action is required on the SIP side other than to terminate local procedures if any are in progress								
SIP parameter values									
ISDN parameter values	RELEASE COMPLETE; cause value: (PIXIT), location (PIXIT)								
Comments	ISDN	ISDN SUT SIP-I							
	SETUP	<b>→</b>							
	RELEASE COMPLETE	<b>→</b>							

TP605002	SIP reference: R	FC 32	261	(	ISDN/ISDN reference: Q.1912.5 clause 7.7.1, 1)				
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT								
SIP selection criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT after receiving the SETUP but before an INVITE has been sent. On receipt of a RELEASE message  no action is required on the SIP side other than to terminate local procedures if any are in progress								
SIP parameter values									
ISDN parameter values	RELEASE; cause value: (PIXIT), location (PIXIT)								
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>							
	RELEASE	<b>→</b>							
	RELEASE COMPLETE (+								

TP605003	SIP reference: R	FC 32	261			ISDN/ISDN reference: 0.1912.5 clause 7.7.1, 1)		
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT							
SIP selection criteria								
ISDN selection criteria								
Test purpose	Ensure that the SUT after receiving the SETUP but before an INVITE has been sent. On receipt of a DISCONNECT message  no action is required on the SIP side other than to terminate local procedures if any are in progress							
SIP parameter values								
ISDN parameter values	DISCONNECT; cause value: (PIXIT), location (PIXIT)							
Comments	ISDN		SUT			SIP-I		
	SETUP	<b>→</b>						
	DISCONNECT	<b>→</b>						
	RELEASE	+						
	RELEASE COMPLETE →							

TP605004	SIP reference: R	FC 326	51		ISDN/ISDN reference: Q.1912.5 clause 7.7.1 2)			
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT							
SIP selection criteria	NOT PICS 4/10							
ISDN selection criteria								
Test purpose  SIP parameter values	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE ISDN message before a 200 OK (any) response message has been received which establishes a confirmed dialogue  The SUT shall hold the release procedure until a SIP 200 OK INVITE response has been received  The SUT shall send a BYE request							
ISDN parameter values	RELEASE COMPLETE; cause value: (PIXIT), location (PIXIT)							
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		→	INVITE(IAM)			
	RELEASE COMPLETE	<b>→</b>						
				<b>←</b>	200 OK INVITE(ANM)			
				→	ACK			
				→	BYE(REL)			
				+	200 OK BYE(RLC)			

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

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

TP605007	SIP reference: R	FC 3261		Q	ISDN/ISDN reference: 0.1912.5 clause 7.7.1 2) 3)			
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT							
SIP selection	NOT PICS 4/10							
criteria								
ISDN selection								
criteria								
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE ISDN message before a 200 OK SIP response message has been received  The SUT shall hold the release procedure. A CANCEL is sent when any SIP response was been received  On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent							
SIP parameter values								
ISDN parameter values	RELEASE COMPLETE; c	ause valu	e: (PIXI	Γ), location	(PIXIT)			
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		→	INVITE(IAM)			
				<b>←</b>	100 Trying			
	RELEASE COMPLETE	<b>→</b>						
				→	CANCEL(REL)			
				←	200 OK INVITE(ANM)			
				→	ACK			
				+	200 OK CANCEL			
				→	BYE(REL)			
				<b>←</b>	200 OK BYE(RLC)			

TP605008	SIP reference: R	FC 320	61	C	ISDN/ISDN reference: 0.1912.5 clause 7.7.1 2) 3)	
TSS reference	ISDN-SIP /Basic call/Rece	eipt_of_	_RELEASE_	_or_DISCO	NNECT	
SIP selection	NOT PICS 4/10					
criteria						
ISDN selection						
criteria						
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE message ISDN <b>before</b> a 200 OK SIP response message has been received  The SUT shall hold the release procedure. A CANCEL is sent when any SIP response					
	was been received	10 10100	acc procedu		22 is cont union any on response	
					les, the SUT shall send an ACK for request after the ACK has been sent	
SIP parameter						
values						
ISDN parameter values	RELEASE; cause value:	(PIXIT)	), location	(PIXIT)		
Comments	ISDN		SUT		SIP-I	
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)	
				+	100 Trying	
	RELEASE	<b>→</b>				
	RELEASE COMPLETE	<b>←</b>		<b>→</b>	CANCEL(REL)	
				+	200 OK INVITE(ANM)	
				<b>→</b>	ACK	
				+	200 OK CANCEL	
				<b>→</b>	BYE(REL)	
		<b>→</b>		+	200 OK BYE(RLC)	

TP605009	SIP reference: R	FC 3261		ISDN/ISDN reference:				
			C	2.1912.5 clause 7.7.1 2) 3)				
TSS reference	ISDN-SIP /Basic call/Rece	eipt_of_RELEAS	_or_DISCO	NNECT				
SIP selection	NOT PICS 4/10							
criteria								
ISDN selection criteria								
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a DISCONNECT message ISDN before a 200 OK SIP response message has been received  The SUT shall hold the release procedure. A CANCEL is sent when any SIP response was been received  On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent							
SIP parameter values								
ISDN parameter values	DISCONNECT; cause va	lue: (PIXIT), loc	ntion (PIXIT)					
Comments	ISDN	SU <sup>-</sup>	•	SIP-I				
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)				
			<b>←</b>	100 Trying				
	DISCONNECT	<b>→</b>						
	RELEASE	+	<b>→</b>	CANCEL(REL)				
	RELEASE COMPLETE							
			<b>→</b>	ACK				
			+	200 OK CANCEL				
			<b>→</b>	BYE(REL)				
			+	200 OK BYE(RLC)				

TP605010	SIP reference: R	FC 32	261	Q	ISDN/ISDN reference: .1912.5 clause 7.7.1 2) 3)	
TSS reference	ISDN-SIP /Basic call/Rec	eipt_o	f_RELEASE_	_or_DISCO	NNECT	
SIP selection criteria	NOT PICS 4/10					
ISDN selection criteria						
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE message ISDN message before an early dialogue with the message defined as SIP_MESSAGE_VA has been established  The SUT shall hold the release procedure until a SIP_MESSAGE_VA response has been received  The SUT shall send a CANCEL or BYE request.					
SIP parameter values						
ISDN parameter values	RELEASE COMPLETE; c	ause	value: (PIXI	T), location	(PIXIT)	
Comments	ISDN		SUT		SIP-I	
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)	
				+	100 Trying	
	RELEASE COMPLETE	<b>→</b>				
				←	SIP_MESSAGE_VA	
				CAS	SE A	
				→	CANCEL(REL)	
					200 OK CANCEL	
				<b>←</b>	487 Request Terminated	
				<b>→</b>	ACK	
				CAS		
				<b>→</b>	BYE(REL)	
				+	200 OK BYE(RLC)	
				+	487 Request Terminated	
				<b>→</b>	ACK	

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

TP605012	SIP reference: RFC 3261			ISDN/ISDN reference: Q.1912.5 clause 7.7.1 2) 3)			
TSS reference	ISDN-SIP /Basic call/Reco	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT					
SIP selection criteria	NOT PICS 4/10						
ISDN selection criteria							
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a DISCONNECT message ISDN before an early dialogue with the message defined as SIP_MESSAGE_VA has been established  • The SUT shall hold the release procedure until a SIP_MESSAGE_VA response has been received  • The SUT shall send a CANCEL or BYE request						
SIP parameter values							
ISDN parameter values	DISCONNECT; cause va	lue: (l	PIXIT), <b>locat</b> i	ion (PIXIT)			
Comments	ISDN		SUT	1	SIP-I		
Comments	SETUP	<b>→</b>	301	<b>→</b>	INVITE(IAM)		
	SETUP	7		<del> </del>	100 Trying		
	DISCONNECT	<b>→</b>			100 Trying		
	RELEASE	<del>′</del>					
	RELEASE COMPLETE	<b>→</b>					
	RELETION CONTINUES	Ť		+	SIP_MESSAGE_VA		
	CASE A				on _moonto		
				<b>→</b>	CANCEL(REL)		
				+	200 OK CANCEL		
				+	487 Request Terminated		
				<b>→</b>	ACK		
	CASE B						
				→	BYE(REL)		
				+	200 OK BYE(RLC)		
				+	487 Request Terminated		
				<b>→</b>	ACK		

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

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

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

TP605016	SIP reference: R	FC 32	61		ISDN/ISDN reference: Q.1912.5 clause 7.7.1 3)		
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT						
SIP selection	NOT PICS 4/10						
criteria							
ISDN selection							
criteria							
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE message ISDN message after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established  • The SUT shall send a CANCEL or BYE request						
SIP parameter							
values							
ISDN parameter	RELEASE COMPLETE; c	cause v	value: (PIXI <sup>-</sup>	Γ), <b>locatio</b> n	(PIXIT)		
values							
Comments	ISDN		SUT		SIP-I		
	SETUP	→		→	INVITE(IAM)		
				<b>←</b>	SIP_MESSAGE_VA		
	RELEASE COMPLETE	→					
	CASE A						
				→	CANCEL(REL)		
				<del>-</del>	200 OK CANCEL		
				<b>←</b>	487 Request Terminated		
				<b>→</b>	ACK		
	CASE B						
				<b>→</b>	BYE(REL)		
				<b>←</b>	200 OK BYE(RLC)		
			-	+	487 Request Terminated		
				<b>→</b>	ACK		

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

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

TP605019	SIP reference: RFC 3261				Q.	ISDN/ISDN reference: .1912.5 clause 7.7.1 2) 4)	
TSS reference	ISDN-SIP /Basic call/Rece	eipt_of	f_RELEASE_	or_DIS	1003	NNECT	
SIP selection	PICS 4/10						
criteria							
ISDN selection criteria							
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message on receipt RELEASE COMPLETE message with Cause value CV_ISDN, location LOC_ISDN before a 200 OK response (any) message has been received which establishes a confirmed dialogue  The SUT shall hold the release procedure until a SIP 200 OK INVITE response has been received  The SUT shall send a BYE request. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP						
SIP parameter values	BYE:Reason header value RELEASE COMPLETE	e CV_	SIP, encapsı	ulated R	REL	constructed from the ISDN	
ISDN parameter values	RELEASE COMPLETE; c	ause	value: (PIXI	T), loca	tion	(PIXIT)	
Comments	ISDN		SUT			SIP-I	
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)	
	RELEASE COMPLETE	<b>→</b>					
					<b>←</b>	200 OK INVITE(ANM)	
					<b>→</b>	ACK	
			-		<b>→</b>	BYE(REL)	
					<b>←</b>	200 OK BYE(RLC)	

TP605020	SIP reference: R	FC 3261	Q	ISDN/ISDN reference: .1912.5 clause 7.7.1 2) 4)					
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT								
SIP selection	PICS 4/10	, = =							
criteria									
ISDN selection criteria									
Test purpose  SIP parameter	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message on receipt RELEASE message with Cause value CV_ISDN, location LOC_ISDN before a 200 OK response (any) message has been received which establishes a confirmed dialogue  The SUT shall hold the release procedure until a SIP 200 OK INVITE response has been received  The SUT shall send a BYE request. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP								
values	RELEASE	e Cv_SiP, encaps	uiated REL	constructed from the ISDN					
ISDN parameter values	RELEASE; cause value:	(PIXIT), location	(PIXIT)						
Comments	ISDN	SUT		SIP-I					
	SETUP	→	→	INVITE(IAM)					
	RELEASE	→							
	RELEASE COMPLETE ←								
	← 200 OK INVITE(ANM)								
			<b>→</b>	ACK					
			→	BYE(REL)					
			+	200 OK BYE(RLC)					

TP605021	SIP reference: R	FC 32	261		Q	ISDN/ISDN reference: .1912.5 clause 7.7.1 2) 4)		
TSS reference	ISDN-SIP /Basic call/Rece	eipt_o	f_RELEASE_	or_DIS	1003	NNECT		
SIP selection criteria	PICS 4/10							
ISDN selection criteria								
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message on receipt DISCONNECT message with Cause value CV_ISDN, location LOC_ISDN before a 200 OK response (any) message has been received which establishes a confirmed dialogue  The SUT shall hold the release procedure until a SIP 200 OK INVITE response has been received  The SUT shall send a BYE request. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP							
SIP parameter values						constructed from the ISDN		
ISDN parameter values	DISCONNECT; cause val	lue: (	PIXIT), locati	on (Pl	XIT)			
Comments	ISDN		SUT			SIP-I		
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)		
	ALERTING	+			<b>←</b>	180 Ringing(ACM		
	CONNECT ← 200 OK INVITE(ANM)							
	→ ACK							
			Conversat	ion				
	DISCONNECT	<b>→</b>			<b>→</b>	BYE(REL)		
	RELEASE	+			<b>←</b>	200 OK BYE(RLC)		
	RELEASE COMPLETE	<b>→</b>						

TP605022	SIP reference: R	FC 3261		ISDN/ISDN reference: Q.1912.5 clause 7.7.1 2) 3					
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT								
SIP selection	PICS 4/10								
criteria									
ISDN selection									
criteria									
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE message with Cause value CV_ISDN, location LOC_ISDN before a 200 OK response message has been received  The SUT shall hold the REL message a CANCEL is sent when any SIP response was been received  On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the								
CID navameter	Reason header field			acceptance of from the ICDN					
SIP parameter values	RELEASE COMPLETE	e Cv_SIP, enca	osulated REL	constructed from the ISDN					
ISDN parameter values	RELEASE COMPLETE; c	ause value: (Pl	XIT), locatio	n (PIXIT)					
Comments	ISDN	SU	Т	SIP-I					
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)					
			<b>←</b>	100 Trying					
	RELEASE COMPLETE	<b>→</b>							
			→	CANCEL(REL)					
			+	200 OK INVITE(ANM)					
			→	71011					
			<b>←</b>	200 OK CANCEL					
			→	- : = (: ·==)					
			<b>←</b>	200 OK BYE(RLC)					

TP605023	SIP reference: R	FC 3261	C	ISDN/ISDN reference: 0.1912.5 clause 7.7.1 2) 3					
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT								
SIP selection	PICS 4/10		<del></del>						
criteria									
ISDN selection									
criteria									
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE message with Cause value CV_ISDN, location LOC_ISDN message before a 200 OK response message has been received  The SUT shall hold the release procedure. A CANCEL is sent when any SIP response was been received  On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP								
SIP parameter				constructed from the ISDN					
values	RELEASE								
ISDN parameter values	RELEASE; cause value:	(PIXIT), <b>location</b>	(PIXIT)						
Comments	ISDN	SUT		SIP-I					
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)					
			+	100 Trying					
	RELEASE	<b>→</b>							
	RELEASE COMPLETE	<del>(</del>	→	CANCEL(REL)					
			+	200 OK INVITE(ANM)					
			→	ACK					
			<b>←</b>	200 OK CANCEL					
	→ BYE(REL)								
		<b>→</b>	←	200 OK BYE(RLC)					

TP605024	SIP reference: R	FC 326	61		Q	ISDN/ISDN reference: .1912.5 clause 7.7.1 2) 3		
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT							
SIP selection criteria	PICS 4/10							
ISDN selection criteria								
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a DISCONNECT message with Cause value CV_ISDN, location LOC_ISDN before a 200 OK response message has been received  The SUT shall hold the release procedure. A CANCEL is sent when any SIP response was been received  On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP  BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN							
SIP parameter values	DISCONNECT	e Cv_s	or, encapsi	ulated K		constructed from the 15DN		
ISDN parameter values	DISCONNECT; cause va	lue: (P	IXIT), <b>locat</b> i	ion (PI)	(IT)			
Comments	ISDN		SUT			SIP-I		
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)		
					<del>(</del>	100 Trying		
	DISCONNECT	<b>→</b>						
	RELEASE	<b>←</b>			<b>→</b>	CANCEL(REL)		
	RELEASE COMPLETE →							
					<b>→</b>	ACK		
	← 200 OK CANCEL							
	→ BYE(REL)							
					<del>(</del>	200 OK BYE(RLC)		

TP605025	SIP reference: R	FC 32	61		ISDN/ISDN reference: Q.1912.5 clause 7.7.1 3)		
TSS reference	ISDN-SIP /Basic call/Rec	eipt_of	_RELEASE	_or_DISCOI	NNECT		
SIP selection	PICS 4/10						
criteria							
ISDN selection							
criteria							
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE message with Cause value CV_ISDN, location LOC_ISDN before an early dialogue with the message defined as SIP_MESSAGE has been established  The SUT shall hold the release procedure until a SIP_MESSAGE_VA response has been received  The SUT shall send a CANCEL request or a BYE request. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP						
SIP parameter		e CV_S	SIP, encaps	ulated REL	constructed from the ISDN		
values	RELEASE COMPLETE	_					
ISDN parameter values	RELEASE COMPLETE; o	ause	value: (PIXI	T), location	(PIXIT)		
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
				<del>-</del>	100 Trying		
	RELEASE COMPLETE	<b>→</b>					
				+	SIP_MESSAGE_VA		
	CASE A						
				<b>→</b>	CANCEL(REL)		
				+	200 OK CANCEL		
				+	487 Request Terminated		
				→	ACK		
	CASE B						
				<b>→</b>	BYE(REL)		
				+	200 OK BYE(RLC)		
				+	487 Request Terminated		
				<b>→</b>	ACK		

TP605026	SIP reference: F	RFC 3261		ISDN/ISDN reference: Q.1912.5 clause 7.7.1 3)
TSS reference	ISDN-SIP /Basic call/Rec	eipt_of_RELEASE	_or_DISCO	NNECT
SIP selection criteria	PICS 4/10			
ISDN selection criteria				
Test purpose	sending an INVITE mess: CV_ISDN, location LOC SIP_MESSAGE has been The SUT shall hold to been received The SUT shall send to	age. On receipt of _ISDN before an n established he release procec a CANCEL reque	a RELEASE early dialogu lure until a <b>S</b> l st or a BYE r	e complete called party number, E message with Cause value are with the message defined as IP_MESSAGE_VA response has request. The cause Value Indicator to the Reason header field defined as
SIP parameter		e CV SIP, encap	sulated REL	constructed from the ISDN
values	RELEASE			
ISDN parameter values	RELEASE; cause value:	(PIXIT), location	(PIXIT)	
Comments	ISDN	SUT	•	SIP-I
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)
			<del>(</del>	100 Trying
	RELEASE	<b>→</b>		, ,
	RELEASE COMPLETE	+		
			+	SIP_MESSAGE_VA
	CASE A			
			<b>→</b>	CANCEL(REL)
			+	200 OK CANCEL
			+	487 Request Terminated
			<b>→</b>	ACK
	CASE B			
			<b>→</b>	BYE(REL)
			+	200 OK BYE(RLC)
		1 1		
			<b>←</b>	487 Request Terminated

TP605027	SIP reference: R	FC 32	261		ISDN/ISDN reference: Q.1912.5 clause 7.7.1 3)			
TSS reference	ISDN-SIP /Basic call/Rec	eipt_o	f_RELEASE_	_or_DISCO	NNECT			
SIP selection criteria	PICS 4/10							
ISDN selection	_							
criteria								
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a DISCONNECT message with Cause value CV_ISDN, location LOC_ISDN before an early dialogue with the message defined as SIP_MESSAGE has been established  The SUT shall hold the release procedure until a SIP_MESSAGE_VA response has been received  The SUT shall send a CANCEL request or a BYE request. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP							
SIP parameter	BYE:Reason header valu	e CV_	SIP, encapsi	ulated REL	constructed from the ISDN			
values	DISCONNECT							
ISDN parameter	DISCONNECT; cause va	ilue: (l	PIXIT), <b>locat</b> i	ion (PIXIT)				
values Comments	ISDN	1	SUT		SIP-I			
Comments		+	501					
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	DIGGOVINEGE			+	100 Trying			
	DISCONNECT	<b>→</b>						
	RELEASE	+						
	RELEASE COMPLETE	<b>→</b>						
				<b>←</b>	SIP_MESSAGE_VA			
	CASE A							
				<b>→</b>	CANCEL(REL)			
				+	200 OK CANCEL			
				+	487 Request Terminated			
				<b>→</b>	ACK			
	CASE B							
				→	BYE(REL)			
				<del>-</del>	200 OK BYE(RLC)			
				+	487 Request Terminated			
				<b>→</b>	ACK			

TP605028	SIP reference: RFC 3261 ISDN/ISDN reference: Q.1912.5 clause 7.7.1 3)							
TSS reference	ISDN-SIP /Basic call/Rece	eipt_of	RELEASE	or DISCO	,			
SIP selection	PICS 4/10							
criteria								
ISDN selection								
criteria								
Test purpose  SIP parameter	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE message with Cause value CV_ISDN, location LOC_ISDN after a 200 OK response message has been received  The SUT shall send a BYE request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP  BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN							
values	RELEASE COMPLETE		on , onoupo		toonouracted from the 16211			
ISDN parameter values	RELEASE COMPLETE; c	ause	value: (PIXI	T), locatio	n (PIXIT)			
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ALERTING	<b>←</b>		<b>←</b>	180 Ringing(ACM			
	CONNECT ← 200 OK INVITE(ANM)							
	RELEASE COMPLETE →							
	→ ACK							
				→	BYE(REL)			
				<b>←</b>	200 OK BYE(RLC)			

TP605029	SIP reference: R	FC 32	261	ISDN/ISDN reference: Q.1912.5 clause 7.7.1 3)						
TSS reference	ISDN-SIP /Basic call/Rece	eipt_o	f_RELEASE_	or_DIS	CON	NECT				
SIP selection	PICS 4/10									
criteria										
ISDN selection criteria										
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE message with Cause value CV_ISDN, location LOC ISDN after a 200 OK response message has been received  The SUT shall send a BYE request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP									
SIP parameter	BYE:Reason header value	e CV_	SIP, encapsi	ulated R	EL d	constructed from the ISDN				
values	RELEASE									
ISDN parameter	RELEASE; cause value:	(PIXI	Γ), <b>location</b>	(PIXIT)						
values										
Comments	ISDN		SUT			SIP-I				
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)				
	ALERTING	+			<b>←</b>	180 Ringing(ACM				
	CONNECT	+			<b>←</b>	200 OK INVITE(ANM)				
	RELEASE	<b>→</b>								
	RELEASE COMPLETE	RELEASE COMPLETE ← → ACK								
					<b>→</b>	BYE(REL)				
					<b>←</b>	200 OK BYE(RLC)				

TP605030	SIP reference: R	FC 326	i1	ISDN/ISDN reference: Q.1912.5 clause 7.7.1 3)					
TSS reference	ISDN-SIP /Basic call/Rece	eipt_of_	RELEASE_	_or_DISC0	ONNECT				
SIP selection	PICS 4/10								
criteria									
ISDN selection criteria									
SIP parameter values ISDN parameter values	sending an INVITE messa CV_ISDN, location LOC_ • The SUT shall send a Indicator parameter d defined as CV_SIP	ge. On ISDN a BYE re efined a	receipt of a after a 200 equest after as CV_ISD	a DISCON OK respor r the ACK N shall be ulated REI	e complete called party number, NECT message with <b>Cause value</b> use message has been received has been sent. The cause Value mapped to the Reason header field constructed from the ISDN				
Comments	ISDN		SUT		SIP-I				
Commonto	SETUP	<b>→</b>		→					
	ALERTING	<del>-</del>		<del>′</del>					
	CONNECT								
	DISCONNECT								
	RELEASE ← → ACK								
	RELEASE COMPLETE	<b>→</b>		→	BYE(REL)				
				+	200 OK BYE(RLC)				

TP605031	SIP reference: R	FC 32	61	ı	ISDN/ISDN reference: Q.1912.5 clause 7.7.1 3)			
TSS reference	ISDN-SIP /Basic call/Rece	eipt_of	_RELEASE_	_or_DISCOI	NNECT			
SIP selection criteria	PICS 4/10							
ISDN selection criteria								
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE message with Cause value CV_ISDN, location LOC_ISDN after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established  The SUT shall send a CANCEL or BYE request. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP							
SIP parameter values	BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN RELEASE COMPLETE							
ISDN parameter values	RELEASE COMPLETE; c	ause	value: (PIXI	T), location	(PIXIT)			
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
				<del>(</del>	SIP_MESSAGE_VA			
	RELEASE COMPLETE	<b>→</b>						
	CASE A							
				→	CANCEL(REL)			
				<del>(</del>	200 OK CANCEL			
				<del>(</del>	487 Request Terminated			
				<b>→</b>	ACK			
	CASE B							
				<b>→</b>	BYE(REL)			
				+	200 OK BYE(RLC)			
				•				
				<del>`</del>	487 Request Terminated			

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

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

#### Table 29

Values fo	r test purpose TP605010, TP605011, TP605012; TP605016, TP605017, TP605018 TP605025; TP605026; TP605027; TP605031; TP605032 TP605033	
VA	SIP MESSAGE_VA	
VA_1	180 Ringing	
VA_2	181 Call Is Being Forwarded	
VA_3	182 Queued	
VA_4	183 Session Progress	Ī

### Table 30

	Values for test purposes 306021 - 306033				
←SIP Mess	age	← REL			
Reason hea	nder field	Cause Indicators parameter			
CV_SIP		CV_ISDN			
VA_1	Normal call clearing # 16	Normal call clearing # 16			
VA_2	Normal, unspecified # 31	Normal, unspecified # 31			
VA_3	Temporary failure # 41	Temporary failure # 41			
VA_4	Invalid message, unspecified # 95	Invalid message, unspecified # 95			
VA_5	Recovery on timer expiry # 102	Recovery on timer expiry # 102			
VA_6	Protocol error, unspecified # 111	Protocol error, unspecified # 111			

Table 31: Mapping of Cause Indicators parameter into SIP Reason header fields

Cause indications parameter field	Value of parameter field	component of SIP Reason header field	Component value
-	-	Protocol	"Q.850"
Cause Value	"XX" (Note 1)	Protocol-cause	"cause= XX" (Note 1)
-	-	Reason-text	Should be filled with the definition text as stated in Q.850 (Note 2)

NOTE 1: "XX" is the Cause Value as defined in Q.850.

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

#### A.1.1.2.6 Receipt of a backward final response or BYE Message

TP606001	SIP reference: RFC 3261			ISDN/ISDN reference: Q.1912.5 clause 7.7.2				
TSS reference	ISDN-SIP /Basic call/Rec	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE						
SIP selection criteria								
ISDN selection criteria								
Test purpose	receipt of a BYE messag included	sends a DISCONNECT message constructed from the encapsulated REL in the						
SIP parameter values	BYE: ISUP REL encapsu	BYE: ISUP REL encapsulated in the MIME body						
ISDN parameter values	DISCONNECT: Cause va	alue const	ructed from the	encap	sulated REL			
Comments	ISDN		SUT		SIP-I			
	SETUP	→		<b>→</b>	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM			
	CONNECT	+		+	200 OK INVITE(ANM)			
	→ ACK							
			Conversation					
	DISCONNECT	+		+	BYE(REL)			
	RELEASE	<b>→</b>	•	<b>→</b>	200 OK BYE(RLC)			
	RELEASE COMPLETE	+						

TP606002	SIP reference: RI	FC 3261			_	DN/ISDN reference: .1912.5 clause 7.7.2	
TSS reference	ISDN-SIP /Basic call/Rec	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE					
SIP selection criteria							
ISDN selection criteria							
Test purpose	receipt of a BYE message included  sends a DISCONNEC	Ensure that the SUT after receiving the SETUP sends out a INVITE message and on receipt of a BYE message where a Reason header field with Q.850 Cause Value is included  sends a DISCONNECT message constructed from the encapsulated REL in the received BYE MIME body.					
SIP parameter values	BYE: ISUP REL encapsul	lated in t	he MIME	body, R	eason	header value = (PIXIT)	
ISDN parameter values	DISCONNECT: Cause va	alue cons	structed fr	om the	encaps	sulated REL	
Comments	ISDN		SL	JT		SIP-I	
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)	
	ALERTING	+			+	180 Ringing(ACM	
	CONNECT						
		→ ACK					
		Conversation					
	DISCONNECT	+			+	BYE(REL)	
	RELEASE	<b>→</b>			<b>→</b>	200 OK BYE(RLC)	
	RELEASE COMPLETE	+		•			

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

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

TP606003	SIP reference: RFC	3261			DN/ISDN reference: 1.1912.5 clause 7.7.6			
TSS reference	ISDN-SIP /Basic call/Receip	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE						
SIP selection criteria								
ISDN selection criteria								
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT in state N2, before having received an backward message, on receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with Q.850 Cause Value is <b>not</b> included defined as SIP_Failure_VA  • sends a DISCONNECT or RELEASE message with the Cause value set to the value of the encapsulated REL							
SIP parameter values	SIP_Failure_VA:ISUP REL encapsulated in the MIME body							
ISDN parameter values	DISCONNECT/RELEASE:	cause va	lue: mapped	from th	e encapsulated REL			
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		→	INVITE(IAM)			
	SETUP ACK	+			, , ,			
	CASE A			+	SIP_Failure_VA(REL)			
	RELEASE ← → ACK							
	RELEASE COMPLETE →							
	CASE B							
	DISCONNECT	+	•					
	RELEASE	<b>→</b>	<u> </u>					
	RELEASE COMPLETE	+						

TP606004	SIP reference: RFC	3261			SDN/ISDN reference: 1.1912.5 clause 7.7.6		
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE						
SIP selection criteria							
ISDN selection criteria							
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT in state N3, before having received an backward message, on receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with Q.850 Cause Value is <b>not</b> included defined as SIP_Failure_VA  • sends a DISCONNECT or RELEASE message with the Cause value set to the value of the encapsulated REL.						
SIP parameter values	SIP_Failure_VA: ISUP REL	encaps	ulated in the M	IIME bo	ody		
ISDN parameter values	DISCONNECT/RELEASE: 0	cause v	alue: mapped	from th	e encapsulated REL		
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
	CALL PROCEEDING	+					
	CASE A			+	SIP_Failure_VA(REL)		
	RELEASE ← → ACK RELEASE COMPLETE →						
CASE B							
	DISCONNECT	+					
	RELEASE	<b>→</b>					
	RELEASE COMPLETE	<b>←</b>					

TP606005	SIP reference: RF	C 3261			DN/ISDN reference: .1912.5 clause 7.7.6		
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE						
SIP selection criteria							
ISDN selection criteria							
Test purpose  SIP parameter values	Ensure that the SUT after receiving the SETUP message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party sends out an INVITE message. Ensure that the SUT in state N3 on receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with Q.850 Cause Value is not included defined as SIP_Failure_VA  sends a DISCONNECT or RELEASE message with the Cause value set to the value of the encapsulated REL.  SIP_Failure_VA: no ISUP REL encapsulated in the MIME body						
ISDN parameter values	DISCONNECT/RELEASE	: cause	value: mapped	from th	e encapsulated REL		
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
	CALL PROCEEDING	+					
	CASE A			+	SIP_Failure_VA(REL)		
	DISCONNECT	+		<b>→</b>	ACK		
	RELEASE	<b>→</b>					
	RELEASE COMPLETE	+					
	CASE B						
	RELEASE	+					
	RELEASE COMPLETE	<b>→</b>					

Table 33

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

TP606006	SIP reference: RFC 3	261			SDN/ISDN reference: 1.1912.5 clause 7.7.6
TSS reference	ISDN-SIP /Basic call/Receipt_	of_back	ward_final_	respons	se_or_BYE
SIP selection	PICS 4/12				
criteria					
ISDN selection					
criteria —					
Test purpose	SETUP message sends out a	n INVITE	message.	Ensure	orked to ISDN after receiving an that the SUT in state N2, before
	where a Reason header field SIP_Failure_VA	message with Q.8	e, on receip 50 Cause V	alue is	ailure message (4xx, 5xx, 6xx)  not included defined as
	<ul> <li>sends a DISCONNECT of ISDN.</li> </ul>	r RELEA	SE messaç	ge with	the Cause value set to CV_
SIP parameter					
values					
ISDN parameter values	DISCONNECT/RELEASE; ca	use valu	ie: CV_ISD	N (PIXI	T)
Comments	ISDN		SUT		SIP-I
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)
	SETUP ACK	+			
	CASE A			<b>←</b>	SIP_Failure_VA
	RELEASE	<b>←</b>		→	ACK
	RELEASE COMPLETE	<b>→</b>			
	CASE B				
	DISCONNECT	<b>←</b>			
	RELEASE	<b>→</b>			
	RELEASE COMPLETE	<b>←</b>			

TP606007	SIP reference: RFC	3261			SDN/ISDN reference: 1.1912.5 clause 7.7.6				
TSS reference	ISDN-SIP /Basic call/Receip	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE							
SIP selection	PICS 4/12								
criteria									
ISDN selection									
criteria									
Test purpose	SETUP message sends out having received an backwar where a Reason header field SIP_Failure_VA	an INVI d messa d with Q	ΓE message. ge, on receipt 850 Cause V	Ensure t of a Fa alue is	orked to ISDN after receiving an that the SUT in state N3, before ailure message (4xx, 5xx, 6xx)  not included defined as the Cause value set to CV_				
SIP parameter									
values									
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	CALL PROCEEDING	+							
	CASE A			+	SIP_Failure_VA				
	RELEASE	<b>←</b>		→	ACK				
	RELEASE COMPLETE	<b>→</b>							
	CASE B		<u>-</u>						
	DISCONNECT	<del>-</del>			·				
	RELEASE	<b>→</b>	-						
	RELEASE COMPLETE	+							

Table 34

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

TP606008	SIP reference: RFC 3261			ISDN/ISDN reference: Q.1912.5 clause 7.7.6						
TSS reference	ISDN-SIP /Basic call/Red	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE								
SIP selection criteria	NOT PICS 4/12									
ISDN selection criteria										
Test purpose	of a Failure message (4x is <b>not</b> included defined a	Ensure that the SUT after receiving the SETUP sends out an INVITE message. On receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with Q.850 Cause Value is <b>not</b> included defined as SIP_Failure_VA  • no action is taken on the ISDN.								
SIP parameter values										
ISDN parameter values										
Comments	ISDN		SUT			SIP-I				
	SETUP	SETUP → INVITE(IAM)								
		← SIP_Failure_VA								
				-	<b>→</b>	ACK				
		Further SIP procedures apply								

Table 35

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

TP606009	SIP reference: RFC 3261			ISDN/ISDN reference: Q.1912.5 clause 7.7.6				
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE							
SIP selection criteria								
ISDN selection criteria								
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message, on receipt of a Failure message 487 Request terminated where a Reason header field with Q.850 Cause Value is not included  no action is taken on the ISDN if a CANCEL request was previously sent before answer to an INVITE							
SIP parameter values								
ISDN parameter values								
Comments	ISDN		SI	JT		SIP-I		
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)		
					+	100 Trying		
	RELEASE	<b>→</b>			<b>→</b>	CANCEL(REL)		
	RELEASE COMPLETE	+			+	200 OK CANCEL		
					+	487 Request terminated		
					<b>→</b>	ACK		

TP606010	SIP reference: RFC	3261			DN/ISDN reference: 1912.5 clause 7.7.6		
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE						
SIP selection criteria							
ISDN selection criteria							
Test purpose	of a Failure message 491 R	Ensure that the SUT after receiving the SETUP sends out an INVITE message, on receipt of a Failure message <b>491 Request Pending</b> where a Reason header field with Q.850 Cause Value is <b>not</b> included					
SIP parameter values							
ISDN parameter values							
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
				+	100 Trying		
				+	491 Request Pending		
				→	ACK		

TP606011	SIP reference: RFC	erence: RFC 3261			SDN/ISDN reference: .1912.5 clause 7.7.6			
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE							
SIP selection	NOT PICS 4/11							
criteria								
ISDN selection criteria								
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message, a SIP message defined as SIP MESSAGE_VA has been received, on receipt of a Failure message (4xx, 5xx, 6xx) defined as SIP_Failure_VA where a Reason header field with Q.850 Cause Value is <b>not</b> included  • sends a RELEASE or DISCONNECT message with the Cause value set to CV_ISDN.							
SIP parameter values								
ISDN parameter values	RELEASE/DISCONNECT; c	ause val	ue: CV_ISD	N				
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
				+	SIP MESSAGE_VA			
	CASE A			+	SIP_Failure_VA			
	RELEASE	<b>←</b>		<b>→</b>	ACK			
	RELEASE COMPLETE	<b>→</b>						
	CASE B							
	DISCONNECT	+	<u> </u>					
	RELEASE	→						
	RELEASE COMPLETE	+						

Table 36

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

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

Table 37

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

TP606013	SIP reference: RF0	C 3261			SDN/ISDN reference: 0.1912.5 clause 7.7.6
TSS reference	ISDN-SIP /Basic call/Recei	pt_of_bac	kward_final_	respon	se_or_BYE
SIP selection criteria	PICS 4/11				
ISDN selection criteria					
Test purpose	that the SUT in state N2, be defined as SIP_MESSAGE 5xx, 6xx) defined as SIP_F Value is included  • sends a DISC messag mapped to the ISDN C value set to CV_ ISDN	efore havir LVA has tallure_VA e. The Callause Valu	ng received a been received where a Re use Value in e field in the	an back d on re- ason ho the hea	at an INVITE message. Ensure award message, a SIP message ceipt of a Failure message (4xx, eader field with Q.850 Cause ader field set to CV_SIP is REL message with the Cause
SIP parameter values	SIP_Failure_VA Reason he	eader CV_	SIP (PIXIT)		
ISDN parameter values	DISCONNECT/RELEASE (	cause valu	e CV_ ISDN	(PIXIT	7)
Comments	ISDN		SUT		SIP-I
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)
	SETUP ACK	+			, , ,
				<del>-</del>	SIP MESSAGE_VA
	CASE A			+	SIP_Failure_VA
	RELEASE	+		<b>→</b>	ACK
	RELEASE COMPLETE	<b>→</b>			
	CASE B		•		
	DISCONNECT	+			
	RELEASE	→			
	RELEASE COMPLETE	<b>←</b>			

TP606014	SIP reference: RFC	3261			SDN/ISDN reference: 1.1912.5 clause 7.7.6	
TSS reference	ISDN-SIP /Basic call/Receip	t_of_bac	kward_final_	respon	se_or_BYE	
SIP selection criteria	PICS 4/11					
ISDN selection criteria						
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT in state N3, before having received an backward message, a SIP message defined as SIP_MESSAGE_VA has been received on receipt of a Failure message (4xx, 5xx, 6xx) defined as SIP_Failure_VA where a Reason header field with Q.850 Cause Value is included  • sends a REL message. The Cause Value in the header field set to CV_SIP is mapped to the ISDN Cause Value field in the ISDN REL message with the Cause value set to CV_ISDN.					
SIP parameter values	SIP_Failure_VA Reason hea	ader CV_	SIP (PIXIT)			
ISDN parameter values	DISCONNECT/RELEASE c	ause valu	ie CV_ ISDN	(PIXIT		
Comments	ISDN		SUT		SIP-I	
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)	
	CALL PROCEEDING	+				
				+	SIP MESSAGE_VA	
	CASE A			+	SIP_Failure_VA	
	RELEASE	+		<b>→</b>	ACK	
	RELEASE COMPLETE →					
	CASE B					
	DISCONNECT	+				
	RELEASE	<b>→</b>				
	RELEASE COMPLETE	<b>←</b>				

TP606015	SIP reference: RF0	3261			SDN/ISDN reference: 0.1912.5 clause 7.7.6				
TSS reference	ISDN-SIP /Basic call/Recei	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE							
SIP selection	PICS 4/11	PICS 4/11							
criteria									
ISDN selection									
criteria									
Test purpose	Ensure that the SUT after receiving the SETUP message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party sends out an INVITE message. Ensure that the SUT in state N3, having received an backward message indicating that sufficient number of digits has been received to route the call to the called party, on receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with Q.850 Cause Value Cause Value is included  sends a REL message. The Cause Value in the header field set to C V_SIP is mapped to the ISDN Cause Value field in the ISDN REL message with the Cause value set to CV ISDN								
SIP parameter values	SIP_Failure_VA Reason he	eader C\	/_ SIP (PIXIT)						
ISDN parameter values	DISCONNECT/RELEASE	cause va	alue CV_ ISDN	(PIXIT	)				
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	CALL PROCEEDING	+							
				+	SIP MESSAGE_VA				
	CASE A			+	SIP_Failure_VA				
	RELEASE ← → ACK								
	RELEASE COMPLETE	→							
	CASE B								
	DISCONNECT	<b>←</b>							
	RELEASE	→							
	RELEASE COMPLETE	<b>←</b>							

Table 38

	Values for test purpose TP606013, 606014, 606015				
VA	SIP MESSAGE_VA				
VA_1	180 Ringing				
VA_2	181 Call Is Being Forwarded				
VA_3	182 Queued				
VA_4	183 Session Progress				

Table 39

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

TP606016	SIP reference: RFC 3261			DN/ISDN reference: .1912.5 clause 7.7.6				
TSS reference	ISDN-SIP /Basic call/Receipt_	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE						
SIP selection criteria	NOT PICS 4/17							
ISDN selection criteria								
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT in state N2, before having received an backward message, on receipt of a response message (3xx) defined as SIP_Response_VA, the SUT  • sends a DISC message with the Cause value 127 Interworking							
SIP parameter values				-				
ISDN parameter values	REL; cause value: CV_ISDN							
Comments	ISDN	;	SUT	SIP-I				
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)				
	SETUP ACK	<del>-</del>						
	CASE A		+	SIP_Response_VA				
	RELEASE	<del>(</del>	<b>→</b>	ACK				
	RELEASE COMPLETE	<b>→</b>						
	CASE B							
	DISCONNECT	<del>-</del>						
	RELEASE	<b>→</b>						
	RELEASE COMPLETE	<b>←</b>						

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

TP606018	SIP reference: RFC 3	261			DN/ISDN reference: .1912.5 clause 7.7.6		
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE						
SIP selection	NOT PICS 4/17						
criteria							
ISDN selection							
criteria							
Test purpose	Ensure that the SUT after receiving the SETUP message containing the complete <b>called</b> party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party sends out an INVITE message. Ensure that the SUT in state N3, on receipt of a response message (3xx) defined as <b>SIP_Response_VA</b> , the SUT  sends a DISC message with the <b>Cause value</b> 127 Interworking						
SIP parameter values							
ISDN parameter values	DISC; cause value: CV_ISDN	1					
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
	CALL PROCEEDING	+					
	CASE A			<b>←</b>	SIP_Response_VA		
	RELEASE	+		<b>→</b>	ACK		
	RELEASE COMPLETE	<b>→</b>					
	CASE B						
	DISCONNECT	+					
	RELEASE	<b>→</b>					
	RELEASE COMPLETE	+					

TP606019	SIP reference: RFC 3261			ISDN/ISDN reference: Q.1912.5 clause 7.7.6			
TSS reference	ISDN-SIP /Basic call/Red	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE					
SIP selection criteria	PICS 4/17						
ISDN selection criteria							
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT in state N2, before having received an backward message on receipt of a response message (3xx) defined as SIP_Response_VA, the SUT  • sends an INVITE using the value of the Contact header field in the received SIP_Response_VA in the Request URI						
SIP parameter	SIP_Response_VA Conta			tion			
values	INVITE: Request URI of I	new dest	ination				
ISDN parameter							
values							
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		→	INVITE(IAM)		
	SETUP ACK	+					
				+	SIP_Response_VA		
				<b>→</b>	ACK		
				<b>→</b>	INVITE(IAM)		
	ALERTING	+		+	180 Ringing(ACM		
	CONNECT	+		+	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
			Conversation	n			
	DISCONNECT	+		+	BYE(REL)		
	RELEASE	<b>→</b>		<b>→</b>	200 OK BYE(RLC)		
	RELEASE COMPLETE	+					

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

TP606021	SIP reference: RF	reference: RFC 3261 ISDN/ISDN reference: Q.1912.5 clause 7.7.6						
TSS reference	ISDN-SIP /Basic call/Rece	eipt_of_	backward_final_	respons	se_or_BYE			
SIP selection	PICS 4/17	PICS 4/17						
criteria								
ISDN selection criteria								
Test purpose	party number where the party number to indicate to call to the called party sends out an INVITE mes message (3xx) defined as	sends out an INVITE message. Ensure that the SUT in state N3, on receipt of a response message (3xx) defined as SIP_Response_VA, the SUT  • sends an INVITE using the value of the Contact header field in the received						
SIP parameter		SIP_Response_VA Contact: URI of new destination						
values	INVITE: Request URI of n							
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		→	INVITE(IAM)			
	CALL PROCEEDING	+						
				+	SIP_Response_VA			
				→	ACK			
				→	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM			
	CONNECT	+		+	200 OK INVITE(ANM)			
				→	ACK			
			Conversation	1				
	DISCONNECT	+		<b>+</b>	BYE(REL)			
	RELEASE	<b>→</b>		→	200 OK BYE(RLC)			
	RELEASE COMPLETE	+						

Table 40

Values for test purpose TP606016, TP606017, TP606018 TP606019, TP606020, TP606021				
VA	SIP_Response_VA			
VA_1	300 Multiple Choices			
VA_2	301 Moved Permanently			
VA_3	302 Move Temporarily			
VA_4	305 Use Proxy			
VA_5	380 Alternative Service			

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

TP607001	SIP reference: RFC	3261			DN/ISDN reference: 912.5 clause 7.7.6.1
TSS reference	ISDN-SIP /Basic call/Autono	mous_r	elease		
SIP selection criteria	PICS 3/2				
ISDN selection criteria					
Test purpose	Ensure that the SUT a On re INVITE (i.e. there are no oth configured to propagate ove shall not clear immediat RELEASE or DISCONN	er pend rlap sigr ely the l	ing INVITE transac nalling into the SIP pearer channel and	tions netwo	for this call), if the SUT is ork, the SUT instead start timer TOIW3. The
SIP parameter values			<u> </u>		·
ISDN parameter values					
Comments	ISDN		SUT		SIP-I
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)
			Start timer T <sub>OIW3</sub>	+	484 Address incomplete
				<b>→</b>	ACK
			Timeout T <sub>OIW3</sub>	•	
	CASE A				
	RELEASE	+			
	RELEASE COMPLETE	<b>→</b>			
	CASE B				
	DISCONNECT	+			
	RELEASE	<b>→</b>			
	RELEASE COMPLETE	+			

TP607002	SIP reference: RFC	3261			N/ISDN reference: 012.5 clause 7.7.6.1		
TSS reference	ISDN-SIP /Basic call/Autonomous_release						
SIP selection criteria	NOT PICS 3/4						
ISDN selection criteria							
Test purpose	Ensure that the SUT a On receipt of a 484 Address Incomplete response for the current INVITE (i.e. there are no other pending INVITE transactions for this call), if the O-IWU is not configured to propagate overlap signalling into the SIP network then the timer shall not be started and the  • DISCONNECT message shall be sent immediately to the ISDN network						
SIP parameter values		,					
ISDN parameter values							
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
				+	484 Address incomplete		
	CASE A			<b>→</b>	ACK		
	RELEASE						
	RELEASE COMPLETE	<b>→</b>					
	CASE B						
	DISCONNECT	+					
	RELEASE	<b>→</b>					
	RELEASE COMPLETE	+					

TP607003	SIP reference: RFC 3261				ISDN/ISDN reference: Q.1912.5 clause 7.7.3					
TSS reference	ISDN-SIP /Basic call/Autonomous_release									
SIP selection	PICS 4/4 AND 4/5									
criteria										
ISDN selection criteria										
Test purpose	Ensure that the SUT when the internal resource reservation is unsuccessful and preconditions used, the SUT  sends a CANCEL or BYE to the SIP network.  sends a RELEASE to the ISDN terminal									
SIP parameter values										
ISDN parameter values										
Comments	ISDN		SUT		SIP-I					
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)					
					183 Session Progress					
				PRACK						
					200 OK PRACK					
	Internal resource									
		res	servation uns	uccessful						
	CASE A									
	RELEASE	← → CANCEL(REL)								
	RELEASE COMPLETE	<b>→</b>		+						
				+	487 Request Terminated					
				<b>→</b>	ACK					
	CASE B									
	RELEASE	+		<b>→</b>	BYE(REL)					
	RELEASE COMPLETE	<b>→</b>		+	200 OK BYE(RLC)					
				+	487 Request Terminated					
	→ ACK									

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

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

TP701001	SIP reference: RFC 3		ISUP reference: Q.1912.5			
				Q.7	31 clause 3.5.2.1.1	
TSS reference	ISDN-(ISUP)-SIP/SS/CLIP					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Calling party number (network provided)  Verify that the IUT can successfully originate a call having a calling party number with the screening indicator set to "network provided" and the presentation restricted indicator set to "presentation allowed"					
SIP parameter	INVITE: Content-Type: applicat	ion/ISUP; IAI	d encapsula	ted i	n the MIME body	
values	180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body					
ISDN parameter values						
Comments	ISDN	S	SUT		SIP-I	
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)	
	ALERTING	<del>(</del>		<del>(</del>	180 Ringing(ACM)	
	CONN	<del>-</del>		<b>←</b>	200 OK INVITE(ANM)	
				<b>→</b>	ACK	
	Communication					
	DISC	<b>→</b>		<b>→</b>	BYE(REL)	
	REL	+		<del>(</del>	200 OK BYE(RLC)	
	REL_COM	<b>→</b>				

TP701002	SIP reference: RFC 326	61	ISUP reference: Q.1912.5				
			Q.7	'31 clause 3.5.2.1.1			
TSS reference	ISDN-(ISUP)-SIP/SS/CLIP						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Calling party number (network provided) with calling sub-address  Verify that the IUT can successfully originate a call having a calling party number with the screening indicator set to "network provided" and an access transport parameter containing the calling sub-address						
SIP parameter	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body						
values	180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body						
ISDN parameter values							
Comments	ISDN	SU	JT	SIP-I			
	SETUP →		<b>→</b>	INVITE(IAM)			
	ALERTING ← 180 Ringing(ACM)						
	CONN ← 200 OK INVITE(ANM)  → ACK						
	Communication  DISC → BYE(REL)						
	REL +		+	200 OK BYE(RLC)			
	REL_COM →						

TP701003	SIP referenc	e: RFC 3261			ISUP reference: Q.1912.5 31 clause 3.5.2.1.1			
TSS reference	ISDN-(ISUP)-SIP/SS/0	CLIP		Q.1	51 Clause 5.5.2.1.1			
SIP selection criteria		<del></del> :						
ISUP selection criteria								
Test purpose	Calling party number (user provided, verified and passed)  Verify that the IUT can successfully originate a call having the calling party number with the screening indicator set to "user provided, verified and passed"							
SIP parameter	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body							
values	180 Ringing: Content-	180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body						
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ALERTING	<del>(</del>		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
	Communication							
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	REL	+		+	200 OK BYE(RLC)			
	REL_COM	<b>→</b>						

TP701004	SIP reference: RFC 3261			I:	SUP reference: Q.1912.5		
				Q.73	31 clause 3.5.2.1.1		
TSS reference	ISDN-(ISUP)-SIP/SS/CLIP						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Calling party number (user provided, verified and passed) with calling sub-address Verify that the IUT can successfully originate a call having a calling party number with the screening indicator set to "user provided, verified and passed" and an access transport parameter containing the calling sub-address						
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body						
ISDN parameter values							
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ALERTING ← 180 Ringing(ACM)						
	CONN ← 200 OK INVITE(ANM)  → ACK						
	Communication						
	DISC	<b>→</b>		<b>→</b>	BYE(REL)		
	REL	+		+	200 OK BYE(RLC)		
	REL_COM	<b>→</b>		•			

TP701005	SIP reference: RFC 3261			ISUP reference: Q.1912.5			
					Q.7	31 clause 3.5.2.1.1	
TSS reference	ISDN-(ISUP)-SIP/SS/CLIP						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Calling party number (user provided, not verified)  Verify that the IUT can successfully originate a call having a default calling party number with the screening indicator set to "network provided" and a generic number containing the additional calling party number with the screening indicator set to "user provided, not verified"						
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body						
ISDN parameter values						•	
Comments	ISDN		SU	Т		SIP-I	
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)	
	ALERTING	<b>←</b>			+	180 Ringing(ACM)	
	CONN	<b>←</b>			+	200 OK INVITE(ANM)	
	→ ACK						
	Communication						
	DISC	<b>→</b>			<b>→</b>	BYE(REL)	
	REL	<b>←</b>			+	200 OK BYE(RLC)	
	REL_COM	<b>→</b>					

TP701006	SIP reference: RFC 3261			ISUP reference: Q.1912.5 Q.731 clause 3.5.2.1.1				
TSS reference	ISDN-(ISUP)-SIP/SS/0	CLIP						
SIP selection criteria								
ISUP selection criteria								
Test purpose	Verify that the IUT can with the screening indi- additional calling party	Calling party number (user provided, not verified) with calling sub-address Verify that the IUT can successfully originate a call having a default calling party number with the screening indicator set to "network provided", a generic number containing the additional calling party number with the screening indicator set to "user provided, not verified" and an access transport parameter containing the calling sub-address						
SIP parameter values	INVITE: Content-Type: 180 Ringing: Content-				in the MIME body lated in the MIME body			
ISDN parameter values		71 11	,	'	,			
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
		Communication						
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	REL	+		+	200 OK BYE(RLC)			
	REL_COM	<b>→</b>						

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

TP702001	SIP reference: RF	FC 3261		_	SUP reference: Q.1912.5 31 clause 4.5.2.1.1		
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Restricted calling party number (network provided)  Verify that the IUT can successfully originate a call having a calling party number with the screening indicator set to "network provided" and the address presentation restricted indicator set to "presentation restricted"						
SIP parameter values	INVITE: Content-Type: app 180 Ringing: Content-Type:						
ISDN parameter values					•		
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ALERTING	+		+	180 Ringing(ACM)		
	CONN	+		+	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
			Communicati	on			
	DISC	<b>→</b>	_	<b>→</b>	BYE(REL)		
	REL	+		+	200 OK BYE(RLC)		
	REL_COM	<b>→</b>					

TP702002	SIP reference: RFC 3261			ISUP reference: Q.1912.5			
TSS reference	ICDN (ICLID) CID/CC/CLID			Q.7	31 clause 4.5.2.1.1		
SIP selection	ISDN-(ISUP)-SIP/SS/CLIR						
criteria							
ISUP selection							
criteria							
Test purpose	Restricted calling party number (network provided) with calling sub-address Verify that the IUT can successfully originate a call having a calling party number with the screening indicator set to "network provided", the address presentation restricted indicator set to "presentation restricted" and an access transport parameter containing the calling sub-address						
SIP parameter	INVITE: Content-Type: appli	ication/ISU	P; IAM encapsu	ılated i	n the MIME body		
values	180 Ringing: Content-Type:	application	/ISUP; ACM en	capsul	ated in the MIME body		
ISDN parameter values							
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ALERTING	+		+	180 Ringing(ACM)		
	CONN	+		+	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
		С	ommunication				
	DISC	<b>→</b>		<b>→</b>	BYE(REL)		
	REL	+		+	200 OK BYE(RLC)		
	REL_COM	<b>→</b>					

TP702003	SIP reference	: RFC 3261		-	SUP reference: Q.1912.5 31 clause 4.5.2.1.1			
ISDN parameter values	ISDN-(ISUP)-SIP/SS/CI	LIR						
ISDN parameter values								
ISDN parameter values								
ISDN parameter values	Verify that the IUT can seem the screening indicators	Restricted calling party number (user provided, verified and passed)  Verify that the IUT can successfully originate a call having the calling party number with the screening indicator set to "user provided, verified and passed" and the address presentation restricted indicator set to "presentation restricted"						
ISDN parameter values	INVITE: Content-Type: 6 180 Ringing: Content-Ty							
ISDN parameter values								
Comments	ISDN SETUP ALERTING CONN DISC	÷ + +	SUT	+ + + + +	SIP-I INVITE(IAM) 180 Ringing(ACM) 200 OK INVITE(ANM) ACK BYE(REL)			
	REL_COM	<b>←</b>		+	200 OK BYE(RLC)			

TP702004	SIP reference: F	RFC 3261			SUP reference: Q.1912.5 31 clause 4.5.2.1.1		
TSS reference	ISDN-(ISUP)-SIP/SS/CLIF	₹					
SIP selection criteria							
ISUP selection criteria							
Test purpose	Restricted calling party number (user provided, verified and passed) with calling sub-address  Verify that the IUT can successfully originate a call having a calling party number with the screening indicator set to "user provided, verified and passed", the address presentation restricted indicator set to "presentation restricted" and an access transport parameter containing the calling sub-address						
SIP parameter	INVITE: Content-Type: ap			ulated i	in the MIME body		
values	180 Ringing: Content-Type	e: applicat	ion/ISUP; ACM er	ncapsu	lated in the MIME body		
ISDN parameter values							
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ALERTING	+		+	180 Ringing(ACM)		
	CONN	+		+	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
			Communication				
	DISC	<b>→</b>		<b>→</b>	BYE(REL)		
	REL	+		+	200 OK BYE(RLC)		
	REL_COM	<b>→</b>					

TP702005	SIP reference: RFC 32	61		SUP reference: Q.1912.5 31 clause 4.5.2.1.1			
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR		Q.1	51 Clause 4.5.2.1.1			
SIP selection criteria	iobit (ioo: y oii yoo,oeiit						
ISUP selection criteria							
Test purpose	Restricted calling party number (user provided, not verified)  Verify that the IUT can successfully originate a call having a default calling party number with the screening indicator set to "network provided" and a generic number containing the additional calling party number with the screening indicator set to "user provided, not verified", both having the address presentation restricted indicator set to "presentation restricted"						
SIP parameter	INVITE: Content-Type: application	n/ISUP; IAM	encapsulated i	in the MIME body			
values	180 Ringing: Content-Type: appl	ication/ISUP;	ACM encapsu	lated in the MIME body			
ISDN parameter values							
Comments	ISDN	SI	JT	SIP-I			
	SETUP -	<b>&gt;</b>	<b>→</b>	INVITE(IAM)			
	ALERTING	-	+	180 Ringing(ACM)			
	CONN	-	+	200 OK INVITE(ANM)			
			→	ACK			
	Communication						
	DISC -	•	<b>→</b>	BYE(REL)			
	REL	<del>-</del>	+	200 OK BYE(RLC)			
	REL_COM -	•					

TP702006	SIP reference: RFC	3261			ISUP reference: Q.1912.5
				0.	731 clause 4.5.2.1.1
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR			Q.	731 Clause 4.3.2.1.1
SIP selection	13DIN-(13UP)-31P/33/CLIR				
criteria					
ISUP selection					
criteria					
Test purpose	Restricted calling party nun address	•	-	•	,
	Verify that the IUT can success with the screening indicator se				default calling party number eric number containing the
	additional calling party number verified", both having the additional calling party number verified.	ress pr	esentation re	estricted inc	licator set to "presentation
	restricted" and an access tra				
SIP parameter	INVITE: Content-Type: applic				
values	180 Ringing: Content-Type: a	pplicat	ion/ISUP; AC	CM encapsu	lated in the MIME body
ISDN parameter					
values				1	
Comments	ISDN		SUT		SIP-I
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)
	ALERTING	+		<b>←</b>	180 Ringing(ACM)
	CONN	+		←	200 OK INVITE(ANM)
				→	ACK
			Communica	ation	
	DISC	<b>→</b>		<b>→</b>	BYE(REL)
	REL	+		+	200 OK BYE(RLC)
	REL_COM	1			

TP702007	SIP reference: RFC 3	3261	_	SUP reference: Q.1912.5 731 clause 4.2.1					
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR								
SIP selection criteria									
ISUP selection criteria	DLE	DLE							
Test purpose	Presentation of the address - interaction with MCID  Verify that the information conveyed in an incoming call (especially the calling party number and the additional calling party number in the generic number) is registered in the network regardless of whether the calling user has activated the CLIR service or not, if the called user has MCID activated								
SIP parameter values	INVITE: Content-Type: applicated 180 Ringing: Content-Type: ap								
ISDN parameter values				•					
Comments	ISDN	SI	JT	SIP-I					
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)					
	ALERTING	+	+	180 Ringing(ACM)					
	CONN	<del>-</del>	+	200 OK INVITE(ANM)					
			→	ACK					
		Commu							
	DISC	<b>→</b>	<b>→</b>	BYE(REL)					
	REL	<b>←</b>	+	200 OK BYE(RLC)					
	REL_COM	<b>→</b>							

TP702008	SIP reference: RFC 3261			I	SUP reference: Q.1912.5		
					Q.	731 clause 4.2.1	
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR						
SIP selection criteria							
ISUP selection criteria	DLE						
Test purpose	Presentation of the address - called party has override category  Verify that the calling party number and the additional calling party number in the  generic number are passed to the access regardless of whether the calling user has  activated the CLIR service or not if the called user has the override category						
SIP parameter	INVITE: Content-Type: applica	tion/IS	SUP; IAM e	ncapsula	ted ir	n the MIME body	
values	180 Ringing: Content-Type: ap	plicati	on/ISUP; /	ACM enca	psul	ated in the MIME body	
ISDN parameter values							
Comments	ISDN		SUT	-		SIP-I	
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)	
	ALERTING	+			<del>(</del>	180 Ringing(ACM)	
	CONN	+			<del>(</del>	200 OK INVITE(ANM)	
	→ ACK						
		Communication					
	DISC	<b>+</b>			<b>→</b>	BYE(REL)	
	REL	+	_		<del>(</del>	200 OK BYE(RLC)	
	REL_COM	<b>→</b>					

TP702009	SIP reference: RFC 3261			ISUP reference: Q.1912.5 Q.731 clause 4.2.1				
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR							
SIP selection criteria								
ISUP selection criteria	DLE	DLE						
Test purpose	Presentation of the address - called party has not override category Verify that the calling party number is not passed to the access							
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body							
ISDN parameter values								
Comments	ISDN		SU	Т		SIP-I		
	SETUP	<b>→</b>		-	<b>&gt;</b>	INVITE(IAM)		
	ALERTING	+		•	F	180 Ringing(ACM)		
	CONN	+		•	Т	200 OK INVITE(ANM)		
				-	*	ACK		
			Commun	ication				
	DISC	<b>→</b>		-	<b>&gt;</b>	BYE(REL)		
	REL	+		•	1	200 OK BYE(RLC)		
	REL_COM	<b>→</b>						

TP7020010	SIP reference: RFC 3261			ISUP reference: Q.1912.5 Q.731 clause 4.2.1				
TSS reference	ISDN-(ISUP)-SIP/SS/C	CLIR						
SIP selection								
criteria								
ISUP selection	DLE							
criteria								
Test purpose	Verify that the calling	Presentation of the address - called party has not override category Verify that the calling party number and the additional calling party number in the generic number are not passed to the access						
SIP parameter	INVITE: Content-Type:	: application/ISL	JP; IAM encap	sulated i	n the MIME body			
values	180 Ringing: Content-	Type: application	n/ISUP; ACM e	encapsu	lated in the MIME body			
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		→	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
		Communication						
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	REL	<del>(</del>		+	200 OK BYE(RLC)			
	REL_COM	<b>→</b>						

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

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

TP703002	SIP reference: RFC 3261			ISUP reference: Q.1912.5 Q.731 clause 5.5.2.5.1 i)				
TSS reference	ISDN-(ISUP)-SIP/SS/COLP							
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	Verify that the IUT can provid	Connected number (user provided, verified and passed) Verify that the IUT can provide a connected number with the screening indicator set to  'user provided, verified and passed", if the user provided COL is valid						
SIP parameter	INVITE: Content-Type: applic	ation/IS	SUP; IAM end	apsulated i	in the MIME body			
values		180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body						
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	CASE A							
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
	CASE B							
	CONN	+		+	200 OK INVITE(CON)			
				<b>→</b>	ACK			
			Communicat	tion				
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	REL	+		<b>←</b>	200 OK BYE(RLC)			
	REL_COM	<b>→</b>						

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

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

TP703005	SIP reference: RF0	SUP reference:					
					Q.1912.5		
				Q.73	1 clause 5.5.2.5.1 i)		
TSS reference	ISDN-(ISUP)-SIP/SS/COLP						
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Connected number (network						
	Verify that the IUT can provide						
	set to "network provided", if the				d and an access transport		
	parameter containing the cor	nected	sub-address	5			
SIP parameter	INVITE: Content-Type: applic						
values	180 Ringing: Content-Type: a						
	200 OK INVITE: Content-Typ	e: appl	ication/ISUP;	; ANM enca <sub>l</sub>	osulated in the MIME body		
ISDN parameter							
values		1	T				
Comments	ISDN		SUT		SIP-I		
	SETUP	→		<b>→</b>	INVITE(IAM)		
	CASE A						
	ALERTING	+		+	180 Ringing(ACM)		
	CONN	+		+	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
	CASE B						
	CONN ← 200 OK INVI						
				<b>→</b>	ACK		
			Communica	ition			
	DISC	<b>→</b>		<b>→</b>	BYE(REL)		
	REL	+		+	200 OK BYE(RLC)		
	REL_COM	<b>→</b>					

TP703006	SIP reference: RF	C 3261			SUP reference: Q.1912.5 I1 clause 5.5.2.5.1 i)			
TSS reference	ISDN-(ISUP)-SIP/SS/COLP							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Connected number (user provided, not verified)  Verify that the IUT can provide a default connected number with the screening indicator set to "network provided" and a generic number containing the additional connected number with the screening indicator set to "user provided, not verified"							
SIP parameter	INVITE: Content-Type: appl							
values	180 Ringing: Content-Type: 200 OK INVITE: Content-Ty							
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		→	INVITE(IAM)			
	CASE A							
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
	CASE B							
	CONN	+		+	200 OK INVITE(CON)			
				→	ACK			
			Communic	ation				
	DISC	<b>→</b>		→	BYE(REL)			
	REL	+		<del>-</del>	200 OK BYE(RLC)			
	REL_COM	<b>→</b>						

TP703007	SIP reference:	RFC 3261			ISUP reference: Q.1912.5 31 clause 5.5.2.5.1 i)				
TSS reference	ISDN-(ISUP)-SIP/SS/CC	)LP							
SIP selection criteria									
ISUP selection criteria									
Test purpose	Verify that the IUT can p set to "network provided with the screening indica	Connected number (user provided, not verified) with connected sub-address Verify that the IUT can provide a default connected number with the screening indicator set to "network provided", a generic number containing the additional connected number with the screening indicator set to "user provided, not verified" and an access transport parameter containing the connected sub-address							
SIP parameter values	INVITE: Content-Type: a 180 Ringing: Content-Ty	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body							
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	CASE A								
	ALERTING	+		+	180 Ringing(ACM)				
	CONN	<b>←</b>		<b>←</b>	200 OK INVITE(ANM)				
				→	ACK				
	CASE B								
	CONN	+		+	200 OK INVITE(CON)				
				<b>→</b>	ACK				
		Co	mmunication	1					
	DISC	<b>→</b>	-	<b>→</b>	BYE(REL)				
	REL	+		+	200 OK BYE(RLC)				
	REL_COM	<b>→</b>							

TP703008	SIP reference: RFC 3261				ISUP reference: Q.1912.5 31 clause 5.5.2.5.1 i)			
TSS reference	ISDN-(ISUP)-SIP/SS/COLP			4				
SIP selection criteria								
ISUP selection criteria								
Test purpose	Verify that an exchange with	COLP - interaction with MSN  Verify that an exchange with MSN can provide the connected party multiple subscriber number or full ISDN number as the connected number on call answer						
SIP parameter values	180 Ringing: Content-Type: a	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body						
ISDN parameter values								
Comments	ISDN		SU	IT	SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	CASE A							
	ALERTING	+		<b>←</b>	180 Ringing(ACM)			
	CONN	+		<b>←</b>	200 OK INVITE(ANM)			
				→	ACK			
	CASE B							
	CONN	+		←	200 OK INVITE(CON)			
		ACK						
			Commun					
	DISC	<b>→</b>		→	BYE(REL)			
	REL	+		<b>←</b>	200 OK BYE(RLC)			
	REL_COM	→						

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

TP704001	SIP reference: RFC 3261			ISUP reference: Q.1912.5 Q.731 clause 6.5.2.1.2				
TSS reference	ISDN-(ISUP)-SIP/SS/0	COLR						
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	Verify that a local exch system when a <b>conne</b> presentation restricted	Presentation of restricted COL  Verify that a local exchange will not pass the information on to the access signalling system when a connected number is received in the ANM or CON and its address presentation restricted indicator is set to "presentation restricted", i.e. that presentation is denied on the user-network interface (UNI)						
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body							
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	CASE A							
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
	CASE B							
	CONN	<del>(</del>		+	200 OK INVITE(CON)			
				<b>→</b>	ACK			
			Communication	<u> </u>				
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	REL	<del>(</del>		+	200 OK BYE(RLC)			
	REL_COM	<b>→</b>						

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

TP704003	SIP reference: RF	C 3261			ISUP reference: Q.1912.5 '31 clause 6.5.2.1.2				
TSS reference	ISDN-(ISUP)-SIP/SS/COLR								
SIP selection criteria									
ISUP selection criteria	DLE								
Test purpose	Verify that the IUT can provided, verified and set to "presentation restricted"	Restricted connected number (user provided, verified and passed)  Verify that the IUT can provide a connected number with the screening indicator set to  "user provided, verified and passed" and with the address presentation restricted indicator set to "presentation restricted", if the user provided COL is valid							
SIP parameter	INVITE: Content-Type: appli								
values	180 Ringing: Content-Type: 200 OK INVITE: Content-Type								
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		→	INVITE(IAM)				
	CASE A								
	ALERTING	+		<b>←</b>	180 Ringing(ACM)				
	CONN	+		<b>←</b>	200 OK INVITE(ANM)				
				→	ACK				
	CASE B								
	CONN	+		<b>←</b>	200 OK INVITE(CON)				
				→	ACK				
			Communicatio	n					
	DISC	<b>→</b>		<b>→</b>	BYE(REL)				
	REL	+		+	200 OK BYE(RLC)				
	REL_COM	<b>→</b>							

TP704004	SIP referen	ce: RFC 3261			ISUP reference: Q.1912.5 731 clause 6.5.2.1.			
TSS reference	ISDN-(ISUP)-SIP/SS/	COLR						
SIP selection criteria								
ISUP selection criteria								
Test purpose	Restricted connected number (user provided, verified and passed) with connected sub-address  Verify that the IUT can provide a connected number with the screening indicator set to "user provided, verified and passed" and with the address presentation restricted indicator set to "presentation restricted", if the user provided COL is valid. Additionally, an access transport parameter containing the connected sub-address shall also be provided							
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body							
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	CASE A							
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
	CASE B							
	CONN	+		+	200 OK INVITE(CON)			
				<b>→</b>	ACK			
		C	ommunicatior	1				
	DISC	<b>→</b>	<u>-</u>	<b>→</b>	BYE(REL)			
	REL	+		+	200 OK BYE(RLC)			
	REL_COM	<b>→</b>						

TP704005	SIP reference: RFC	3261			SUP reference: Q.1912.5 731 clause 6.5.2.5				
TSS reference	ISDN-(ISUP)-SIP/SS/COLR								
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Verify that the IUT can provided set to "network provided" and	Restricted connected number (network provided) Verify that the IUT can provide a default connected number with the screening indicator set to "network provided" and with the address presentation restricted indicator set to "presentation restricted", if the user provided COL is not valid							
SIP parameter	INVITE: Content-Type: applic	ation/I	SUP; IAM encap	sulated	in the MIME body				
values	180 Ringing: Content-Type: a 200 OK INVITE: Content-Typ								
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	CASE A								
	ALERTING	+		<b>←</b>	180 Ringing(ACM)				
	CONN	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
	CASE B								
	CONN	+		+	200 OK INVITE(CON)				
				→	ACK				
			Communication						
	DISC	<b>→</b>		<b>→</b>	BYE(REL)				
	REL	+		+	200 OK BYE(RLC)				
	REL_COM	<b>→</b>							

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

TP704007	SIP reference: RFC 3261		Q.191		ce: 6.5.2.1.2		
TSS reference	ISDN-(ISUP)-SIP/SS/COL	R					
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Restricted connected number (user provided, not verified) with connected sub-address  Verify that the IUT can provide a default calling party number with the screening indicator set to "network provided", a generic number containing the additional connected number with the screening indicator set to "user provided, not verified" - both having the address presentation restricted indicator set to "presentation restricted" and additionally an access transport parameter containing the connected sub-address						
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body						
ISDN parameter values		71 11	,		,		
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
	CASE A						
	ALERTING	+		+	180 Ringing(ACM)		
	CONN	+		+	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
	CASE B						
	CONN	+		+	200 OK INVITE(CON)		
				<b>→</b>	ACK		
			Communication	1			
	DISC	<b>→</b>		<b>→</b>	BYE(REL)		
	REL	+		+	200 OK BYE(RLC)		
	REL_COM	<b>→</b>					

## A.1.1.2.5 Terminal Portability (TP)

TP705001	SIP reference: RFC 3261			ISUP reference: Q.1912.5 4.5.2.1.1 a)/Q.733			
TSS reference	ISDN-(ISUP)-SIP/SS/	TP					
SIP selection							
criteria							
ISUP selection criteria							
Test purpose	Terminal portability, requested by the calling party To verify that the calling party can suspend and resume an outgoing call and that user initiated SUS and RES messages are sent to the succeeding exchange						
SIP parameter					-		
values							
ISDN parameter							
values							
Comments	ISDN		SUT		SIP		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ALERTING	<b>←</b>		<b>←</b>	180 Ringing(ACM)		
	CONN	+		+	200 OK INVITE(ANM)		
		(	Communication	า			
	SUSPEND	<b>→</b>		<b>→</b>	INFO(SUS)		
				+	200 OK INFO		
	RESUME	→		<b>→</b>	INFO(RES)		
				+	200 OK INFO		
		(	Communication	า			
	DISC	<b>→</b>		<b>→</b>	BYE(REL)		
	REL	+		+	200 OK BYE		
	REL_COM	<b>→</b>					

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

TP705003	SIP reference	e: RFC 3261			SUP reference: Q.1912.5 4.5.2.1.2/Q.733
TSS reference	ISDN-(ISUP)-SIP/SS/T	Р			
SIP selection criteria					
ISUP selection criteria					
Test purpose		released with	cause #102 (re	covery	Resume after Suspend on timer expiry) by the IUT if turns the call
SIP parameter	,				
values					
ISDN parameter					
values					
Comments	ISDN		SUT		SIP
	SETUP	→		<b>→</b>	INVITE(IAM)
	ALERTING	<b>+</b>		+	180 Ringing(ACM)
	CONN	+		+	200 OK INVITE(ANM)
		C	ommunication	n	
	SUSPEND	<b>→</b>		<b>→</b>	INFO(SUS)
				+	200 OK INFO
			T2 expiry	•	
				<b>→</b>	BYE(REL)
				+	200 OK BYE

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

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

TP705006	SIP reference: RFC 3261			ISUP reference: Q.1912.5 4.5.2.5.1 b)/Q.733					
TSS reference	ISDN-(ISUP)-SIP/SS/TP								
SIP selection criteria									
ISUP selection criteria									
Test purpose	To verify that the called p	Terminal portability, requested by the called party  To verify that the called party can suspend and resume an incoming call and that user initiated SUS and RES messages are sent to the preceding exchange							
SIP parameter		-	•						
values									
ISDN parameter values									
Comments	ISDN		SUT		SIP				
	SETUP	+		+	INVITE(IAM)				
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)				
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)				
		Co	ommunicatio	1 /					
	NOTIFY(suspend)	<b>→</b>		<b>→</b>	INFO(SUS)				
				+	200 OK INFO				
	NOTIFY(resume)	<b>→</b>		<b>→</b>	INFO(RES)				
	,			+	200 OK INFO				
		Co	ommunicatio	n					
	DISC	+		+	BYE(REL)				
	REL	<b>→</b>		<b>→</b>	200 OK BYE				
	REL_COM	+							

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

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

TP706001	SIP reference: RFC 3261				ISUP reference:
				44504	Q.1912.5
				1.1.5.2.1.	1.1; 1.1.5.2.1.1.3; 1.1.5.2.2-
					4.1/Q.737
TSS reference	ISDN-(ISUP)-SIP/SS/U	US1			
SIP selection criteria					
ISUP selection criteria					
Test purpose	UUS1 implicit - reques	st			
	To verify that the IUT c	an successfull	y initiate/trar	nsit a call v	with an UUS 1 implicit request,
	having the user-to-use	r information	parameter i	in the IAM	, without the user-to-user
	indicators parameter				
SIP parameter					
values					
ISDN parameter values	SETUP: User-to-user in	nformation			
Comments	ISDN		SUT		SIP
	SETUP(UUInf)	<b>→</b>		<b>→</b>	INVITE(IAM UUInf)
	ALERTING	+		+	180 Ringing(ACM)
	CONN(UUInf)	+		+	200 OK INVITE(ANM)
				<b>→</b>	ACK
		С	ommunicat	ion	
	DISC	+		+	BYE(REL)
	REL	<b>→</b>		→	200 OK BYE
	REL_COM	+			

TP706002	SIP reference	e: RFC 3261		.1.5.2	ISUP reference: Q.1912.5 2.5.2.3; 1.1.5.2.2-4.2/Q.737
TSS reference	ISDN-(ISUP)-SIP/SS/U	IUS1			
SIP selection criteria					
ISUP selection criteria					
Test purpose	request, continue norm	an, after succ al call set up i	essfully initiati f the first back	ng/tran: ward m	siting a call with an UUS1 implicit nessage is received with the liscarded by the network" (see
SIP parameter values	,				
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP(UUInf)	<b>→</b>		<b>→</b>	INVITE(IAM UUInf)
	ALERTING	+		+	180 Ringing(ACM UUInd)
	CONN	+		+	200 OK INVITE(ANM)
				→	ACK
		(	ommunication	on	
	DISC	+		+	BYE(REL)
	REL	<b>→</b>		<b>→</b>	200 OK BYE
	REL_COM	+			
NOTE: The use	r-to-user information is dis	scarded becar	use the followi	ng netv	vork does not support it.

TP706003	SIP reference	e: RFC 3261			ISUP reference: Q.1912.5
				1.1.5.2	.5.2.3; 1.1.5.2.3-5.2/Q.737
TSS reference	ISDN-(ISUP)-SIP/SS/U	US1			
SIP selection					
criteria					
ISUP selection criteria					
Test purpose		an successfully	y initiate/trans	sit a call	with an UUS1 implicit request,
SIP parameter	•		•		,
values					
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP(UUInf)	<b>→</b>		<b>→</b>	INVITE(IAM UUInf)
	ALERTING	+		+	180 Ringing(ACM)
	CONN	+		+	200 OK INVITE(ANM)
				<b>→</b>	ACK
		C	ommunication	on	
	DISC	+		<b>←</b>	BYE(REL)
	REL	<b>→</b>		→	200 OK BYE
İ	REL_COM	<b>+</b>			

- 1) the remote network is unable to pass the service 1 in any message.2) the remote user may not be able to interpret incoming UUS information.

TP706004	SIP reference	SIP reference: RFC 3261			ISUP reference: Q.1912.5 1.1.5.2.1.1.1; 1.1.5.2.1.1.3; 1.1.5.2.3- 5.1/Q.737				
TSS reference	ISDN-(ISUP)-SIP/SS/U	JS1							
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	To verify that the IUT ca and transfer/include the	UUS1 implicit - acceptance To verify that the IUT can successfully transit/accept a call with an UUS1 implicit request, and transfer/include the user-to-user information parameter in the ACM, CPG, ANM or CON as implicit acceptance (no user-to-user indicators)							
SIP parameter values									
ISDN parameter values									
Comments	ISDN		SUT		SIP				
	SETUP(UUInf)	<b>→</b>		<b>→</b>	INVITE(IAM UUInf)				
	CASE A								
	ALERTING(UUInf)	+		+	180 Ringing(ACM UUInf)				
	CASE B								
	ALERTING	+		+	180 Ringing(ACM)				
	NOTIFY(UUInf)			+	183 Session Progress(CPG UUInf)				
	CASE C								
	CONN(UUInf)	+		+	200 OK INVITE(CON UUInf)				
				<b>→</b>	ACK				
	CASE D								
	ALERTING	+		+	180 Ringing(ACM)				
	CONN(UUInf)	+		+	200 OK INVITE(ANM UUInf)				
				<b>→</b>	ACK				
		Co	ommunicatio	n					
	DISC	+		+	BYE(REL)				
	REL	<b>→</b>		<b>→</b>	200 OK BYE				
	REL_COM	+							

TP706005	SIP reference: RF0	3261			-	SUP reference: Q.1912.5
				1.1	.5.2.5	.2.3; 1.1.5.2.3-5.2/Q.737
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1					
SIP selection criteria						
ISUP selection criteria						
Test purpose	UUS1 implicit - discard with indication generated To verify that the IUT can successfully transit/accept a call with an UUS1 implicit request and set the user-to-user indicators to "user-to-user information discarded by the network" in the first backward message, if the network is unable to support it (see note).					
SIP parameter values						
ISDN parameter values						
Comments	SIP		SU	T		ISDN
	INVITE(IAM UUInf)	<b>→</b>			<b>→</b>	SETUP
	180 Ringing(ACM UUInd)	+			+	ALERTING
	200 OK INVITE(ANM)	+			+	CONN
	ACK	<b>→</b>				
			Commun	ication		
	BYE(REL)	+		-	+	DISC
	200 OK BYE	<b>→</b>			<b>→</b>	REL
					+	REL_COM
NOTE: The user	to-user information is discarde	ed beca	use the ne	etwork do	es no	t support it.

TP706006	SIP reference: RFC	3261	1.1	_	SUP reference: Q.1912.5 .1.2; 1.1.5.2.2-4.1/Q.737
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1		•		·
SIP selection					
criteria					
ISUP selection criteria					
Test purpose	UUS1 explicit non-essential To verify that the IUT can such non-essential request, by inclu- and the user-to-user indicate	cessfully uding/tra	/ initiate/transit a	er-to-u	ser information parameter
SIP parameter values					
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP(FAC uus1reqness)				INVITE(IAM UUInd, UUInf)
	ALERTING(FAC uus1rr)				180 Ringing(ACM)
	CONN				200 OK INVITE(ANM)
					ACK
			Communication		
	BYE(REL)	+		+	DISC
	200 OK BYE	<b>→</b>		<b>→</b>	REL
				+	REL_COM

TP706007	SIP reference: RFC 3261				ISUP reference: Q.1912.5 1.1.5.2.5.2.3; 1.1.5.2.2-4.2/Q.737		
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1						
SIP selection criteria							
ISUP selection							
criteria							
Test purpose	UUS1 explicit non-essential - explicit rejection received To verify that the IUT can successfully initiate/transit a call with an UUS1 explicit non-essential request, and continue normal call set up if the UUS1 service is explicitly rejected (the user-to-user indicators parameter is received as "service not provided" in the ACM or CPG or ANM or CON) (see note).						
SIP parameter		, ,		,			
values							
ISDN parameter values							
Comments	ISDN		SUT		SIP		
	SETUP(FAC uus1reqness)	<b>→</b>		<b>→</b>	INVITE(IAM UUInd, UUInf)		
	CASE A						
	ALERTING(FAC uus1rr)	<b>←</b>		+	180 Ringing(ACM UUInd s1 prov)		
	CONN	<b>←</b>		+	200 OK INVITE		
	CASE B						
				+	183 Session Progress(ACM)		
	ALERTING(FAC uus1rr)	+		+	180 Ringing(CPG UUInd s1 prov)		
	CONN	+		+	200 OK INVITE		
	CASE C						
	ALERTING	+		+	180 Ringing(ACM)		
	CONN(FAC uus1rr)	<b>←</b>		+	200 OK INVITE(ANM UUInd s1 prov)		
	CASE D						
	CONN(FAC uus1rr)	+		+	200 OK INVITE(ANM UUInd s1 prov)		
			municat	ion			
	BYE(REL)	+		+	DISC		
	200 OK BYE	<b>→</b>		<b>→</b>	REL		
				+	REL_COM		
1) the net	-to-user information is discarde twork is unable to pass the expinote user may not be able to in	licit ser	vice 1 in				

TP706008	SIP reference: RFC	3261	1	ISUP reference: Q.1912.5 1.1.5.2.5.2.3; I.1.5.2.2-4.2/Q.737			
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1						
SIP selection criteria							
ISUP selection criteria							
Test purpose	To verify that the IUT can suc	UUS1 explicit non-essential - implicit (no explicit) rejection received  To verify that the IUT can successfully initiate/transit a call with an UUS1 explicit non-essential request, and continue normal call set up if no indication is provided in the					
SIP parameter values							
ISDN parameter values							
Comments	ISDN	S	UT	SIP			
	SETUP(FAC uus1reqness)	<b>→</b>	→	INVITE(IAM UUInd, UUInf)			
	ALERTING(FAC uus1rr)	+	+	180 Ringing(ACM)			
	CONN(FAC uus1reterr)	+	+	200 OK INVITE(ANM)			
			<b>→</b>	ACK			
		Commu	ınication				
	BYE(REL)	+	+	DISC			
	200 OK BYE	<b>→</b>	→	REL			
			+	REL_COM			
1) the net	to-user information is discarde work is unable to pass the expl	licit service 1 in					
2) the ren	note user may not be able to in	terpret incoming	g UUS informat	ion.			

TP706009	SIP reference: RFC 3261	1.1.5	ISUP reference: Q.1912.5 1.1.5.2.1.1.2; 1.1.5.2.3-5.1/Q.737					
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1							
SIP selection criteria								
ISUP selection criteria								
Test purpose	UUS1 explicit non-essential - acceptance To verify that the IUT can successfully transit/accept a call with an UUS1 explicit non-essential request, by transferring/including the user-to-user indicators parameter in the ACM, CPG, ANM or CON set to "service provided"							
SIP parameter values	, ,		·					
ISDN parameter values								
Comments	SIP		SUT		ISDN			
	INVITE(IAM UUInd, UUInf)	<b>→</b>		<b>^</b>	SETUP(FAC uus1reqness)			
					CASE A			
	180 Ringing(ACM UUInd s1 prov)	+		4	ALERTING(FAC uus1rr)			
	200 OK INVITE	+		+	CONN			
					CASE B			
	183 Session Progress(ACM)	+						
	180 Ringing(CPG UUInd s1 prov)	+		+	ALERTING(FAC uus1rr)			
	200 OK INVITE	+		+	CONN			
					CASE C			
	180 Ringing(ACM)	+		+	ALERTING			
	200 OK INVITE(ANM UUInd s1 prov)	+		+	CONN(FAC uus1rr)			
					CASE D			
	200 OK INVITE(ANM UUInd s1 prov)							
	,	Co	mmunicatio	n				
	DISC	+		<b>←</b>	BYE(REL)			
	REL	→		<b>→</b>	200 OK BYE			
	REL_COM			+				

TP706010	SIP reference: RFC	3261	1.4		ISUP reference: Q.1912.5 5.2.2; 1.1.5.2.2-5.2/Q.737
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1		1.	1.3.2.3	5.2.2, 1.1.5.2.2-3.2/Q.737
SIP selection criteria					
ISUP selection criteria					
Test purpose	UUS1 explicit non-essentia To verify that the IUT can tran request, and reject the servic the ACM, CPG, ANM or CON	nsfer/aco e by not	cept a call with an providing any <b>us</b>	UUS	
SIP parameter values		`	,		
ISDN parameter values					
Comments	SIP		SUT		ISDN
	INVITE(IAM UUInd, UUInf)	<b>→</b>		<b>→</b>	SETUP(FAC uus1reqness)
	180 Ringing(ACM)	+		+	ALERTING
	200 OK INVITE(ANM)	+		+	CONN
	ACK	<b>→</b>			
		(	Communication		
	DISC	+		+	BYE(REL)
	REL	<b>→</b>		<b>→</b>	200 OK BYE
	REL_COM			+	
NOTE: The nety	vork or the user cannot support	UUS1.			

TP706011	SIP reference: RFC	3261			SUP reference: Q.1912.5
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1			1.1.3.2.1	.1.2; 1.1.5.2.2-5.1/Q.737
SIP selection criteria	13514-(1361-)-011-736/0061				
ISUP selection criteria					
Test purpose	To verify that the IUT can successential request, by including parameter, the user-to-user in preference indicator in the for	cessfully /transfer ndicator	ring in the <b>IAN</b> <b>s</b> set to "requ	<b>I</b> f the <b>use</b> est, esse	er-to-user information ential" and the ISDN user part
SIP parameter values					
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP(FAC uus1reqess)				INVITE(IAM UUInd, UUInf)
	ALERTING(FAC uus1rr)				180 Ringing(ACM)
	CONN				200 OK INVITE(ANM)
					ACK
		С	ommunication	n	
	BYE(REL)	+		+	DISC
	200 OK BYE	<b>→</b>		<b>→</b>	REL
				<b>←</b>	REL_COM

TP706012	SIP reference: RF	C 3261			ISUP reference:
					Q.1912.5
			1.1	1.5.2.5	5.2.2; 1.1.5.2.2-5.2/Q.737
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	UUS1 explicit essential - in To verify that the service car parameter or the service 1 fie "not provided") is received in (see note).	n be rejec eld in the	cted if no indication user inc	n (no dicato	user-to-user indicators rs set to "no information" or
SIP parameter					
values					
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP(FAC uus1regess)	<b>→</b>		<b>→</b>	INVITE(IAM UUInd, UUInf)
	ALERTING(FAC uus1rr)	+		+	180 Ringing(ACM)
	CONN(FAC uus1reterr)	+		+	200 OK INVITE(ANM)
	·			<b>→</b>	ACK
		(	Communication		
	BYE(REL)	+		+	DISC
	200 OK BYE	<b>→</b>		<b>→</b>	REL
				+	REL_COM
NOTE: The netv	vork does not understand the s	ervice 1	request. In this ca	ase the	e call should be released.

TP706013	SIP reference: RFC 3261	1.1.	ISUP reference: Q.1912.5 1.1.5.2.1.1.2; 1.1.5.2.2-5.1/Q.737				
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1						
SIP selection criteria							
ISUP selection criteria							
Test purpose	UUS1 explicit essential - acceptance To verify that the IUT can successfully complete a call with an UUS1 explicit essential request having the user-to-user indicators parameter in the ACM, CPG, ANM or CON set to "service provided"						
SIP parameter values	correction promises						
ISDN parameter values							
Comments	SIP		SUT		ISDN		
	INVITE(IAM UUInd, UUInf)	<b>→</b>		<b>→</b>	SETUP(FAC uus1reqess)		
	CASE A						
	180 Ringing(ACM UUInd s1 prov)	+		+	ALERTING(FAC uus1rr)		
	200 OK INVITE	+		+	CONN		
	CASE B						
	183 Session Progress(ACM)	+					
	180 Ringing(CPG UUInd s1 prov)	+		+	ALERTING(FAC uus1rr)		
	200 OK INVITE	+		+	CONN		
	CASE C						
	180 Ringing(ACM)	+		+	ALERTING		
	200 OK INVITE(ANM UUInd s1 prov)	+		+	CONN(FAC uus1rr)		
	CASE D						
	200 OK INVITE(ANM UUInd s1 prov)						
		Coi	mmunicatio	n			
	DISC	+		<b>←</b>	BYE(REL)		
	REL	<b>→</b>		<b>→</b>	200 OK BYE		
	REL_COM			+			

TP706014	SIP reference: RFC	3261		1.	ISUP reference: Q.1912.5 .1.5.2.5.2.2; 1.1.5.2.2-5.2/Q.737		
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1		•		•		
SIP selection criteria							
ISUP selection criteria							
Test purpose	UUS1 explicit essential - rejection To verify that the service can be rejected with a REL having the Cause value 29 "facility rejected" or 69 "requested facility not implemented", either with diagnostics (specifying the name of the user-to-user indicator parameter) (see note).						
SIP parameter values							
ISDN parameter values							
Comments	ISDN		SUT		SIP		
	SETUP(FAC uus1reqess)	<b>→</b>		<b>→</b>	INVITE(IAM UUInd, UUInf)		
	RELEASE	+		+	500 Server Internal Error(REL#29)		
	RELEASE COLMPLETE	<b>→</b>		<b>→</b>	ACK		
NOTE: The netw	ork or the called user cannot s	upport tl	ne service.				

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

TP706101	SIP reference: R	FC 3261	1		ISUP reference: Q.1912.5 .1.2; 1.2.5.2.2-5.1/Q.737
TSS reference	ISDN-(ISUP)-SIP/SS/UUS	2			
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	essential request, having the essential". To verify that the	successful he <b>user-to</b> e IUT can	lly originate/trans <b>b-user indicator</b> successfully co	sit a call <b>s</b> in the mplete a	with an UUS2 explicit non- IAM set to "request, not a call with an UUS2 explicit arameter in the ACM or CPG
SIP parameter					
values					
ISDN parameter					
values					
Comments	ISDN		SUT		SIP
	SETUP	→		→	INVITE(IAM UU2 not ess)
	ALERTING	<b>←</b>		<b>←</b>	180 Ringing(ACM)
	USER INFO	→		→	INFO(USR)
				<b>←</b>	200 OK INFO
	USER INFO	<b>←</b>		<b>←</b>	INFO(USR)
				→	200 OK INFO
	CONN	+		+	200 OK INVITE(ANM)
			Communication	า	
	DISC	+		+	BYE(REL)
	RELEASE	<b>→</b>		<b>→</b>	200 OK BYE
	REL_COM	+			

TP706102	SIP reference: RFC 3261				ISUP reference: Q.1912.5 1.2.5.2.5.2.2; 1.2.5.2.2-5.2/Q.737		
TSS reference	ISDN-(ISUP)-SIP/SS/UUS2			•			
SIP selection criteria							
ISUP selection criteria							
Test purpose	UUS2 explicit non-essential - explicit rejection (service not provided) To verify that the UUS2 service can be rejected and the user-to-user indicators in the ACM or CPG are set to "service 2 not provided" (see note).						
SIP parameter values			•		,		
ISDN parameter values							
Comments	ISDN		SUT		SIP		
	SETUP(FAC uus2regness)	<b>→</b>		<b>→</b>	INVITE(IAM UU2 ness		
	CASE A						
	ALERTING(FAC uus1rr)	+		+	180 Ringing(ACM UUInd s2 prov)		
	CONN	+		+	200 OK INVITE		
	CASE B						
				+	183 Session Progress(ACM)		
	ALERTING(FAC uus1rr)	+		+	180 Ringing(CPG UUInd s1 prov)		
	CONN	+		+	200 OK INVITE		
		Com	munica	tion			
	BYE(REL)	+		+	DISC		
	200 OK BYE(RLC)	<b>→</b>		<b>→</b>	REL		
				+	REL_COM		
NOTE: The nety	vork or the user cannot support	UUS2.					

TP706103	SIP reference: RFC	3261		ISUP reference: Q.1912.5				
				1.2.5.2.5	.2.3; 1.2.5.2.2-5.2/Q.737			
TSS reference	ISDN-(ISUP)-SIP/SS/UUS2							
SIP selection criteria								
ISUP selection criteria								
Test purpose	UUS2 explicit non-essential - implicit rejection (no indication)  To verify that the IUT can successfully complete a call with an UUS2 explicit non-essential request, if no indication is provided in the backward direction (see note).							
SIP parameter values					,			
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP(FAC uus1regness)	<b>→</b>		<b>→</b>	INVITE(IAM UU2 ness)			
	ALERTING(FAC uus1rr)	+		+	180 Ringing(ACM)			
	CONN(FAC uus1reterr)	<del>-</del>		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
			Communicati	on				
	BYE(REL)	<del>-</del>		+	DISC			
	200 OK BYE	<b>→</b>		→	REL			
				+	REL_COM			
NOTE: The netw	ork or the user cannot support	UUS2.						

TP706104	SIP referenc	e: RFC 3261		ISUP reference: Q.1912.5 1.2.5.2.1.1.2; 1.2.5.2.2-5.1/Q.737					
TSS reference	ISDN-(ISUP)-SIP/SS/U	JUS2							
SIP selection criteria									
ISUP selection criteria									
Test purpose	To verify that the IUT of essential request, having	UUS2 explicit essential - request To verify that the IUT can successfully originate/transit a call having an UUS2 explicit essential request, having the user-to-user indicators set to "request, essential" and the ISDN user part preference indicator of the forward call indicators in the IAM set to "ISUP required"							
SIP parameter values									
ISDN parameter values									
Comments	ISDN		SUT		SIP				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM UU2 ess)				
	ALERTING	+		+	180 Ringing(ACM)				
	USER INFO	→		<b>→</b>	INFO(USR)				
				+	200 OK INFO				
	USER INFO	<b>←</b>		+	INFO(USR)				
				<b>→</b>	200 OK INFO				
	CONN	+		+	200 OK INVITE(ANM)				
		Communication							
	DISC	+		+	BYE(REL)				
	RELEASE	→		<b>→</b>	200 OK BYE				
	REL_COM	+							

TP706105	SIP reference	e: RFC 3261		ISUP reference: Q.1912.5 1.2.5.2.1.1.2; 1.2.5.2.2-5.1/Q.737				
TSS reference	ISDN-(ISUP)-SIP/SS/UI	US2			,			
SIP selection criteria								
ISUP selection criteria								
Test purpose	UUS2 explicit essential - acceptance To verify that the IUT can successfully complete a call having an UUS2 explicit essential request having the user-to-user indicators parameter in the ACM or CPG set to "service provided"							
SIP parameter values								
ISDN parameter								
values								
Comments	ISDN		SUT		SIP			
	SETUP	+		+	INVITE(IAM UU2 ess)			
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)			
	USER INFO	+		+	INFO(USR)			
				<b>→</b>	200 OK INFO			
	USER INFO	<b>→</b>		<b>→</b>	183 Session Progress(USR)			
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)			
		Co	mmunicati	ion	,			
	DISC	+		<b>←</b>	BYE(REL)			
	RELEASE	<b>→</b>		<b>→</b>	200 OK BYE			
	REL_COM	<del>-</del>						

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

TP706107	SIP refe	erence:	RFC 3261		ISUP reference: Q.1912.5			
TSS reference	ISDN-(ISUP)-SIF	0/99/11119	52		1.2 5.2.5.2.1; 1.2.5.2.2-5.2/Q.737			
SIP selection criteria	13014-(1301 )-311	733/00	<u> </u>					
ISUP selection criteria								
Test purpose	UUS2 explicit essential - implicit rejection To verify that the service can be rejected if no indication is received (no user-to-user indicators parameter) in the first backward message (implicit rejection of service 2) (see note).							
SIP parameter values	180 Ringing: the	encapsu	lated ACM	does	not contain an user-to-user response indicator			
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM UU2 ess)			
				+	180 Ringing(ACM)			
	RELEASE	+		<b>→</b>	CANCEL(REL)			
	REL_COMP	<b>→</b>		+	200 OK CANCEL			
				+	487 Request Terminated			
				<b>→</b>	ACK			
NOTE: The remo	ote network does n	ot under	stand the s	ervice	2 request or the remote user cannot support			

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

TP706201	SIP reference: RF	FC 3261		1	ISUP reference: Q.1912.5 .3.5.2.1.1.2; 1.3.5.2.2-5.1/Q.737				
TSS reference	ISDN-(ISUP)-SIP/SS/UUS3	3							
SIP selection	,								
criteria									
ISUP selection									
criteria									
Test purpose	UUS3 explicit non-essent								
	To verify that the IUT can successfully originate/transit a call with an UUS3 explicit non- essential request, having the <b>user-to-user indicators</b> in the <b>IAM</b> set to "request, not essential"  To verify that the IUT can successfully complete a call with an UUS3 explicit non-essential request, having the Service 3 field in the <b>user-to-user indicators</b> parameter in the <b>ANM</b> or <b>CON</b> set to "service provided"								
SIP parameter values									
ISDN parameter values									
Comments	ISDN		SUT		SIP				
	SETUP(UU3 req not ess)	<b>→</b>		<b>→</b>	INVITE(IAM UU3 not ess)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONN(UU3 ret res)	+		+	200 OK INVITE(ANM UU3 prov)				
	,	Co	mmunicat	ion	· · · · · · · · · · · · · · · · · · ·				
	USER INFO	<b>→</b>		→	INFO(USR)				
				+	200 OK INFO				
	USER INFO	+		+	INFO(USR)				
				→	200 OK INFO				
	DISC	<b>→</b>		<b>→</b>	BYE(REL)				
	RELEASE	+		+	200 OK BYE				
	REL_COM	<b>→</b>							

TP706202	SIP reference: RFC 3261				ISUP reference: Q.1912.5 .3.5.2.1.1.2; 1.3.5.2.2-5.1/Q.737					
TSS reference	ISDN-(ISUP)-SIP/SS/UUS	3								
SIP selection criteria										
ISUP selection criteria										
Test purpose	To verify that the IUT can essential request, having i essential" and the ISDN us to "ISUP required all the w To verify that the IUT can request having in the ANN	UUS3 explicit essential - request and acceptance To verify that the IUT can successfully originate/transit a call with an UUS3 explicit essential request, having in the IAM the user-to-user indicators set to "request, essential" and the ISDN user part preference indicator in the forward call indicators set to "ISUP required all the way" To verify that the IUT can successfully complete a call with an UUS3 explicit essential request having in the ANM or CON the Service 3 field of the user-to-user indicators parameter set to "service provided"								
SIP parameter values	personal control of									
ISDN parameter values										
Comments	ISDN		SUT		SIP					
	SETUP(UU3 reg ess)	<b>→</b>		<b>→</b>	INVITE(IAM UU3 ess)					
	ALERTING	+		+	180 Ringing(ACM)					
	CONN(UU3 ret res)	+		+	200 OK INVITE(ANM UU3 prov)					
	,	Co	ommunicati	on						
	USER INFO	<b>→</b>		<b>→</b>	INFO(USR)					
				+	200 OK INFO					
	USER INFO	+		+	INFO(USR)					
				<b>→</b>	200 OK INFO					
	USER INFO	<b>→</b>		<b>→</b>	INFO(USR)					
				+	200 OK INFO					
	USER INFO	+		+	INFO(USR)					
				<b>→</b>	200 OK INFO					
		Co	ommunicati	on						
	DISC	<b>→</b>		<b>→</b>	BYE(REL)					
	RELEASE	+		+	200 OK BYE					
	REL_COM	<b>→</b>								

TP706203	SIP reference: RFC 3261				ISUP reference: Q.1912.5 1.3.5.2.5.2.2; 1.3.5.2.2-5.2/Q.737					
TSS reference	ISDN-(ISUP)-SIP/SS	S/UUS3								
SIP selection criteria										
ISUP selection criteria										
Test purpose	To verify that the se rejected", #69 "requ	UUS3 explicit essential - explicit rejection To verify that the service can be rejected with a REL having the Cause value #29 "facility rejected", #69 "requested facility not implemented", either with diagnostics (user-to-user indicators name) (see note).								
SIP parameter values										
ISDN parameter values										
Comments	ISDN		SUT		SIP					
	SETUP	+		+	INVITE(IAM UU2 ess)					
	RELEASE	<b>→</b>		<b>→</b>	500 Server Internal error(REL#29, 69)					
	REL_COM	+		+	ACK					
NOTE: The netv	vork or the called user	cannot su	pport the se	ervice.						

TP706204	SIP reference: R	FC 3261		ISUP reference: Q.1912.5					
	1.3.5.2.1.1.2; 1.3.5.2.2-5.1/Q.737								
TSS reference	ISDN-(ISUP)-SIP/SS/UUS	ISDN-(ISUP)-SIP/SS/UUS3							
SIP selection									
criteria									
ISUP selection									
criteria	1,1100								
Test purpose	call	tiai - req	uest and a	acceptano	ce during the active phase of the				
	request, with a FAR having	g the <b>fac</b> i	ility indica	tor paran	an UUS3 explicit non-essential neter set to "user-to-user service"				
					et to "request, not essential".				
					B explicit non-essential request with ser-to-user service" and the				
	Service 3 field in the user-								
SIP parameter	Service 3 field in the user-	to-user i	illulcator 5	paramete	set to service provided				
values									
ISDN parameter									
values									
Comments	ISDN		SUT		SIP				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONN	+		+	200 OK INVITE				
		С	ommunic	ation					
	FAC(UU3 req not ess)	<b>→</b>		<b>→</b>	INFO(FAR req UU3 not ess)				
				+	200 OK INFO				
	FAC(UU3 ret res)	<b>←</b>		<b>←</b>	INFO(FAR resp UU3 prov)				
	USER INFO	<b>→</b>		<b>→</b>	INFO(USR)				
				<b>←</b>	200 OK INFO				
	USER INFO	<b>←</b>		<b>←</b>	INFO(USR)				
				<b>→</b>	200 OK INFO				
	USER INFO	<b>→</b>		<b>→</b>	INFO(USR)				
				+	200 OK INFO				
	USER INFO	+		+	INFO(USR)				
				<b>→</b>	200 OK INFO				
			ommunic	ation					
	DISC	<b>→</b>		<b>→</b>	BYE(REL)				
	RELEASE	+		+	200 OK BYE				
	REL_COM	<b>→</b>							

TP706205	SIP reference: RFC 3261			ISUP reference: Q.1912.5 1.3.5.2.5.2.2/Q.737				
TSS reference	ISDN-(ISUP)-SIP/SS/UUS	S3						
SIP selection								
criteria								
ISUP selection								
criteria	111100		!!-!(!(!					
Test purpose	FRJ)	ntiai - e	xplicit rejecti	on auring	call (service not provided - in			
	To verify that the UUS3 e	To verify that the UUS3 explicit non-essential service can be rejected during the active phase of the call and the Service 3 field in the <b>user-to-user indicators</b> in the <b>FRJ</b> are set						
SIP parameter values								
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE			
			Communica	tion				
	FAC(UU3 req not ess)	<b>→</b>		<b>→</b>	INFO(FAR req UU3 not ess)			
				+	200 OK INFO			
	FAC(UU3 ret err)	+		+	INFO(FRJ resp UU3 not prov)			
				<b>→</b>	200 OK INFO			
			Communica	tion				
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	RELEASE	+		+	200 OK BYE			
	REL_COM	<b>→</b>						

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

## A.1.1.2.7 Closed User Group (CUG)

TP707001	SIP reference: RI	C 3261		ISUP reference:				
					Q.1912.5			
TCC reference	ICDNI (ICLID) CID/CC/CLIC			1.3	5.2.1.1 i) a)/Q.735			
TSS reference	ISDN-(ISUP)-SIP/SS/CUG							
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	CUG without outgoing ac	cess in I	AM					
	To verify that the IUT can s							
					oing access not allowed" in			
	the optional forward call i							
	forward call indicators in	the IAM	should be set to "IS	UP re	quired all the way"			
SIP parameter								
values								
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM, CUG -OA)			
	ALERTING	+		+	180 Ringing			
	CONN	+		+	200 OK INVITE			
				<b>→</b>	ACK			
			Communication					
	DISC	<b>→</b>		<b>→</b>	BYE			
	REL	+		+	200 OK BYE			
	REL_COM	<b>→</b>						

TP707002	SIP reference: RFC 3261				ISUP reference: Q.1912.5 1.5.2.5.1; Table 1-2/Q.735	
TSS reference	ISDN-(ISUP)-SIP/S	S/CUG				
SIP selection criteria						
ISUP selection criteria						
Test purpose	CUG call without outgoing access; class of called user CUG without IA, no ICB activated  To verify that the IUT can successfully establish a CUG call					
SIP parameter values						
ISDN parameter values						
Comments	ISDN		SUT		SIP	
	SETUP	+		+	INVITE(IAM, CUG -OA)	
	ALERTING	→		<b>→</b>	180 Ringing(ACM)	
	CONN	→		<b>→</b>	200 OK INVITE(ANM)	
			Communication	1		
	DISC	+		+	BYE(REL)	
	REL	→		<b>→</b>	200 OK BYE	
	REL_COM	+				

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

TP707003	SIP reference: RFC 3261			1.	ISUP reference: Q.1912.5 5.2.5.1; Table 1-2/Q.735			
TSS reference	ISDN-(ISUP)-SIP/SS	ISDN-(ISUP)-SIP/SS/CUG						
SIP selection criteria								
ISUP selection criteria								
Test purpose	activated	CUG call without outgoing access; class of called user CUG with IA and no ICB activated  To verify that the IUT can successfully establish a CUG call						
SIP parameter values	,							
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	+		+	INVITE(IAM, CUG -OA)			
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)			
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)			
			Communication					
	DISC	+		+	BYE(REL)			
	REL	<b>→</b>		<b>→</b>	200 OK BYE			
	REL_COM	+						

TP707004	SIP reference: RFC 3261			1.	ISUP reference: Q.1912.5 .5.2.5.1; Table 1-2/Q.735	
TSS reference	ISDN-(ISUP)-SIP/S	SS/CUG				
SIP selection criteria						
ISUP selection criteria						
Test purpose	CUG call with outgoing access; class of called user CUG with IA and no ICB activated  To verify that the IUT can successfully establish a CUG call with outgoing access					
SIP parameter values			•		<u> </u>	
ISDN parameter values						
Comments	ISDN		SUT		SIP	
	SETUP	+		+	INVITE(IAM, CUG +OA)	
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)	
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)	
			Communication		` '	
	DISC	+		+	BYE(REL)	
	REL	→		<b>→</b>	200 OK BYE	
	REL_COM	+				

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

TP707006	SIP reference:	: RFC 3261		ISUP reference: Q.1912.5 Q.73	
TSS reference	ISDN-(ISUP)-SIP/SS/CU	JG			
SIP selection criteria					
ISUP selection criteria					
Test purpose	CUG call without outgoing access; class of called user non-CUG To verify that the IUT rejects the CUG call with a 500 Server Internal Error encapsulated the REL cause #87 "User not member of CUG"				
SIP parameter values					
ISDN parameter values					
Comments	ISDN	SUT		SIP	
			+	INVITE(IAM CUG -OA)	
			<b>→</b>	500 Server Internal Error(REL#87)	
			+	ACK	

TP707007	SIP reference: RFC 3261			ISUP reference: Q.1912.5 1.5.2.5.1; Table 1-2/Q.735		
TSS reference	ISDN-(ISUP)-SIP/SS/0	CUG				
SIP selection criteria						
ISUP selection criteria						
Test purpose	CUG call with outgoi To verify that the IUT of					
SIP parameter values						
ISDN parameter values						
Comments	ISDN		SUT		SIP-I	
	SETUP	+		+	INVITE(IAM CUG, +OA)	
	ALERTING	→		<b>→</b>	180 Ringing(ACM)	
	CONN	→		<b>→</b>	200 OK INVITE(ANM)	
				+	ACK	
			Communication	•		
	DISC	+		+	BYE(REL)	
	REL	→		<b>→</b>	200 OK BYE(RLC)	
	REL_COM	+				

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

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

TP707010	SIP reference	e: RFC 3261		ISUP reference: Q.1912.5 1.5.2.5.1; Table 1-2/Q.735			
TSS reference	ISDN-(ISUP)-SIP/SS/C	UG					
SIP selection criteria							
ISUP selection criteria							
Test purpose	CUG call without outgoing access; class of called user other CUG without IA To verify that the IUT rejects the CUG call with a 500 Server Internal Error encapsulated the REL cause #87 "User not member of CUG"						
SIP parameter values							
ISDN parameter values							
Comments	ISDN (CUG -IA)	SUT		SIP-I			
			+	INVITE(IAM CUG -OA)			
			<b>→</b>	500 Server Internal Error(REL#87)			
			+	ACK			

TP707011	SIP reference:	RFC 3261		ISUP reference: Q.1912.5 1.5.2.5.1; Table 1-2/Q.735			
TSS reference	ISDN-(ISUP)-SIP/SS/CU	JG					
SIP selection criteria							
ISUP selection criteria							
Test purpose	CUG call with outgoing access; class of called user other CUG without IA To verify that the IUT rejects the CUG call with a 500 Server Internal Error encapsulated the REL cause #87 "User not member of CUG"						
SIP parameter values							
ISDN parameter values							
Comments	ISDN (CUG -IA)	SUT		SIP-I			
			+	INVITE(IAM CUG +OA)			
			<b>→</b>	500 Server Internal Error(REL#87)			
			+	ACK			

TP707012	SIP reference	e: RFC 3261		ISUP reference: Q.1912.5 1.5.2.5.1; Table 1-2/Q.735		
TSS reference	ISDN-(ISUP)-SIP/SS/C	UG		·		
SIP selection criteria						
ISUP selection criteria						
Test purpose	CUG call without outgoing access; class of called user: other CUG with IA To verify that the IUT rejects the CUG call with a 500 Server Internal Error encapsulated the REL cause #87 "User not member of CUG"					
SIP parameter values						
ISDN parameter values						
Comments	ISDN (CUG A +IA)	SUT		SIP-I		
			+	INVITE(IAM CUG B -OA)		
			<b>→</b>	500 Server Internal Error(REL#87)		
			+	ACK		

TP707013	SIP reference: RFC 3261			1.	ISUP reference: Q.1912.5 .5.2.5.1; Table 1-2/Q.735			
TSS reference	ISDN-(ISUP)-SIP/SS/C	CUG						
SIP selection criteria								
ISUP selection criteria								
Test purpose		CUG call with outgoing access; class of called user other CUG with IA  To verify that the IUT can successfully establish a non-CUG call						
SIP parameter	-		-					
values								
ISDN parameter values								
Comments	ISDN (CUG A +IA)		SUT		SIP-I			
	SETUP	+		+	INVITE(IAM, CUG B +OA)			
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)			
	CONN	→		<b>→</b>	200 OK INVITE(ANM)			
			Communication	•				
	DISC	+		+	BYE(REL)			
	REL	<b>→</b>		<b>→</b>	200 OK BYE			
	REL_COM	+						

## A.1.1.2.8 SUB-addressing (SUB)

TP708001	SIP reference:	RFC 3261			ISUP reference: Q.1912.5 8.5.2.1.1/Q.731			
TSS reference	ISDN-(ISUP)-SIP/SS/SU	В						
SIP selection criteria								
ISUP selection								
criteria								
Test purpose	Sending the called sub-address in the access transport parameter To verify that the IUT can include the called sub-address in the access transport parameter in the IAM							
SIP parameter values	INVITE: IAM encapsulated in the MIME body ATP with called sub-address included							
ISDN parameter values	SETUP: called sub-addr	ess included						
Comments	ISDN		SUT		SIP-I			
	SETUP(SUB)	<b>→</b>		<b>→</b>	INVITE(IAM, ATP(SUB))			
	ALERTING	<del>-</del>		+	180 Ringing			
	CONN	+		+	200 OK INVITE			
				<b>→</b>	ACK			
	Communication							
	DISC	<b>→</b>	•	<b>→</b>	BYE			
	REL	+	•	+	200 OK BYE			
	REL_COM	<b>→</b>	•					

TP708002	SIP reference	e: RFC 3261			SUP reference: Q.1912.5 3.5.2.5.1/Q.731				
TSS reference	ISDN-(ISUP)-SIP/SS/SI	UB							
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	To verify that a call may	Receiving the called sub-address in the access transport parameter  To verify that a call may be successfully established if the IAM contains the sub-address in the access transport parameter and that the called sub-address is passed on to the user network interface.							
SIP parameter values	INVITE: IAM encapsula	ted in the MIM	E body ATP wi	th called	d sub-address included				
ISDN parameter values	SETUP: called sub-add	ress included							
Comments	ISDN		SUT		SIP-I				
	SETUP(SUB)	<del>-</del>		+	INVITE(IAM, ATP(SUB))				
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)				
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)				
		+	ACK						
	DISC	+		+	BYE(REL)				
	REL	<b>→</b>		<b>→</b>	200 OK BYE(RLC)				
	REL_COM	+							

# A.1.1.2.9 Malicious Call Identification (MCID)

TP709001	SIP reference	ce: RFC 3261			ISUP reference: Q.1912.5 7.5.2.1.1/Q.731.7
TSS reference	ISDN-(ISUP)-SIP/SS/	MCID			
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Successful MCID red				
	set to "MCID request"	by sending an I	RS with MCID		g the MCID request indicator se indicator set to "MCID
	included" and the call				
SIP parameter	INFO: The encapsula				
values	INFO: The encapsula Calling party number			oonse in	dicator "included" and the
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)
				+	INFO(IDR requested)
				<b>→</b>	200 OK INFO
				<b>→</b>	INFO(IRS included)
				+	200 OK INFO
	ALERTING	+		+	180 Ringing(ACM)
	CONN	+		+	200 OK INVITE(ANM)
				<b>→</b>	ACK
		(	Communication	)	
	DISC	<b>→</b>		<b>→</b>	BYE(REL)
	REL	+		+	200 OK BYE
	REL_COM	<b>→</b>			

TP709002	SIP reference	SIP reference: RFC 3261			ISUP reference: Q.1912.5 7.5.2.1.1/Q.731.7			
TSS reference	ISDN-(ISUP)-SIP/SS/N	1CID	•					
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Successful MCID requ							
	set to "MCID request" Included and the calli	To verify that the IUT can successfully reply to an IDR having the MCID request indicator set to "MCID request" by sending an IRS with MCID response indicator set to "MCID included" and the calling party number included						
SIP parameter	INFO: The encapsulate							
values	INFO: The encapsulate Calling party number o			onse in	dicator "included" and the			
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	+		+	INVITE(IAM)			
				<b>→</b>	INFO(IDR requested)			
				+	200 OK INFO			
				+	INFO(IRS included)			
				<b>→</b>	200 OK INFO			
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)			
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
		(	Communication	1				
	DISC	+		+	BYE(REL)			
	REL	<b>→</b>		<b>→</b>	200 OK BYE			
	REL_COM	+						

TP709003	SIP reference	ce: RFC 3261			ISUP reference: Q.1912.5 7.5.2.1.1/Q.731.7			
TSS reference	ISDN-(ISUP)-SIP/SS/	MCID						
SIP selection criteria								
ISUP selection criteria								
Test purpose	To verify that the IUT been received. The IU "MCID request" by se and the calling party	Successful MCID request - after ACM To verify that the IUT will accept and reply correctly to an MCID request after ACM has been received. The IUT should reply to an IDR having the MCID request indicator set to "MCID request" by sending an IRS with MCID response indicator set to "MCID included" and the calling party number included (see note).						
SIP parameter values	INFO: The encapsula	INFO: The encapsulated IDR contains the MCID Request indicator "requested" INFO: The encapsulated IRS contains the MCID response indicator "included" and the Calling party number of the originating User						
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	→		<b>→</b>	INVITE(IAM)			
I	ALERTING	<b>←</b>		+	180 Ringing(ACM)			
Ì				+	INFO(IDR requested)			
				<b>→</b>	200 OK INFO			
				<b>→</b>	INFO(IRS included)			
				+	200 OK INFO			
	CONN	<b>+</b>		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
		(	Communication	l				
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	REL	+		+	200 OK BYE			
	REL_COM	<b>→</b>						
NOTE: This situ	ation may occur e.g. if the	ne call has been	forwarded befo	ore reac	hing the destination.			

TP709004					ISUP reference: Q.1912.5 7.5.2.1.1/Q.731.7			
TSS reference	ISDN-(ISUP)-SIP/SS/M	CID						
SIP selection criteria								
ISUP selection criteria								
Test purpose	To verify that the IUT whosen received. The IUT "MCID request" by sendand the calling party n	Successful MCID request - after ACM To verify that the IUT will accept and reply correctly to an MCID request after ACM has been received. The IUT should reply to an IDR having the MCID request indicator set to "MCID request" by sending an IRS with MCID response indicator set to "MCID included" and the calling party number included (see note).						
SIP parameter values	INFO: The encapsulate	INFO: The encapsulated IDR contains the MCID Request indicator "requested" INFO: The encapsulated IRS contains the MCID response indicator "included" and the Calling party number of the originating User						
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	<b>←</b>		+	INVITE(IAM)			
	ALERTING	→		<b>→</b>	180 Ringing(ACM)			
				<b>→</b>	INFO(IDR requested)			
				+	200 OK INFO			
				+	INFO(IRS included)			
				<b>→</b>	200 OK INFO			
	CONN	→		<b>→</b>	200 OK INVITE(ANM)			
				+	ACK			
			Communication	1				
	DISC	+		+	BYE(REL)			
	REL	<b>→</b>		<b>→</b>	200 OK BYE			
	REL_COM	+						
NOTE: This situa	ation may occur e.g. if the	call has beer	forwarded befo	ore reac	hing the destination.			

TP709005	SIP reference: RFC 3261			ISUP reference: Q.1912.5 7.5.2.1.1/Q.731.7			
TSS reference	ISDN-(ISUP)-SIP/SS	S/MCID					
SIP selection criteria							
ISUP selection criteria							
Test purpose	Successful MCID request with calling sub-address To verify that the IUT can successfully reply to an IDR having the MCID request indicator set to "MCID request" by sending an IRS with MCID response indicator set to "MCID included", the calling party number and a calling sub-address in the access transport parameter						
SIP parameter values	INFO: The encapsulated IDR contains the MCID Request indicator "requested" INFO: The encapsulated IRS contains the MCID response indicator "included", the Calling party number and the calling sub-address of the originating User						
ISDN parameter values							
Comments	ISDN		SUT		SIP		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
				+	INFO(IDR requested)		
				<b>→</b>	200 OK INFO		
				<b>→</b>	INFO(IRS included)		
				+	200 OK INFO		
	ALERTING	<b>←</b>		<b>←</b>	180 Ringing(ACM)		
	CONN	+		+	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
			Communication	1			
	DISC	<b>→</b>		<b>→</b>	BYE(REL)		
	REL	+		+	200 OK BYE		
	REL_COM	<b>→</b>					

TP70906	SIP referen	ce: RFC 3261			SUP reference: Q.1912.5 7.5.2.1.2/Q.731.7
TSS reference	ISDN-(ISUP)-SIP/SS/	/MCID			
SIP selection criteria					
ISUP selection criteria	PICS 9/1				
Test purpose	MCID request - MCII To verify that the IUT indicator set to "MCI	rejects a MCID		ding a <b>IF</b>	RS with the MCID response
SIP parameter	INFO: The encapsula	ted IDR contain	s the MCID Red	quest inc	dicator "requested"
values	INFO: The encapsula	ited IRS contain	s the MCID res	ponse in	dicator "not included"
ISDN parameter					
values					
Comments	ISDN		SUT		SIP
	SETUP	<b>→</b>		→	INVITE(IAM)
				+	INFO(IDR requested)
				<b>→</b>	200 OK INFO
				<b>→</b>	INFO(IRS not included)
				+	200 OK INFO
	ALERTING	+		+	180 Ringing(ACM)
	CONN	+		+	200 OK INVITE(ANM)
				<b>→</b>	ACK
			Communication	1	
	DISC	<b>→</b>		<b>→</b>	BYE(REL)
	REL	+		+	200 OK BYE
	REL_COM	<b>→</b>			

TP70907	SIP reference:	RFC 3261		•	SUP reference: Q.1912.5 7.5.2.1.2/Q.731.7
TSS reference	ISDN-(ISUP)-SIP/SS/MC	ID			
SIP selection criteria					
ISUP selection criteria	PICS 9/1				
Test purpose	MCID request - MCID no To verify that the IUT rejoindicator set to "MCID no	ects a MCID	request by send	ding a <b>IF</b>	RS with the MCID response
SIP parameter	INFO: The encapsulated	IDR contain	s the MCID Rec	uest inc	dicator "requested"
values	INFO: The encapsulated	IRS contain	is the MCID resp	onse in	dicator "not included"
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP	+		+	INVITE(IAM)
				<b>→</b>	INFO(IDR requested)
				+	200 OK INFO
				+	INFO(IRS not included)
				<b>→</b>	200 OK INFO
	ALERTING	→		<b>→</b>	180 Ringing(ACM)
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)
				+	ACK
			Communication		
	DISC	<del>-</del>		+	BYE(REL)
	REL	<b>→</b>		<b>→</b>	200 OK BYE
	REL_COM	+			

TP70908	SIP reference: RFC	3261		ISUP reference: Q.1912.5 7.5.2.5.2/Q.731.7
TSS reference	ISDN-(ISUP)-SIP/SS/MCID			
SIP selection criteria				
ISUP selection criteria	PICS 5/9			
Test purpose	MCID timer (T39) expiry To verify that call setup is con expiry, after having sent the II			RS is received within timer T39 tor set to "MCID requested"
SIP parameter values	INFO: The encapsulated IDR			
ISDN parameter values				
Comments	ISDN		SUT	SIP
	SETUP	<b>→</b>	→	INVITE(IAM)
			<b>+</b>	INFO(IDR requested)
			→	200 OK INFO
		T39	expiry	
	ALERTING	+	<b>+</b>	180 Ringing(ACM)
	CONN	+	<b>+</b>	200 OK INVITE(ANM)
			→	ACK
		Comn	nunication	
	DISC	<b>→</b>	→	BYE(REL)
	REL	+	+	200 OK BYE
	REL_COM	<b>→</b>		

## A.1.1.2.10 Conference call (CONF)

TP710001	SIP	reference: RFC	3261	ISUP reference:			
				Q.1912.5			
				Q.734 clause 1.5.2.1.1.2			
TSS reference	ISDN-(ISUP)	-SIP/SS/CONF					
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Verify that the	e IUT can success	sfully begin the	conference from an active call and notify			
		arties correctly					
				on "conference established" is received in			
			call at the end te	erminal and check that all network resources			
	are released						
SIP parameter	INFO: Conte	nt-Type: application	on/ISUP; CPG e	encapsulated in the MIME body			
values							
ISDN parameter		ONF invoke comp					
values	FAC: BeginC	ONF return result	component				
Comments	1		T				
ISDN	SUT	1	SIP 1	SIP 2			
SETUP(CRx)	<b>→</b>	<b>→</b>	INVITE				
ALERTING	+	+	180 Ringing				
CONN	<b>←</b>	<del>(</del>	200 OK INVITI				
FAC(BeginCONF_i		→	INFO(CPG cor	,			
FAC(BeginCONF_r	r) ←	←	200 OK INFO				
		Conference	ce communicat	tion			
DISC(CRx)	<b>→</b>	<b>→</b>	BYE				
RELEASE	← 200 OK BYE						
REL_COMP	<b>→</b>						
		on indicator set to or should be set to		stablished" should be sent by the IUT in the			

TP710002	SIP reference: RFC 3261		ISUP reference: Q.1912.5 Q.734 clause 1.5.2.1.1.2				
TSS reference	ISDN-(ISUF	P)-SIP/SS/CONI	=	Į.			
SIP selection		,					
criteria							
ISUP selection							
criteria							
Test purpose	to add a concept to the concept.	conferee to a conference from conference. No . The conference.	nfer n IS tify e is	ence and notif DN to SIP 1. Es subscriber at SI released by cal	y the implicate the stablished a IP 1 by send I clearing by	ed pa conr ding to the	from idle state and is able arties correctly. nection to SIP 2 and add this nim/her "other party added" served user at IADN (see
SIP parameter	INFO: Cont	ent-Type: applic	atic	n/ISUP; CPG e	ncapsulated	d in t	ne MIME body
values							<u> </u>
ISDN parameter		CONF invoke c					
values		CONF return re					
	DISC: addC	CONF return res	ult c	component			
Comments							
ISDN	SU	IT		SIP 1			SIP 2
SETUP(CRx)	<b>→</b>	-	<del>}</del>	INVITE			
ALERTING	+	•	<b>-</b>	180 Ringing			
CONN	+	•	<b>F</b>	200 OK INVIT			
FAC(BeginCONF_ir		-	<del>}</del>	INFO(CPG cor	nf est)		
FAC(BeginCONF_rr	r) <b>←</b>	•	<b>-</b>	200 OK INFO			
SETUP(CRy)	<b>→</b>					<b>→</b>	INVITE
ALERTING	+					<b>←</b>	180 Ringing
CONN	+					<b>←</b>	200 OK INVITE
FAC(AddCONF_inv						<b>→</b>	INFO(CPG conf est)
DISC(AddCONF_rr,						<b>←</b>	200 OK INFO
RELEASE	<b>→</b>		<b>→</b>	INFO(CPG par	rty add)		
REL_COMP	+	•	<del>-</del>	200 OK INFO			
		Confer	enc	e communicat	ion		
DISC(CRx)	<b>→</b>	-	<del>}</del>	BYE			
RELEASE	+	•	<b>F</b>	200 OK BYE			
REL_COMP	<b>→</b>						
						<b>→</b>	BYE
						+	200 OK BYE
new affect	ted conferee		c n	otification indi	cator set to	"oth	d be sent by the IUT to the er party added" to the non-ress".

TP710003	SIP reference: RFC 3261			3261	0		UP reference: Q.1912.5
TSS reference	ISDN (ISLID	)-SIP/SS/CON	ıE		Q.	134	clause 1.5.2.1.1.3
SIP selection	13014-(1304	)-31P/33/CON	NF				
criteria							
ISUP selection criteria							
Test purpose	Verify that the implied parti		ccess	sfully isolate a c	onferee from	the	conference and notify the
			m IS	DN to SIP 1. Ad	dd SIP 2 to th	ne co	onference and notify
							he CPG. Isolate a conferee
							6. Reattach the conferee.
CID manage et au							at ISDN (see note).
SIP parameter values	INFO: Conte	nt-Type: appl	icatio	on/ISUP; CPG e	ncapsulated	ın ti	ne MIME body
ISDN parameter	EAC: Bogin	ONE invoko	oomr	onont			
values		CONF invoke CONF return r					
- 31400		NF invoke co					
		ONF return re					
	FAC: isolate	CONF invoke	com	ponent			
		CONF return					
		hCONF invok					
	FAC: reattac	hCONF retur	n res	ult component			
Comments	0.11	<u>-</u>		loip 4			loup o
ISDN CETUD(CD::)	SUT	1	_	SIP 1			SIP 2
SETUP(CRx)	<b>→</b>	+	<b>→</b>	INVITE			
ALERTING CONN	<del>-</del>	+	<del>-</del>	180 Ringing 200 OK INVITI	_		
FAC( <b>BeginCONF</b> _ir		+	<b>→</b>	INFO(CPG cor			
FAC(BeginCONF_ri			<del>/</del>	200 OK INFO	ii est)		
TAC(Beginoon _n	1)			200 OK IIVI O			
SETUP(CRy)	<b>→</b>					<b>→</b>	INVITE
ALERTING	<del>-</del>					<u>-</u>	180 Ringing
CONN	<del>(</del>					<del>-</del>	200 OK INVITE
FAC( <b>AddCONF</b> _inv						<del>}</del>	INFO(CPG conf est)
DISC(AddCONF_rr,					•	<del>(</del>	200 OK INFO
RELEASE	→		<b>→</b>	INFO(CPG par	rty add)		
REL_COMP	<del>(</del>		<del>(</del>	200 OK INFO			
		Confe	erenc	e communicat	ion		•
FAC(IsolConf_inv,C					•	<del>)</del>	INFO(CPG isol)
FAC(IsolConf_rr,CF	₹x) <b>←</b>					<del>(</del>	200 OK INFO
		1	<b>→</b>	INFO(CPG par	rty isol)		
		<u></u>	<b>←</b>	200 OK INFO	1.1. 61= :		
E40/B	00 ) 15	Private con	mur	ication ISDN v			III I I I I I I I I I I I I I I I I I
FAC(ReattConf_inv		1				<del>)</del>	INFO(CPG reatt)
FAC(ReattConf_rr,C	CRx)	1	<b>—</b>	INIEO/ODO		<del>(</del>	200 OK INFO
	<del></del>	+	<b>→</b>	INFO(CPG par	rty reatt)		
		Confe	← rene	200 OK INFO	ion		
DISC(CBv)	احا	Conte		1	IUII		
DISC(CRx) RELEASE	<b>→</b>	+	<b>→</b>	BYE 200 OK BYE			
REL_COMP	<b>→</b>	+	_	ZUU UN DIE			
IVEE_OOINIE	7	1		1		<b>→</b>	BYE
						<del>7</del>	200 OK BYE
NOTE: The gene	ric notification	on indicator	set to	ı "isolated" withi			should be sent by the IUT
to the affe	ected confered	e and the <b>gen</b>	eric	notification inc	dicator set to	ot "ot	her party isolated" should be set to "progress".
							the conference.

TP710004	SIP reference: RFC 3261			IS	SUP reference: Q.1912.5		
					Q.734 clause 1.5.2.1.1.5		
TSS reference:	ISDN-(ISUP)	-SIP/SS/CONF					
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	of the conference	ees and notify th	e implied parties	correctly (see n	een the served user and onotes).		
SIP parameter	INFO: Conte	nt-Type: applicat	tion/ISUP; CPG	encapsulated in	the MIME body		
values							
ISDN parameter		ONF invoke con					
values		ONF return resu					
		NF invoke comp					
		NF return resul					
		CONF invoke co					
	CONNECT: 9	plitCONF return	result compone	nt			
Comments	Ta		loip 4	т	loin e		
ISDN	SU		SIP 1		SIP 2		
SETUP(CRx)	<b>→</b>	<b>→</b>	INVITE				
ALERTING	<del>(</del>	<b>←</b>	180 Ringing				
CONN	<b>←</b>	<b>←</b>	200 OK INVIT				
FAC( <b>BeginCONF</b> _i		→	INFO(CPG co	nf est)			
FAC( <b>BeginCONF</b> _I	rr) <del>(</del>	<b>←</b>	200 OK INFO				
SETUP(CRy)	→			→	INVITE		
ALERTING	<b>←</b>			<del>(</del>	180 Ringing		
CONN	<b>←</b>			+	200 OK INVITE		
FAC( <b>AddCONF</b> _inv				<b>→</b>	INFO(CPG conf est)		
DISC(AddCONF_rr				<del>(</del>	200 OK INFO		
RELEASE	→	→	INFO(CPG pa	rty add)			
REL_COMP	+	←	200 OK INFO				
		Confere	nce communica	tion			
SETUP( <b>SplitConf_</b>				<b>→</b>	INFO(CPG conf disc)		
CONN(SplitIConf_	rr,CRy)			+	200 OK INFO		
		→	INFO(CPG pa	rty split)			
		+	200 OK INFO				
		Private commi	unication ISDN v	with SIP 1			
FAC( <b>AddCONF</b> _inv	v,CRy <b>)</b> →			<b>→</b>	INFO(CPG conf est)		
DISC(AddCONF_rr	,CRy <b>) ←</b>			+	200 OK INFO		
RELEASE	<b>→</b>	→	INFO(CPG pa	rty add)			
REL_COMP	+	<del>(</del>	200 OK INFO				
	•	Confere	nce communica	tion	•		
DISC(CRx)	<b>→</b>	<b>→</b>	BYE				
RELEASE	+	<b>+</b>	200 OK BYE				
REL_COMP	<b>→</b>		_				
				<b>→</b>	BYE		

The **generic notification indicator** set to "conference disconnected" should be sent by the IUT to the affected conferee and the **generic notification indicator** set to "other party split" should be sent to the non-affected conferees. The event indicator in the **CPG** should be set to "progress". The nonaffected conferees should not be able to participate in the communication of the private communication.

NOTE 2: See also figure 1-5/Q.734.

TP710005	SIP refe	erence: RFC	3261	IS	SUP reference: Q.1912.5
				Q.734	4 clause 1.5.2.1.1.6
TSS reference	ISDN-(ISUP)-SIF	P/SS/CONF			
SIP selection	, ,				
criteria					
ISUP selection					
criteria					
Test purpose	requested by the Establish a confe subscriber at SIF	served user, erence from IS 1 by sending SIP 2. The c	, and notify the ir SDN to SIP 1. Ac g him/her "other	nplied parties co dd SIP 2 to the o party added" in	n the conference, if breatly. conference and notify the CPG. Release the aring by the served user at
SIP parameter	INFO: Content-T		ion/ISUP; CPG e	encapsulated in	the MIME body
values			· 	<u>.                                    </u>	<u>-</u>
ISDN parameter	FAC: BeginCON				
values	FAC: BeginCON				
	FAC: addCONF				
	DISC: addCONF	return result	component		
	FAC: dropCONF				
_	FAC: dropCONF	return result	component		
Comments			1		T
ISDN	SUT	•	SIP 1		SIP 2
SETUP(CRx)	<b>→</b>	→	INVITE		
ALERTING	+	←	180 Ringing		
CONN	+	<del>-</del>	200 OK INVIT		
FAC( <b>BeginCONF</b> _		→	INFO(CPG co	nf est)	
FAC( <b>BeginCONF</b> _	rr) <b>←</b>	<del>(</del>	200 OK INFO		
SETUP(CRy)	<b>→</b>			<b>→</b>	INVITE
ALERTING	+			<del>(</del>	180 Ringing
CONN	+			<del>(</del>	200 OK INVITE
FAC(AddCONF_in	v,CRy <b>)</b> →			<b>→</b>	INFO(CPG conf est)
DISC(AddCONF_r				<del>(</del>	200 OK INFO
RELEASE	<b>→</b>	→	INFO(CPG pa	rty add)	
REL_COMP	+	←	200 OK INFO		
		Conferen	ce communica	ion	
FAC( <b>DropCONF</b> _ir				<b>→</b>	BYE
FAC(DropCONF_r	r,CRx <b>) ←</b>			+	200 OK BYE
		<b>→</b>	INFO(CPG pa	rty disc)	
		+	200 OK INFO		
		Coi	mmunication		
DISC(CRx)	<b>→</b>	→	BYE		
RELEASE	+	<del>(</del>	200 OK BYE		

NOTE: The IUT should release the leg towards the conferee according to normal call release procedures, i.e. send a REL to a conferee connected to the conference. The **generic notification indicator** set to "other party disconnected" should be sent to the non-affected conferees. The event indicator in the CPG should be set to "progress".

TP710006	SIP refere	ence: RFC	3261	IS	GUP reference: Q.1912.5
				Q.734	4 clause 1.5.2.1.1.7
TSS reference	ISDN-(ISUP)-SIP/S	S/CONF			
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	requested by the of Establish a confere subscriber at SIP 1	conferee, a nce from IS by sending	and notify the in SDN to SIP 1. Acg g him/her "other	mplied parties odd SIP 2 to the operty added" in	from the conference, if correctly conference and notify the CPG. Release request learing by the served user at
SIP parameter	INFO: Content-Typ	e: applicati	ion/ISUP: CPG e	encapsulated in	the MIME body
values	G. Gomon Typ	J. 4PP.1041		apoulatou III	2003
ISDN parameter	FAC: BeginCONF i	nvoke com	ponent		
values	FAC: BeginCONF r	eturn resu	It component		
	FAC: addCONF inv	oke compo	onent .		
	DISC: addCONF re	turn result	component		
	FAC: partyDISC inv				
	FAC: partyDISC ref	turn result	component		
Comments					
ISDN	SUT		SIP 1		SIP 2
SETUP(CRx)	<b>→</b>	→	INVITE		
ALERTING	+	+	180 Ringing		
CONN	<del>(</del>	+	200 OK INVIT		
FAC( <b>BeginCONF</b> _		→	INFO(CPG co	nf est)	
FAC( <b>BeginCONF</b> _	rr) <del>(</del>	+	200 OK INFO		
SETUP(CRy)	→			→	INVITE
ALERTING	+			+	180 Ringing
CONN	+			+	200 OK INVITE
FAC(AddCONF_in				→	INFO(CPG conf est)
DISC(AddCONF_r	r,CRy <b>) ←</b>			+	200 OK INFO
RELEASE	<b>→</b>	<b>→</b>	INFO(CPG pa	rty add)	
REL_COMP	+	+	200 OK INFO		
		Conferen	ce communica	tion	•
FAC( <b>PartyDisc_</b> inv		+	BYE		
FAC(PartyDisc_rr,	CRy) →	→	200 OK BYE		
<u>-</u>				<b>→</b>	INFO(CPG party disc)
				<b>←</b>	200 OK INFO
		Coi	mmunication	•	
DISC(CRx)	<b>→</b>			<b>→</b>	BYE
RELEASE	+			<del>(</del>	200 OK BYE
REL_COMP	→				

NOTE: The IUT should release the leg towards the conferee according to normal call release procedures, i.e. send a RLC in response to the REL to a conferee connected to the conference through ISUP. The generic notification indicator set to "other party disconnected" should be sent to the non-affected conferees. The event indicator in the CPG should be set to "progress".

TP710007	SIP	SIP reference: RFC 3261		3261		IS	UP reference: Q.1912.5
					a	.734	clause 1.5.2.1.1.8
TSS reference	ISDN-(ISUP)	)-SIP/SS/CONF	F				0.0000 110.211110
SIP selection		,					
criteria							
ISUP selection							
criteria							
Test purpose							m the conference, if
		y the served us	er,	and initiate the n	ormal call i	relea	se procedure towards each
	conferee	, ,		DNI OID 4 A I	101001		
							onference and notify
				e served user at			he CPG. The conference is
SIP parameter	INFO: Conta	ent-Type: applic	ratio	on/ISUP; CPG er	ncansulate	d in t	he MIME body
values	. Conte	ant Type, applic	Jan	, OI G 61	ισαρουιαισι	וווג	no mini body
ISDN parameter	FAC: Begin(	CONF invoke c	omi	ponent			
values		CONF return re					
	FAC: addCC	NF invoke con	npo	nent			
	DISC: addC0	ONF return res	ult	component			
Comments							
ISDN	SI			SIP 1			SIP 2
SETUP(CRx)	<b>→</b>		<del>}</del>	INVITE			
ALERTING	+		<b>(</b>	180 Ringing			
CONN	+		<u> </u>	200 OK INVITE			
FAC( <b>BeginCONF</b> _in			<del>}</del>	INFO(CPG con	f est)		
FAC(BeginCONF_rr	·) ←	•	<u> </u>	200 OK INFO			
SETUP(CRy)	<b>→</b>					<b>→</b>	INVITE
ALERTING	+					<del>(</del>	180 Ringing
CONN	<b>+</b>					<del>(</del>	200 OK INVITE
FAC(AddCONF_inv,						<b>→</b>	INFO(CPG conf est)
DISC(AddCONF_rr,						<del>(</del>	200 OK INFO
RELEASE	<b>→</b>		<u> </u>	INFO(CPG par	ty add)		
REL_COMP	<b>←</b>		<u> </u>	200 OK INFO			
DIOC(OD.)	1 -			ce communicati	on		T
DISC(CRx)	<b>→</b>		<u>≯</u>	BYE			
RELEASE	<del>+</del>		<u> </u>	200 OK BYE		<b>L</b>	INICO(CDC months disa)
REL_COMP	<b>→</b>					<b>→</b>	INFO(CPG party disc)
		+				<del>-</del>	200 OK INFO
						<b>→</b>	BYE
NOTE: The IUT s	hould sond P	El to all confo	roc	 s connected to th	ao oonforan	_	200 OK BYE
NOTE. THE IUTS	noula sena <b>R</b>	LEL TO All CONTE	iee	s connected to tr	ie conierer	ice.	

TP710008	SIP r	eference: RFC	3261	IS	UP reference: Q.1912.5
				0.7	34 clause 1.6.15
TSS reference	ISDN-(ISUP)-S	SIP/SS/CONF		Q.7	04 Clau3C 1.0.10
SIP selection	10011 (1001 )	311 7007001 <b>1</b> 1			
criteria					
ISUP selection					
criteria					
Test purpose	Verify that no r	etrieve notificat	tion is sent to a u	user put on hold	and subsequently added to
	a conference of	all, but that the	IUT sends the "	conference esta	olished" notification to the
	held user				
SIP parameter	INFO: Content	-Type: applicati	ion/ISUP; CPG	encapsulated in t	he MIME body
values					
ISDN parameter		NF invoke com			
values		NF return resu			
		F invoke compo			
Comments	וטואט: addCOI	NF return result	component		
Comments	CUT	•	CID 4		CID 2
ISDN CETUD(CD:x)	SUT →		SIP 1 INVITE		SIP 2
SETUP(CRx) ALERTING	<del>-</del>	<b>&gt;</b>			
CONN	<del>-</del>	<del>+</del>	180 Ringing 200 OK INVIT	_	
FAC( <b>BeginCONF</b> _ir		<b>→</b>			
			INFO(CPG co	nr est)	
FAC(BeginCONF_rr	r) <del>-</del>		200 OK INFO		
CETUD/CD:/	_				INIVITE
SETUP(CRy) ALERTING	<b>→</b>			<b>→</b>	INVITE 180 Ringing
CONN	<del>-</del>			<del>-</del>	200 OK INVITE
CONN					200 OK INVITE
HOLD	<b>→</b>			<b>→</b>	INFO(CPG hold)
ПОСО	7			<del></del>	200 OK INFO
					200 OK INFO
FAC( <b>AddCONF</b> _inv	,CRy) →	+ +		<b>→</b>	INFO(CPG conf est)
DISC(AddCONF_III)		+ +	1	<del>-</del>	200 OK INFO
RELEASE	CRy) <b>←</b>	<b>&gt;</b>	INFO(CPG pa	_	ZOU OK INFO
REL_COMP	<del> </del>		200 OK INFO	rty auu)	
NEL_CONF		-	ce communica	tion	
DISC(CRx)	<b>→</b>	Conferen	BYE		1
RELEASE	<del></del>	<b>+</b>	200 OK BYE		
REL_COMP	<b>→</b>		200 ON DIE	<b>→</b>	INFO(CPG party disc)
INLL_CONIF				<del></del>	200 OK INFO
				<b>→</b>	BYE
				<del>-</del>	200 OK BYE
				7	ZUU UN BTE

TP710009	SIP refer	SIP reference: RFC 3261			GUP reference: Q.1912.5 734 clause 1.6.15	
TSS reference	ISDN-(ISUP)-SIP/S	SS/CONE		Q.734 Clause 1.0.13		
SIP selection	10014-(1001 )-011 /0	30/00141				
criteria						
ISUP selection						
criteria						
Test purpose	To verify that no ho	old and no r	etrieve notificati	on is sent to the	conferees when the	
	conference control					
SIP parameter	INFO: Content-Typ	e: applicati	on/ISUP; CPG e	encapsulated in	the MIME body	
values	,,			•	•	
ISDN parameter	FAC: BeginCONF					
values	FAC: BeginCONF					
	FAC: addCONF in					
_	DISC: addCONF re	eturn result	component			
Comments			_			
ISDN	SUT		SIP 1		SIP 2	
SETUP(CRx)	<b>→</b>	<b>→</b>	INVITE			
ALERTING	+	<b>←</b>	180 Ringing			
CONN	+	<b>←</b>	200 OK INVIT			
FAC(BeginCONF_		→	INFO(CPG co	nf est)		
FAC(BeginCONF_	rr) <del>(</del>	<del>(</del>	200 OK INFO			
SETUP(CRy)	<b>→</b>			<b>→</b>	INVITE	
ALERTING	<del>-</del>			<del>-</del>	180 Ringing	
CONN	<del>-</del>			<del>-</del>	200 OK INVITE	
FAC( <b>AddCONF</b> _in				<b>→</b>	INFO(CPG conf est)	
DISC(AddCONF_r	, ,,			<del>-</del>	200 OK INFO	
RELEASE	→ ·	<b>→</b>	INFO(CPG pa	rty add)	200 011 1111 0	
REL_COMP	<del>-</del>	<del>-</del>	200 OK INFO	ity ddd,		
IXEL_OOM			ce communicat	ion		
HOLD	<b>→</b>	220.311				
RETRIVE	<b>→</b>					
DISC(CRx)	<b>→</b>	<b>→</b>	BYE			
RELEASE	+	+	200 OK BYE			
REL_COMP	<b>→</b>			→	INFO(CPG party disc)	
				+	200 OK INFO	
				<b>→</b>	BYE	
				+	200 OK BYE	

# A.1.1.2.11 Explicit Call Transfer (ECT)

TP711001	SIP referer	nce: F	RFC 3261	l		Q.	reference: 1912.5					
T00 (	1001 (10110) 010 (00	· /= 0.T	-		7.5	.2.1.1	.1 a)/Q.732.7					
TSS reference	ISDN-(ISUP)-SIP/SS	S/ECT										
SIP selection												
criteria ISUP selection												
criteria												
Test purpose	Canability of starin	~ ~ ~	l aandin	4L	ne additional calling p		number in the cell					
rest purpose	transfer number	g and	senain	gu	ie additional calling p	arty i	number in the call					
	transfer number  To verify that the ILIT is able to store the additional calling party number in the generic											
	To verify that the IUT is able to store the additional calling party number in the <b>generic</b>											
	<b>number</b> when the <b>calling party number</b> and the <b>generic number</b> have been received from the remote user. This information is sent by the IUT to the other remote user in the											
	call transfer number in either the FAC or CPG when the call transfer is activated											
SIP parameter					e additional calling par							
values					d CPG generic notifica							
	INFO C: encapsulate	ed FA	C contair	ns c	eneric notification call	transf	er active, call transfer					
	number deriv	number derived from the additional calling party number of user B (SIP-I 1)										
ISDN parameter		AC: ECT invoke request component										
values		DISCONNECT: ECT invoke return result component										
Comments	ISDN 2		SUT		SIP-I 1		SIP-I 3					
	SETUP	+		<b>←</b>	INVITE(IAM)							
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)							
	CONN	<b>→</b>			200 OK INVITE(ANM)	)						
				<b>←</b>	ACK							
	HOLD	<b>→</b>		<b>→</b>	INVITE(CPG hold)							
	_				200 OK INVITE							
				<b>→</b>	ACK							
					-							
	SETUP	<b>→</b>				→	INVITE(IAM)					
	ALERTING	+					180 Ringing(ACM)					
	CONN	+					200 OK INVITE(ANM)					
							ACK					
	FAC(ECT invoke)	→										
	DISCONNECT(rr)	+		<b>→</b>	INFO (FAC ect active	)						
	RELEASE	→			200 OK INFO	,						
	RELEASE COMPL	+				<b>→</b>	INFO (FAC ect active)					
							200 OK INFO					
					ı		1					
					BYE(REI	)	BYE(REL)					
		+					200 OK BYE(RLC					

TP711002	SIP referen	ce: F	RFC 326	1			reference: .1912.5				
						-	1.1 a)/Q.732.7				
TSS reference	ISDN-(ISUP)-SIP/SS	/ECT	-			7.0.2.1.1	1.1 ajr 4.1 02.1				
SIP selection	10211 (1001 ) 011 700	,									
criteria											
ISUP selection											
criteria											
Test purpose	Capability of storing and sending the calling party number in the call transfer number  To verify that the IUT is able to store the calling party number when only this CLI has been received from the remote user. This information is sent by the IUT to the other										
	remote user in the ca	all tra	ınsfer nı	ımk	er in either the FA	C or ĆPG	when the call transfer is				
SIP parameter values	INVITE B SDP sendo INFO C: encapsulate number derive	NVITE: encapsulated IAM contains the calling party number of user B NVITE B SDP sendonly, encapsulated CPG generic notification remote hold NFO C: encapsulated FAC contains generic notification call transfer active, call transfer number derived from the calling party number of user B (SIP-I 1)									
ISDN parameter values		FAC: ECT invoke request component DISCONNECT: ECT invoke return result component									
Comments	ISDN 2		SUT		SIP-I 1		SIP-I 3				
	SETUP	+		<b>←</b>	INVITE(IAM)						
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)	)					
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(AI	VM)					
				+	ACK						
	HOLD	<b>→</b>		<b>→</b>	INVITE(CPG hold)	)					
					200 OK INVITE						
				<b>→</b>	ACK						
	SETUP	<b>→</b>				<b>→</b>	INVITE(IAM)				
	ALERTING	+				+	180 Ringing(ACM)				
	CONN	+				<b>←</b>	200 OK INVITE(ANM)				
						<b>→</b>	ACK				
	FAC(ECT invoke)	<b>→</b>									
	DISCONNECT(rr)	+			INFO (FAC ect act	tive)					
	RELEASE	<b>→</b>		<b>←</b>	200 OK INFO						
	RELEASE COMPL	+					INFO (FAC ect active)				
						+	200 OK INFO				
					RYF(	REL) ⋺	BYE(REL)				
		+					200 OK BYE(RLC				

TP711003	SIP referen	SIP reference: RFC 3261						ISUP reference: Q.1912.5				
						752	-	.1 b)/Q.732.7				
TSS reference	ISDN-(ISUP)-SIP/SS	/FCT	-			7.0.2.	•••	. 1 bjr Q. 1 02.1				
SIP selection	10211 (1001 ) 011 700	,										
criteria												
ISUP selection												
criteria												
Test purpose	Capability of storing	g and	d sendin	g th	ne additiona	I connected	nu	mber in the call				
	transfer number											
	To verify that the IUT											
								ave been received from				
	the remote user. This											
SIP parameter	transfer number in e											
values	INVITE B SDP sendo 200 OK INVITE: enc											
values												
		NFO B: encapsulated FAC contains generic notification call transfer active, call transfer number derived from the additional connected of user C (SIP-I 2)										
ISDN parameter		AC: ECT invoke request component										
values		DISCONNECT: ECT invoke return result component										
Comments	ISDN 2		SUT		SIP-I 1	-		SIP-I 3				
	SETUP	+		<b>←</b>	INVITE(IAN	1)						
	ALERTING	<b>→</b>			180 Ringing							
	CONN	<b>→</b>			200 OK INV							
				<b>←</b>	ACK	,						
	HOLD	<b>→</b>		<b>→</b>	INVITE(CP	G hold)						
					200 OK INV							
				<b>→</b>	ACK							
	SETUP	<b>→</b>					<b>→</b>	INVITE(IAM)				
	ALERTING	+					+	180 Ringing(ACM)				
	CONN	+					+	200 OK INVITE(ANM)				
							<b>^</b>	ACK				
	FAC(ECT invoke)	<b>→</b>										
	DISCONNECT(rr)	+			INFO (FAC							
	RELEASE	<b>→</b>		<b>←</b>	200 OK INF	0						
	RELEASE COMPL	+						INFO (FAC ect active)				
							<b>←</b>	200 OK INFO				
						BYE(REL)						
					200 OI	K BYE(RLC	<b>←</b>	200 OK BYE(RLC				

TP711004	SIP referen	ce: F	RFC 3261				Q.	reference: 1912.5 I.1 b)/Q.732.	
TSS reference	ISDN-(ISUP)-SIP/SS	ISDN-(ISUP)-SIP/SS/ECT							
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	To verify that the IUT	Capability of storing and sending the connected number in call transfer number To verify that the IUT is able to store connected number when only this COL has been received from the remote user. This information is sent by the IUT to the other remote use							
								Ill transfer is activated	
SIP parameter values	INVITE B SDP sendo 200 OK INVITE: enco	NVITE B SDP sendonly, encapsulated CPG generic notification remote hold 200 OK INVITE: encapsulated ANM containing the connected number NFO B: encapsulated FAC contains generic notification call transfer active, call transfer							
	number derived from								
ISDN parameter		FAC: ECT invoke request component							
values	DISCONNECT: ECT	DISCONNECT: ECT invoke return result component							
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3	
	SETUP	<b>←</b>		<b>+</b>	INVITE	(IAM)			
	ALERTING	<b>→</b>				nging(ACM)			
	CONN	<b>→</b>		<b>→</b>	200 Ok	(INVITE(ANM)			
				+	ACK				
	HOLD	<b>→</b>		<b>→</b>	INVITE	(CPG hold)			
				+	200 OK	INVITE			
				<b>→</b>	ACK				
	SETUP	<b>→</b>					<b>→</b>	INVITE(IAM)	
	ALERTING	+					+	180 Ringing(ACM)	
	CONN	+					<b>←</b>	200 OK INVITE(ANM)	
							<b>→</b>	ACK	
	FAC(ECT invoke)	<b>→</b>							
	DISCONNECT(rr)	+		<b>→</b>	INFO (I	FAC ect active)			
	RELEASE	<b>→</b>			200 Ok				
	RELEASE COMPL	+					<b>→</b>	INFO (FAC ect active)	
								200 OK INFO	
					•			•	
						BYE(REL)	<b>→</b>	BYE(REL)	
					20	00 OK BYE(RLC	<b>←</b>	200 OK BYE(RLC	

TP711005	SIP referen	SIP reference: RFC 3261				-	reference:			
					75.	Q.1912.5 7.5.2.1.1.2.1/Q.732.7				
TSS reference	ISDN-(ISUP)-SIP/SS/ECT									
SIP selection	10014 (1001 ) 011 700	/								
criteria										
ISUP selection										
criteria										
Test purpose	To verify that the local prevention procedure with call transfer ref To verify that the local transfer if a LOP with loop exists", and the	Loop prevention procedure - initiation and successful response  To verify that the local exchange controlling the ECT can successfully initiate the loop prevention procedure by sending LOP with loop prevention indicator set to "request" and with call transfer reference for both calls  To verify that the local exchange controlling the ECT can successfully perform a call transfer if a LOP with loop prevention indicator set to "response" is received and "no loop exists", and the call identity matches the one used by the IUT								
SIP parameter	INFO: encapsulated									
values		LOP	response,	ca	III transfer reference, res	pon	se indicator: "no loop			
ICDN parameter	exists"									
ISDN parameter values										
Comments	ISDN 2		SUT		SIP-I 1		SIP-I 3			
	SETUP	+			INVITE(IAM)		0 10			
	ALERTING	<b>→</b>		_	180 Ringing(ACM)					
	CONN	<b>→</b>			200 OK INVITE(ANM)					
					ACK					
					-					
	HOLD	<b>→</b>	-3	•	INVITE(CPG hold)					
					200 OK INVITE					
			-3	•	ACK					
	SETUP	<b>→</b>				<b>→</b>	INVITE(IAM)			
	ALERTING	+					180 Ringing(ACM)			
	CONN	1				+	200 OK INVITE(ANM)			
						<b>→</b>	ACK			
			-3	•	INFO(LOP request)					
			€	•	200 OK INFO					
							INFO(LOP request)			
						+	200 OK INFO			
				_						
					INFO(LOP response)	-				
			7	•	200 OK INFO					
				-		Z	INFO(LOD response)			
				_			INFO(LOP response) 200 OK INFO			
						7	ZOU OK IINFU			
	FAC(ECT invoke)	<b>→</b>				+				
	DISCONNECT(rr)	+			INFO (FAC ect active)	1				
	RELEASE	<b>→</b>			200 OK INFO	1				
	RELEASE COMPL	+				<b>→</b>	INFO (FAC ect active)			
		-		1			200 OK INFO			
		-1	L				<u> </u>			
					BYE(REL)	<b>→</b>	BYE(REL)			
					200 OK BYE(RLC		200 OK BYE(RLC			

TP711006	SIP referen	RFC 3261	I			Q.	reference: 1912.5 2.2 a)/Q.732.7				
TSS reference	ISDN-(ISUP)-SIP/SS	ISDN-(ISUP)-SIP/SS/ECT									
SIP selection criteria											
ISUP selection criteria											
Test purpose	To verify that the local by sending <b>FAC</b> with	Facility message with generic notification sent to the remote user  To verify that the local exchange controlling the ECT can successfully initiate a call transfor sending FAC with the generic notification set to "call transfer, active" or "call transfer laterting" and the service activation parameter set to "call transfer"									
SIP parameter	INFO B: encapsulate										
values	INFO C: encapsulate										
ISDN parameter values	FAC: ECT invoke red	FAC: ECT invoke request component DISCONNECT: ECT invoke return result component									
Comments	ISDN 2 SUT SIP-I 1 SIP-I 3										
	SETUP	+		+	INVITE	(IAM)					
	ALERTING	<b>→</b>				iging(ACM)					
	CONN	→ 200 OK INVITE(ANM)									
		← ACK									
	HOLD	<b>→</b>		<b>→</b>	INVITE	(CPG hold)					
						INVITE					
				→	ACK						
	SETUP	<b>→</b>					4	INVITE(IAM)			
	ALERTING	<del>-</del>						180 Ringing(ACM)			
	CONN	÷									
		→ ACK									
	FAC(ECT invoke)	FAC(ECT invoke)									
	DISCONNECT(rr)	+		<b>→</b>	INFO (F	AC ect active)					
	RELEASE	<b>→</b>			200 OK						
	RELEASE COMPL	<del>-</del>					<b>→</b>	INFO (FAC ect active)			
							<b>←</b>	200 OK INFO			
								T			
				<u> </u>		BYE(REL)		BYE(REL)			
					20	0 OK BYE(RLC	<b>←</b>	200 OK BYE(RLC			

TP711007	SIP reference	ce: F	RFC 326	1		ISI	_	reference:		
						Q.1912.5 7.5.2.1.1.2.2 a)/Q.732.7				
TSS reference	ISDN-(ISUP)-SIP/SS/ECT									
SIP selection	( = = , =									
criteria										
ISUP selection										
criteria										
Test purpose		Call progress message with generic notification sent to the remote user To verify that the local exchange (controlling the ECT) can successfully initiate a call								
	transfer by sending C	transfer by sending CPG with the generic notification set to "call transfer, active" ar service activation parameter set to "call transfer"								
OID								, ,		
SIP parameter values	INFO C: encapsulated INFO B: encapsulated	J CF	G contai	ns (	generic i	notification call tra	ansi	er active		
values	INFO B. encapsulated									
ISDN parameter	FAC: ECT invoke requ				CHEIR III	ouncation can trai	1310	i dolly6		
values	DISCONNECT: ECT				ult com	oonent				
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3		
	SETUP	+		<b>←</b>	INVITE	(IAM)				
	ALERTING	<b>→</b>		<b>→</b>	180 Rir	nging(ACM)				
	CONN	<b>→</b>		<b>→</b>	200 OK	(INVITE(ANM)				
				+	ACK					
	HOLD	<b>→</b>				(CPG hold)				
						INVITE				
				<b>→</b>	ACK					
	SETUP	<b>→</b>					_	INVITE(IAM)		
	ALERTING	<del>/</del>						180 Ringing(ACM)		
	ALLINING							100 Kinging(ACIVI)		
	FAC(ECT invoke)	<b>→</b>								
	DISCONNECT(rr)	<b>←</b>				FAC ect alert)				
	RELEASE	<b>→</b>		<b>←</b>	200 OK	INFO				
	RELEASE COMPL	+						INFO (CPG ect active)		
							+	200 OK INFO		
							-	200 OK INVITE(ANM)		
								ACK		
				<b>→</b>	INFO (	AC ect active)	-	/ CIC		
		1			200 OK					
		1				BYE(REL)	_	DVE(DEL)		
		1			20			200 OK BYE(RLC		
	1				20	ON DIE(KLC	_	ZUU UN DIE(RLU		

TP711008	SIP referen	SIP reference: RFC 3261					_	reference:				
		Q.1912.5 7.5.2.1.1.2.2 b)/Q.732.7										
TSS reference	ISDN-(ISUP)-SIP/SS	/ECT						•				
SIP selection	, ,											
criteria												
ISUP selection												
criteria												
Test purpose	Facility message send upon receipt of the ANM when the ECT is invoked while one call is alerting											
	To verify that, in case the ECT is invoked while one call is alerting, as soon as the local											
	exchange (controlling the ECT) receives the <b>ANM</b> , it can successfully send to the other											
	remote user the FAC with service activation set to "call transfer" and the generic											
	notification set to "call transfer, active"  INFO B encapsulated FAC contains generic notification call transfer active											
SIP parameter	INFO B encapsulated	d FAC	C contail	ns g	eneric n	otification call tra	nsfe	er active				
values												
ISDN parameter values												
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3				
	SETUP	+			INVITE							
	ALERTING	→				nging(ACM)						
	CONN	→				(INVITE(ANM)						
				+	ACK							
	HOLD	<b>→</b>		<b>→</b>	INVITE	(CPG hold)						
						INVITE						
					ACK							
	SETUP	<b>→</b>					<b>→</b>	INVITE(IAM)				
	ALERTING	<del>-</del>						180 Ringing(ACM)				
	ALLINING	+					Ť	100 Tunging(7tow)				
	FAC(ECT invoke)	<b>→</b>										
	DISCONNECT(rr)	+		<b>→</b>	INFO (	AC ect alert)	1					
	RELEASE	<b>→</b>			200 Ok							
	RELEASE COMPL	+					<b>→</b>	INFO (CPG ect active)				
								200 OK INFO				
							<del>  -</del>					
								200 OK INVITE(ANM)				
							→	ACK				
						FAC ect active)						
				+	200 Ok	(INFO						
						BYE(REL)	<b>→</b>	BYE(REL)				
					20	0 OK BYE(RLC	+	200 OK BYE(RLC				

TP711009	SIP referer	ice: F	RFC 3261	Į:	ISUP reference:							
				Q.1912.5 7.5.2.1.1.2.2 b)/Q.732.7								
TSS reference	ISDN-(ISUP)-SIP/SS	/ECT										
SIP selection	( /											
criteria												
ISUP selection												
criteria												
Test purpose	Capability of sending the additional connected number in the call transfer number parameter when the ECT is invoked while one call is alerting  To verify that, in case the ECT is invoked while one call is alerting, the FAC sent to the											
							fer number parameter					
	with the information received in the <b>generic number</b> parameter if both the <b>connected</b>											
	number and an additional connected number in the generic number are received in the ANM  200 OK INVITE: encapsulated ANM contains the connected number and the additional											
SIP parameter		apsul	ated ANM	l c	ontains the connected r	numb	er and the additional					
values	connected number		•									
	INFO B: encapsulate	d FA	C contains	s g	generic notification call t tional connected numbe	ranst	er active and call					
ISDN parameter	manaier number dem	veu II	on the at	ul	nonai comiecteu numbe	J1						
values												
Comments	ISDN 2		SUT		SIP-I 1		SIP-I 3					
	SETUP	+		<del>-</del>	INVITE(IAM)							
	ALERTING	<b>→</b>			180 Ringing(ACM)							
	CONN	<b>→</b>			200 OK INVITE(ANM)							
					ACK							
	HOLD	<b>→</b>		<del>→</del>	INVITE(CPG hold)							
					200 OK INVITE							
					ACK							
				_	7.01.							
	SETUP	<b>→</b>				<b>→</b>	INVITE(IAM)					
	ALERTING	<del>-</del>					180 Ringing(ACM)					
	7122111110	+-				Ť	10011gg(/1011.)					
	FAC(ECT invoke)	<b>→</b>				+						
	DISCONNECT(rr)	+		<del>&gt;</del>	INFO (FAC ect alert)							
	RELEASE	<b>→</b>			200 OK INFO							
	RELEASE COMPL	+				→	INFO (CPG ect active)					
		1					200 OK INFO					
						Ť						
						+	200 OK INVITE(ANM)					
		+					ACK					
		+		<b>→</b>	INFO (FAC ect active)	Ť						
		+			200 OK INFO	$\top$						
		+		_		+						
			1		1	- 1						
					BYE(REL	)	BYE(REL)					
					200 OK BYE(RLO		200 OK BYE(RLC					

TP711010	SIP referer	nce: F	RFC 326	1		ISI	_	reference:		
						Q.1912.5 7.5.2.1.1.2.2 b)/Q.732.7				
TSS reference	ISDN-(ISUP)-SIP/SS	/ECT						•		
SIP selection										
criteria										
ISUP selection criteria										
Test purpose	when the ECT is inv	Capability of sending the connected number in the call transfer number parameter when the ECT is invoked while one call is alerting  To verify that, in case the ECT is invoked while one call is alerting, the FAC sent to the								
								fer number parameter		
				e cc	nnected	<b>i number</b> param	etei	r if only the <b>connected</b>		
OID	number is received			N 4				Lat. LPC 1		
SIP parameter	200 OK INVITE: enc	apsui	ated AN	IVI C	ontains t	ne connected nu	ımb	er and the additional		
values	connected number	-1 - 1	04-:			- (:f: (: 1)				
	INFO B: encapsulate	ed FA	C contai	ns (	generic n	iotification call tra	anst	er active and call		
ICDN parameter	transfer number deri	vea tr	om the o	conr	nected n	umber				
ISDN parameter values										
Comments	ISDN 2		SUT	1	SIP-I 1		I	SIP-I 3		
Comments	SETUP	+		-	INVITE	(IAM)		0.1.13		
	ALERTING	<b>→</b>				nging(ACM)				
	CONN	→ →		2	100 KII	(INVITE(ANM)				
	COMM	7			ACK	(AINVIIE				
				_	ACK					
	LIOLD			_	INIVATE	(ODO 11-I)				
	HOLD	<b>→</b>				(CPG hold)				
						INVITE				
				7	ACK					
							Ļ			
	SETUP	<b>→</b>						INVITE(IAM)		
	ALERTING	+					+	180 Ringing(ACM)		
	= 4.0/= 0= 4.0									
	FAC(ECT invoke)	<b>→</b>		<u> </u>						
	DISCONNECT(rr)	<b>←</b>				FAC ect alert)				
	RELEASE	<b>→</b>		+	200 OK	INFO				
	RELEASE COMPL	<b>←</b>						INFO (CPG ect active)		
							<b>←</b>	200 OK INFO		
								200 OK INVITE(ANM)		
							<b>→</b>	ACK		
				<b>→</b>	INFO (F	AC ect active)				
				+	200 OK	INFO				
		•			•			•		
						BYE(REL)	<b>→</b>	BYE(REL)		
					20			200 OK BYE(RLC		

TP711011		SIP reference: RFC 3261						reference: 1912.5 2.3.1/Q.732.7			
TSS reference	ISDN-(ISUP)-SIP/SS/ECT										
SIP selection criteria											
ISUP selection criteria											
Test purpose	To verify that the IUT	Call transfer number - conversion to international number To verify that the IUT converts the call transfer number to international format. The nature of address indicator shall be set to "international number"									
SIP parameter values											
ISDN parameter values											
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3			
	SETUP	<b>←</b>			INVITE						
	ALERTING	→				iging(ACM)					
	CONN	<b>→</b>				INVITE(ANM)					
				+	ACK						
	HOLD	<b>→</b>				(CPG hold)					
					200 OK	INVITE					
				<b>→</b>	ACK						
	SETUP	<b>→</b>						INVITE(IAM)			
	ALERTING	+						180 Ringing(ACM)			
	CONN	+						200 OK INVITE(ANM)			
							<b>→</b>	ACK			
	FAC(ECT invoke)	→									
	DISCONNECT(rr)	+		<b>→</b>	INFO (F	FAC ect active)					
	RELEASE	<b>→</b>		+	200 OK	INFO					
	RELEASE COMPL	+						INFO (FAC ect active)			
							<b>←</b>	200 OK INFO			
						BYE(REL)	<b>→</b>	BYE(REL)			
					20			200 OK BYE(RLC			

FAC(ECT invoke) → INFO (FAC ect active)  RELEASE → € 200 OK INFO  RELEASE COMPL ← INFO (FAC ect active)	TP711012	SIP referer	ice: R	FC 3261			Q.	reference: .1912.5					
SIP selection criteria  ISUP selection criteria  Test purpose  Call transfer number - removal of own country code To verify that the IUT removes the country code in the address signals of the call trannumber if it is the network's own country code. The nature of address indicator shall to "national (significant) number"  INVITE SIP-1 1: encapsulated IAM contains calling party number NoA "international number with the networks own country code.  INFO SIP-1 1: encapsulated ANM contains connected number NoA "international number with the networks own country code.  INFO SIP-1 1: encapsulated FAC contains the call transfer number derived from connumber NoA "national number"  INFO SIP-1 3: encapsulated FAC contains the call transfer number derived from connumber NoA "national number"  ISDN parameter values  Comments  ISDN 2 SUT SIP-1	TCC	IODAL (IOUE), OLD (CO	VECT			7.3;	7.5.	2.4.1/Q.732.7					
criteria         Test purpose       Call transfer number - removal of own country code         To verify that the IUT removes the country code in the address signals of the call transfer number if it is the network's own country code. The nature of address indicator shall be to "national (significant) number"         SIP parameter values       INVITE SIP-1 1: encapsulated IAM contains calling party number NoA "international number with the networks own country code.         200 OK INVITE SIP-1 3: encapsulated FAC contains the call transfer number derived from continumber NoA "national number"         IIFO SIP-1 1: encapsulated FAC contains the call transfer number derived from continumber NoA "national number"         ISDN parameter values         Comments       ISDN 2       SUT       SIP-1 3         SETUP       ★       INVITE(IAM)       INVITE(ANM)         ALERTING       →       180 Ringing(ACM)       CONN       ★       ACK         HOLD       →       INVITE(CPG hold)       ★       ACK         HOLD       →       INVITE(CPG hold)       ★       ★       ACK         SETUP       →       INVITE(CPG hold)       ★       ★       ACK         SETUP       →       INVITE(ANM)       ★       ACK       ACK       BRINGING(ACM)       ★       CON INVITE(AM)       ACK       ACK       BRINGING(ACM)       CON INVI		ISDN-(ISUP)-SIP/SS	/ECT										
Test purpose  Call transfer number - removal of own country code To verify that the IUT removes the country code in the address signals of the call transfunder if it is the network's own country code. The nature of address indicator shall be to "national (significant) number"  INVITE SIP-I 1: encapsulated IAM contains calling party number NoA "international number with the networks own country code. 200 OK INVITE SIP-I 3: encapsulated ANM contains connected number NoA "international number with the networks own country code.  INFO SIP-I 1: encapsulated FAC contains the call transfer number derived from connumber NoA "national number" INFO SIP-I 3: encapsulated FAC contains the call transfer number derived from callin party number NoA "national number"  ISDN parameter values  Comments  ISDN 2 SUT SIP-I 1 SIP-I 3  SETUP ← INVITE(IAM)  ALERTING → ISON NIVITE (ACK)  ACK  HOLD → INVITE(CPG hold)  ← 200 OK INVITE  SETUP → INVITE(CPG hold)  ← 200 OK INVITE  ACK  SETUP → INVITE(IAM)  ALERTING ← ISON Ringing(ACM)  CONN ← ISON RINGING (ACM)  CONN ← ISON RINGING (ACM)  FACK  SETUP → INVITE(IAM)  ALERTING ← ISON RINGING (ACM)  CONN ← ISON RINGING (ACM)  FACK  FAC(ECT invoke) → INFO (FAC ect active)  RELEASE → ← 200 OK INFO  RELEASE → ← 200 OK INFO	criteria												
Test purpose  Call transfer number - removal of own country code To verify that the IUT removes the country code in the address signals of the call transumber if it is the network's own country code. The nature of address indicator shall be to "national (significant) number"  SIP parameter values  INVITE SIP-I 1: encapsulated IAM contains calling party number NoA "international number with the networks own country code. 200 OK INVITE SIP-I 3: encapsulated ANM contains connected number NoA "interna number with the networks own country code.  INFO SIP-I 1: encapsulated FAC contains the call transfer number derived from connumber NoA "national number"  INFO SIP-I 3: encapsulated FAC contains the call transfer number derived from callin party number NoA "national number"  ISDN parameter values  Comments  ISDN 2  SUT  SIP-I 1  SIP-I 3  SIP-I													
number with the networks own country code. 200 OK INVITE SIP-I 3: encapsulated ANM contains connected number NoA "interna number with the networks own country code. INFO SIP-I 1: encapsulated FAC contains the call transfer number derived from connumber NoA "national number" INFO SIP-I 3: encapsulated FAC contains the call transfer number derived from callin party number NoA "national number"  ISDN parameter values  Comments  ISDN 2 SUT SIP-I 1 SIP-I 3 SETUP ← INVITE(IAM) ALERTING → 180 Ringing(ACM) CONN → 200 OK INVITE(ANM) ← ACK  HOLD → INVITE(CPG hold) ← 200 OK INVITE → ACK  SETUP → INVITE(IAM) ALERTING ← 200 OK INVITE → ACK  SETUP → INVITE(IAM) ALERTING ← 180 Ringing(ACM) CONN ← 200 OK INVITE → ACK  SETUP → INVITE(IAM) ALERTING ← 180 Ringing(ACM) CONN ← 200 OK INVITE(ACM) ALERTING ← 180 Ringing(ACM) CONN ← 200 OK INVITE(ACM) ACK  FAC(ECT invoke) → INFO (FAC ect active) RELEASE → ← 200 OK INFO RELEASE → ← 200 OK INFO		To verify that the IUT number if it is the ne to "national (signification).	that the IUT removes the country code in the address signals of the <b>call transfer</b> if it is the network's own country code. The nature of address indicator shall be senal (significant) number"										
ISDN parameter values		number with the netw 200 OK INVITE SIP- number with the netw INFO SIP-I 1: encap- number NoA "national INFO SIP-I 3: encap-	vorks of the vorks	own count capsulate own count d FAC cor ber" d FAC cor	ry o d A ry o ntaii	code. NM contains connected code. ns the call transfer number	ed nu mber	umber NoA "international derived from connected					
SETUP       ←       ←       INVITE(IAM)         ALERTING       →       180 Ringing(ACM)         CONN       →       200 OK INVITE(ANM)         HOLD       →       INVITE(CPG hold)         ←       200 OK INVITE         →       ACK         SETUP       →       INVITE(IAM)         ALERTING       ←       180 Ringing(ACM)         CONN       ←       200 OK INVITE(A         FAC(ECT invoke)       →       ACK         FAC(ECT invoke)       →       INFO (FAC ect active)         RELEASE       →       ←       200 OK INFO         RELEASE COMPL       ←       1NFO (FAC ect active)			The state of the s										
ALERTING  CONN  → 200 OK INVITE(ANM)  ← ACK  HOLD  → INVITE(CPG hold)  ← 200 OK INVITE  → ACK  SETUP  → ACK  SETUP  → INVITE(IAM)  ALERTING  ← 180 Ringing(ACM)  ← 200 OK INVITE  → ACK  SETUP  → INVITE(IAM)  ALERTING  ← 180 Ringing(ACM)  ← 200 OK INVITE(A  → ACK  FAC(ECT invoke)  DISCONNECT(rr)  ← INFO (FAC ect active)  RELEASE  ← 200 OK INFO  RELEASE  → INFO (FAC ect active)	Comments			SUT	S	SIP-I 1		SIP-I 3					
CONN → 200 OK INVITE(ANM)  HOLD → INVITE(CPG hold)  ← 200 OK INVITE  → ACK  SETUP → INVITE(IAM)  ALERTING ← 180 Ringing(ACM)  CONN ← 200 OK INVITE(A  → ACK  FAC(ECT invoke) → INFO (FAC ect active)  RELEASE → € 200 OK INFO  RELEASE COMPL ← INFO (FAC ect active)  RELEASE COMPL ← INFO (FAC ect active)		SETUP	+	•	-	NVITE(IAM)							
HOLD		ALERTING	<b>→</b>	-	1	80 Ringing(ACM)							
HOLD		CONN	→										
SETUP       →       ACK         ALERTING       ←       180 Ringing(ACM)         CONN       ←       200 OK INVITE(A         FAC(ECT invoke)       →       ACK         FAC(ECT invoke)       →       INFO (FAC ect active)         RELEASE       →       ←       200 OK INFO         RELEASE COMPL       ←       INFO (FAC ect active)         INFO (FAC ect active)       →       INFO (FAC ect active)													
SETUP       →       INVITE(IAM)         ALERTING       ←       180 Ringing(ACM)         CONN       ←       200 OK INVITE(A         FAC(ECT invoke)       →       ACK         FAC(ECT invoke)       →       INFO (FAC ect active)         RELEASE       →       ←       200 OK INFO         RELEASE COMPL       ←       INFO (FAC ect active)		HOLD	<b>→</b>	•	- 2	00 OK INVITE							
ALERTING CONN CONN CONN CONN CONN CONN CONN CO				7	<b>▶</b>	iCK							
CONN ← 200 OK INVITE(A  → ACK  FAC(ECT invoke) → INFO (FAC ect active)  RELEASE → ← 200 OK INFO  RELEASE COMPL ← INFO (FAC ect active)  RELEASE → INFO (FAC ect active)		SETUP	<b>→</b>				<b>→</b>	INVITE(IAM)					
FAC(ECT invoke) → ACK  FAC(ECT invoke) → INFO (FAC ect active)  RELEASE → ← 200 OK INFO  RELEASE COMPL ← INFO (FAC ect active)		ALERTING	+				+	180 Ringing(ACM)					
FAC(ECT invoke) → INFO (FAC ect active)  RELEASE → ← 200 OK INFO  RELEASE COMPL ← INFO (FAC ect active)  RELEASE COMPL ← INFO (FAC ect active)		CONN	+					200 OK INVITE(ANM)					
DISCONNECT(rr) ← → INFO (FAC ect active)  RELEASE → ← 200 OK INFO  RELEASE COMPL ← → INFO (FAC ect active)		FAC(FCT invoke)	-				+	AOR					
RELEASE → ← 200 OK INFO  RELEASE COMPL ← → INFO (FAC ect ac					<b>1</b> 11	VEO (EAC activo)	-						
RELEASE COMPL ← INFO (FAC ect ac							-						
					+	OU OIN IINI O	-	INFO (FAC ect active)					
<b>←</b> 200 OK INFO		RELEAGE COM E											
BYE(REL) → BYE(REL)					1	BYF/RFI	\ <del>  \</del>	BYF(RFL)					
200 OK BYE(RLC			++	<del></del>	+								

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

TP712001	SIP reference: RFC 326	61		ISUP reference: Q.1912.5 2.5.2.1.1/Q.732				
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose  SIP parameter values	"Call is diverting" indication received in ACM To verify that a call can be successfully established, if diversion occurs. The encapsulated ACM contains the generic notification indicator set to "call is diverting", the call diversion information and the redirection number Applicable redirection reason in the call diversion information: "busy"  CFB(n); CFB(u,l)  "unconditional"  CFU  "deflection immediate response"  CD(i,l)  183 Session Progress encapsulated ACM generic notification indicator "call is diverting"							
ISDN parameter values	NOTIFY: Notification indicator "ca	ll is div	erting"					
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	NOTIFY	<b>←</b>		<b>←</b>	183 Session Progress(ACM)			
	ALERTING	<b>←</b>	· · · · · · · · · · · · · · · · · · ·	+	180 Ringing(CPG)			
	CONNECT	<b>←</b>		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)			
	RELEASE	<b>←</b>		+	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>						

TP712002	SIP reference: RFC	ISUP reference: Q.1912.5 2.5.2.1.1/Q.732						
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose  SIP parameter values	"Call diversion may occur" received in ACM To verify that a call can be successfully established, if diversion may occur. The ACM indicates that "call diversion may occur" in the optional backward call indicators. The following CPG contains the generic notification indicator set to "call is diverting", the call diversion information and the redirection number, if diversion occurs Applicable redirection reason in the call diversion information: "busy"  CFB(u,e) "no reply"  CFNR "deflection during alerting"  CD(a) "deflection immediate response"  CD(i,e)  180 Ringing: encapsulated ACM optional backward call indicator "call diversion may occur" 183 Session Progress: encapsulated CPG contains generic notification "call is diverting",							
ISDN parameter values	call diversion information, red	irection na	IIIDCI					
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
				+	183 Session Progress(CPG)			
	CONNECT	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)			
	RELEASE	<del>-</del>		+	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>						

TP712003	SIP reference: RFC 32	61	ISUP reference: Q.1912.5				
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Redirection number - presenta subscription option To verify that the originating exch calling access signalling system, information is coded "010 presen The redirection number restriction	ange makes if the notification allowe	s the redirect ation subscr	tion number available to the iption option of the call diversion ection number".			
SIP parameter values	183 Session Progress encapsula 200 OK INVITE: encapsulated Al	ted ACM ge	neric notifica	ation indicator "call is diverting"			
ISDN parameter values	CONNECT: redirection number	vivi realitecti	on namber i	estitionary presentation anowed			
Comments	ISDN	SI	JT	SIP-I			
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)			
	NOTIFY	+	+	183 Session Progress(ACM)			
	ALERTING	+	<b>←</b>	180 Ringing(CPG)			
	CONNECT	+	<b>←</b>	200 OK INVITE(ANM)			
			<b>→</b>	ACK			
	DISCONNECT	<b>→</b>	<b>→</b>	BYE(REL)			
	RELEASE	+	+	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>					

TP712004	SIP reference: RFC 3261			ISUP reference: Q.1912.5 2.4.2; Table 2-1/Q.732						
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV		•							
SIP selection										
criteria										
ISUP selection criteria										
Test purpose	subscription option To verify that the originating exchange the calling access signalling system,	To verify that the originating exchange does not make the redirection number available to the calling access signalling system, if the notification subscription option of the call diversion information is coded "001 presentation not allowed", "011 presentation allowed								
SIP parameter		183 Session Progress encapsulated ACM call diversion information notification								
values	subscription option "presentation a									
ISDN parameter	NOTIFY: notification indicator "call is									
values	ALERTING: no redirection number CONNECT: no redirection number									
Comments	ISDN	SU	Т	SIP-I						
	SETUP -	•	<b>→</b>	INVITE(IAM)						
	NOTIFY		+	183 Session Progress(ACM)						
	ALERTING	-	+	180 Ringing(CPG)						
	CONNECT ← 200 OK INVITE(ANM)									
			<b>→</b>	ACK						
	DISCONNECT	<b>—</b>	<b>→</b>	BYE(REL)						
	RELEASE	-	+	200 OK BYE(RLC)						
	RELEASE COMPLETE	<b></b>								

TP712005	SIP reference: RFC 32	61			ISUP reference:		
					Q.1912.5 2.4.2; Table 2-1/Q.732		
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV				2.4.2, Table 2-1/Q./32		
SIP selection							
criteria							
ISUP selection criteria							
Test purpose  SIP parameter values	Redirection number - presentation restricted - according to redirection number restriction parameter  To verify that the originating exchange does not make the redirection number available to the calling access signalling system, if the redirection number restriction parameter indicates "01 Presentation restricted"  The notification subscription option of the call diversion information is coded "010 Presentation allowed with redirection number"  183 Session Progress encapsulated ACM call diversion information notification subscription option "presentation allowed with redirection number"						
values	200 OK INVITE: encapsulated restricted						
ISDN parameter values	CONNECT: no redirection number	er					
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
	NOTIFY	+		+	183 Session Progress(ACM)		
	ALERTING	+		+	180 Ringing(CPG)		
	CONNECT	← 200 OK INVITE(ANM)					
				<b>→</b>	ACK		
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)		
	RELEASE	+		+	200 OK BYE(RLC)		
	RELEASE COMPLETE	<b>→</b>					

TP712006	SIP reference: RFC 3261		ISUP reference: Q.1912.5 2.4.2; Table 2-1/Q.732				
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV						
SIP selection							
criteria							
ISUP selection							
criteria							
SIP parameter values ISDN parameter values	Redirection number - presentation restricted - no redirection number restriction parameter received To verify that the originating exchange does not make the redirection number available to the calling access signalling system, if no redirection number restriction parameter is received The notification subscription option of the call diversion information is coded "010 Presentation allowed with redirection number"  183 Session Progress encapsulated ACM call diversion information notification subscription option "presentation allowed with redirection number" 200 OK INVITE: encapsulated ANM without redirection number restriction parameter CONNECT: redirection number						
Comments	ISDN	SUT		SIP-I			
	SETUP →		<b>→</b>	INVITE(IAM)			
	NOTIFY <b>←</b>		+	183 Session Progress(ACM)			
	ALERTING		+	180 Ringing(CPG)			
	CONNECT		+	200 OK INVITE(ANM)			
			<b>→</b>	ACK			
	DISCONNECT →		<b>→</b>	BYE(REL)			
	RELEASE		+	200 OK BYE(RLC)			
	RELEASE COMPLETE →						

TP712007	SIP reference: RFC 32	61		ISUP reference: Q.1912.5 2.4.2/Q.732			
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Multiple diversions - redirection number not send by the last diversion  To verify that the originating exchange does not make any redirection number available to the calling access signalling system, if the last diverting exchange does not send one (see note).						
SIP parameter	183 Session Progress encapsula	ted AC	M call dive	ersion i	nformation notification		
values  ISDN parameter	subscription option "presentation allowed with redirection number", redirection number 183 Session Progress encapsulated ACM call diversion information notification subscription option "presentation allowed with redirection number", no redirection number 200 OK INVITE: encapsulated ANM redirection number restriction "presentation allowed"						
values	ALERTING: no redirection number CONNECT: no redirection number 1.00 number 1.0						
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
	NOTIFY	+		+	183 Session Progress(ACM)		
				+	183 Session Progress(CPG)		
	ALERTING	+		+	180 Ringing(CPG)		
	CONNECT	+		+	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)		
	RELEASE	+		+	200 OK BYE(RLC)		
	RELEASE COMPLETE	<b>→</b>					

The first diverting exchange sends the redirection number and allows for its presentation. The NOTE: second (last) diversion allows for the presentation of the redirection number, but does not send it, i.e. only **call diversion information** is present in the message and the redirection number is missing. The **redirection number restriction** parameter is also received as "presentation allowed".

TP712008	SIP reference: RFC 3261			ISUP reference: Q.1912.5 2.4.2/Q.732			
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV						
SIP selection							
criteria							
ISUP selection criteria							
Test purpose	Multiple diversions - redirection restrictive notification subscripti To verify that the originating exchanumber according to the contents the call diversion information, if t ("presentation allowed" in the redir	on op nge ha of the he foa	otion andles the most rest warded-to	prese rictive user a	ntation of the <b>redirection</b> notification subscription option of allows presentation of the number		
SIP parameter	183 Session Progress encapsulated ACM call diversion information notification						
values	subscription option "presentation allowed with redirection number", redirection number 183 Session Progress encapsulated ACM call diversion information notification subscription option "presentation allowed without redirection number", redirection number 200 OK INVITE: encapsulated ANM redirection number restriction "presentation allowed"						
ISDN parameter	ALERTING: no redirection number						
values	CONNECT: no redirection number						
Comments	ISDN		SUT		SIP-I		
	<b>-</b>	<b>→</b>		<b>→</b>	INVITE(IAM)		
	NOTIFY	<b>←</b>		+	183 Session Progress(ACM)		
				+	183 Session Progress(CPG)		
	ALERTING	<b>←</b>		+	180 Ringing(CPG)		
	CONNECT	<b>←</b>		+	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)		
	_	<del>(</del>		+	200 OK BYE(RLC)		
	RELEASE COMPLETE	<b>→</b>					
forwardin	nessages each containing the <b>call di</b> ligs have occurred (from option B - im ation takes place).						

TP712009	SIP	SIP reference: RFC 3261				ISUP reference: Q.1912.5 2.5.2.5.1.1/Q.732			
TSS reference	ISDN-(ISUP)-	-SIP/SS/CDI\	/						
SIP selection criteria									
ISUP selection criteria									
Test purpose	To verify that	the IUT acce	pts a	y the diverted-to exchange and can successfully establish a	div	erted call			
SIP parameter values	183 Session information, r			ulated ACM generic notification	"cal	I is diverted", redirection			
ISDN parameter values									
Comments	ISDN 2	SUT		SIP-I 1		SIP-I 3			
			<b>←</b>	INVITE(IAM)					
		CDIV							
			<b>→</b>	183 Session Progress(ACM)					
					<b>→</b>	INVITE(IAM)			
					+	180 Ringing(ACM)			
			<b>→</b>	180 Ringing(ACM)					
						200 OK INVITE(ANM)			
				200 OK INVITE(ANM)	→	ACK			
			<b>←</b>	ACK					
			_	BYE(REL)		BYE(REL)			
			→	200 OK BYE(RLC	<b>←</b>	200 OK BYE(RLC			

TP712010	SIP reference: RFC 32	61		ISUP reference: Q.1912.5 2.5.2.5.1.1/Q.732			
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Setting of redirection number r (pres. allowed) To verify that the IUT includes the CPG, ANM or CON set to "prese	e redire	ction nun	nber re	estriction indicator in the ACM,		
SIP parameter values	200 OK INVITE: encapsulated ANM redirection number restriction "presentation allowed"						
ISDN parameter values							
Comments	ISDN		SUT		SIP-I		
	SETUP	<del>-</del>		+	INVITE(IAM)		
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)		
	CONNECT	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)		
		← ACK					
			•				
	DISCONNECT	<b>+</b>		+	BYE(REL)		
	RELEASE	<b>→</b>		<b>→</b>	200 OK BYE(RLC)		
	RELEASE COMPLETE	<b>+</b>					

TP712011	SIP reference: RFC 3261			ISUP reference: Q.1912.5 2.5.2.5.1.1/Q.732
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV			2.0.2.0, Q.7.02
SIP selection criteria				
ISUP selection criteria				
Test purpose	Setting the redirection num (pres. restricted) To verify that the IUT includes CPG, ANM or CON set to "pre	the <b>redirecti</b>	on number r	estriction indicator in the ACM,
SIP parameter values	·		•	restriction "presentation restricted"
ISDN parameter values				
Comments	ISDN		SUT	SIP-I
	SETUP	+	<b>+</b>	INVITE(IAM)
	ALERTING	→	→	180 Ringing(ACM)
	CONNECT	<b>→</b>	→	200 OK INVITE(ANM)
			+	ACK
	DISCONNECT	+	<b>←</b>	BYE(REL)
	RELEASE	<b>→</b>	<b>→</b>	200 OK BYE(RLC)
	RELEASE COMPLETE	+		

TP712012	SIP reference: RFC 3261				Q.	reference: 1912.5 2 b) 2)/Q.732
TSS reference	ISDN-(ISUP)	-SIP/SS/CDI\	/			
SIP selection						
criteria						
ISUP selection criteria						
Test purpose	Verify that the	e IUT sets the	adc	rated by the diverting exchangeress presentation restricted ind ed user releases his/her number	icato	
SIP parameter	INVITE SIP-I	3:encapsulat	ted I/	AM original called number prese	entat	ion allowed
values				-		
ISDN parameter						
values						
Comments	ISDN 2	SUT		SIP-I 1		SIP-I 3
			<b>←</b>	INVITE(IAM)		
		CDIV				
			<b>→</b>	183 Session Progress(ACM)		
					<b>→</b>	INVITE(IAM)
					+	180 Ringing(ACM)
			<b>→</b>	180 Ringing(ACM)		
					+	200 OK INVITE(ANM)
			→	200 OK INVITE(ANM)		ACK
				ACK		
		l.				ı
			+	BYE(REL)	<b>→</b>	BYE(REL)
			<b>→</b>	200 OK BYE(RLC	+	200 OK BYE(RLC

TP712013	SIP reference: RFC 3261			3261	ISUP reference: Q.1912.5			
TSS reference	ISDN-(ISUP)	-SIP/SS/CDI\	/					
SIP selection criteria								
ISUP selection criteria								
Test purpose	Verify that the number according option	Redirecting number generated by the diverting exchange /erify that the IUT sets the address presentation restricted indicator of the redirecting number according to the "served user releases his/her number to the diverted-to user" option The redirecting indicator in the redirection information shall be set to "011 Call diverted"						
SIP parameter values	INVITE SIP-I	3: redirecting	nun	nber, redirection information	on			
ISDN parameter values								
Comments	ISDN 2	SUT		SIP-I 1		SIP-I 3		
			+	INVITE(IAM)				
		CDIV						
			<b>→</b>	183 Session Progress(AC	CM)			
					<b>→</b>	INVITE(IAM)		
					+	180 Ringing(ACM)		
			<b>→</b>	180 Ringing(ACM)				
					+	200 OK INVITE(ANM)		
			<b>→</b>	200 OK INVITE(ANM)	<b>→</b>	ACK		
			+	ACK				
				<u> </u>		<u> </u>		
			+	BYE(REL)	→	BYE(REL)		
			<b>→</b>	200 OK BYE(RLC	+	200 OK BYE(RLC		

TP712014	SIP reference: RFC 3261				Q.	reference: 1912.5 2 b) 5)/Q.732	
TSS reference	ISDN-(ISUP)-	SIP/SS/CDI	/				
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	To verify that indicator rece - not required way" - preferred all - required all	the IUT can ived in the for all the way" the way" shathe way shathe	succ orwa shall all be Il be	essfully divert a rd call indicato be changed to eleft unchanged left unchanged	<b>rs</b> with the value "ISDN user part	DN ι e "IS pref	ferred all the
SIP parameter	INVITE SIP-I	3 : encapsula	ated	IAM forward cal	I indicator ISDN	use	r part required all the
values	way	-					
ISDN parameter							
values							
Comments	ISDN 2	SUT		SIP-I 1			SIP-I 3
			<b>←</b>	INVITE(IAM)			
		CDIV					
			<b>→</b>	183 Session Pr	rogress(ACM)		
						<b>→</b>	INVITE(IAM)
						+	180 Ringing(ACM)
			<b>→</b>	180 Ringing(A0	CM)		
						+	200 OK INVITE(ANM)
			<b>→</b>	200 OK INVITE	(ANM)	<b>→</b>	ACK
			+	ACK			
		•	•				
			+	BYE(REL)		<b>→</b>	BYE(REL)
			<b>→</b>	200 OK BYE(R	LC	+	200 OK BYE(RLC

TP712015	SIP reference: RFC	3261	2	ISUP reference: Q.1912.5 .5.2.5.1.2 c) ii); iii)/Q.732			
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV		•				
SIP selection criteria							
ISUP selection							
criteria							
Test purpose	Call diversion may occur in To verify that the IUT includes "call diversion may occur" in t	s an <b>optional b</b>	ackward ca	II indicator with the indication CD(a), CFB(u,e) and CD(i,e)			
SIP parameter				ator "subscriber free" optional			
values	backward call indicator "call d						
ISDN parameter values	ALERTING: no mapping of or	otional backwar	d call indicat	or value			
Comments	ISDN	S	UT	SIP-I			
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)			
	ALERTING	+	+	180 Ringing(ACM)			
	CONNECT	+	+	200 OK INVITE(ANM)			
		→ ACK					
	DISCONNECT	<b>→</b>	<b>→</b>	BYE(REL)			
	RELEASE	+	+	200 OK BYE(RLC)			
	RELEASE COMPLETE	+					

## A.1.1.2.13 Call HOLD (HOLD)

TP713001	SIP reference: RFC 3261			ISUP reference: Q.1912.5 2.5.2.1.1.1; 2.5.2.1.1.2/Q.733		
TSS reference	ISDN-(ISUP)-SIP/SS/	HOLD				
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Call hold after answer To verify that a call cat that notifications are s "progress"	n be placed on	hold and can b	e retriev	ed again by the local user and ent indicator set to	
SIP parameter						
values						
ISDN parameter						
values						
Comments	ISDN		SUT		SIP	
	SETUP	→		→	INVITE(IAM)	
	ALERTING	+		+	180 Ringing(ACM)	
	CONN	+		<b>←</b>	200 OK INVITE(ANM)	
				<b>→</b>	ACK	
			Communicatio	n		
	HOLD	<b>→</b>		<b>→</b>	INFO(CPG hold)	
				+	200 OK INFO	
	RETRIVE	<b>→</b>		→	INFO(CPG retrieve)	
				+	200 OK INFO	
			Communicatio	n		
	DISC	<b>→</b>		<b>→</b>	BYE(REL)	
	REL	+		+	200 OK BYE	
	REL_COM	<b>→</b>				

TP713002	SIP reference: RFC 3261			ISUP reference: Q.1912.5 2.5.2.1.1.1; 2.5.2.1.1.2/Q.733				
TSS reference	ISDN-(ISUP)-SIP/SS/HO	OLD						
SIP selection criteria								
ISUP selection criteria								
Test purpose	Call hold after answer, To verify that a call can and that notifications are	be placed or	n hold and can b		ed again by the remote user			
SIP parameter values								
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
			Communication	)				
	HOLD	<b>←</b>		<b>←</b>	INFO(CPG hold)			
				<b>→</b>	200 OK INFO			
	RETRIVE	<del>-</del>		<b>←</b>	INFO(CPG retrieve)			
				<b>→</b>	200 OK INFO			
			Communication					
	DISC	<b>→</b>		→	BYE(REL)			
	REL	<b>←</b>		<b>←</b>	200 OK BYE			
	REL_COM	<b>→</b>						

TP713003	SIP reference: RFC 3261			ISUP reference: Q.1912.5 2.2.1; 2.5.2.1.1.1; 2.5.2.1.1.2/Q.7				
TSS reference	ISDN-(ISUP)-SIP/SS/	HOLD						
SIP selection								
criteria								
ISUP selection criteria	PICS 8/1							
Test purpose	To verify that an outgo	Call hold after alerting, requested by the local user To verify that an outgoing call can be placed on HOLD after alerting has commenced and can be retrieved afterwards by the local user and that notifications are sent with CPG nessages						
SIP parameter								
values								
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	HOLD	<b>→</b>		<b>→</b>	INFO(CPG hold)			
				+	200 OK INFO			
	RETRIVE	<b>→</b>		<b>→</b>	INFO(CPG retrieve)			
				+	200 OK INFO			
	CONN	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
		C	ommunication	า				
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	REL	+		+	200 OK BYE			
	REL_COM	<b>→</b>						

TP713004	SIP reference: RFC 3261				SUP reference: Q.1912.5 .1; 2.5.2.5.1/Q.733
TSS reference	ISDN-(ISUP)-SIP/SS/	HOLD			
SIP selection criteria					
ISUP selection criteria	PICS 8/1				
Test purpose	Call hold after alerting To verify that an incorrupt the remote user				n be retrieved afterwards by
SIP parameter values					
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP	→		<b>→</b>	INVITE(IAM)
	ALERTING	<b>←</b>		+	180 Ringing(ACM)
	HOLD	<b>+</b>		+	INFO(CPG hold)
				→	200 OK INFO
	RETRIVE	<del>-</del>		+	INFO(CPG retrieve)
				→	200 OK INFO
	CONN	<del>-</del>		+	200 OK INVITE(ANM)
				<b>→</b>	ACK
			Communication	)	
	DISC	<b>→</b>		<b>→</b>	BYE(REL)
	REL	<del>-</del>		+	200 OK BYE
	REL_COM	→			

TP713005	SIP reference: RFC 3261			I	SUP reference: Q.1912.5 2.3/Q.764
TSS reference	ISDN-(ISUP)-SIP/SS/Ho	OLD			
SIP selection criteria					
ISUP selection criteria					
Test purpose	Call hold after answer To verify that a call in the hold service				ved user ser who activated the Call
SIP parameter values					
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP	→		<b>→</b>	INVITE(IAM)
	ALERTING	<b>←</b>		+	180 Ringing(ACM)
	CONN	+		+	200 OK INVITE(ANM)
				<b>→</b>	ACK
			Communication	n	
	HOLD	<b>→</b>		<b>→</b>	INFO(CPG hold)
				+	200 OK INFO
	DISC	<b>→</b>		<b>→</b>	BYE(REL)
	REL	+		+	200 OK BYE
	REL_COM	<b>→</b>			

TP713006	SIP reference	SIP reference: RFC 3261			SUP reference: Q.1912.5 2.3/Q.764
TSS reference	ISDN-(ISUP)-SIP/SS/F	HOLD			
SIP selection criteria					
ISUP selection criteria					
Test purpose	Call hold after answe To verify that a call in t Call hold service				ved user user who did not activate the
SIP parameter values					
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP	→		<b>→</b>	INVITE(IAM)
	ALERTING	<b>+</b>		+	180 Ringing(ACM)
	CONN	+		+	200 OK INVITE(ANM)
				<b>→</b>	ACK
			Communication	on	
	HOLD	→		<b>→</b>	INFO(CPG hold)
				+	200 OK INFO
	DISC	<del>-</del>		+	BYE(REL)
	REL	<b>→</b>		+	200 OK BYE
	REL_COM	+			

TP713007	SIP reference: RFC 3261			ISUP reference: Q.1912.5 2.3/Q.764		
TSS reference	ISDN-(ISUP)-SIP/SS/HC	DLD				
SIP selection						
criteria						
ISUP selection						
criteria —						
Test purpose	Call hold after alerting, To verify that a held call without retrieving the cal	can be release			rved user ivated the Call hold service	
SIP parameter						
values						
ISDN parameter values						
Comments	ISDN		SUT		SIP	
	SETUP	+		+	INVITE(IAM)	
	ALERTING	→		<b>→</b>	180 Ringing(ACM)	
	HOLD	→		<b>→</b>	INFO(CPG hold)	
				+	200 OK INFO	
	DISC	<b>→</b>		<b>→</b>	CANCEL/BYE	
	RELEASE	+		+	200 OK CANCEL/BYE	
	REL_COMP	<b>→</b>		+	487 Request Terminated	
				<b>→</b>	ACK	

TP713008	SIP reference	e: RFC 3261		ISUP reference: Q.1912.5 2.3/Q.764		
TSS reference	ISDN-(ISUP)-SIP/SS/H	IOLD				
SIP selection criteria						
ISUP selection criteria						
Test purpose	Call hold after answer To verify that a call in the Call hold service				ved user user who did not activate the	
SIP parameter values						
ISDN parameter values						
Comments	ISDN		SUT		SIP	
	SETUP	+		+	INVITE(IAM)	
	ALERTING	→		<b>→</b>	180 Ringing(ACM)	
	HOLD	<b>←</b>		+	INFO(CPG hold)	
				<b>→</b>	200 OK INFO	
	DISC	+		+	CANCEL	
	RELEASE	→		<b>→</b>	200 OK CANCEL/BYE	
	REL_COMP	+		<b>→</b>	487 Request Terminated	
				<b>←</b>	ACK	

## A.1.1.2.14 Call Waiting (CW)

TP714001	SIP reference	e: RFC 3261		ISUP reference: Q.1912.5 1.5.2.1.1/Q.733			
TSS reference	ISDN-(ISUP)-SIP/SS/C	:W					
SIP selection							
criteria							
ISUP selection criteria							
Test purpose	Call waiting indication To verify that a call can call		ly established i	f the AC	M indicates that it is a waiting		
SIP parameter		ated ACM cont	ains the Generi	c notific	ation parameter value "call is		
values	a waiting call"						
ISDN parameter values							
Comments	ISDN		SUT		SIP		
	SETUP	→		<b>→</b>	INVITE(IAM)		
	ALERTING	<b>+</b>		+	180 Ringing(ACM)		
	CONN	+		+	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
		(	Communication	)			
	DISC	<b>→</b>		<b>→</b>	BYE(REL)		
	REL	+		+	200 OK BYE		
	REL_COM	<b>→</b>					

TP714002	SIP reference	: RFC 3261		ISUP reference: Q.1912.5 1.5.2.1.1/Q.733					
TSS reference	ISDN-(ISUP)-SIP/SS/CV	N							
SIP selection criteria									
ISUP selection criteria									
Test purpose		Call waiting indication in CPG To verify that a call can be successfully established if the CPG indicates that it is a waiting call							
SIP parameter values	180 Ringing: encapsulated ACM the called party status is set to "no indication" 183 Session Progress: encapsulated CPG Alerting contains the Generic notification parameter value "call is a waiting call"								
ISDN parameter values									
Comments	ISDN		SUT		SIP				
	SETUP	→		<b>→</b>	INVITE(IAM)				
				+	183 Session Progress(ACM)				
	ALERTING	<b>←</b>		+	180 Ringing(CPG)				
	CONN	<b>←</b>		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
		Cor	nmunicatio	n					
	DISC	<b>→</b>		<b>→</b>	BYE(REL)				
	REL	+		+	200 OK BYE				
	REL_COM	<b>→</b>							

TP714003	SIP reference: R	FC 3261		ISUP reference: Q.1912.5 1.5.2.5.1/Q.733					
TSS reference	ISDN-(ISUP)-SIP/SS/CW								
SIP selection criteria									
ISUP selection criteria									
Test purpose	To verify that a call can be waiting service (with notific	Call waiting indication in ACM or CPG To verify that a call can be successfully established if the user has subscribed to the call waiting service (with notification) and if he is currently busy, but answers the waiting call. The indication shall be sent either in an ACM or a CPG							
SIP parameter values	180 Ringing: encapsulated a waiting call"	180 Ringing: encapsulated ACM contains the Generic notification parameter value "call is a waiting call"							
ISDN parameter values	-								
Comments	ISDN		SUT		SIP				
	SETUP	+		+	INVITE(IAM)				
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)				
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)				
				+	ACK				
			Communication						
	DISC	+		+	BYE(REL)				
	REL				200 OK BYE				
	REL_COM	+							

TP714004	SIP reference: R	₹FC 32	61		ISUP reference: Q.1912.5 1.5.2.5.2/Q.733			
TSS reference	ISDN-(ISUP)-SIP/SS/CW							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Call waiting rejected To verify that the IUT send waiting call	To verify that the IUT sends a <b>REL</b> with cause #21 (call rejected) if a busy user rejects the						
SIP parameter values	480 Temporarily unavailab	ole: end	capsulated	REL c	ause 21			
ISDN parameter values	RELEASE COMPLETE: ca	ause 2	1					
Comments	ISDN		SUT		SIP			
	SETUP	+		+	INVITE(IAM)			
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)			
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)			
				+	ACK			
		C	ommunicat	ion				
	SETUP	+		+	INVITE(IAM)			
	ALERTING	→		<b>→</b>	180 Ringing(ACM waiting call)			
	RELEASE COMPLETE	→		→	480 Temporarily unavailable(REL#21)			
				+	ACK			
			ommunicat					
	DISC	<b>←</b>		+	BYE(REL)			
	REL				200 OK BYE			
	REL_COM	←						

## A.1.1.2.15 Three Party Service (3PTY)

TP715001	SIP re	SIP reference: RFC 3261				Q	reference: .1912.5 2.1/Q.734.2	
TSS reference	ISDN-(ISUP)-S	ISDN-(ISUP)-SIP/SS/3PTY						
SIP selection	10011 (1001 ) 0	11 /00/	01 11					
criteria								
ISUP selection								
criteria								
Test purpose	To verify that the successfully join remote parties and the IUT should	Served user initiates 3PTY  To verify that the IUT, where the served user with two active calls is located, can successfully join these calls to form a three-way conversation, and notify the implied remote parties accordingly.  The IUT should send CPG messages with the generic notification indicator set to "conference established" to both implied parties. The event indicator in the CPG should						
	be set to "progr				p pa			
SIP parameter								
values								
ISDN parameter								
values								
Comments	ISDN		SUT		SIP-I 1		SIP-I 2	
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ALERTING	+		<b>←</b>	180 Ringing(ACM)			
	CONN	<b>←</b>		<b>←</b>	200 OK INVITE(ANM)			
			Commun	icat				
	HOLD	<b>→</b>		<b>→</b>	INVITE(CPG hold)			
				<del>(</del>	200 OK INVITE			
				<b>→</b>	ACK			
	SETUP	→				→	INVITE(IAM)	
	ALERTING	+				<b>←</b>	180 Ringing(ACM)	
	CONN	+				+	200 OK INVITE(ANM)	
	FAC(est3pty)	<b>→</b>		<b>→</b>	INFO(CPG conf est)			
				<del>(</del>	200 OK INFO			
						<b>→</b>	INFO(CPG conf est)	
						+	200 OK INFO	
				3	PTY communication			
	DISC	+		<b>←</b>	BYE(REL)			
	RELEASE	<b>→</b>		<b>→</b>	200 OK BYE			
	REL_COM	+				<b>→</b>	INFO(CPG conf disc)	
						+	200 OK INFO	
	DISC	<b>→</b>				<b>→</b>	BYE(REL)	
	RELEASE	+				+	200 OK BYE	
	REL_COM	<b>→</b>						

TP715002	SIP re	feren	ce: RFC 3261	İ			reference: .1912.5				
				5.2.1.1.3 a)/Q.734.2							
TSS reference	ISDN-(ISUP)-SI	P/SS/	3PTY								
SIP selection											
criteria											
ISUP selection											
criteria											
Test purpose		e IUT	(controlling th	ne conferen	ice) on a 3PTY	call	can successfully create				
	private communication with one of the remote users. The appropriate notification (depending on A-B active-held or A-C active-idle connection) is sent in <b>CPG</b> messages to										
	the two users										
SIP parameter values											
ISDN parameter values											
Comments	ISDN		SUT	SIP-I 1			SIP-I 2				
	SETUP	<b>→</b>	<b>→</b>	INVITE(I	AM)						
	ALERTING	+	<b>←</b>	180 Ring	jing(ACM)						
	CONN	+	<b>←</b>	200 OK I	NVITE(ANM)						
	HOLD	→  -		INVITE(0	INVITE(CPG hold)						
			<b>←</b>	200 OK I	NVITE						
			→	ACK							
	SETUP	<b>→</b>				<b>→</b>	INVITE(IAM)				
	ALERTING	+				+	180 Ringing(ACM)				
	CONN	+				+	200 OK INVITE(ANM)				
	FAC(est3pty)	<b>→</b>	<b>→</b>	INFO(CF	PG conf est)						
			<del>(</del>	200 OK I	NFO						
						<b>→</b>	INFO(CPG conf est)				
						+	200 OK INFO				
			3	PTY com	munication		•				
	FAC(end3pty)	<b>→</b>	<b>→</b>	INFO(CF	PG conf disc)						
	FAC(ret res)	+	+	200 OK I	NFO						
	,					<b>→</b>	INFO(CPG conf disc)				
						+	200 OK INFO				
			•	Com	munication IS	DN -	SIP-I 2				
	DISC	+	+	BYE(REI	L)						
	RELEASE	<b>→</b>	<b>→</b>	200 OK I							
	REL_COM	+									
	DISC	<b>→</b>				<b>→</b>	BYE(REL)				
	RELEASE	+				+	200 OK BYE				
	REL_COM	<b>→</b>									

TP715003	SIP reference: RFC 3261				Į;		reference:	
	2					Q.1912.5 .5.2.1.1.3 b)/Q.734.2		
TSS reference	ISDN-(ISUP)-SIP/SS/3PTY							
SIP selection								
criteria								
SUP selection								
criteria								
Test purpose	Served user disconnects one remote user and retains the other To verify that the IUT (controlling the conference) on a 3PTY call can successfully disconnect one remote user and retain and notify the other user appropriately using CPG messages The IUT should send to the appropriate remote users CPG messages with a generic notification indicator (depending on A-B active-held or A-C active-idle connection). The event indicator in the CPG should be set to "progress" (see note).							
SIP parameter								
values								
ISDN parameter								
values		1		Ta.a			T=.=	
Comments	ISDN		SUT	SIP-I 1			SIP-I 2	
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAI				
	ALERTING	+	+	180 Ringin				
	CONN	<b>←</b>	<del>(</del>	200 OK IN	VITE(ANM)			
	HOLD	<b>→</b>		INIVITE (OF	)C h al d\			
	HOLD	7	<b>→</b>	INVITE(CF				
			<b>←</b>	200 OK IN	VIIE			
	OFTUD	+	7	ACK			IN IV (ITT (I A B 4)	
	SETUP	<b>→</b>		_		<b>→</b>	INVITE(IAM)	
	ALERTING					<del>(</del>	180 Ringing(ACM)	
	CONN	<b>+</b>	_			+	200 OK INVITE(ANM)	
	FAC(est3pty)	<b>→</b>	<b>→</b>	INFO(CPG				
			+	200 OK IN	FO	<u> </u>		
						<b>→</b>	INFO(CPG conf est)	
						<b>←</b>	200 OK INFO	
			3	PTY commu	unication		1	
	DISC	<b>→</b>				<b>→</b>	BYE(REL)	
	RELEASE	+		1		+	200 OK BYE	
	REL_COM	<b>→</b>	<b>→</b>	INFO(CPG		ļ		
			+	200 OK IN				
			→	INFO(CPG				
			+	200 OK IN	FO	<u> </u>		
	DISC	+	+	BYE(REL)				
	RELEASE	<b>→</b>	→	200 OK BY	Έ <u></u>			
	REL_COM	+						

TP715004	SIP reference: RFC 3261					Q	reference: .1912.5	
	2.5.2.1.1.3/Q.734.2							
TSS reference	ISDN-(ISUP)-S	ISDN-(ISUP)-SIP/SS/3PTY						
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	To verify that the two remote The IUT should indicator (depe	Served user disconnects both remote users and terminates the call To verify that the IUT (controlling the conference) can send the appropriate notification to the two remote users when disconnecting both remote users on the 3PTY call The IUT should send to the appropriate remote users a CPG with a generic notification indicator (depending on A-B active-held or A-C active-idle connection). The event indicator in the CPG is set to "progress"						
SIP parameter values			•	<u> </u>				
ISDN parameter								
values								
Comments	ISDN		SUT	SIP-I 1			SIP-I 2	
	SETUP(CRx)	<b>→</b>	<b>→</b>		AM)			
	ALERTING	+	+		ging(ACM)			
	CONN	+	-		INVITE(ANM)			
	HOLD	<b>→</b>	<b>→</b>	INVITE(	CPG hold)			
			+	200 OK	INVITE			
			<b>→</b>	ACK				
	SETUP(CRy)	<b>→</b>				<b>→</b>	INVITE(IAM)	
	ALERTING	+				+	180 Ringing(ACM)	
	CONN	+				+	200 OK INVITE(ANM)	
	FAC(est3pty)	<b>→</b>	<b>→</b>		PG conf est)			
			<b>(</b>	200 OK	INFO			
						<b>→</b>	INFO(CPG conf est)	
						+	200 OK INFO	
				3 PTY com	munication			
	DISC(CRx)	<b>→</b>	<b>→</b>					
	RELEASE	+	+	200 OK	BYE			
	REL_COM	<b>→</b>				<b>→</b>	INFO(CPG conf disc)	
						+	200 OK INFO	
	DISC(CRy)	<b>→</b>				<b>→</b>	BYE(REL)	
	RELEASE	+				+	200 OK BYE	
	REL_COM	<b>→</b>						

TP715005		reference: RFC 3261			ISUP reference: Q.1912.5 2.2.1/Q.734.2		
TSS reference	ISDN-(ISUP)-S	IP/SS/	3PTY				
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	To verify that the after receiving on the lut should (depending on the lut).	Remote user disconnects 3PTY call To verify that the IUT (controlling the conference) can successfully continue the 3PTY call after receiving disconnection by one of the remote users, and send the appropriate notification to the remaining party The IUT should send to the other remote user CPG with a generic notification indicator (depending on A-B active-held or A-C active-idle connection). The event indicator in the CPG is set to "progress" (see note).					
SIP parameter	'			<u>/</u>			
values							
ISDN parameter							
values							
Comments	ISDN		SUT	SIP-I 1			SIP-I 2
	SETUP	<b>→</b>	→				
	ALERTING	+	<b>←</b>		ging(ACM)		
	CONN	+	←	200 OK	INVITE(ANM)		
	HOLD	<b>→</b>	→		CPG hold)		
			<b>←</b>		INVITE		
			→	ACK			
	SETUP	→				<b>→</b>	INVITE(IAM)
	ALERTING	+				<b>←</b>	180 Ringing(ACM)
	CONN	+				+	200 OK INVITE(ANM)
	FAC(est3pty)	<b>→</b>	→		PG conf est)		
			<b>+</b>	200 OK	INFO		
						<b>→</b>	INFO(CPG conf est)
						+	200 OK INFO
				3 PTY com	munication		
	DISC	+				<b>←</b>	BYE(REL)
	RELEASE	<b>→</b>				<b>→</b>	200 OK BYE
	REL_COM	+	→		GP (conf disc)		
			+				
			→	INFO(C	GP (hold)		
			+	200 OK	INFO		
	DISC(CRx)	<b>→</b>	→				
	RELEASE	+	<b>+</b>				
	REL COM	<b>→</b>					
NOTE: The "rem		tion sh	ould be sen	t in a CPG t	o the other rem	ote u	ser, followed by the

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

TP715006	SIP reference:	RFC 3261		ISUP reference: Q.1912.5 2.4; 2.2.1/Q.734.2			
TSS reference	ISDN-(ISUP)-SIP/SS/3P	TY	•				
SIP selection criteria							
ISUP selection criteria							
Test purpose	on to the access signalling	n receive the ng system. T ollowing noti ward direction ed"	he IUT should be fications in the <b>g</b> e	able t	related to 3PTY, and pass it to transparently transfer the notification indicator in both		
SIP parameter values	,						
ISDN parameter values							
Comments	ISDN		SUT		SIP		
	SETUP	+		+	INVITE(IAM)		
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)		
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)		
			Communication				
				+	INVITE(CPG hold)		
				<b>→</b>	200 OK INVITE		
				+	ACK		
	NOTIFY(conf est)	+		+	INFO(CPG conf est)		
				<b>→</b>	200 OK INFO		
		3 P	TY communicat	ion			
	NOTIFY(conf disc)	+		<b>+</b>	INFO(CPG conf disc)		
	,			<b>→</b>	200 OK INFO		
	NOTIFY(hold)	+		+	INFO(CPG hold)		
	, ,			<b>→</b>	200 OK INFO		
	DISC	+		+	BYE(REL)		
	REL	→		<b>→</b>	200 OK BYE		
	REL_COM	+					

## History

	Document history					
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