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Technical Specification

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

Reference

RTS/TISPAN-06028-3-NGN-R1

Keywords

BICC, ISUP, SIP, testing, TSS&TP

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Contents

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Foreword

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

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

Part 1: "Protocol Implementation Conformance Statement (PICS)";

Part 2: "Test Suite Structure and Test Purposes (TSS&TP) for Profile A and B";

Part 3: "Test Suite Structure and Test Purposes (TSS&TP) for Profile C";

Part 4: "Abstract Test Suite (ATS) and partial Protocol Implementation eXtra Information for Testing (PIXIT) for Profiles A and B";

Part 5: "Abstract Test Suite (ATS) and partial Protocol Implementation eXtra Information for Testing (PIXIT) for Profile C".

1 Scope

The present document specifies the network Test Suite Structure and Test Purposes (TSS and TP) Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control Protocol (BICCP)or 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.

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2.1 Normative references

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

[1] ITU-T Recommendation Q.1912.5: "Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part".

[2] ETSI EN 383 001: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control (BICC) Protocol or ISDN User Part (ISUP) [ITU-T Recommendation Q.1912.5, modified]".

[3] ITU-T Recommendation Q.850 (1998): "Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part".

[4] IETF RFC 3261 (2002): "SIP: Session Initiation Protocol".

[5] IETF RFC 3312 (2002): "Integration of Resource Management and Session Initiation Protocol (SIP)".

[6] ISO/IEC 9646-1 (1994): "Information technology - Open Systems Interconnection - Conformance testing methodology and framework - Part 1: General concepts".

[7] ISO/IEC 9646-3 (1992): "Information technology - Open Systems Interconnection - Conformance testing methodology and framework - Part 3: The Tree and Tabular Combined Notation (TTCN)".

[8] ISO/IEC 9646-7 (1995): "Information technology - Open Systems Interconnection - Conformance testing methodology and framework - Part 7: Implementation Conformance Statements".

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

2.2 Informative references

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

[i.1] ITU-T Recommendation Q.730: "ISDN user part supplementary services".

[i.2] ITU-T Recommendation Q.731: "Stage 3 description for the number identification supplementary services using SS No.7".

[i.3] ITU-T Recommendation Q.731.7: "Malicious call identification (MCID)".

[i.4] ITU-T Recommendation Q.732: "Call diversion services".

[i.5] ITU-T Recommendation Q.732.7: "Explicit Call Transfer".

[i.6] ITU-T Recommendation Q.733: "Stage 3 description for call completion supplementary services using Signalling System No. 7: Terminal portability (TP)".

[i.7] ITU-T Recommendation Q.734: "Stage 3 description for multiparty supplementary services using Signalling System No. 7 : Conference calling".

[i.8] ITU-T Recommendation Q.734.2: "Three-party service".

[i.9] ITU-T Recommendation Q.735: "Closed user group (CUG)".

[i.10] ITU-T Recommendation Q.737: "User-to-user signalling (UUS)".

[i.11] ITU-T Recommendation Q.784: "ISUP basic call test specification".

[i.12] ITU-T Recommendations Q.764: "Signalling System No. 7 - ISDN User Part signalling procedures".

3 Definitions and abbreviations

3.1 Definitions

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

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

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

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

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

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

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

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

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

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

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

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

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

SIP number: number conforming to the numbering and structure specified in ITU-T Recommendation E.164 [9]

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

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

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

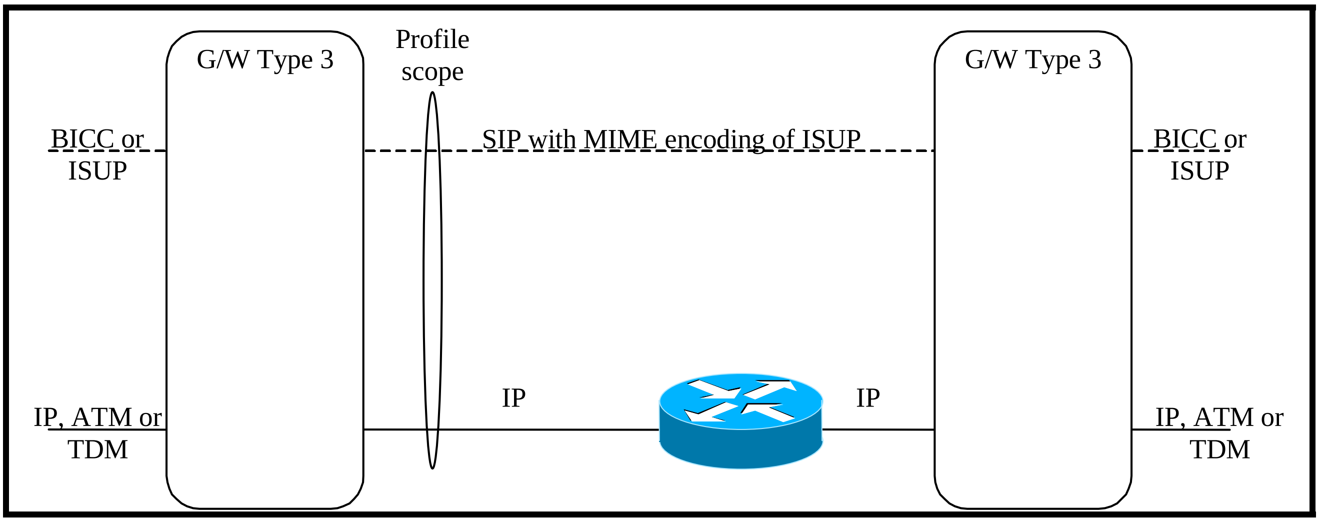

Figure 1: Profile scope for SIP with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateways

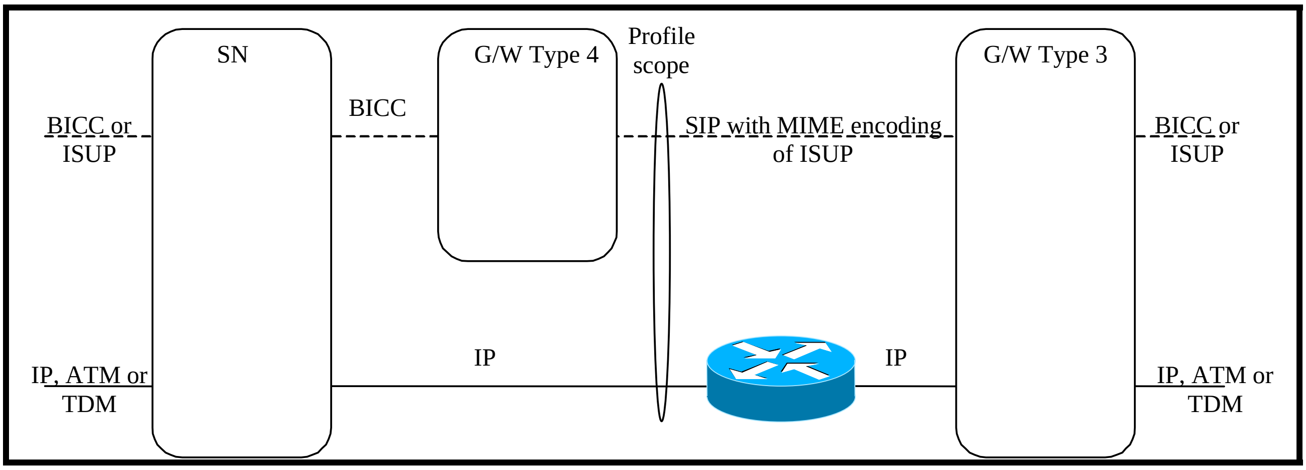

Figure 2: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 3 and 4 gateways

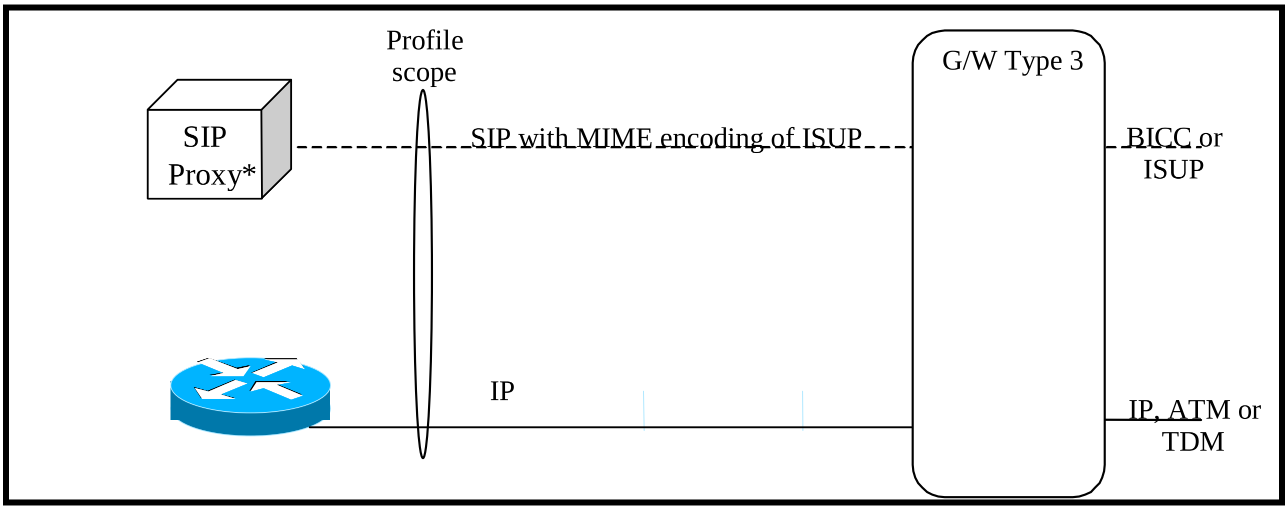

Figure 3: Profile scope for SIP with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateways

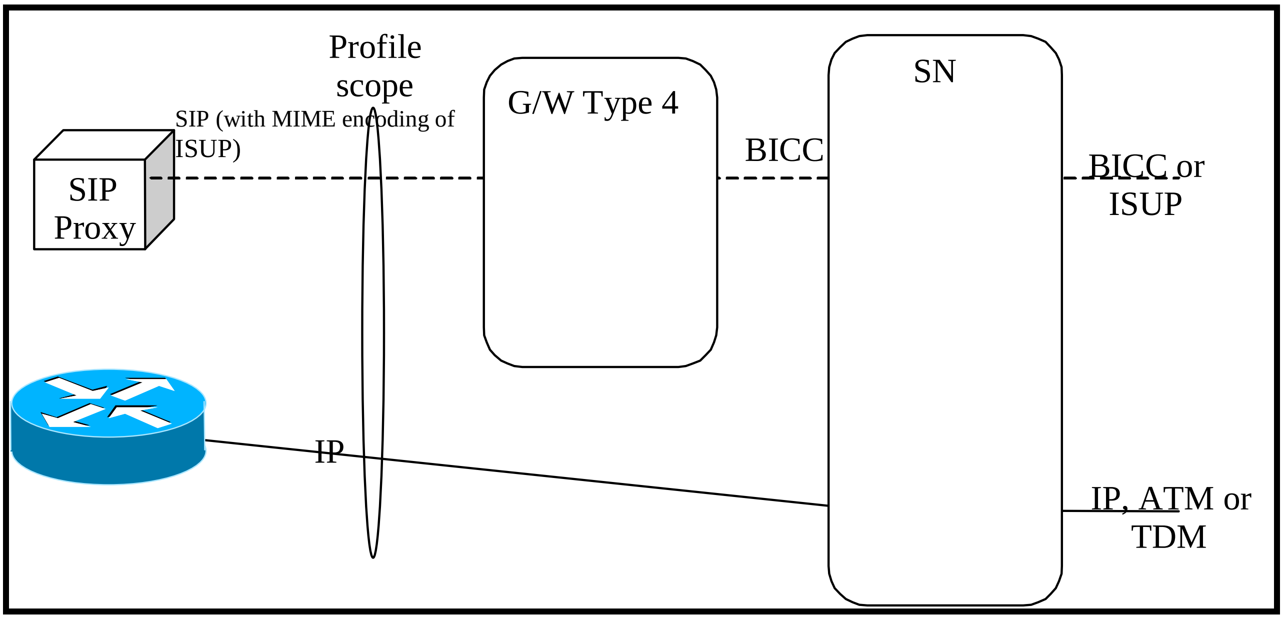

Figure 4: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 4 gateway

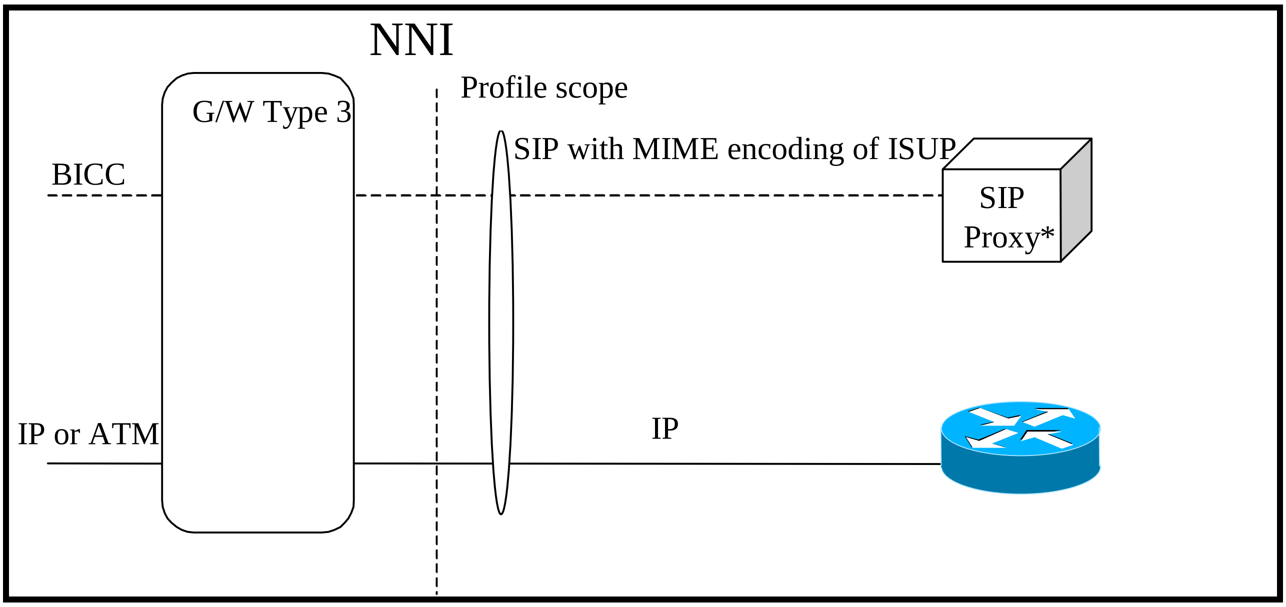

Figure 5: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateway

3.2 Abbreviations

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

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

BC	Bearer Capability
BCI	Backward Call Indicators
BICC	Bearer Independent Call Control protocol
BICCP	Bearer Independent Call Control Protocol
BLA	BLocking Acknowledgement message
BLO	BLOcking message
CC	Country Code
CCBS	Completion of Communication to Busy Subscriber
CD	Call Deflection
CDIV	Call DIVersion
CFB	Call Forwarding Busy
CFN	ConFusioN message
CFNR	Communications Forwarding No Reply
CFU	Call Forwarding Unconditional
CGB	Circuit Group Blocking
CGBA	Circuit Group Blocking Acknowledgement message
CGU	Circuit Group Unblocking message
CGUA	Circuit Group Unblocking Acknowledgement message
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
COL	COnnected Line
COLP	COnnected Line identification Presentation
COLR	COnnected Line identification Restriction
CON	CONnect message
CONF	CONFerence calling
COT	COnTinuity message
CPG	Call Progress Message
CPS	Calling Party's Category
CTNb	ConnecTed Number
CUG	Closed User Group
CW	Call Waiting
DISC	DISConnect message
DLE	Destination Local Exchange
DSS1	Digital Subscriber System no. 1
ECT	Explicit Call transfer
FAA	FAcility Accepted message
FAC	FACility message
FAR	FAcility Request message
FCI	Forward Call Indicators
FRJ	Facility ReJect message
GRA	circuit Group Reset Acknowledgement message
GRS	Group ReSet
HLC	High Layer Compatibility
HOLD	Call HOLD
IA	Incomming Access
IAM	Initial Address Message
ICB	Incomming Call Barred
IDR	IDentification Request message
I-IWU	Incoming InterWorking Unit
I-MGCF	Incoming Media Gateway Control Function
IRS	Identification ResponSe message
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
ITU	International Telecommunication Union
IUT	Implementation Under Test
LOP	LOop Prevention message
MCID	Malicious Call IDentification
MGCF	Media Gateway Control Function
MIME	Multi-purpose Internet Mail Extension
MOT	Means Of Testing
NCI	Nature of Connection Indicators
NDC	National Destination Code

OA	Outgoing Access
OBCI	Optional Backward Call Indicators
O-IWU	Outgoing InterWorking Unit
OLE	Originating Local Exchange
O-MGCF	Outgoing Media Gateway Control Function
OSI	Open Systems Interconnection
PCMA	Pulse Code Modulation A-law
PCMU	Pulse Code Modulation µ-law
PCO	Point of Control and Observation
PICS	Protocol Implementation Conformance Statement
PIXIT	Protocol Implementation eXtra Information for Testing
PT	Pay load Type
PTC	Parallel Test Component
REL	RELease message
RES	RESUME
RLC	ReLease Complete message
RSC	ReSet Circuit
RTP	Real Time Protocol
SAM	Subsequent Address Message
SDP	Session Description Protocol
SGM	SeGmentation Message
SIP	Session Initiation Protocol
SIP-I	Session Initiation Protocol with encapsulated ISUP
SN	Subscriber Number
SS	Supplementary Services
SUB	SUBaddressing
SUS	SUSPEND
SUT	System Under Test
TMR	Transmission Medium Requirement
TON	Type Of Number
TP	Test Purpose
TSS	Test Suite Structure
UNI	User-Network Interface
UPA	User Part Available message
UPT	User Part Test message
URI	Uniform Resource Identifier
USI	User Service Information parameter
USR	User-to User message
UUS	User to User Signalling

4 Test Suite Structure (TSS)

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

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

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

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

4.3 Supplementary services supported by encapsulation

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

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

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

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

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

4.6 Interworking SIP-I/ISDN Supplementary Services

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

5 Test Purposes (TP)

5.1 Introduction

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

5.1.1 Test Purpose (TP) naming convention

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

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

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

5.1.2 Source of test purpose definition

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

5.1.3 Test purpose structure

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

5.2 Test purposes for the basic cal

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

5.2.1.1 Sending of the Initial Address Message (IAM)

TP101001	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clause 6.1.2 (i,1)**		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	NOT PICS 4/4 AND NOT PICS 4/5				
ISUP selection criteria	NOT PICS 1/6				
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, **with a SDP offer:** • the SUT shall delete µ-law (PCMU), if present, from the media description that it will send back in the SDP answer; • the SUT shall immediately send out the IAM.				
SIP parameter values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 8				
ISUP parameter values	IAM USI: A-law or absent				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101002	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clause 6.1.2 (i,2ai)**		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	PICS 4/4 AND PICS 4/5				
ISUP selection 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 values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 8				
ISUP parameter values	IAM Continuity Indicator: **COT to be expected**, USI: A-law or absent COT; Continuity Indicator: **continuity**				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
			Preconditions met		
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP101003</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 6.1.2 (i,2ai)</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer 100rel extensions and preconditions extensions in the SIP Require header:
• the SUT shall delete µ-law (PCMU), if present, from the media description that it will send back in the SDP answer;
• the IAM shall be sent out immediately on the BICC side with the coding of the Nature of Connection Indicators parameter: "COT to be expected".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">SIP INVITE: Audio RTP/AVP 0 8
200 OK: Audio RTP/AVP 8</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM Continuity Indicator: COT to be expected, USI: A-law or absent
COT; Continuity Indicator: continuity</td></tr>
<tr><td rowspan="14">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>COT</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Preconditions met</td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
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Q.1912.5 [1], clause 6.1.2 (i,2aii)</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer 100rel extensions and preconditions extensions in the SIP Supported header:
• the SUT shall delete µ-law (PCMU), if present, from the media description that it will send back in the SDP answer;
• the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "continuity check required on this circuit" or set to "continuity check performed on previous circuit".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">SIP INVITE: Audio RTP/AVP 0 8
200 OK: Audio RTP/AVP 8</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM Continuity Indicator: continuity check required on this circuit or continuity check performed on previous circuit, USI: A-law or absent
COT Continuity Indicator: continuity check successful</td></tr>
<tr><td rowspan="14">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>183 Session Progress</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>COT</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Preconditions met</td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP101005	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clause 6.1.2 (i,2aii)**		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	PICS 4/4 AND PICS 4/5				
ISUP selection criteria	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, **with a SDP offer** 100rel extensions and preconditions extensions in the SIP Require header: • the SUT shall delete μ-law (PCMU), if present, from the media description that it will send back in the SDP answer; • the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "**continuity check required on this circuit**" or set to "**continuity check performed on previous circuit**".				
SIP parameter values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 8				
ISUP parameter values	IAM Continuity Indicator: **continuity check required on this circuit or continuity check performed on previous circuit**, USI: A-law or absent COT Continuity Indicator: **continuity check successful**				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	→			
	PRACK	←			
	200 OK PRACK	→			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
			Preconditions met		
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101006	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clause 6.1.2 (i,2b)**		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	PICS 4/4 AND PICS 4/5				
ISUP selection criteria	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 deferred until all preconditions have been met.				
SIP parameter values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 8				
ISUP parameter values	IAM USI: A-law or absent				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	IAM
	200 OK UPDATE	←			
			Preconditions met		
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP101007</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 6.1.2 (i,2b)</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">NOT PICS 1/6 AND PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer 100rel extensions and preconditions extensions in the SIP Require header:
• the SUT shall delete µ-law (PCMU), if present, from the media description that it will send back in the SDP answer;
• the IAM shall be deferred until all preconditions have been met.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">SIP INVITE: Audio RTP/AVP 0 8
200 OK: Audio RTP/AVP 8</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM USI: A-law or absent</td></tr>
<tr><td rowspan="14">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Preconditions met</td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP101008</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 6.1.2 (i,1)</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/4 AND NOT 4/5</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/6</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer:
• 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 SUT shall immediately send out the IAM.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">SIP INVITE: Audio RTP/AVP 0 8
200 OK: Audio RTP/AVP 0</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM USI: µ-law</td></tr>
<tr><td rowspan="8">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP101009	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 6.1.2 (i,2ai)
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	PICS 4/4 AND PICS 4/5				
ISUP selection criteria	PICS 1/4 AND PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, **with a SDP offer** 100rel extensions and preconditions extensions in the SIP Supported header: • the SUT shall delete A-law (PCMA) if both A-law (PCMA) and µ-law (PCMU) were present in the offer of the media description, that it will send it back in the SDP answer: • the IAM shall be sent out immediately on the BICC side with the coding of the Nature of Connection Indicators parameter: "**COT to be expected**".				
SIP parameter values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 0				
ISUP parameter values	IAM USI: µ-law; Nature of Connection Indicators parameter: "**COT to be expected**" COT; Continuity Indicator: **continuity**				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
		Preconditions met			
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP101010</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 6.1.2 (i,2ai)</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4 AND PICS 1/6 AND PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="5">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".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">SIP INVITE: Audio RTP/AVP 0 8
200 OK: Audio RTP/AVP 0</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM USI: µ-law; Nature of Connection Indicators parameter: "COT to be expected" COT; Continuity Indicator: continuity</td></tr>
<tr><td rowspan="14">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>COT</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Preconditions met</td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP101011	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause 6.1.2 (i,2aii)
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/	
SIP selection criteria	PICS 4/4 AND PICS 4/5	
ISUP selection criteria	PICS 1/5 AND PICS 1/6 AND PICS 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 A-law (PCMA) if both A-law (PCMA) and µ-law (PCMU)) were present in the offer of the media description, that it will send it back in the SDP answer; • the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "**continuity check required on this circuit**" or is set to "**continuity check performed on previous circuit**".	
SIP parameter values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 0	
ISUP parameter values	IAM: USI: µ-law; Continuity check indicator "**continuity check required on this circuit**" or **continuity** check performed on previous circuit COT: Continuity Indicator: **continuity check successful**	
Comments		

SIP-I		SUT		ISUP
INVITE(IAM)	→		→	IAM
183 Session Progress	←			
PRACK	→			
200 OK PRACK	←			
UPDATE	→		→	COT
200 OK UPDATE	←			
		Preconditions met		
180 Ringing(ACM)	←		←	ACM
200 OK INVITE(ANM)	←		←	ANM
ACK	→			
		Conversation		
BYE(REL)	→		→	REL
200 OK BYE(RLC)	←		←	RLC

TP101012	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 6.1.2 (i,2aii)
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	PICS 4/4 AND PICS 4/5				
ISUP selection criteria	PICS 1/5 AND PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, **with a SDP offer** 100rel extensions and preconditions extensions in the SIP Require header: • the SUT shall delete A-law (PCMA) if both A-law (PCMA) and µ-law (PCMU)) were present in the offer of the media description, that it will send it back in the SDP answer; • the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "**continuity check required on this circuit**" or is set to "**continuity check performed on previous circuit**".				
SIP parameter values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 0				
ISUP parameter values	IAM: USI: µ-law; Continuity check indicator "**continuity check required on this circuit**" **continuity** check performed on previous circuit COT: Continuity Indicator: **continuity check successful**				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
			Preconditions met		
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101013	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 6.1.2 (i,2b)	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	PICS 4/4 AND PICS 4/5				
ISUP selection criteria	PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, **with a SDP offer** 100rel extensions and preconditions extensions in the SIP Supported header: • the SUT shall delete A-law (PCMA) if both A-law (PCMA) and µ-law (PCMU) were present in the offer of the media description, that it will send it back in the SDP answer; • the IAM shall be deferred until all preconditions have been met.				
SIP parameter values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 0				
ISUP parameter values	IAM USI: µ-law				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	IAM
	200 OK UPDATE	←			
			Preconditions met		
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101014	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 6.1.2 (i,2b)	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	PICS 4/4 AND PICS 4/5				
ISUP selection criteria	PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, **with a SDP offer** 100rel extensions and preconditions extensions in the SIP Require header: • the SUT shall delete A-law (PCMA) if both A-law (PCMA) and µ-law (PCMU) were present in the offer of the media description, that it will send it back in the SDP answer; • the IAM shall be deferred until all preconditions have been met.				
SIP parameter values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 0				
ISUP parameter values	IAM USI: µ-law				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	IAM
	200 OK UPDATE	←			
			Preconditions met		
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101015	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 6.1.3.2 Q.1912.5 [1], clause 6.1.3.3 Q.1912.5 [1], clause 6.1.3.4
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria					
ISUP selection criteria	NOT PICS 1/9 AND NOT PICS 4/4 and NOT PICS 4/5				
Test purpose	Ensure that the SUT on receipt of an INVITE message sends an IAM message, where: • the **Calling party's category** is generated from the Calling Party's Category present in the encapsulated IAM; • the **Nature of Connection Indicators (NCI)** is generated by the MGCF using the Nature of Connection Indicators received in the encapsulated IAM; • the appropriate values of the **Forward Call Indicator** parameter are generated by the MGCF using the Forward Call Indicators parameter present within the received encapsulated IAM.				
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

P101016	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 6.1.3.5
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	NOT PICS 4/4 and NOT PICS 4/5				
ISUP selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message with an encapsulated IAM message. The TMR and USI shall be taken from the encapsulated ISUP: • sends an IAM message, with the Transmission Medium Requirement (TMR) taken from the encapsulated ISUP.				
SIP parameter values	SIP INVITE				
ISUP parameter values	IAM; USI; ISDN_BC_ITR; TMR				
Comments	**SIP-I**		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

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 ISDN_BC_ITR = 64 kbits/s unrestricted	ISUP_TMR = 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

<table>
<tr><th>TP101017</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clause 6.1.3.5</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/4 and NOT PICS 4/5</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in the Idle state on receipt of a INVITE message with an encapsulated IAM message the HLC shall be taken from the encapsulated ISUP:
• sends an IAM message, with the HLC taken from the encapsulated ISUP.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE ;</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM; Access transport parameter HLC: HLC_VALUE; USI</td></tr>
<tr><td rowspan="8">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

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

TP101018	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 6.1.3.9**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	NOT PICS 4/4 and NOT PICS 4/5				
ISUP selection criteria	PICS 4/3				
Test purpose	Ensure that the MGCF acting as an independent exchange and shall perform the normal BICC/ISUP Hop Counter procedure using the Hop Counter taken from the encapsulated IAM if the Hop Counter parameter is available. The initial and successively mapped values of Hop Counter should be large enough to accommodate the maximum number of hops that might be expected of a validly routed call.				
SIP parameter values	Max-Forwards header				
ISUP parameter values	IAM: Hop Counter parameter value				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101019	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 6.1.3.1**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	PICS 1/9 AND NOT PICS 4/4 and NOT PICS 4/5				
ISUP selection criteria	NOT PICS 1/7				
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		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101020	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clause 6.1.3.1**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/	
SIP selection criteria	PICS 1/9 AND NOT PICS 4/4 and NOT PICS 4/5	
ISUP selection criteria	PICS 1/7	
Test purpose	Ensure that the SUT on receipt of an INVITE message with a Called party number contained in the user info component of the Request-URI. Send an IAM Message with the called party number coded as follows: • Nature of address indicator: Analyse the information contained in received URI with user=phone, and if it is in the format: **+CC NDC SN** where CC is not the country code of the network in which the next hop terminates, then set Nature of Address indicator to "**International number**", remove "+" and use the remaining digits to fill the Address signals. • Internal Network Number Indicator: routing to internal network number not allowed • **Numbering plan Indicator 001 ISDN (Telephony) numbering plan** • Address Signals **CC NDC SN**	
SIP parameter values		
ISUP parameter values	IAM: Called party number	
Comments		

SIP-I		SUT		ISUP
INVITE(IAM)	→		→	IAM
180 Ringing(ACM)	←		←	ACM
200 OK INVITE(ANM)	←		←	ANM
ACK	→			
		Conversation		
BYE(REL)	→		→	REL
200 OK BYE(RLC)	←		←	RLC

TP101021	**SIP reference: RFC 3261 [4]**	**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/	
SIP selection criteria	NOT PICS 4/4 AND NOT PICS 4/5 AND PICS 1/9	
ISUP selection criteria		
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for µ-Law and a-Law, **then independent from the received order of preference:** • the G.711 a-law codec shall be returned in the SDP answer as preferred codec.	
SIP parameter values	Offer: m=audio 4711 RTP/AVP 0 8 Answer: m=audio 4712 RTP/AVP 8 0	
ISUP parameter values		
Comments		

SIP-I		SUT		ISUP
INVITE(IAM)	→		→	IAM
180 Ringing(ACM)	←		←	ACM
200 OK INVITE(ANM)	←		←	ANM
ACK	→			
		Conversation		
BYE(REL)	→		→	REL
200 OK BYE(RLC)	←		←	RLC

TP101022	**SIP reference: RFC 3261 [4]**		**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9				
ISUP selection criteria	PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer 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 BICC side with the coding of the Nature of Connection Indicators parameter: "**COT to be expected**"; • the G.711 a-law codec shall be returned in the SDP answer as preferred codec.				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 0 8 Answer: m=audio 4712 RTP/AVP 8 0				
ISUP parameter values	IAM: Continuity Indicator: **COT to be expected**, USI: A-law or absent COT: Continuity Indicator: **continuity**				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101023	**SIP reference: RFC 3261 [4]**		**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9				
ISUP selection criteria	PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for µ-Law and a-Law 100rel extensions and preconditions extensions in the SIP Require header, **then independent from the received order of preference:** • 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**"; • the G.711 a-law codec shall be returned in the SDP answer as preferred codec.				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 0 8 Answer: m=audio 4712 RTP/AVP 8 0				
ISUP parameter values	IAM: Continuity Indicator: **COT to be expected**, USI: A-law or absent COT: Continuity Indicator: **continuity**				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP101024</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 1/9</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT on receipt of an INVITE message with a SDP offer for µ-Law and a-Law 100rel extensions and preconditions extensions in the SIP Supported header:, then independent from the received order of preference:
• the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "continuity check required on this circuit" or " continuity check performed on previous circuit";
• the G.711 a-law codec shall be returned in the SDP answer as preferred codec.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">Offer: m=audio 4711 RTP/AVP 0 8
Answer: m=audio 4712 RTP/AVP 8 0</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: Continuity Indicator: continuity check required on this circuit or continuity check performed on previous circuit, USI: A-law or absent
COT: Continuity Indicator: continuity check successful</td></tr>
<tr><td rowspan="13">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>COT</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP101025</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 1/9</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT on receipt of an INVITE message with a SDP offer for µ-Law and a-Law 100rel extensions and preconditions extensions in the SIP Require header, then independent from the received order of preference:
• the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "continuity check required on this circuit"" continuity check performed on previous circuit";
• the G.711 a-law codec shall be returned in the SDP answer as preferred codec.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">Offer: m=audio 4711 RTP/AVP 0 8
Answer: m=audio 4712 RTP/AVP 8 0</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: Continuity Indicator: continuity check required on this circuit or continuity check performed on previous circuit, USI: A-law or absent
COT: Continuity Indicator: continuity check successful</td></tr>
<tr><td rowspan="13">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>COT</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP101026</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 1/9</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/5 AND NOT PICS 1/6 AND NOT PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT on receipt of an INVITE message with a SDP offer for µ-Law and a-Law 100rel extensions and preconditions extensions in the SIP Supported header:, then independent from the received order of preference:
• the shall be deferred until all preconditions have been met;
• the G.711 a-law codec shall be returned in the SDP answer as preferred codec.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">Offer: m=audio 4711 RTP/AVP 0 8
Answer: m=audio 4712 RTP/AVP 8 0</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="13">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP101027	**SIP reference: RFC 3261 [4]**				**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9				
ISUP selection criteria	PICS 1/5 AND NOT PICS 1/6 AND NOT PICS 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for µ-Law and a-Law 100rel extensions and preconditions extensions in the SIP Require header, **then independent from the received order of preference:** • the shall be deferred until all preconditions have been met; • the G.711 a-law codec shall be returned in the SDP answer as preferred codec.				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 0 8 Answer: m=audio 4712 RTP/AVP 8 0				
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	IAM
	200 OK UPDATE	←			
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

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

TP101029	**SIP reference: RFC 3261 [4]**			**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9				
ISUP selection criteria	PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer 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 BICC side with the coding of the Nature of Connection Indicators parameter: "**COT to be expected**"; • the G.711 a-law codec shall be returned in the SDP answer.				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 8 Answer: m=audio 4711 RTP/AVP 8				
ISUP parameter values	IAM: Continuity Indicator: **COT to be expected**, USI: A-law or absent COT: Continuity Indicator: continuity				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
			Preconditions met		
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101030	**SIP reference: RFC 3261 [4]**			**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9				
ISUP selection criteria	PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer 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 BICC side with the coding of the Nature of Connection Indicators parameter: "**COT to be expected**"; • the G.711 a-law codec shall be returned in the SDP answer.				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 8 Answer: m=audio 4711 RTP/AVP 8				
ISUP parameter values	IAM: Continuity Indicator: **COT to be expected**, USI: A-law or absent COT: Continuity Indicator: **continuity**				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
			Preconditions met		
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP101031</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 1/9</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no µ-Law 100rel extensions and preconditions extensions in the SIP Supported header:, then independent the normal offer answer procedures apply:
• the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "continuity check required on this circuit" or "continuity check performed on previous circuit";
• the G.711 a-law codec shall be returned in the SDP answer.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">Offer: m=audio 4711 RTP/AVP 8
Answer: m=audio 4711 RTP/AVP 8</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: Continuity Indicator: continuity check required on this circuit or continuity check performed on previous circuit, USI: A-law or absent
COT: Continuity Indicator: continuity check successful</td></tr>
<tr><td rowspan="15">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>COT</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Preconditions met</td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP101032</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 1/9</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="5">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 required on this circuit" or "continuity check performed on previous circuit";
• the G.711 a-law codec shall be returned in the SDP answer.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">Offer: m=audio 4711 RTP/AVP 8
Answer: m=audio 4711 RTP/AVP 8</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: Continuity Indicator: continuity check required on this circuit or continuity check performed on previous circuit, USI: A-law or absent
COT: Continuity Indicator: continuity check successful</td></tr>
<tr><td rowspan="14">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>COT</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Preconditions met</td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP101033</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 1/9</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">NOT PICS 1/6 AND NOT PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">Offer: m=audio 4711 RTP/AVP 8
Answer: m=audio 4711 RTP/AVP 8</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="13">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP101034	**SIP reference: RFC 3261 [4]**			**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/			
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9			
ISUP selection criteria	NOT PICS 1/6 AND NOT PICS 4/1			
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no µ-Law 100rel extensions and preconditions extensions in the SIP Require header, **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 values	Offer: m=audio 4711 RTP/AVP 8 Answer: m=audio 4711 RTP/AVP 8			
ISUP parameter values				
Comments				

SIP-I		SUT		ISUP
INVITE(IAM)	→			
183 Session Progress	←			
PRACK	→			
200 OK PRACK	←			
UPDATE	→		→	IAM
200 OK UPDATE	←			
180 Ringing(ACM)	←		←	ACM
200 OK INVITE(ANM)	←		←	ANM
ACK	→			
	Conversation			
BYE(REL)	→		→	REL
200 OK BYE(RLC)	←		←	RLC

TP101035	**SIP reference: RFC 3261 [4]**			**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/			
SIP selection criteria	NOT PICS 4/4 AND NOT PICS 4/5 AND PICS 1/9			
ISUP selection criteria	PICS 1/7			
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer m line **without a-law codec:** • **the u-law codec shall be rejected.**			
SIP parameter values	Offer: m=audio 4711 RTP/AVP 0 m=audio 4712 RTP/AVP 8 Answer: m=audio 0 RTP/AVP 0			
ISUP parameter values				
Comments				

SIP-I		SUT		ISUP
INVITE(IAM)	→		→	IAM
180 Ringing(ACM)	←		←	ACM
200 OK INVITE(ANM)	←		←	ANM
ACK	→			
	Conversation			
BYE(REL)	→		→	REL
200 OK BYE(RLC)	←		←	RLC

TP101036	**SIP reference: RFC 3261 [4]**		**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9				
ISUP selection criteria	PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer m line **without a-law codec** 100rel extensions and preconditions extensions in the SIP Supported header: • 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**"; • **the u-law codec shall be rejected.**				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 0 m=audio 4712 RTP/AVP 8 Answer: m=audio 0 RTP/AVP 0				
ISUP parameter values	IAM: Continuity Indicator: **COT to be expected**, USI: A-law or absent COT: Continuity Indicator: **continuity**				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
		Preconditions met			
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101037	**SIP reference: RFC 3261 [4]**		**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9				
ISUP selection criteria	PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer m line **without a-law codec** 100rel extensions and preconditions extensions in the SIP Require header: • the IAM shall be sent out immediately on the BICC side with the coding of the Nature of Connection Indicators parameter: "**COT to be expected**"; • **the u-law codec shall be rejected.**				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 0 m=audio 4712 RTP/AVP 8 Answer: m=audio 0 RTP/AVP 0				
ISUP parameter values	IAM: Continuity Indicator: **COT to be expected**, USI: A-law or absent COT: Continuity Indicator: **continuity**				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
		Preconditions met			
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP101038</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 1/9</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT on receipt of an INVITE message with a SDP offer m line without a-law codec 100rel extensions and preconditions extensions in the SIP Supported header:
• the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "continuity check required on this circuit" or "continuity check performed on previous circuit";
• the u-law codec shall be rejected.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">Offer: m=audio 4711 RTP/AVP 0
m=audio 4712 RTP/AVP 8
Answer: m=audio 0 RTP/AVP 0</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: Continuity Indicator: continuity check required on this circuit or continuity check performed on previous circuit, USI: A-law or absent
COT: Continuity Indicator: continuity check successful</td></tr>
<tr><td rowspan="14">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>COT</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Preconditions met</td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP101039	**SIP reference: RFC 3261 [4]**	**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/	
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9	
ISUP selection criteria	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1	
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer m line **without a-law codec** 100rel extensions and preconditions extensions in the SIP Require header: • the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "**continuity check required on this circuit**" or "**continuity check performed on previous circuit**"; • **the u-law codec shall be rejected.**	
SIP parameter values	Offer: m=audio 4711 RTP/AVP 0 m=audio 4712 RTP/AVP 8 Answer: m=audio 0 RTP/AVP 0	
ISUP parameter values	IAM: Continuity Indicator: **continuity check required on this circuit or continuity check performed on previous circuit**, USI: A-law or absent COT: Continuity Indicator: **continuity check successful**	
Comments		

SIP-I		SUT		ISUP
INVITE(IAM)	→		→	IAM
183 Session Progress	←			
PRACK	→			
200 OK PRACK	←			
UPDATE	→		→	COT
200 OK UPDATE	←			
		Preconditions met		
180 Ringing(ACM)	←		←	ACM
200 OK INVITE(ANM)	←		←	ANM
ACK	→			
		Conversation		
BYE(REL)	→		→	REL
200 OK BYE(RLC)	←		←	RLC

TP101040	**SIP reference: RFC 3261 [4]**	**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/	
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9	
ISUP selection criteria	NOT PICS 1/6 AND NOT 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 deferred until all preconditions have been met; • **the u-law codec shall be rejected.**	
SIP parameter values	Offer: m=audio 4711 RTP/AVP 0 m=audio 4712 RTP/AVP 8 Answer: m=audio 0 RTP/AVP 0	
ISUP parameter values		
Comments		

SIP-I		SUT		ISUP
INVITE(IAM)	→			
183 Session Progress	←			
PRACK	→			
200 OK PRACK	←			
UPDATE	→		→	IAM
200 OK UPDATE	←			
		Preconditions met		
180 Ringing(ACM)	←		←	ACM
200 OK INVITE(ANM)	←		←	ANM
ACK	→			
		Conversation		
BYE(REL)	→		→	REL
200 OK BYE(RLC)	←		←	RLC

TP101041	**SIP reference: RFC 3261 [4]**				**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9				
ISUP selection criteria	NOT PICS 1/6 AND NOT PICS 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer m line **without a-law codec** 100rel extensions and preconditions extensions in the SIP Require header: • the IAM shall be deferred until all preconditions have been met; • **the u-law codec shall be rejected.**				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 0 m=audio 4712 RTP/AVP 8 Answer: m=audio 0 RTP/AVP 0				
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	IAM
	200 OK UPDATE	←			
			Preconditions met		
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101042	**SIP reference: RFC 3261 [4]**				**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	NOT PICS 4/4 AND NOT 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 and based on operator policy then:** • **the call is refused with a 415 Unsupported media type response.**				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 8 m= audio 4712 RTP/AVP 8				
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	415 Unsupported media type	←			
	ACK	→			

TP101043	**SIP reference: RFC 3261 [4]**				**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (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 Supported header: **and based on operator policy then:** • **the call is refused with a 415 Unsupported media type response.**				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 8 m= audio 4712 RTP/AVP 8				
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	415 Unsupported media type	←			
	ACK	→			

TP101044	**SIP reference: RFC 3261 [4]**				**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (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 values	Offer: m=audio 4711 RTP/AVP 8 m= audio 4712 RTP/AVP 8				
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	415 Unsupported media type	←			
	ACK	→			

<table>
<tr><th>TP101045</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="6">SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6">NOT PICS 4/4 AND NOT PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">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;
• if the SDP offer contains several audio type media streams, the IWU shall only consider one, and reject the other streams.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">Offer: m=audio 4711 RTP/AVP 8
m= audio 4712 RTP/AVP 8
m= video 4713 RTP/AVP 31

Answer: m=audio 4711 RTP/AVP 8
m=audio 0 RTP/AVP 8
m=video 0 RTP/AVP 31</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6"></td></tr>
<tr><td rowspan="8">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td colspan="2">ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td colspan="2">IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td colspan="2">ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td colspan="2">ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="2"></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td colspan="2">REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td colspan="2">RLC</td></tr>
</table>

<table>
<tr><th>TP101046</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="6">SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6">NOT PICS 4/4 AND NOT PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6">PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="6">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">Offer: m=audio 4711 RTP/AVP 8
m= audio 4712 RTP/AVP 8
m= video 4713 RTP/AVP 31

Answer: m=audio 4711 RTP/AVP 8
m=audio 0 RTP/AVP 8
m=video 0 RTP/AVP 31</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">IAM: Continuity Indicator: COT to be expected, USI: A-law or absent
COT: Continuity Indicator: continuity</td></tr>
<tr><td rowspan="14">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td colspan="2">ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td colspan="2">IAM</td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td colspan="2">COT</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td colspan="2">Preconditions met</td><td></td><td colspan="2"></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td colspan="2">ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td colspan="2">ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="2"></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td colspan="2">REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td colspan="2">RLC</td></tr>
</table>

<table>
<tr><th>TP101047</th><th>SIP reference: RFC 3261 [4]</th><th colspan="5">ISUP reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="6">SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6">PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6">PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="6">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 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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">Offer: m=audio 4711 RTP/AVP 8
m= audio 4712 RTP/AVP 8
m= video 4713 RTP/AVP 31

Answer: m=audio 4711 RTP/AVP 8
m=audio 0 RTP/AVP 8
m=video 0 RTP/AVP 31</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">IAM: Continuity Indicator: COT to be expected, USI: A-law or absent
COT: Continuity Indicator: continuity</td></tr>
<tr><td rowspan="14">Comments</td><td colspan="2">SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td colspan="2">INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td colspan="2">183 Session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td colspan="2">PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td colspan="2">200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td colspan="2">UPDATE</td><td>→</td><td></td><td>→</td><td>COT</td></tr>
<tr><td colspan="2">200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td colspan="2"></td><td colspan="2">Preconditions met</td><td></td><td></td></tr>
<tr><td colspan="2">180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td colspan="2">200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td colspan="2">ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td colspan="2"></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td colspan="2">BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td colspan="2">200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP101048	**SIP reference: RFC 3261 [4]**	**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/	
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19	
ISUP selection criteria	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1	
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer **with more than 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 required on this circuit**" or "**continuity check performed on previous circuit**"; • if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected; • if the SDP offer contains several audio type media streams, the IWU shall only consider one, and reject the other streams.	
SIP parameter values	Offer: m=audio 4711 RTP/AVP 8 m= audio 4712 RTP/AVP 8 m= video 4713 RTP/AVP 31 Answer: m=audio 4711 RTP/AVP 8 m=audio 0 RTP/AVP 8 m=video 0 RTP/AVP 31	
ISUP parameter values	IAM: Continuity Indicator: **continuity check required on this circuit or continuity check performed on previous circuit**, USI: A-law or absent COT: Continuity Indicator: **continuity check successful**	
Comments		

SIP-I		SUT		ISUP
INVITE(IAM)	→		→	IAM
183 Session Progress	←			
PRACK	→			
200 OK PRACK	←			
UPDATE	→		→	COT
200 OK UPDATE	←			
	Preconditions met			
180 Ringing(ACM)	←		←	ACM
200 OK INVITE(ANM)	←		←	ANM
ACK	→			
		Conversation		
BYE(REL)	→		→	REL
200 OK BYE(RLC)	←		←	RLC

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

<table>
<tr><th>TP101050</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">NOT PICS 1/6 AND NOT PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams 100rel extensions and preconditions extensions in the SIP Supported header: and based on operator policy then:
• the IAM shall be deferred until all preconditions have been met;
• if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected;
• if the SDP offer contains several audio type media streams, the IWU shall only consider one, and reject the other streams.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">Offer: m=audio 4711 RTP/AVP 8
m= audio 4712 RTP/AVP 8
m= video 4713 RTP/AVP 31

Answer: m=audio 4711 RTP/AVP 8
m=audio 0 RTP/AVP 8
m=video 0 RTP/AVP 31</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="14">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="2">Preconditions met</td><td></td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP101051</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">NOT PICS 1/6 AND NOT PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams 100rel extensions and preconditions extensions in the SIP Require header and based on operator policy then:
• the IAM shall be deferred until all preconditions have been met;
• if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected;
• if the SDP offer contains several audio type media streams, the IWU shall only consider one, and reject the other streams.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">Offer: m=audio 4711 RTP/AVP 8
m= audio 4712 RTP/AVP 8
m= video 4713 RTP/AVP 31

Answer: m=audio 4711 RTP/AVP 8
m=audio 0 RTP/AVP 8
m=video 0 RTP/AVP 31</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="14">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="2">Preconditions met</td><td></td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

5.2.1.2 Sending of the Subsequent Address Message (SAM)

TP102001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 6.2 a)**
TSS reference	SIP-ISUP/Basic call/ Sending of the Subsequent Address Message (SAM)/				
SIP selection criteria	PICS 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 **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		SUT		ISUP
	INVITE	→		→	IAM
	INVITE	→		→	SAM
	INVITE	→		→	SAM
	180 Ringing	←		←	ACM
	200 OK INVITE	←		←	ANM
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP102002	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 6.2 b)**
TSS reference	SIP-ISUP/Basic call/ Sending of the Subsequent Address Message (SAM)/				
SIP selection criteria	PICS 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 **fewer** than the number of digits already accumulated for the call: • then the SUT shall immediately send a **484 Address Incomplete** response for this INVITE; • in this case no SAM is sent to BICC/ISUP procedures.				
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	INVITE(IAM)	→			
	484 Address incomplete	←		→	REL
	ACK	→		←	RLC

5.2.1.3 Sending of COT

TP103001	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 6.3**	
TSS reference	SIP-ISUP/Basic call/COT				
SIP selection criteria	PICS 4/4 AND PICS 4/5				
ISUP selection criteria	PICS 1/4 AND PICS 4/1				
Test purpose	Ensure that the when the SUT determines that all the preconditions on the incoming SIP side have been met and any continuity procedures on the outgoing BICC side have been successfully completed: • the SUT shall send the COT message where the Continuity Indicator in the COT message shall be set to "**Continuity**".				
SIP parameter values					
ISUP parameter values	COT continuity indicator: **Continuity**				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP103002	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 6.3**	
TSS reference	SIP-ISUP/Basic call/ COT				
SIP selection criteria	PICS 4/4 AND PICS 4/5				
ISUP selection criteria	PICS 1/5 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 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: Continuity check successful;				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
			Preconditions met		
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

5.2.1.4 Receipt of the Address Complete Message (ACM)

TP104001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 6.5 2)**
TSS reference	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT on receipt of an ACM message where the Called party status indicator is set to "no indication": • 183 Session Progress response is sent from the I-IWU; • the received ACM is encapsulated in the 183 Session Progress.				
SIP parameter values					
ISUP parameter values	ACM Called party status: no indication;				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress (ACM)	←		←	ACM(no indication)
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP104002	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 6.5 1)**
TSS reference	SIP-ISUP/Basic call/ 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 " subscriber free" where the ISUP indicator parameter set to ISUP_ID, the ISDN access indicator set to ISDN_ACCES_ID and the OBCI in-band information set to OBCI_INBAND then: • the **180 Ringing** SIP response is sent. Ensure that the in-band information can be transmitted to the calling user; • the received ACM is encapsulated in the 180 Ringing.				
SIP parameter values					
ISUP parameter values	ACM FCI: ISUP_ID, ISDN_ACCESS_ID, OBCI: OBCI_INBAND;				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

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)

<table>
<tr><th>TP105001</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 6.6</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the event information parameter event indicator is set to "Alerting":
• the 180 Ringing SIP response is sent;
• The received CPG is encapsulated in the 180 Ringing.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">ACM: Called party status "no indication"
CPG; event information parameter event indicator: Alerting</td></tr>
<tr><td rowspan="9">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>183 Session Progress (ACM)</td><td>←</td><td></td><td>←</td><td>ACM(no indication)</td></tr>
<tr><td>180 Ringing(CPG)</td><td>←</td><td></td><td>←</td><td>CPG(ALERTING)</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP105002</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clause 6.6</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the event information parameter event indicator is set to "Progress":
• 183 Session Progress response is sent from the I-IWU;
• the received CPG is encapsulated in the 183 Session Progress.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">ACM: Called party status "no indication"
CPG; event information parameter event indicator: Progress</td></tr>
<tr><td rowspan="9">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>183 Session Progress (ACM)</td><td>←</td><td></td><td>←</td><td>ACM(no indication)</td></tr>
<tr><td>183 Session (CPG)</td><td>←</td><td></td><td>←</td><td>CPG(PROGRESS)</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP105003</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clause 6.6</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">ACM: Called party status "no indication"
CPG; event information parameter event indicator: in-band-information or an appropriate pattern is now available</td></tr>
<tr><td rowspan="9">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>183 Session Progress (ACM)</td><td>←</td><td></td><td>←</td><td>ACM(no indication)</td></tr>
<tr><td>183 Session (CPG)</td><td>←</td><td></td><td>←</td><td>CPG (Inbad Info available)</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

5.2.1.6 Receipt of the Answer message (ANM)

<table>
<tr><th>TP106001</th><th>SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clause 6.7</th></tr>
<tr><td>TSS reference</td><td colspan="2">SIP-ISUP/Basic call/ Receipt of the Answer message (ANM).</td></tr>
<tr><td>SIP selection criteria</td><td colspan="2"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="2"></td></tr>
<tr><td>Test purpose</td><td colspan="2">Ensure that the SUT, having received the ACM message, on receipt of an ANM message:
• sends a 200 OK INVITE;
• the received ANM is encapsulated in the 200 OK INVITE.
The bearer path shall be connected in both directions when both of the following conditions are satisfied:
• the BICC outgoing bearer set-up procedure, (see ITU-T Recommendation Q.1902.4 [Error! Reference source not found.]) is successfully completed; and
• the I-IWU determines (using the procedures defined in RFC 3312 [5]) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable).
In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "notification not required", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312 [5]) that sufficient preconditions have been met for the session to proceed.</td></tr>
<tr><td>SIP parameter values</td><td colspan="2">200 OK INVITE with encapsulated ANM</td></tr>
<tr><td>ISUP parameter values</td><td colspan="2">ANM</td></tr>
<tr><td>Comments</td><td colspan="2"><table>
<tr><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table></td></tr>
</table>

5.2.1.7 Receipt of the Connect message (CON)

<table>
<tr><th>TP107001</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clause 6.4, 6.7</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Receipt of the CONNECT message (CON).</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">SDP offer was received in the initial INVITE. Ensure that the SUT, on receipt of an CON message:
• sends a 200 OK INVITE;
• the received CON is encapsulated in the 200 OK INVITE.
The bearer path shall be connected in both directions when both of the following conditions are satisfied:
• the BICC outgoing bearer set-up procedure, (see ITU-T Recommendation Q.1902.4 [Error! Reference source not found.]) is successfully completed; and
• the I-IWU determines (using the procedures defined in RFC 3312 [5]) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable).
In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "notification not required", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312 [5]) that sufficient preconditions have been met for the session to proceed.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="7">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>200 OK INVITE(CON)</td><td>←</td><td></td><td>←</td><td>CON</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

5.2.1.8 Receipt of the REL message

<table>
<tr><th>TP108001</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clause 6.11.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP /Basic call/ Receipt of the Release message (REL)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">SIP Statue-Code: SIP_FAILURE_VA (PIXIT)</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">REL; cause value: CV_ISUP (PIXIT)</td></tr>
<tr><td rowspan="4">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>SIP_FAILURE_VA(REL)</td><td>←</td><td></td><td>←</td><td>REL</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>RLC</td></tr>
</table>

Table 1

Values for test purpose TP108001		
←SIP Message SIP_FAILURE_VA		← REL Cause Indicators parameter CV_ISUP
VA_1	404 Not Found	Cause Value No. 1 ("unallocated (unassigned) number")
VA_2	500 Server Internal Error	Cause Value No. 2 ("no route to network")
VA_3	500 Server Internal Error	Cause Value No. 3 ("no route to destination")
VA_4	500 Server Internal Error	Cause Value No. 4 ("Send special information tone")
VA_5	404 Not Found	Cause Value No. 5 ("Misdialled trunk prefix")
VA_6	500 Server Internal Error	Cause Value No. 8 ("Pre-emption")
VA_7	500 Server Internal Error	Cause Value No. 9 ("Pre-emption-circuit reserved for reuse")
VA_8	486 Busy Here	Cause Value No. 17 ("user busy")
VA_9	480 Temporarily unavailable	Cause Value No. 18 ("no user responding")
VA_10	480 Temporarily unavailable	Cause Value No. 19 ("no answer from the user")
VA_11	480 Temporarily unavailable	Cause Value No. 20 ("subscriber absent")
VA_12	480 Temporarily unavailable	Cause Value No. 21 ("all rejected")
VA_13	410 Gone	Cause Value No. 22 ("number changed")
VA_14	480 Temporarily unavailable	Cause Value No. 25 ("Exchange routing error")
VA_15	502 Bad Gateway	Cause Value No. 27 ("destination out of order")
VA_16	484 Address Incomplete	Cause Value No. 28 ("invalid number format (address incomplete")
VA_17	500 Server Internal Error	Cause Value No. 29 ("facility rejected")
VA_18	480 Temporarily unavailable	Cause Value No. 31 ("normal unspecified") (Class default)
VA_19	486 Busy here if Diagnostics indicator includes the (CCBS indicator = CCBS possible) else 480 Temporarily unavailable	Cause Value in the Class 010 (No circuit/channel available, Cause Value No. 34)
VA_20	500 Server Internal Error	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38 to 47) (47 is class default)
VA_21	500 Server Internal Error	Cause Value No. 50 ("requested facility not subscribed")
VA_22	500 Server Internal Error (SIP-I only)	Cause Value No. 55 ("incoming calls barred within CUG")
VA_23	500 Server Internal Error	Cause Value No. 57 ("bearer capability not authorized")
VA_24	500 Server Internal Error	Cause Value No. 58 ("bearer capability not presently")
VA_25	500 Server Internal Error	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_26	500 Server Internal Error	Cause Value in the Class 100 (service or option not implemented Cause Value No. 65 79) (79 is class default)
VA_27	500 Server Internal Error	Cause Value No. 87 ("user not member of CUG")
VA_28	500 Server Internal Error	Cause Value No. 88 ("incompatible destination")
VA_29	500 Server Internal Error	Cause Value No. 90 ("Non-existent CUG")
VA_30	404 Not Found	Cause Value No. 91 ("invalid transit network selection")
VA_31	500 Server Internal Error	Cause Value No. 95 ("invalid message") (Class default)
VA_32	500 Server Internal Error	Cause Value No. 97 ("Message type non-existent or not implemented")
VA_33	500 Server Internal Error	Cause Value No. 99 ("information element/parameter non-existent or not implemented")
VA_34	480 Temporarily unavailable	Cause Value No. 102 ("recovery on timer expiry")
VA_35	500 Server Internal Error	Cause Value No. 103 ("Parameter non-existent or not implemented, pass on")
VA_36	500 Server Internal Error	Cause Value No. 110 ("Message with unrecognized Parameter, discarded")
VA_37	500 Server Internal Error	Cause Value No. 111 ("protocol error, unspecified") (Class default)
VA_38	480 Temporarily unavailable	Cause Value No. 127 ("interworking unspecified") (Class default)

TP108002	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message where the CPS indicator is set to "**no indication**", on receipt of an ISUP REL: • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; • the SUT shall send the appropriate SIP status defined as **SIP_FAILURE_VA** with the encapsulated REL message.				
SIP parameter values	SIP Statue-Code: SIP_FAILURE_VA (PIXIT)				
ISUP parameter values	REL; **cause value**: CV_ISUP (PIXIT)				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress(ACM)	←		←	ACM(no indication)
	SIP_FAILURE_VA(REL)	←		←	REL
	ACK	→		→	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 [4]**			**ISUP reference: Q.1912. [1],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_FAILURE_VA (PIXIT)			
ISUP parameter values	REL; **cause value**: CV_ISUP (PIXIT)			
Comments	SIP-I		SUT	ISUP
	INVITE(IAM)	→	→	IAM
	180 Ringing(ACM)	←	←	ACM
	SIP_FAILURE_VA(REL)	←	←	REL
	ACK	→	→	RLC

TP108004	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 6.11.2**
TSS reference	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message where the CPS indicator is set to "**no indication**", having received a CPG message where the **event information parameter event indicator** is set to "Alerting", a 180 Ringing message is sent, on receipt of an ISUP REL: • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA with the encapsulated REL message.			
SIP parameter values	SIP Statue-Code: SIP_FAILURE_VA (PIXIT)			
ISUP parameter values	REL; **cause value**: CV_ISUP (PIXIT)			
Comments	SIP-I		SUT	ISUP
	INVITE(IAM)	→	→	IAM
	183 Session Progress(ACM)	←	←	ACM(no indication)
	180 Ringing(CPG)	←	←	CPG(ALERTING)
	SIP_FAILURE_VA(REL)	←	←	REL
	ACK	→	→	RLC

Table 3

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

<table>
<tr><td>TP108005</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="3">ISUP reference:
Q.1912.5 [1], clause 6.11.2</td></tr>
<tr><td>TSS reference</td><td colspan="6">SIP-ISUP /Basic call/ Receipt of the Release message (REL)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message, having received a ANM", a 200 OK message is sent, on receipt of an ISUP REL, where the cause value defined as CV_ISUP:
• the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side;
• the SUT shall send a BYE message with the encapsulated REL message.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="6"></td></tr>
<tr><td rowspan="8">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td colspan="2">ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td colspan="2">IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td colspan="2">ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td colspan="2">ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="2"></td></tr>
<tr><td>BYE(REL)</td><td>←</td><td></td><td>←</td><td colspan="2">REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>→</td><td></td><td>→</td><td colspan="2">RLC</td></tr>
</table>

<table>
<tr><td>TP108006</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="3">ISUP reference:
Q.1912.5 [1], clause 6.11.2</td></tr>
<tr><td>TSS reference</td><td colspan="6">SIP-ISUP /Basic call/ Receipt of the Release message (REL)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a CON message , a 200 OK message is sent, on receipt of an ISUP REL, where the cause value defined as CV_ISUP:
• the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side;
• the SUT shall send a BYE message with the encapsulated REL message.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">REL; cause value: CV_ISUP (PIXIT)</td></tr>
<tr><td rowspan="7">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td colspan="2">ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td colspan="2">IAM</td></tr>
<tr><td>200 OK INVITE(CON)</td><td>←</td><td></td><td>←</td><td colspan="2">CON</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="2"></td></tr>
<tr><td>BYE(REL)</td><td>←</td><td></td><td>←</td><td colspan="2">REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>→</td><td></td><td>→</td><td colspan="2">RLC</td></tr>
</table>

Table 4

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

TP108007	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/				
SIP selection criteria	PICS 4/21				
ISUP selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, on receipt of an ISUP REL with cause value 23 the SUT shall: • the SUT immediately requests the redirection to the new destination according the ISUP/BICC procedures.				
SIP parameter values					
ISUP parameter values	REL; **cause value**: 23				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM(Destination 1)
				←	REL(new Destination)
				→	RLC
				→	IAM(Destination 2)
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
		Conversation			
	BYE(REL)	←		←	REL
	200 OK BYE(RLC)	→		→	RLC

5.2.1.9 Autonomous release at I-IWU

TP109001	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 6.11.3
TSS reference	SIP-ISUP/Basic call/ Autonomous release at I-IWU				
SIP selection criteria					
ISUP selection criteria	PICS 4/6				
Test purpose	Ensure that when a an automatic repeat attempt initiated by the SUT is not successful (because the call is not routable), the SUT shall: • send a 480 Temporarily Unavailable response to the SIP side. No actions on the ISUP (BICC) side are required.				
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	480 Temporarily unavailable (REL)	←		←	RSC
	ACK	→		→	RLC

TP109002	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 6.11.3**
TSS reference	SIP-ISUP/Basic call/ Autonomous release at I-IWU				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that when the SUT receives unrecognized backward ISUP or BICC signalling information and determines that the call needs to be released based on the coding, the SUT: • shall send a 500 Server Internal Error response on the SIP side.				
SIP parameter values					
ISUP parameter values	Unknown message: Message compatibility "Release call"				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
				←	???
	500 Server internal error(REL)	←		→	REL
	ACK	→		←	RLC

TP109003	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 6.11.3**
TSS reference	SIP-ISUP/Basic call/ Autonomous release at I-IWU				
SIP selection criteria					
ISUP selection criteria	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					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	484 Address incomplete	←			
	ACK	→			

<table>
<tr><th>TP109004</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 6.11.3</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Autonomous release at I-IWU</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 3/4</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT on receipt of subsequent INVITE message:
• 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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="5">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>484 Address incomplete</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><th>TP109005</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 6.11.3</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Autonomous release at I-IWU</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in congestion on receipt of INVITE message:
• sends an 480 Temporarily Unavailable message.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="4">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>480 Temporarily unavailable</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
</table>

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

TP109007	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clause 6.11.3**		
TSS reference	SIP-ISUP/Basic call/ Autonomous release at I-IWU				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the call is released due to expiry of T7 within the BICC/ISUP procedures: • sends 484 Address Incomplete.				
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
			T7 expiry		
	484 Address incomplete	←		→	REL
	ACK	→		←	RLC

TP109008	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clause 6.11.3**		
TSS reference	SIP-ISUP/Basic call/ Autonomous release at I-IWU				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the call is released due expiry of T9 within the BICC/ISUP procedures: • sends 480 Temporarily Unavailable.				
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
			T9 expiry		
	480 Temporarily unavailable	←		→	REL
	ACK	→		←	RLC

5.2.1.10 Receipt of the Release message BYE / CANCEL

TP110001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 6.11.1**
TSS reference	SIP-ISUP/Basic call/ Receipt of the BYE-CANCEL message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT on receipt of SIP BYE , the SUT shall send an ISUP REL with the cause value # 16 to the ISUP side.				
SIP parameter values					
ISUP parameter values	REL: Cause value #16, Location "Network beyond an interworking point"				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP110002	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 6.11.1**
TSS reference	SIP-ISUP/Basic call/ Receipt of the BYE-CANCEL message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT on receipt of SIP CANCEL, the I-IWU shall send an ISUP REL with the cause value # 31 to the ISUP side.				
SIP parameter values	CANCEL without encapsulated ISUP message				
ISUP parameter values	REL: Cause value #31, Location "Network beyond an interworking point"				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	CANCEL	→		→	REL
	200 OK CANCEL	←		←	RLC
	487 Request Terminated	←			
	ACK	→			

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

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

TP111001	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], 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 BYE message if the SUT has already received an ACK for the 200 OK INVITE message which had it sent.				
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	←		←	RSC
	200 OK BYE(RLC)	→		→	RLC

TP111002	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 6.11.4 and 5
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a GRS message sends: • a BYE message if the SUT has already received an ACK for the 200 OK INVITE message which had it sent.				
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE	←		←	GRS
	200 OK BYE	→		→	GRA

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

TP111004	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 6.11.4 and 5**
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a RSC message sends 200 OK INVITE if the SUT has not yet received an ACK for the 200 OK INVITE: • the SUT shall wait until it receives the ACK for the 200 OK INVITE before sending the BYE.				
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
				←	RSC
	ACK	→		→	RLC
	BYE(REL)	←			
	200 OK BYE(RLC)	→			

TP111005	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], 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 BYE.				
SIP parameter values:					
ISUP parameter values:					
Comments:	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
				←	GRS
	ACK	→		→	GRA
	BYE	←			
	200 OK BYE	→			

TP111006	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 6.11.4**
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
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 Message Type Indicator "hardware failure oriented"				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
				←	CGB(hardware failure)
	ACK	→		→	CGBA
	BYE	←			
	200 OK BYE	→			

TP111007	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 6.11.4 and 5**
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a RSC message sends: • a 500 Server Internal Error on the SIP side.				
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	500 Server Internal Error(REL)	←		←	RSC
	ACK	→		→	RLC

<table>
<tr><th>TP111008</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 6.11.4 and 5</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a GRS message sends:
• a 500 Server Internal Error on the SIP side.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="5">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>500 Server Internal Error</td><td>←</td><td></td><td>←</td><td>GRS</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>GRA</td></tr>
</table>

<table>
<tr><th>TP111009</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 6.11.4</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT, when at least one backward ISUP message relating to the call has already been received on receipt of a CGB message, with the Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented", sends:
• a 500 Server Internal Error on the SIP side.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="5">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>500 Server Internal Error</td><td>←</td><td></td><td>←</td><td>CGB(hardware failure)</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>CGBA</td></tr>
</table>

TP111010	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clause 6.11.4 and 5**		
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving more than one INVITE sending an IAM message for each call association on receipt of a GRS message were the Range and Status Parameter value is bigger than *"1":* • the SUT shall send a BYE requests for each call association.				
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM) 1	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
	INVITE(IAM) 2	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
	BYE 1	←		←	GRS
	200 OK BYE	→		→	GRA
	BYE 2	←			
	200 OK BYE	→			

TP111011	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clause 6.11.4 and 5**		
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving more than one INVITE sending an IAM message for each call association on receipt of a CGB message, with the Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" were the Range and Status Parameter value is bigger than "1": • the SUT shall send a BYE requests for each call association.				
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM) 1	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
	INVITE(IAM) 2	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
	BYE 1	←		←	CGB(hardware failure)
	200 OK BYE	→		→	CGBA
	BYE 2	←			
	200 OK BYE	→			

5.2.1.12 Receipt of the Suspend message (SUS) network initiated

TP112001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 6.9**
TSS reference	SIP-ISUP/Basic call/ receipt of a SUSPEND message with the **suspend indicator** set to "network initiated"				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT, on receipt of a SUSPEND message with the **suspend indicator** set to "network initiated": • is transferred in an INFO message.				
SIP parameter values					
ISUP parameter values	SUS; **Suspend indicator**: network initiated				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	INFO(SUS)	←		←	SUS(network)
	200 OK INFO	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

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

5.2.1.13 Receipt of the RESume message (RES) network initiated

TP113001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 6.10**
TSS reference	SIP-ISUP/Basic call/				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT, on receipt of a RESUME message containing the suspend/resume indicator set to "network initiated": • the RES is transferred in an INFO message.				
SIP parameter values					
ISUP parameter values	RES; **Suspend indicator**: network initiated				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	INFO(SUS)	←		←	SUS(network)
	200 OK INFO	→			
	INFO(RES)	←			RES(network)
	200 OK INFO	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

5.2.1.14 Receipt of Confusion message

TP114001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause A.1.1.3**
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Confusion message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the 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.				
SIP parameter values	180 Ringing containing an ACM with an unknown parameter				
ISUP parameter values	INFO with encapsulated CFN				
Comments	**SIP-I**				**ISUP**
	INVITE	→		→	IAM
	183 Session Progress(CFN)	←		←	CFN
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Communication		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

5.2.1.15 Segmentation

TP115001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause A.1.1.3.1**
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Receipt of a user part test message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that a call can be successfully completed if segmentation applies in backward direction.				
SIP parameter values	180 Ringing - encapsulated ACM: Backward call indicator absent or set to "no additional information will be sent" No action takes place on the SIP side				
ISUP parameter values	ACM: optional forward call indicator: additional information will be sent in a segmentation message SGM: optional parameters				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
				←	SGM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE	←		←	RLC

5.2.2 Interworking from ISUP to SIP-I

5.2.2.1 Sending of the INVITE message

TP301001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.1 1 a)**
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** 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 sending complete indication				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><td>TP301002</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1 1 b)</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP /Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM; Called party number complete number</td></tr>
<tr><td rowspan="8">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><td>TP301003</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1 1 c)</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP /Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM; Called party number: complete number</td></tr>
<tr><td rowspan="8">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP301004	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.1 1 d)**
TSS reference	ISUP-SIP /Basic call/Sending of the INVITE message			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the **complete called party number** where the end of address signalling is determined by the expiration timer T_{OIW1} after the receipt of the latest address message: • sends the INVITE message.			
SIP parameter values				
ISUP parameter values				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→			
			T_{OIW1} expiry		
				→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP301005	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.1 A)**
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message			
SIP selection criteria				
ISUP selection criteria	PICS 1/5			
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message with the **complete called party number** containing the Continuity Check indicator in the Nature of Connection Indicators parameter is set to indicate **"continuity check not required"**: • sends a INVITE message.			
SIP parameter values				
ISUP parameter values				
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP301006</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1 A)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/4 AND NOT PICS 4/5 AND NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/5</td></tr>
<tr><td>Test purpose</td><td colspan="5">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".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="9">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>COT</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP301007</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1 A)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/4 AND NOT PICS 4/5 AND NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/5</td></tr>
<tr><td>Test purpose</td><td colspan="5">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".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="9">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>COT</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP301008	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1 A)
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection 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 Continuity message is received with the Continuity Indicators parameter set to "**continuity check failed**".				
SIP parameter values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	→			
	COT	→			

TP301009	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1 A)
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection 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		SUT		SIP-I
	IAM	→			
			T8 expiry		
	REL	←			
	RLC	→			

TP301010	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clause 7.1 B)**		
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 4/15				
ISUP selection criteria	PICS 1/5 AND PICS 4/2				
Test purpose	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		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP301011	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clause 7.1 B)**		
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 4/15				
ISUP selection criteria	PICS 1/5 AND PICS 4/2				
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message where the Continuity Check indicator in the Nature of Connection Indicators parameter in the IAM is set to indicate **"continuity check required on this circuit"**: • sends an INVITE message with precondition using the SDP offer in the INVITE. The SDP offer or answer carrying the confirmation of a precondition being met is sent when the Continuity message with the Continuity Indicators parameter set to **"continuity check successful"** was received and the requested preconditions are met in the SIP network.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	183 Session Progress
				→	PRACK
				←	200 OK PRACK
	COT(successful)	→		→	UPDATE
				←	200 OK UPDATE
			Preconditions met		
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP301012</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1 B)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/5 AND PICS 4/2</td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="15">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>183 Session Progress</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK PRACK</td></tr>
<tr><td>COT(successful)</td><td>→</td><td></td><td>→</td><td>UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK UPDATE</td></tr>
<tr><td></td><td colspan="3">Preconditions met</td><td></td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP301013</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1 B)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/5 AND PICS 4/2</td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="8">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>100 Trying</td></tr>
<tr><td colspan="5"></td></tr>
<tr><td>COT(unsuccessful)</td><td>→</td><td></td><td>→</td><td>CANCEL</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK CANCEL</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>487 Request Terminated</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
</table>

<table>
<tr><th>TP301014</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1 B)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/5 AND PICS 4/2</td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="9">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>100 Trying</td></tr>
<tr><td colspan="5">T8 expires</td></tr>
<tr><td>REL(#47)</td><td>←</td><td></td><td>→</td><td>CANCEL</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td>←</td><td>200 OK CANCEL</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>487 Request Terminated</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
</table>

<table>
<tr><th>TP301015</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1 C)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4</td></tr>
<tr><td>Test purpose</td><td colspan="5">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:
o Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received;
o Bearer Set-up indication - for the forward bearer set-up case where the incoming Connect Type is "notification not required" was received.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="9">Comments</td><td>BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>COT(successful)</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP301016	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.1 C)**	
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria	NOT PICS 4/15				
ISUP selection criteria	PICS 1/4				
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "**COT to be expected**": • The sending of the INVITE is delayed until all the following conditions are satisfied: o Continuity message, with the Continuity Indicators parameter set to "**continuity**" shall be received; o APM with Action indicator set to "Connected" - for the forward bearer set-up cases (with, or without bearer control tunnelling) where the incoming Connect Type is "notification required", and for the fast set-up (backward) case.				
SIP parameter values					
ISUP parameter values					
Comments	BICC		SUT		SIP-I
	IAM	→			
	COT(successful)	→			
	APM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP301017	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.1 C)**	
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
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: o Continuity message, with the Continuity Indicators parameter set to "**continuity**" shall be received; o Bearer Set-up Connect indication - for the backward bearer set-up case was received.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	→			
	COT(successful)	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP301018</th><th>SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clause 7.1 C) 2.4</th></tr>
<tr><td>TSS reference</td><td colspan="2">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="2">NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="2">PICS 1/4</td></tr>
<tr><td>Test purpose</td><td colspan="2">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="2"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="2"></td></tr>
<tr><td>Comments</td><td colspan="2"><table>
<tr><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>COT(successful)</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table></td></tr>
</table>

<table>
<tr><th>TP301019</th><th>SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clause 7.1 C)</th></tr>
<tr><td>TSS reference</td><td colspan="2">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="2">NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="2">PICS 1/4</td></tr>
<tr><td>Test purpose</td><td colspan="2">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="2"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="2"></td></tr>
<tr><td>Comments</td><td colspan="2"><table>
<tr><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td colspan="5">T8 expires</td></tr>
<tr><td>REL(#41)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>RLC</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
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<table>
<tr><th>TP301020</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1 D)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4 AND PICS 4/2</td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="14">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>183 Session Progress</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK PRACK</td></tr>
<tr><td>COT(successful)</td><td>→</td><td></td><td>→</td><td>UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK UPDATE</td></tr>
<tr><td></td><td colspan="3">Preconditions met</td><td></td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP301021</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1 D) 2.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4 AND PICS 4/2</td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="14">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>183 Session Progress</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK PRACK</td></tr>
<tr><td>COT(successful)</td><td>→</td><td></td><td>→</td><td>UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK UPDATE</td></tr>
<tr><td></td><td colspan="3">Preconditions met</td><td></td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP301022	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.1 D) 2.3**
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 4/15				
ISUP selection criteria	PICS 1/4				
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "**COT to be expected**" sends an INVITE message with precondition using the SDP offer in the INVITE. The precondition signalling is concluded upon sending the (within an SDP offer-answer exchange) confirmation of a precondition being met. The SDP offer or answer carrying the confirmation of a precondition being met is sent when all of the following conditions are satisfied when: • Continuity message, with the Continuity Indicators parameter set to "**continuity**" shall be received; • Bearer Set-up Connect indication - for the backward bearer set-up case was received.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	183 Session Progress
				→	PRACK
				←	200 OK PRACK
	COT(successful)	→		→	UPDATE
				←	200 OK UPDATE
			Preconditions met		
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP301023	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause 7.1 D) 2.4
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 4/15	
ISUP selection criteria	PICS 1/4 AND PICS 4/2	
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "**COT to be expected**" sends an INVITE message with precondition using the SDP offer in the INVITE. The precondition signalling is concluded upon sending the (within an SDP offer-answer exchange) confirmation of a precondition being met. The SDP offer or answer carrying the confirmation of a precondition being met is sent when all of the following conditions are satisfied when • Continuity message, with the Continuity Indicators parameter set to "**continuity**" shall be received • BNC set-up success indication for cases using bearer control tunnelling was received	
SIP parameter values		
ISUP parameter values		
Comments		

BICC		SUT		SIP-I
IAM	→		→	INVITE(IAM)
			←	183 Session Progress
			→	PRACK
			←	200 OK PRACK
COT(successful)	→		→	UPDATE
			←	200 OK UPDATE
		Preconditions met		
ACM	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

TP301024	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], 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 4/15	
ISUP selection criteria	PICS 1/4 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 **before** an early dialogue has been established.	
SIP parameter values		
ISUP parameter values		
Comments		

BICC		SUT		SIP-I
IAM	→		→	INVITE(IAM)
			←	100 Trying
		T8 expires		
REL(#47)	←		→	CANCEL
RLC	→		←	200 OK CANCEL
			←	487 Request Terminated
			→	ACK

<table>
<tr><th>TP301025</th><th>SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clause 7.1</th></tr>
<tr><td>TSS reference</td><td colspan="2">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="2">PICS 4/4 AND PICS 4/5 AND PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="2">PICS 1/5 AND PICS 4/2</td></tr>
<tr><td>Test purpose</td><td colspan="2">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="2"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="2"></td></tr>
<tr><td>Comments</td><td colspan="2">
<table>
<tr><td>BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>100 Trying</td></tr>
<tr><td colspan="5">internal resource reservation was unsuccessful</td></tr>
<tr><td>REL(#47)</td><td>←</td><td></td><td>→</td><td>CANCEL</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td>←</td><td>200 OK CANCEL</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>487 Request Terminated</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
</table>
</td></tr>
</table>

<table>
<tr><th>TP301026</th><th>SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clause 7.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="2">ISUP-SIP/Basic call/ Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="2">Based on table 6</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="2"></td></tr>
<tr><td>Test purpose</td><td colspan="2">Ensure that the SUT in the Idle state on receipt of a IAM message, with the Transmission Medium Requirement (TMR) parameter set to TMR_VALUE if no USI parameter is contained in the IAM:
• sends an INVITE message containing the media description defined with the "a =" "b =" and "m=" lines set to a_b_m_LINE_VALUE.</td></tr>
<tr><td>SIP parameter values</td><td colspan="2">INVITE : a_b_m_LINE_VALUE</td></tr>
<tr><td>ISUP parameter values</td><td colspan="2">IAM: TMR : ISUP_TMR</td></tr>
<tr><td>Comments</td><td colspan="2">
<table>
<tr><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
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</td></tr>
</table>

TP301027	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1.1
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message				
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_VALUE				
ISUP parameter values	IAM: USI : ISUP_USI				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

Table 6

Values for test purposes TP301026						
	ISUP	**SDP - a_b_m_LINE_VALUE**				
	TMR parameter	m= line			b= line	a= line
	TMR codes	<media>	<transport>	<fmt-list>	<modifier>:<bandwidth-value>	rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>
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

Table 7

Values for test purposes TP301026, TP301027									
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>	<transport>	<fmt-list>	<modifier>:<bandwidth-value>	rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>
VA_01	"speech"	"Speech"	"G.711 µ-law"	Ignore	audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)
	"speech"	"Speech"	"G.711 A-law"	Ignore	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000
VA_02	"3,1 KHz audio"	USI Absent		Ignore	audio	RTP/AVP	0 and/or 8	AS:64	rtpmap:0 PCMU/8000 and/or rtpmap:8 PCMA/8000
VA_03	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 µ-law"		audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)
	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 A-law"		audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000
VA_04	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 A-law"	"Facsimile Group 2/3"	image	tcptl	t38	AS:64	Based on T.38.
	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 A-law"	"Facsimile Group 2/3"	image	udptl	t38	AS:64	Based on T.38.
VA_05	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 µ-law"	"Facsimile Group 2/3"	image	udptl	t38	AS:64	Based on T.38.
	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 µ-law"	"Facsimile Group 2/3"	image	tcptl	t38	AS:64	Based on T.38.
VA_06	"64 kbit/s unrestricted"	"Unrestricted digital inf. W/tone/ann."	N/A	Ignore	audio	RTP/AVP	9	AS:64	Rtpmap:9 G722/8000
VA_07	"64 kbit/s unrestricted"	"Unrestricted digital information"	N/A	Ignore	Audio	RTP/AVP	Dynamic PT	AS:64	rtpmap:<dynamic-PT> CLEARMODE/8000

TP301028	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.1.2	
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message				
SIP selection criteria					
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 **To header field** in the INVITE message.				
SIP parameter values	INVITE: To: …				
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP301029	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.1.2	
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter of the IAM: • to the addr-spec component of the **To header field** which shall include the "user=phone" URI parameter if the To header field contains a sip: URI.				
SIP parameter values	INVITE: To: sip: ….; user=phone				
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP301030	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.1.2**	
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message				
SIP selection criteria	NOT PICS 1/9				
ISUP selection criteria					
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter of the IAM and the and the followed SAM: • to the addr-spec component of the **To header field.**				
SIP parameter values	INVITE: To:				
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→			
	SAM	→			
	SAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP301031	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.1.2**	
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message				
SIP selection criteria	NOT PICS 1/9				
ISUP selection criteria					
Test purpose	Ensure that the SUT is mapping in the Called Party Number parameter contained in the Called Party address information of the IAM and followed SAM: • to the addr-spec component of the **To header field** which shall include the "user=phone" URI parameter if the To header field contains a sip: URI.				
SIP parameter values	INVITE: To: sip: ….; user=phone				
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→			
	SAM	→			
	SAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP301032	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.1.4**	
TSS reference	ISUP-SIP/Basic call/ Sending 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	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP301033	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.1.2**	
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message				
SIP selection criteria	PICS 1/9				
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 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		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP301034	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause 7.1.2
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria	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		SUT		SIP-I
IAM	→		→	INVITE(IAM)
ACM	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

TP301035	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause 7.1.2
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria	PICS 1/9	
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		SUT		SIP-I
IAM	→			
SAM	→			
SAM	→		→	INVITE(IAM)
ACM	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

TP301036	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.1.2**
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message				
SIP selection criteria	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 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	←		←	200 OK INVITE(ANM)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

5.2.2.2 Receipt of the SAM message after INVITE has been send

TP302001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.2**
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after INVITE has been sent				
SIP selection criteria	PICS 3/1				
ISUP selection criteria					
Test purpose	Ensure if the SUT is supporting en bloc addressing towards the SIP network, subsequent SAMs received after the SUT has sent the INVITE are ignored.				
SIP parameter values					
ISUP parameter values	SAM; **subsequent number** (PIXIT)				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	SAM	→			
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP302002</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Receipt of SAM after invite has been sent</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 3/2</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/5</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to indicate "continuity check not required".
sends a INVITE message.
On receipt of a SAM from the ISUP the SUT shall:
1) Stop timer TOIW3 (if it is running);
2) TOIW2 shall be restarted and the SUT shall invoke the following procedures:
a) the Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call;
b) a new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent;
c) the new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question;
d) all other contents of the new INVITE are interworked from the parameters of the original IAM.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="14">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE 1 (IAM)</td></tr>
<tr><td>SAM</td><td>→</td><td></td><td>→</td><td>INVITE 2 (IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>484 Address Incomplete (1)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>SAM</td><td>→</td><td></td><td>→</td><td>INVITE 3 (IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>484 Address Incomplete (2)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing (3) (ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE (3) (ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP302003	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clause 7.2.1**
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after 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	→			
	SAM	→			
	COT	→		→	INVITE1(IAM)
	SAM	→		→	INVITE2(IAM)
				←	484 Address Incomplete (1)
				→	ACK
	ACM	←		←	180 Ringing (2) (ACM)
	ANM	←		←	200 OK INVITE (2) (ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP302004	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clause 7.2.1**
TSS reference	ISUP-SIP/Basic call/Receipt of 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 performed on previous circuit**". Sends the INVITE after the receipt of the Continuity message with the Continuity Indicators parameter "**continuity check successful**". On receipt of a SAM from the ISUP the SUT shall: 1) Stop timer TOIW3 (if it is running); 2) TOIW2 shall be restarted and the SUT shall invoke the following procedures: a) the Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call; b) a new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent; c) the new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question; d) all other contents of the new INVITE are interworked from the parameters of the original IAM.	
SIP parameter values		
ISUP parameter values		
Comments	(see flow below)	

ISUP/BICC		SUT		SIP-I
IAM	→			
SAM	→			
COT	→		→	INVITE 1 (IAM)
SAM	→		→	INVITE 2 (IAM)
			←	484 Address Incomplete (1)
			→	ACK
ACM	←		←	180 Ringing (2) (ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

TP302005	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.2.1**
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after 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	→		
	SAM	→		
	COT	→		

TP302006	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.2.1**
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after 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	→		
	SAM	→		
			T8 expires	
	REL	←		
	RLC	→		

TP302007	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause 7.2.1
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria	PICS 3/2 AND PICS 4/5 AND PICS 4/15	
ISUP selection criteria	PICS 1/5 AND PICS 4/2	
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set "**continuity check required on this circuit**". Sends an INVITE message after the reception of the Continuity message with the Continuity Indicators parameter set to "**continuity check successful**" and after the requested preconditions are met in the SIP network. On receipt of a SAM from the ISUP the SUT shall: 1) Stop timer TOIW3 (if it is running); 2) TOIW2 shall be restarted and the SUT shall invoke the following procedures: a) the Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call; b) a new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent; c) the new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question; d) all other contents of the new INVITE are interworked from the parameters of the original IAM.	
SIP parameter values	INVITE2: Request URI contains digits from the IAM and digits from SAM x and SAM y. The IAM is also contained	
ISUP parameter values		
Comments		

ISUP/BICC		SUT		SIP-I
IAM	→		→	INVITE1(IAM)
SAM x	→			
			←	183 Session Progress without encapsulated ACM
COT	→		→	UPDATE
			←	200 OK UPDATE
SAM y	→		→	INVITE2 (IAM and digits from SAM X + SAM Y)
			←	484 Address Incomplete (1)
			→	ACK
ACM	←		←	180 Ringing2 (ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP302008</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clause 7.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP/Basic call/Receipt of SAM after invite has been sent</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6">PICS 3/2 AND PICS 4/5 AND PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6">PICS 1/5 AND PICS 4/2</td></tr>
<tr><td>Test purpose</td><td colspan="6">Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set or "continuity check performed on previous circuit".
Sends an INVITE message after the reception of the Continuity message with the Continuity Indicators parameter set to "continuity check successful" and after the requested preconditions are met in the SIP network.
On receipt of a SAM from the ISUP the SUT shall:
1) Stop timer TOIW3 (if it is running);
2) TOIW2 shall be restarted and the SUT shall invoke the following procedures:
a) the Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call;
b) a new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent;
c) the new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question;
d) all other contents of the new INVITE are interworked from the parameters of the original IAM.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">INVITE2: Request URI contains digits from the IAM and digits from SAM x and SAM y. The IAM is also contained</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6"></td></tr>
<tr><td rowspan="16">Comments:</td><td colspan="6">The O-IWU should initiate the precondition signalling procedure using the SDP Offer in the INVITE. The precondition signalling is concluded upon sending (within an SDP offer-answer exchange) the confirmation of a precondition being met. The SDP Offer or Answer carrying the confirmation of a precondition being met is sent when the conditions to send an INVITE message are satisfied.</td></tr>
<tr><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td colspan="2">SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE1(IAM)</td></tr>
<tr><td>SAM x</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">183 Session Progress without encapsulated ACM</td></tr>
<tr><td>COT</td><td>→</td><td></td><td>→</td><td colspan="2">UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK UPDATE</td></tr>
<tr><td>SAM</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE2 (IAM and digits from SAM X + SAM Y)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">484 Address Incomplete (1)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing2 (ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="2"></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td colspan="2">BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP302009</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Receipt of SAM after invite has been sent</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 3/2 AND NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4 AND NOT PICS 4/2</td></tr>
<tr><td>Test purpose</td><td colspan="5">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.
On receipt of a SAM from the BICC 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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="11">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>SAM x</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>COT</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>SAM y</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP302010</th><th>SIP reference: RFC 3261 [4]</th><th colspan="4">ISUP reference:
Q.1912.5 [1], clause 7.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Receipt of SAM after invite has been sent</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 3/2 AND NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4 AND PICS 4/2</td></tr>
<tr><td>Test purpose</td><td colspan="5">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.
On receipt of a SAM from the BICC 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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="11">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>SAM x</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>COT</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>SAM y</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP302011</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Receipt of SAM after invite has been sent</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 3/2 AND NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4 AND PICS 4/2</td></tr>
<tr><td>Test purpose</td><td colspan="5">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:
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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="11">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>SAM x</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>COT</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>SAM y</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP302012</th><th colspan="4">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP/Basic call/Receipt of SAM after invite has been sent</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6">PICS 3/2 AND NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6">PICS 1/4 AND PICS 4/2</td></tr>
<tr><td>Test purpose</td><td colspan="6">Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected"
The sending of the INVITE delays until all the following conditions are satisfied:
• Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received;
• BNC set-up success indication for cases using bearer control tunnelling was received
On receipt of a SAM from the BICC the SUT shall:
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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="6"></td></tr>
<tr><td rowspan="11">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td colspan="2">SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>SAM x</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>COT</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(IAM)</td></tr>
<tr><td>SAM y</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="2"></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td colspan="2">BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK BYE(RLC)</td></tr>
</table>

TP302013	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause 7.2.1
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria	PICS 3/2 AND PICS 4/5 AND PICS 4/15	
ISUP selection criteria	PICS 1/4 AND PICS 4/2	
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "**COT to be expected**". Sends the INVITE message. The events: • Continuity message, with the Continuity Indicators parameter set to "**continuity**" was received; • Bearer Set-up indication - for the forward bearer set-up case where the incoming Connect Type is "**notification not required**" was received; are indicating the successful completion of bearer set-up. On receipt of a SAM from the BICC the SUT shall: 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	INVITE2: Request URI contains digits from the IAM and digits from SAM x and SAM y. The IAM is also contained	
ISUP parameter values		
Comments	The O-IWU should initiate the precondition signalling procedure using the SDP Offer in the INVITE. The precondition signalling is concluded upon sending (within an SDP offer-answer exchange) the confirmation of a precondition being met. The SDP Offer or Answer carrying the confirmation of a precondition being met is sent when the conditions to send a INVITE message are satisfied.	

ISUP/BICC		SUT		SIP-I
IAM	→		→	INVITE1(IAM)
SAM x	→			
			←	183 Session Progress without encapsulated ACM
COT	→		→	UPDATE
			←	200 OK UPDATE
SAM y	→		→	INVITE2 (IAM and digits from SAM X + SAM Y)
			←	484 Address Incomplete (1)
			→	ACK
ACM	←		←	180 Ringing2(ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

TP302014	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause 7.2.1
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria	PICS 3/2 AND PICS 4/5 AND PICS 4/15	
ISUP selection criteria	PICS 1/4 AND PICS 4/2	
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "**COT to be expected**". Sends the INVITE message. The events: • Continuity message, with the Continuity Indicators parameter set to "**continuity**" was received; • APM with Action indicator set to "Connected" - for the forward bearer set-up cases (with, or without bearer control tunnelling) where the incoming Connect Type is "notification required", and for the fast set-up (backward) case; are indicating the successful completion of bearer set-up. On receipt of a SAM from the BICC 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	INVITE2: Request URI contains digits from the IAM and digits from SAM x and SAM y. The IAM is also contained	
ISUP parameter values		
Comments	The O-IWU should initiate the precondition signalling procedure using the SDP Offer in the INVITE. The precondition signalling is concluded upon sending (within an SDP offer-answer exchange) the confirmation of a precondition being met. The SDP Offer or Answer carrying the confirmation of a precondition being met is sent when the conditions to send a INVITE message are satisfied	

ISUP/BICC		SUT		SIP-I
IAM	→		→	INVITE1(IAM)
SAM	→			
			←	183 Session Progress without encapsulated ACM
COT	→		→	UPDATE
			←	200 OK UPDATE
SAM	→		→	INVITE2 (IAM with digits from SAM X + SAM Y)
			←	484 Address Incomplete (1)
			→	ACK
ACM	←		←	180 Ringing2(ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

TP302015	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clause 7.2.1**
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria	PICS 3/2 AND PICS 4/5 AND PICS 4/15	
ISUP selection criteria	PICS 1/4 AND PICS 4/2	
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "**COT to be expected**". Sends the INVITE message. The events: • Continuity message, with the Continuity Indicators parameter set to "**continuity**" was received: • Bearer Set-up Connect indication - for the backward bearer set-up case was received. are indicating the successful completion of bearer set-up. On receipt of a SAM from the BICC the SUT shall: 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	INVITE2: Request URI contains digits from the IAM and digits from SAM x and SAM y. The IAM is also contained	
ISUP parameter values		
Comments	The O-IWU should initiate the precondition signalling procedure using the SDP Offer in the INVITE. The precondition signalling is concluded upon sending (within an SDP offer-answer exchange) the confirmation of a precondition being met. The SDP Offer or Answer carrying the confirmation of a precondition being met is sent when the conditions to send a INVITE message are satisfied	

ISUP/BICC		SUT		SIP-I
IAM	→		→	INVITE1(IAM)
SAM	→			
			←	183 Session Progress without encapsulated ACM
COT	→		→	UPDATE
			←	200 OK UPDATE
SAM	→		→	INVITE2 (IAM with digits from SAM X + SAM Y)
			←	484 Address Incomplete (1)
			→	ACK
ACM	←		←	180 Ringing2(ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

TP302016	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clause 7.2.1**
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria	PICS 3/2 AND PICS 4/5 AND PICS 4/15	
ISUP selection criteria	PICS 1/4 AND PICS 4/2	
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing indicating "**COT to be expected**". Sends the INVITE message. The events: • Continuity message, with the Continuity Indicators parameter set to "**continuity**" was received; • BNC set-up success indication for cases using bearer control tunnelling was received. are indicating the successful completion of bearer set-up, On receipt of a SAM from the BICC/ISUP the SUT shall: 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	INVITE2: Request URI contains digits from the IAM and digits from SAM x and SAM y. The IAM is also contained	
ISUP parameter values		
Comments	The O-IWU should initiate the precondition signalling procedure using the SDP Offer in the INVITE. The precondition signalling is concluded upon sending (within an SDP offer-answer exchange) the confirmation of a precondition being met. The SDP Offer or Answer carrying the confirmation of a precondition being met is sent when the conditions to send a INVITE message are satisfied	

ISUP/BICC		SUT		SIP-I
IAM	→		→	INVITE1(IAM)
SAM	→			
			←	183 Session Progress without encapsulated ACM
COT	→		→	UPDATE
			←	200 OK UPDATE
SAM	→		→	INVITE2 (IAM with digits from SAM X + SAM Y)
			←	484 Address Incomplete (1)
			→	ACK
ACM	←		←	180 Ringing2(ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP302017</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clause 7.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Receipt of SAM after invite has been sent</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 3/2</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4</td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="12">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>SAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td colspan="3">T_{oiw2} expired</td><td></td></tr>
<tr><td>SAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3"></td><td></td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP302018</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clause 7.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Receipt of SAM after invite has been sent</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 3/1</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 3/8</td></tr>
<tr><td>Test purpose</td><td colspan="5">The SUT in Idle state, on receipt of an IAM message.
On receipt of a SAM from the BICC/ISUP the SUT shall:
• sends a INVITE message if the minimum number of digits for routing the call has been received in the IAM and the SAM;
• TOIW1 and TIOW2 shall be started and the SUT shall invoke the following procedures:
Ensure that if timer TOIW2 has expired, subsequent SAMs received
after the SUT has sent the INVITE are ignored.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="12">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>SAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td colspan="3">T_{oiw2} expired</td><td></td></tr>
<tr><td>SAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3"></td><td></td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

5.2.2.3 Sending of the ACM message

<table>
<tr><th>TP303001</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1,
Q.764 [i.12], clause 2.1.4.8</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP /Basic call/Sending of the ACM message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 1/3</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 4/9</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number and the sending complete indication.
Sends the INVITE message to called user Sends the ACM message with:
• the CPS indicator set to " no indication (00)";
• the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)";
• the interworking indicator set to " INT_IND_VAL";
• the ISUP indicator set to "ISUP_IND_ID";
• the ISDN access indicator set to "ISDN_ACC_IND_VAL".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM; Called party number: complete number
ACM, CPS indicator no indication (00)
Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)
interworking indicator: INT_IND_VAL (PIXIT)
ISUP indicator: ISUP_IND_ID (PIXIT)
ISDN access indicator ISDN_ACC_IND_VAL (PIXIT)</td></tr>
<tr><td rowspan="9">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM(no indication)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CPG(Alerting)</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP303002</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1,
Q.764 [i.12], clause 2.1.4.8</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP /Basic call/ Sending of the ACM message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 1/3</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 4/9</td></tr>
<tr><td>Test purpose</td><td colspan="5">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".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM; Called party number: complete number
ACM, Backward call indicator is set to the value in the encapsulated ACMCPS 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)</td></tr>
<tr><td rowspan="9">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM(no indication)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CPG(Alerting)</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP303003</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1,
Q.764 [i.12], clause 2.1.4.8</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP /Basic call/Sending of the ACM message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 1/3</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 4/9</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party:
sends the INVITE message to called user;
sends the ACM message with the CPS indicator set to " no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to " INT_IND_VAL", the ISUP indicator set to "ISUP_IND_ID", the ISDN access indicator set to "ISDN_ACC_IND_VAL".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM; Called party number: complete number
ACM, CPS indicator no indication (00)
Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)
interworking indicator: INT_IND_VAL (PIXIT)
ISUP indicator: ISUP_IND_ID (PIXIT)
ISDN access indicator ISDN_ACC_IND_VAL (PIXIT)</td></tr>
<tr><td rowspan="9">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM(no indication)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CPG(Alerting)</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP303004</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clauses 7.1 1) d), 7.3.1, 7.4</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP /Basic call/Sending of the ACM message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 1/3</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">NOT PICS 4/9</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by the expiration timer T_{OIW1} after the receipt of the latest address message:
• sends the INVITE message to called user;
• sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to " INT_IND_VAL", the ISUP indicator set to "ISUP_IND_ID", the ISDN access indicator set to "ISDN_ACC_IND_VAL".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM; Called party number: complete number
ACM, CPS indicator no indication (00)
Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)
interworking indicator: INT_IND_VAL (PIXIT)
ISUP indicator: ISUP_IND_ID (PIXIT)
ISDN access indicator ISDN_ACC_IND_VAL (PIXIT)</td></tr>
<tr><td rowspan="10">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">T_{OIW1} expiry</td><td></td></tr>
<tr><td>ACM(no indication)</td><td>←</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>CPG(Alerting)</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP303005</th><th>SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clauses 7.1, 7.3.1</th></tr>
<tr><td>TSS reference</td><td colspan="2">ISUP-SIP /Basic call/Sending of the ACM message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="2">PICS 1/3</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="2">NOT PICS 4/9</td></tr>
<tr><td>Test purpose</td><td colspan="2">Ensure that the SUT if overlap addressing is to be used toward the SIP network, on receipt of an IAM message containing the minimum number of digits required for routing the call has been received (start timer TOIW2 and invoke the appropriate outgoing SIP signalling procedure):
• sends an INVITE message to the called user and after the expiration of T_{OIW2};
• sends the ACM message with the CPS indicator set to " no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to " INT_IND_VAL", the ISUP indicator set to "ISUP_IND_ID", the ISDN access indicator set to "ISDN_ACC_IND_VAL".</td></tr>
<tr><td>SIP parameter values</td><td colspan="2"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="2">IAM; Called party number: complete number
ACM, CPS indicator no indication (00)
Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)
interworking indicator: INT_IND_VAL (PIXIT)
ISUP indicator: ISUP_IND_ID (PIXIT)
ISDN access indicator ISDN_ACC_IND_VAL (PIXIT)</td></tr>
<tr><td>Comments</td><td colspan="2">
<table>
<tr><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>SAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>SAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td colspan="3">T_{OIW2} expiry</td><td></td></tr>
<tr><td>ACM(no indication)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CPG(Alerting)</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>
</td></tr>
</table>

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

TP303007	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 7.1 1 a), 7.3.2
TSS reference	ISUP-SIP /Basic call/Sending of the ACM message				
SIP selection criteria	PICS 3/1				
ISUP selection criteria	NOT PICS 4/9				
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete **called party number** on receipt of a 183 Session Progress with encapsulated ACM: • sends the ACM message; • the encapsulated ACM message is sent unchanged backward.				
SIP parameter values					
ISUP parameter values	IAM; **Called party number:** complete number				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM(no indication)	←		←	183 Session Progress(ACM)
	CPG(Alerting)	←		←	180 Ringing(CPG)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP303011</th><th colspan="4">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clauses 7.1, 7.3.1, 7.4</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISUP-SIP /Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 1/3</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 4/2 AND NOT PICS 4/9</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by the expiration timer T_{OIW1} after the receipt of the latest address message and the continuity check is performed (ISUP) or COT is expected (BICC):
• sends the INVITE message to called user;
• the SUT shall withhold sending ACM until a successful continuity indication has been received;
• sends the ACM message with the CPS indicator set to " no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "INT_IND_VAL", the ISUP indicator set to " ISUP_IND_ID", the ISDN access indicator set to "ISDN_ACC_IND_VAL".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM; Called party number: complete number
ACM,
CPS indicator no indication (00)
Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)interworking indicator: INT_IND_VAL (PIXIT)
ISUP indicator: ISUP_IND_ID (PIXIT)
ISDN access indicator ISDN_ACC_IND_VAL (PIXIT)</td></tr>
<tr><td rowspan="13">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>183 Session Progress without encapsulated ACM</td></tr>
<tr><td>COT</td><td>→</td><td></td><td>→</td><td>UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK UPDATE</td></tr>
<tr><td></td><td colspan="3">T_{OIW1} expiry</td><td></td></tr>
<tr><td>ACM(no indication)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CPG(Alerting, BCi)</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP303012</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses 7.1, 7.3.1, 7.4</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP /Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 1/3 AND PICS 3/2 AND PICS 4/5 AND PICS 4/4 AND PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 4/2 AND NOT PICS 4/9</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT if overlap addressing is to be used toward the SIP network, on receipt of an IAM message containing the minimum number of digits required for routing the call has been received (start timer TOIW2 and invoke the appropriate outgoing SIP signalling procedure) and the continuity check is performed (ISUP) or COT is expected (BICC). After the expiry of T_{oiw2}:
• sends the ACM message with the CPS indicator set to " no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "INT_IND_VAL", the ISUP indicator set to "ISUP_IND_ID", the ISDN access indicator set to "ISDN_ACC_IND_VAL".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">ACM, Backward call indicator
CPS indicator no indication (00)
Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)
interworking indicator: INT_IND_VAL (PIXIT)
ISUP indicator: ISUP_IND_ID (PIXIT)
ISDN access indicator ISDN_ACC_IND_VAL (PIXIT)
CPG: Event indicator = ALRTING and the BCI from the ACM encapsulated in the received 180 Ringing</td></tr>
<tr><td rowspan="13">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>183 Session Progress without encapsulated ACM</td></tr>
<tr><td>COT</td><td>→</td><td></td><td>→</td><td>UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK UPDATE</td></tr>
<tr><td></td><td colspan="3">T_{OIW2} expiry</td><td></td></tr>
<tr><td>ACM(no indication)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CPG(Alerting, BCi)</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP303013	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clauses 7.1, 7.3.1, 7.4
TSS reference	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria	PICS 1/3	
ISUP selection criteria	PICS 4/2 AND NOT PICS 4/9	
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete **called party number,** the continuity check is performed (ISUP) or COT is expected (BICC) indication receipt of a 180 Ringing message: • Sends the ACM message with: o the **CPS indicator** set to the value in the encapsulated ACM; o the **Called party's category indicator** set to the value in the encapsulated ACM; o the **interworking indicator** set to the value in the encapsulated ACM; o the **ISUP indicator** set to the value in the encapsulated ACM; o the **ISDN access indicator** set to the value in the encapsulated ACM.	
SIP parameter values		
ISUP parameter values	IAM; **Called party number**: complete number ACM, Backward call indicator is set to the value in the encapsulated ACM	
Comments		

ISUP/BICC		SUT		SIP-I
IAM	→		→	INVITE(IAM)
			←	183 Session Progress without encapsulated ACM
COT	→		→	UPDATE
			←	200 OK UPDATE
ACM	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP303014</th><th>SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clauses 7.1, 7.3.1, 7.4</th></tr>
<tr><td>TSS reference</td><td colspan="2">ISUP-SIP /Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="2">PICS 1/3 AND PICS 3/2 AND NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="2">PICS 3/8 AND PICS 4/2 AND NOT PICS 4/9</td></tr>
<tr><td>Test purpose</td><td colspan="2">Ensure that the SUT if overlap addressing is to be used toward the SIP network, on receipt of an IAM message containing the minimum number of digits required for routing the call has been received (start timer T_{OIW2} and invoke the appropriate outgoing SIP signalling procedure) and the continuity check is performed (ISUP) or COT is expected (BICC). After the expiry of T_{oiw2}:
• sends the ACM message with the CPS indicator set to " no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to " INT_IND_VAL", the ISUP indicator set to "ISUP_IND_ID", the ISDN access indicator set to "ISDN_ACC_IND_VAL".</td></tr>
<tr><td>SIP parameter values</td><td colspan="2"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="2">ACM, Backward call indicator
CPS indicator: no indication (00)
Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)
interworking indicator: INT_IND_VAL (PIXIT)
ISUP indicator: ISUP_IND_ID (PIXIT)
ISDN access indicator: ISDN_ACC_IND_VAL (PIXIT)
CPG: Event indicator = ALRTING and the BCI from the ACM encapsulated in the received 180 Ringing</td></tr>
<tr><td>Comments</td><td colspan="2">
<table>
<tr><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>COT</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td colspan="3">T_{OIW2} expiry</td><td></td></tr>
<tr><td>ACM(no indication)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CPG(Alerting)</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>
</td></tr>
</table>

TP303015	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clauses 7.1, 7.3.1; 7.4**	
TSS reference	ISUP-SIP /Basic call/Sending of the ACM message				
SIP selection criteria	PICS 1/3 AND NOT PICS 4/15				
ISUP selection criteria	PICS 4/2 AND NOT PICS 4/9				
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete **called party number**, the continuity check is performed (ISUP) or COT is expected (BICC) indication receipt of a 180 Ringing message: • Sends the ACM message with: o the **CPS indicator** set to the value in the encapsulated ACM; o the **Called party's category indicator** set to the value in the encapsulated ACM; o the **interworking indicator** set to the value in the encapsulated ACM; o the **ISUP indicator** set to the value in the encapsulated ACM; o the **ISDN access indicator** set to the value in the encapsulated ACM.				
SIP parameter values					
ISUP parameter values	IAM; **Called party number**: complete number ACM, Backward call indicator is set to the value in the encapsulated ACM				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→			
	COT	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

5.2.2.4 Sending of the CPG message

TP304001	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clauses 7.1, 7.3.1**	
TSS reference	ISUP-SIP /Basic call/ Sending of the CPG message				
SIP selection criteria	PICS 3/1				
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 **event indicator** set to "Alerting".				
SIP parameter values					
ISUP parameter values	ACM: BCi called party status indicator = no indication CPG: Event Indicator = ALERTING, BCi as received from the encapsulated ACM				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→			
	SAM	→			
	SAM	→		→	INVITE(IAM)
			T_{OIW2} expiry		
	ACM(no indication)	←			
	CPG(Alerting BCi)	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP304002	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 7.1, 7.3.1**
TSS reference	ISUP-SIP /Basic call/ Sending of the CPG message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT, having sent a ACM message with called party status "no indication" on receipt of a 183 Session progress message with a encapsulated ISUP message: • sends the CPG message with the **event indicator** set to "Alerting".				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM(no indication)	←		←	183 Session Progress(ACM)
	CPG(Alerting)	←		←	180 Ringing(CPG)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

5.2.2.5 Sending of the ANM message

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

5.2.2.6 Sending of the CON message

TP306001	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 7.5, 7.5.1
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT, having not sent 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 values	200 OK INVITE;				
ISUP parameter values	CON; **interworking indicator**: INT_IND_VAL (PIXIT) **ISUP indicator**: ISUP_IND_ID (PIXIT) **ISDN access indicator** ISDN_ACC_IND_VAL (PIXIT) **CPS indicator**: no indication				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	CON	←		←	200 OK INVITE(CON)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

5.2.2.7 Receipt of the Release message (REL)

TP307001	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.7.1, 1)
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 but before an INVITE has been sent. On receipt of a REL message: no action is required on the SIP side other than to terminate local procedures if any are in progress.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→			
	REL	→			
	RLC	←			

<table>
<tr><th>TP307002</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.7.1 2)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/ Receipt of the Release message (REL)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="8">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>REL</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>RLC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(CON)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP307003</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.7.1 2) 3)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/ Receipt of the Release message (REL)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message before a 200 OK SIP response message has been received:
• the SUT shall hold the REL message. A CANCEL is sent when any SIP response was been received;
• on subsequently receiving 200 OK INVITE messages , the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="10">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>100 Trying</td></tr>
<tr><td>REL</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>→</td><td>CANCEL(REL)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(CON)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK CANCEL</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP307004	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.7.1 2) 3)**	
TSS reference	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message **before** an early dialogue with the message 100 Trying has been established: • the SUT shall hold the REL message until a **100 Trying** response has been received; • the SUT shall send a CANCEL The received REL is encapsulated in the CANCEL.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	REL	→			
	RLC	←			
				←	100 Trying
				→	CANCEL(REL)
				←	200 OK CANCEL
				←	487 Request terminated
				→	ACK

TP307005	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.7.1 4)**	
TSS reference	ISUP-SIP/Basic call/ Receipt 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** 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	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
	REL	→			
	RLC	←		→	ACK
				→	BYE(REL)
				←	200 OK BYE(RLC)

TP307006	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], 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		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	SIP_MESSAGE_VA
	REL	→			
	RLC	←			
	CASE A				
				→	CANCEL(REL)
				←	200 OK CANCEL
				←	487 Request terminated
				→	ACK
	CASE B				
				→	BYE(REL)
				←	200 OK BYE
				←	487 Request terminated
				→	ACK

Table 8

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

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

TP308001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.7.2**
TSS reference	ISUP-SIP /Basic call/ Sending of the Release message (REL)/				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM sends out an INVITE message and on receipt of a BYE message in the confirmed dialogue: • sends a REL message constructed from the encapsulated REL in the received BYE.				
SIP parameter values					
ISUP parameter values	REL; Cause value "Normal call clearing"				
Comments	ISUP/BICC		SU		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
		Conversation			
	REL	←		←	BYE(REL)
	RLC	→		→	200 OK BYE(RLC)

TP308002	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.7.6**
TSS reference	ISUP-SIP /Basic call/ Sending of the Release message (REL)/				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM sends out an INVITE message. 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		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	100 Trying
	REL	←		←	SIP_Failure_VA(REL)
	RLC	→		→	ACK

Table 9

Values for test purpose TP308002		
VA	**←REL (Cause Value) CV_ ISUP**	**←4XX/5XX/6XX SIP message SIP_Failure_VA**
VA_01	127 Interworking	400 Bad Request
VA_02	127 Interworking	402 Payment Required
VA_03	127 Interworking	403 Forbidden
VA_04	1 Unallocated number	404 Not Found
VA_05	127 Interworking	405 Method Not Allowed
VA_06	127 Interworking	406 Not Acceptable
VA_07	127 Interworking	408 Request Timeout
VA_08	22 Number changed (without diagnostic)	410 Gone
VA_9	127 Interworking	423 Interval Too Brief
VA_10	20 Subscriber absent	480 Temporarily Unavailable
VA_11	127 Interworking	481 Call/Transaction does not exist
VA_12	127 Interworking	482 Loop Detected
VA_13	127 Interworking	483 Too many hops
VA_14	127 Interworking	485 Ambiguous
VA_15	17 User busy	486 Busy Here
VA_16	127 Interworking	488 Not acceptable here
VA_17	127 Interworking	493 Undecipherable
VA_18	127 Interworking	500 Server Internal error
VA_19	127 Interworking	501 Not implemented
VA_20	127 Interworking	502 Bad Gateway
VA_21	127 Interworking	504 Server timeout
VA_22	17 User busy	600 Busy Everywhere
VA_23	21 Call rejected	603 Decline
VA_24	1 Unallocated number	604 Does not exist anywhere
VA_25	127 Interworking	606 Not acceptable

<table>
<tr><th>TP308003</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.7.6</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP /Basic call/ Sending of the Release message (REL)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/10</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT after receiving the IAM sends out an INVITE message, on receipt of a Failure message 487 Request terminated:
• no action is taken on the ISUP if a CANCEL request was previously sent before an answer to an INVITE was received.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="8">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>100 Trying</td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>CANCEL(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK CANCEL</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>487 Request Terminated</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
</table>

<table>
<tr><th>TP308004</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.7.6</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP /Basic call/ Sending of the Release message (REL)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">REL; cause value: CV_ISUP</td></tr>
<tr><td rowspan="5">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>XXX</td><td>←</td><td></td><td>←</td><td>SIP MESSAGE_VA</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>SIP_Failure_VA(REL)</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td>→</td><td>ACK</td></tr>
</table>

Table 10

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

Table 11

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

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

Table 12

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

TP30806	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.7.6**
TSS reference	ISUP-SIP /Basic call/ Sending of the Release message (REL)/				
SIP selection criteria	NOT PICS 4/21				
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM sends out an INVITE message. On receipt of a response message (3xx) defined as **SIP_Response_VA,** the SUT: • sends a REL message with the **Cause value** CV_ ISUP.				
SIP parameter values					
ISUP parameter values	REL; **cause value**: CV_ISUP				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	100 Trying
	REL	←		←	SIP_Response_VA
	RLC	→		→	ACK

Table 13

Values for test purposes TP308006		
VA	**←REL (Cause Value) CV_ ISUP**	**←3XX SIP message SIP_Response_VA**
VA_01	127 Interworking	300 Multiple Choices
VA_02	127 Interworking	301 Moved Permanently
VA_03	127 Interworking	302 Move Temporarily
VA_04	127 Interworking	305 Use Proxy
VA_05	127 Interworking	380 Alternative Service

Mapping of Cause Indicators parameter into SIP Reason header fields.

Table 14

Cause indications parameter field	Value of parameter field	component of SIP Reason header field	Component value
-	-	Protocol	"*Q.850*"
Cause Value	"*XX*" (see note 1)	Protocol-cause	"*cause= XX*" (see note 1)
-	-	Reason-text	Should be filled with the definition text as stated in Q.850 (see note 2)
NOTE 1: "XX" is the Cause Value as defined in ITU-T Recommendation Q.850 [3]. NOTE 2: Due to the fact that the Cause Indications parameter does not include the definition text as defined in table 1/ITU-T Recommendation Q.850 [3] this is based on provisioning in the O-IWU.			

5.2.2.9 Autonomous release at O-IWU

5.2.2.9.1 Receipt of Reset Circuit message (RSC)

<table>
<tr><th>TP309001</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clauses 7.7.1, 1), 7.7.4, 7.7.5</th></tr>
<tr><td>TSS reference</td><td colspan="5">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</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="4">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>RSC</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>RLC</td><td>←</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><th>TP309002</th><th colspan="4">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.5</th></tr>
<tr><td>TSS reference</td><td colspan="6">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</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">CANCEL or BYE: A REL is encapsulated with cause 31</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6"></td></tr>
<tr><td rowspan="15">Comments</td><td>ISUP/BICC</td><td></td><td colspan="2">SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td colspan="2"></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>RSC</td><td>→</td><td colspan="2"></td><td></td><td></td></tr>
<tr><td>RLC</td><td>←</td><td colspan="2"></td><td></td><td></td></tr>
<tr><td></td><td></td><td colspan="2"></td><td>←</td><td>SIP_MESSAGE_VA</td></tr>
<tr><td colspan="4">CASE A</td><td></td><td></td></tr>
<tr><td></td><td></td><td colspan="2"></td><td>→</td><td>CANCEL</td></tr>
<tr><td></td><td></td><td colspan="2"></td><td>←</td><td>200 OK CANCEL</td></tr>
<tr><td></td><td></td><td colspan="2"></td><td>←</td><td>487 Request terminated</td></tr>
<tr><td></td><td></td><td colspan="2"></td><td>→</td><td>ACK</td></tr>
<tr><td colspan="4">CASE B</td><td></td><td></td></tr>
<tr><td></td><td></td><td colspan="2"></td><td>→</td><td>BYE(REL#31)</td></tr>
<tr><td></td><td></td><td colspan="2"></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
<tr><td></td><td></td><td colspan="2"></td><td>←</td><td>487 Request terminated</td></tr>
<tr><td></td><td></td><td colspan="2"></td><td>→</td><td>ACK</td></tr>
</table>

Table 15

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

TP309003	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.5
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message 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		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	RSC	→			
	RLC	←			
				←	200 OK INVITE(CON)
				→	ACK
				→	BYE(REL#31)
				←	200 OK BYE(RLC)

TP309005	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.5
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message with the complete called party number, sending a 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		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
	RSC	→		→	BYE(REL#31)
	RLC	←		←	200 OK BYE(RLC)

TP309006	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.5**	
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message 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		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	SIP_MESSAGE_VA
	RSC	→			
	RLC	←			
	CASE A				
				→	CANCEL
				←	200 OK CANCEL
				←	487 Request terminated
				→	ACK
	CASE B				
				→	BYE(REL#31)
				←	200 OK BYE(RLC)
				←	487 Request terminated
				→	ACK

Table 16

Values for test purpose; TP309006	
VA	**SIP MESSAGE_VA**
VA_1	180 Ringing
VA_2	183 Session Progress

5.2.2.9.2 Receipt of Circuit group reset message (GRS)

TP309007	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 7.7.1, 1), 7.7.4, 7.7.5**
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM but before an INVITE has been sent on receipt of GRS message: no action is required on the SIP side other than to terminate local procedures if any are in progress.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→			
	GRS	→			
	GRA	←			

TP309008	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.5**
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving 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		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	GRS	→			
	GRA	←			
				←	SIP_MESSAGE_VA
	CASE A				
				→	CANCEL
				←	200 OK CANCEL
				←	487 Request terminated
				→	ACK
	CASE B				
				→	BYE(REL#31)
				←	200 OK BYE(RLC)
				←	487 Request terminated
				→	ACK

Table 17

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

<table>
<tr><th>TP309009</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses 7.7.1 3), 7.7.4, 7.7.5</th></tr>
<tr><td>TSS reference</td><td colspan="5">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</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="10">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>100 Trying</td></tr>
<tr><td>GRS</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>GRA</td><td>←</td><td></td><td>→</td><td>CANCEL</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(CON)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK CANCEL</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>BYE(REL#31)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP309011	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.5**		
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message with the complete called party number, sending a INVITE message on receipt GRS message **after** a 200 OK response message has been received: • the SUT shall send a BYE request The GRS is not encapsulated.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
	GRS	→		→	BYE(REL#31)
	GRA	←		←	200 OK BYE(RLC)

TP309012	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.5**		
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt GRS message **after** an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established: • the SUT shall send a CANCEL or BYE request The GRS is not encapsulated.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	SIP_MESSAGE_VA
	GRS	→			
	GRA	←			
	CASE A				
				→	CANCEL
				←	200 OK CANCEL
				←	487 Request terminated
				→	ACK
	CASE B				
				→	BYE(REL#31)
				←	200 OK BYE(RLC)
				←	487 Request terminated
				→	ACK

Table 18

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

TP309013	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.5**
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving more than one IAM's sending an INVITE message for each call association on receipt of a GRS message were the Range Parameter value is bigger than "**1**": • the SUT shall send a BYE requests for each call association The GRS is not encapsulated.				
SIP parameter values	BYE1 contains the CSeq of INVITE1 BYE2 contains the CSeq of INVITE2				
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE1(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
	IAM	→		→	INVITE2(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
	GRS	→			
	GRA	←			
				→	BYE1(REL#31)
				←	200 OK BYE(RLC)
				→	BYE2(REL#31)
				←	200 OK BYE(RLC)

5.2.2.9.3 Receipt of Circuit group blocking message (CGB)

TP3090014	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clauses 7.7.1, 1) , 7.7.4**	
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM but before an INVITE has been sent on receipt of CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented": • no action is required on the SIP side other than to terminate local procedures if any are in progress.				
SIP parameter values					
ISUP parameter values	CGB(hardware failure oriented)				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→			
	CGB	→			
	CGBA	←			

TP309015	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4**	
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" **before** a **SIP MESSAGE_VA** response message has been received: • the SUT shall hold the CGB message until a SIP 200 OK response has been received; • the SUT shall send a CANCEL request The CGB is not encapsulated.				
SIP parameter values					
ISUP parameter values	CGB(hardware failure oriented)				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	CGB	→			
	CGBA	←			
				←	SIP_MESSAGE_VA
	CASE A				
				→	CANCEL
				←	200 OK CANCEL
				←	487 Request terminated
				→	ACK
	CASE B				
				→	BYE(REL#31)
				←	200 OK BYE(RLC)
				←	487 Request terminated
				→	ACK

Table 19

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

TP3090016	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 7.7.1 3), 7.7.4**
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" **before** a 200 OK response message has been received: • on subsequently receiving 200 OK INVITE messages , the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent The CGB is not encapsulated.				
SIP parameter values					
ISUP parameter values	CGB(hardware failure oriented)				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	100 Trying
	CGB	→			
	CGBA	←		→	CANCEL
				←	200 OK INVITE(CON)
				→	ACK
				←	200 OK CANCEL
				→	BYE(REL#31)
				←	200 OK BYE(RLC)

TP309017	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message 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		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
	CGB	→		→	BYE(REL#31)
	CGBA	←		←	200 OK BYE(RLC)

TP309018	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" **after** an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established: • the SUT shall send a CANCEL or BYE request The CGB is not encapsulated.				
SIP parameter values					
ISUP parameter values	CGB(hardware failure oriented)				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	SIP_MESSAGE_VA
	CGB	→			
	CGBA	←			
	CASE A				
				→	CANCEL
				←	200 OK CANCEL
				←	487 Request terminated
				→	ACK
	CASE B				
				→	BYE(REL#31)
				←	200 OK BYE(RLC)
				←	487 Request terminated
				→	ACK

Table 20

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

TP309019	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.5**
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving more than one IAM's sending an INVITE message for each call association on receipt of a CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" were the Range and Status Parameter value is bigger than **"1"**: • the SUT shall send a BYE requests for each call association The CGB is not encapsulated.				
SIP parameter values	BYE1 contains the CSeq of INVITE1 BYE2 contains the CSeq of INVITE2				
ISUP parameter values	CGB(hardware failure oriented)				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE1(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
	IAM	→		→	INVITE2(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
	CGB	→			
	CGBA	←			
				→	BYE1(REL#31)
				←	200 OK BYE(RLC)
				→	BYE2(REL#31)
				←	200 OK BYE(RLC)

5.2.2.10 Receipt of Confusion message

<table>
<tr><th>TP310001</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses A.1.1.3</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/ ISUP Messages for special consideration / Confusion message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">183 Session Progress with encapsulated CFN</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">CFN</td></tr>
<tr><td rowspan="9"></td><td>ISUP</td><td></td><td></td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM with unknown parameter)</td></tr>
<tr><td>CFN</td><td>←</td><td></td><td>←</td><td>183 Session Progress(CFN)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Communication</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

5.2.2.11 Receipt of "Suspend" or "Resume" message

TP311001	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clause A.1.1.3**
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Receipt of **Suspend** message	
SIP selection criteria		
ISUP selection criteria		
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message with the complete called party number, on receipt of a **Suspend initiated by the network**: • ensure that the ISUP message is transported through the SIP network encapsulated in the INFO message; • ensure that the called subscriber can successfully clear back and reanswer the call.	
SIP parameter values		
ISUP parameter values		
Comments		

ISUP/BICC		SUT		SIP-I
IAM	→		→	INVITE(IAM)
ACM	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
SUS	←		←	INFO(SUS)
			→	200 OK INFO
RES	←		←	INFO(RES)
			→	200 OK INFO
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

5.2.2.12 Receipt of a Blocking message

TP312001	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clause A.1.1.3.1**
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Receipt of a Blocking message	
SIP selection 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		SUT		SIP-I
BLO	→			
BLA	←			
UBL	→			
UBA	←			

TP312002	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause A.1.1.3.1**
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Receipt of a Blocking message				
SIP selection 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		SUT		SIP-I
	BLO	→			
	BLA	←			
	BLO	←			
	BLA	→			
	UBL	→			
	UBA	←			

TP312003	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause A.1.1.3.1**
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Receipt of a Blocking message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**CGB and CGU sent** Ensure that the SUT is able to respond on a Circuit group blocking message with a CGBA and on a Circuit group unblocking message (both maintenance oriented) with a CGUA. Ensure the CGB / CGU messages are not encapsulated within SIP messages.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	CGB	→			
	CGBA	←			
	CGU	→			
	CGUA	←			

TP312004	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause A.1.1.3.1**
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Receipt of a Blocking message				
SIP selection 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 [4]**			**ISUP reference: Q.1912.5 [1], clause A.1.1.3.1 Q.784 [i.11], clause 1.3.2.4**	
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Receipt of a Blocking message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that a received IAM will unblock a remotely blocked circuit.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	BLO	→			
	BLA	←			
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

5.2.2.13 Receipt of a user part test message

TP313001	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause A.1.1.3.1 Q.784 [i.11], clause 1.3.2.4**	
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Receipt of a user part test message				
SIP selection criteria					
ISUP selection criteria	PICS 4/22				
Test purpose	Ensure that on receipt of a user part test message the SUT will respond by sending a user part available message. Ensure that the user part test message is not encapsulated within SIP messages.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	UPT	→			
	UPA	←			

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

TP313003	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause A.1.1.3.1**
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Receipt of a user part test message				
SIP selection criteria					
ISUP selection criteria	PICS 4/22				
Test purpose	**T4 Waiting to receive a response to a user part test message.** Ensure that the SUT is able to restart the availability test procedure after expiry of timer T4.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	UPT	←			
	T4 expiry				
	UPT	←			
	UPA	→			

5.2.2.14 Segmentation

TP314001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause A.1.1.3.1**
TSS reference	ISUP-SIP/ ISUP 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 successfully completed if segmentation applies in forward direction.				
SIP parameter values	INVITE - encapsulated IAM: Forward call indicator absent or set to "no additional information will be sent" No action takes place on the SIP side				
ISUP parameter values	IAM: optional forward call indicator: additional information will be sent in a segmentation message SGM: optional parameters				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→			
	SGM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE

5.3 Test purposes for the Supplementary Services

5.3.1 Calling Line Identification Presentation (CLIP)

<table>
<tr><th>TP401001</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1.3</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Calling Party number network provided, transferred in O-MGCF

Ensure that the SUT can successfully transmit a call having a calling party number with the screening indicator set to "network provided" and the presentation restricted indicator set to "presentation allowed".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">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</td></tr>
<tr><td rowspan="7">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP401002</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1.3</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Calling Party number networkprovided, Calling Subaddress transferred in O-MGCF

Ensure that the SUT can successfully transmit a call having a calling party number with the screening indicator set to "network provided" and an access transport parameter containing the calling sub-address.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">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
Access transport parameter including the subaddress information</td></tr>
<tr><td rowspan="7">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP401003	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Calling Party Number user provided transferred in O-MGCF* Ensure that the SUT can successfully transmit a call having the **calling party number** with the screening indicator set to "user provided, verified and passed" and the presentation restricted indicator set to "presentation allowed".				
SIP parameter values					
ISUP parameter values	IAM; **Calling party number 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	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP401004	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Calling Party Number user provided and calling subaddress transferred in O-MGCF* Ensure that the SUT can successfully transmit a call having a **calling party number** with the screening indicator set to "user provided, verified and passed" and an **access transport** parameter containing the **calling sub-address.**				
SIP parameter values					
ISUP parameter values	IAM; **Calling party number 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 Access transport parameter including the subaddress information				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><td>TP401005</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1.3</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Calling Party Number network provided and additional calling party number user provided not verified transferred in O-MGCF.

Ensure that the SUT can successfully transmit a call having a default calling party number with the screening indicator set to "network provided" and a generic number containing the additional calling party number with the screening indicator set to "user provided, not verified" and the presentation restricted indicator set to "presentation allowed".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">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 parameter
Address signals = PIXIT2
Numbering plan indicator = '001'B
Nature of address indicator = '0000011'B
Screening indicator = '00'B
Presentation restricted indicator = presentation allowed, '00'B</td></tr>
<tr><td rowspan="7">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP401006	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Calling Party Number network provided, additional calling party number user provided not verified and calling subaddress transferred in O-MGCF.* Ensure that the SUT can successfully transmit a call having a default calling party number with the screening indicator set to "network provided", 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 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	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP401007	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	PICS 6/8				
Test purpose	*Calling party number discarded to due bilateral agreement in the I-MGCF.* Ensure that the calling party number is discarded in case of bilateral agreements, if the address presentation restricted indicator is set to "presentation allowed" (see note).				
SIP parameter values					
ISUP parameter values	IAM; **No calling party number parameter**				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC
NOTE: This bilateral agreement prohibits the transferral of the calling party number **in any case**. The test with the address presentation restricted indicator set to "presentation restricted" is a CLIR test.					

TP401008	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	PICS 6/7				
Test purpose	*Additional Calling party number is discarded to due bilateral agreements in the I-MGCF* Ensure that the additional calling party number in the **generic number** is discarded in case of bilateral agreements, if the address presentation restricted indicator is set to "presentation allowed".				
SIP parameter values					
ISUP parameter values	IAM; **No calling party number parameter**				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC
NOTE: This bilateral agreement prohibits the transferral of the calling party number in any case. The test with the address presentation restricted indicator set to "presentation restricted" is a CLIR test.					

TP401009	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	PICS 6/6				
Test purpose	*Calling party number is omitted if the presentation restriction indicator is set to address not available in the I-MGCF* Ensure that the **calling party number** is omitted, if the address presentation restricted indicator is set to "address not available".				
SIP parameter values					
ISUP parameter values					
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP401010	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Calling party number is sent as received* 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)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP401011	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Additional calling party number is sent as received* 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 values					
ISUP parameter values					
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP401012	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Additional calling party number is omitted in the I-MGCF* Ensure that if the **calling party number** is not sent, then an additional calling party number in a **generic number** will be omitted.				
SIP parameter values	INVITE: No calling party number included in the encapsulated IAM, additional calling party number included.				
ISUP parameter values	IAM; **No calling party number parameter** **No generic number parameter**				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP401013	SIP reference: RFC 3261 [4]				ISUP reference: Q.731 [i.2], clause 3.5
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	PICS 1/7				
Test purpose	*Convert the Calling party number into the international format in the I-MGCF* Ensure that the SUT can convert the **calling party number** into an international number, setting the nature of address indicator to "international number" and can pass on the address presentation restricted indicator and the screening indicator transparently.				
SIP parameter values					
ISUP parameter values	IAM; **Calling party number parameter** Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000100'B Screening indicator = '11'B Presentation restricted indicator =presentation allowed, '00'B				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP401014	**SIP reference: RFC 3261 [4]**			**ISUP reference: 3.5/Q.731 [i.2]**
TSS reference	ISUP-SIP-ISUP/SS/CLIP			
SIP selection criteria				
ISUP selection criteria	PICS 1/7			
Test purpose	*Converting the additional calling party number to international format in the I-MGCF* Ensure that the SUT can convert the additional calling party number in the **generic 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 values				
ISUP parameter values	**IAM** **Calling party number parameter** Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000100'B Screening indicator = '11'B Presentation restricted indicator =presentation allowed, '00'B **Generic number parameter** Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000100'B Screening indicator = '00'B Presentation restricted indicator =presentation allowed, '00'B			
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

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

<table>
<tr><th>TP401016</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
3.5/Q.731 [i.2]</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/8</td></tr>
<tr><td>Test purpose</td><td colspan="5">Converting the calling party number to national format, if necessary in the O-MGCF

Ensure that the country code in the address signals of the calling party number is removed if it is the network's own country code. The nature of address indicator shall be set to "national (significant) number". The address presentation restricted indicator shall be transferred transparently.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: encapsulated 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</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM
Calling party number parameter
Address signals = PIXIT1
Numbering plan indicator = '001'B
Nature of address indicator = '0000100'B
Screening indicator = '11'B
Presentation restricted indicator = presentation allowed, '00'B</td></tr>
<tr><td rowspan="7">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><td>TP401017</td><td colspan="5">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
3.5/Q.731 [i.2]</td></tr>
<tr><td>TSS reference:</td><td colspan="7">ISUP-SIP-ISUP/SS/CLIP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="7"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="7">PICS 1/8</td></tr>
<tr><td>Test purpose</td><td colspan="7">Converting the additional calling party number to national format, if necessary in the O-MGCF

Ensure that the country code in the address signals of the generic number coded as an "additional calling party number", if the numbering plan indicator is "ISDN Telephony" is removed if it is the network's own country code. The nature of address indicator shall be set to "national (significant) number". The address presentation restricted indicator shall be transferred transparently.</td></tr>
<tr><td>SIP parameter values</td><td colspan="7">INVITE: encapsulated IAM
    Generic number parameter
    Address signals = PIXIT2
    Numbering plan indicator = '001'B
    Nature of address indicator = '0000011'B
    Screening indicator = '11'B
    Presentation restricted indicator = presentation allowed, '00'B</td></tr>
<tr><td>ISUP parameter values</td><td colspan="7">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 parameter
Address signals = PIXIT2
Numbering plan indicator = '001'B
Nature of address indicator = '0000011'B
Screening indicator = '00'B
Presentation restricted indicator = presentation allowed, '00'B</td></tr>
<tr><td rowspan="7">Comments</td><td>SIP-I</td><td></td><td colspan="2">SUT</td><td></td><td colspan="2">ISUP</td></tr>
<tr><td>IAM</td><td>→</td><td colspan="2"></td><td>→</td><td colspan="2">INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td colspan="2"></td><td>←</td><td colspan="2">180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td colspan="2"></td><td>←</td><td colspan="2">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="4">Conversation</td><td colspan="2"></td></tr>
<tr><td>REL</td><td>→</td><td colspan="2"></td><td>→</td><td colspan="2">BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td colspan="2"></td><td>←</td><td colspan="2">200 OK BYE(RLC)</td></tr>
</table>

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

TP401019	**SIP reference: RFC 3261 [4]**				**ISUP reference: 3.5/Q.731 [i.2]**
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Handling of address presentation restricted indicator set to "address not available" in the I-MGCF* Ensure that the screening indicator shall be set to "network provided" if the address presentation restricted indicator in **calling party number** is set to "address not available"(see note).				
SIP parameter values					
ISUP parameter values	IAM; **Calling party number parameter** Address signals = PIXIT1 Numbering plan indicator = ' *'B Nature of address indicator = '*'B Screening indicator = '11'B Presentation restricted indicator =address not available, '10'B				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC
NOTE: The coding "address not available" is a national option (@).					

<table>
<tr><th>TP401020</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clause 7.1.3</th></tr>
<tr><td>**TSS reference**</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIP</td></tr>
<tr><td>**SIP selection criteria**</td><td colspan="5"></td></tr>
<tr><td>**ISUP selection criteria**</td><td colspan="5"></td></tr>
<tr><td>**Test purpose**</td><td colspan="5">*O-MGCF: Calling party number and Additional calling party number not received*

Ensure that when the SUT has received an IAM message whereby Calling Party Number parameter and the Generic Number are not applicable.
Sends an INVITE message without the "P-Asserted-Identity header field", the "From header field" set to "anonymous@anonymous.invalid". No Privacy header field included.</td></tr>
<tr><td>**SIP parameter values**</td><td colspan="5">INVITE: No P-Asserted Identity, From Header: anonymous@anonymous.inv</td></tr>
<tr><td>**ISUP parameter values**</td><td colspan="5">IAM; no Calling party number and no Additional calling party number present</td></tr>
<tr><td rowspan="7">**Comments**</td><td>**ISUP**</td><td></td><td>**SUT**</td><td></td><td>**SIP-I**</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">**Conversation**</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP401021</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clause 7.1.3</th></tr>
<tr><td>**TSS reference:**</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIP</td></tr>
<tr><td>**SIP selection criteria**</td><td colspan="5"></td></tr>
<tr><td>**ISUP selection criteria**</td><td colspan="5"></td></tr>
<tr><td>**Test purpose**</td><td colspan="5">*O-MGCF: Setting of From header*

Ensure that when the SUT has received an IAM message whereby **Calling Party Number** parameter is **not applicable** and the **Generic Number is applicable** whereby the address presentation restriction parameter is set to "presentation allowed" and the Nature of Address Indicator is set to NoAS_VALUE.
Sends an INVITE message without the "P-Asserted-Identity header field", a "From header field" where the user portion of the addr-spec is set to value of the additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN and no "Privacy Header field".</td></tr>
<tr><td>**SIP parameter values**</td><td colspan="5">INVITE: no P-Asserted-Identity, no Privacy header, From header contains the value of the additional calling party number</td></tr>
<tr><td>**ISUP parameter values**</td><td colspan="5">IAM; no Calling party number present, Additional calling party number present</td></tr>
<tr><td rowspan="7">**Comments**</td><td>**ISUP**</td><td></td><td>**SUT**</td><td></td><td>**SIP-I**</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">**Conversation**</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP401022	SIP reference: RFC 3261 [4]		ISUP reference: Q.1912.5 [1], clause 7.1.3		
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*O-MGCF: Setting of P-Asserted header header* Ensure that when the SUT has received an IAM message, the **Calling Party Number is applicable** whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to **presentation allowed** and the **Generic Number is not applicable** Sends an INVITE message with: • the "P-Asserted-Identity header field" where the user portion of the addr-spec is set to value of the calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN; • a "From header field" where the "addr-spec" is set to where the user portion of the addr-spec is set to value of the additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN; • without "Privacy Header field" or "id" is not included.				
SIP parameter values	INVITE: P-Asserted-Identity derived from the calling party number, Privacy=id, From header derived from the additional calling party number				
ISUP parameter values	IAM; Calling party number is present and no Additional calling party number is present				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP401023	SIP reference: RFC 3261 [4]		ISUP reference: Q.1912.5 [1], clause 7.1.3		
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*O-MGCF: Setting of P-Asserted header header and From header* Ensure that when the SUT has received an IAM message, the **Calling Party Number is applicable** whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to **presentation allowed** and the **Generic Number is applicable** Sends an INVITE message with: • the "P-Asserted-Identity header field" , " where the user portion of the addr-spec is set to value of the calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN; • "From header field" " where the user portion of the addr-spec is set to value of the additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN; • and without "Privacy Header field" or "id" is not included.				
SIP parameter values	INVITE: P-Asserted-Identity derived from the calling party number, no Privacy header, From header derived from the additional calling party number				
ISUP parameter values	IAM; Calling party number and Additional calling party number are present				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

Values for test purpose TP401021, TP401022, TP401023		
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: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.1.3**
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	PICS 1/7				
Test purpose	*Calling party derived from the P-Asserted-Identity international number* Ensure when no calling party number is included in the encapsulated IAM or the calling party number in the in the encapsulated IAM is not identical to the P-Asserted-Identity, no Privacy value "id" received. Send an IAM the calling party number is derived from SIP P-Asserted-Identity. The Address Presentation Restricted Indicator is set to Presentation allowed.				
SIP parameter values	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value "id" is not present				
ISUP parameter values	**IAM** message with the **Calling party number parameter** coded Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: "international number"				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP401025	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP			
SIP selection criteria				
ISUP selection criteria	NOT PICS 1/7			
Test purpose	*Calling party derived from the P-Asserted-Identity national (significant) number* Ensure when no calling party number is included in the encapsulated IAM or the calling party number in the in the encapsulated IAM is not identical to the P-Asserted-Identity, no Privacy value "id" received. Send an IAM the calling party number is derived from SIP P-Asserted-Identity. The Address Presentation Restricted Indicator is set to Presentation allowed.			
SIP parameter values	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value "id" is not present			
ISUP parameter values	**IAM** message with the **Calling party number parameter** coded Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: "national (significant) number"			
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP401026	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP			
SIP selection criteria				
ISUP selection criteria	PICS 1/7			
Test purpose	*Additional calling party number derived from the From header international number* Ensure when no additional calling party number is included in the encapsulated IAM or the additional calling party number in the in the encapsulated IAM is not identical to the From header field, no Privacy value "id" received. Send an IAM the additional calling party number is derived from From header field. The Address Presentation Restricted Indicator is set to Presentation allowed.			
SIP parameter values	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value "id" is not present			
ISUP parameter values	**IAM** message with the **Additional Calling party number parameter** coded Address signals = number derived from SIP From header Screening indicator = User provided, not verified" Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: "international number"			
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP401027	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1.3
TSS reference:	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	NOT PICS 1/7				
Test purpose	*Additional calling party number derived from the From header national (significant) number* Ensure when no additional calling party number is included in the encapsulated IAM or the additional calling party number in the 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 values	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value "id" is not present				
ISUP parameter values	**IAM** message with the **Additional Calling party number parameter** coded Address signals = number derived from SIP From header Screening indicator = User provided, not verified" Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: "national (significant) number"				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

5.3.2 Calling Line Identification Restriction (CLIR)

TP402001	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1] Q.731 [i.2], clause 4.5.2.1.1
TSS reference:	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Calling party number network provided presentation restricted is passed.* Ensure that the SUT can successfully transmit 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					
ISUP parameter values	IAM; **Calling party number parameter** Screening indicator = '11'B Address presentation restricted parameter = '01'B **Generic number parameter** not present **Access transport parameter is not** including the subaddress information				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><td>TP402002</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 4.5.2.1.1</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Restricted calling party number (network provided) with calling sub-address
Ensure that the SUT can pass transparently 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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM;
Calling party number parameter
Screening indicator = '11'B
Address presentation restricted parameter = '01'B
Generic number parameter not present
Access transport parameter including subaddress information</td></tr>
<tr><td rowspan="7">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><td>TP402003</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 4.5.2.1.1</td></tr>
<tr><td>TSS reference:</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Restricted calling party number (user provided, verified and passed)
Ensure that the SUT can pass transparently a call having the calling party number with the screening indicator set to "user provided, verified and passed" and the address presentation restricted indicator set to "presentation restricted".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM
Calling party number parameter
Address signals = PIXIT1
Numbering plan indicator = '001'B
Nature of address indicator = '0000011'B
Screening indicator = '01'B
Address presentation restricted parameter = '01'B</td></tr>
<tr><td rowspan="7">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP402004</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 4.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM
Calling party number parameter
Address signals = PIXIT1
Numbering plan indicator = '001'B
Nature of address indicator = '0000011'B
Screening indicator = '01'B
Address presentation restricted parameter = '01'B
Access transport parameter including subaddress information</td></tr>
<tr><td rowspan="7">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP402005</th><th colspan="4">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 4.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">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</td></tr>
<tr><td rowspan="7">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><td>TP402006</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 4.5.2.1.1</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">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</td></tr>
<tr><td rowspan="7">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><td>TP402007</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 4.5.2.1.1</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 6/4</td></tr>
<tr><td>Test purpose</td><td colspan="5">Discarding the calling party number if the presentation is restricted
Ensure that the calling party number is discarded in case of bilateral agreements, if the address presentation restricted indicator is set to "presentation restricted".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM;
No Calling party number parameter</td></tr>
<tr><td rowspan="7">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><td>TP402008</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 4.5.2.1.1</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 6/4 AND PICS 6/5</td></tr>
<tr><td>Test purpose</td><td colspan="5">Discarding the additional calling party number if the presentation is restricted
Ensure that the additional calling party number in the generic number is discarded in case of bilateral agreements, if the address presentation restricted indicator is set to "presentation restricted".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM;
No Calling party number parameter
No Generic number parameter</td></tr>
<tr><td rowspan="7">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><td>TP402009</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 4.5.2.1.1</td></tr>
<tr><td>TSS reference:</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">I-MGCF: Calling party number received in the INVITE is sent in the IAM

Ensure that the calling party number contained in the encapsulated IAM is unchanged sent in the ISUP IAM.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="7">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP402010	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 4.5.2.1.1	
TSS reference:	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*I-MGCF: Additional calling party number received in the INVITE is sent in the IAM* Ensure that the additional calling party number contained in the encapsulated IAM is unchanged sent in the ISUP IAM.				
SIP parameter values					
ISUP parameter values					
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP402011	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.1.3	
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that when the SUT has received an IAM message, the **Calling Party Number is applicable** whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to **presentation restricted** and the **Generic Number is not applicable** Sends an INVITE message with: • the "P-Asserted-Identity header field" where the user portion of the addr-spec is set to value of the additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN; • a "From header field" where the user portion of the addr-spec is set to value of the additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN; • and with "Privacy Header field" set to "id".				
SIP parameter values	INVITE: P-Asserted-Identity, From header field, Privacy "id"				
ISUP parameter values	IAM: Calling party number. No additional calling party number				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><td>TP402012</td><td colspan="4">SIP reference: RFC 3261 [4]</td><td>ISUP reference:
Q.1912.5 [1], clause 7.1.3</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation restricted and the Generic Number is applicable.
Sends an INVITE message with:
• the "P-Asserted-Identity header field", where the user portion of the addr-spec is set to value of the calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN;
• "From header field" where the user portion of the addr-spec is set to value of the additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN;
• and with "Privacy Header field" is set to "id".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: P-Asserted-Identity, From header field, Privacy "id"</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: Calling party number. additional calling party number</td></tr>
<tr><td rowspan="7">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><td colspan="3">Values for test purpose TP401012</td></tr>
<tr><td>NoAS_VALUE</td><td>ISUP parameter values</td><td>SIP parameter values:</td></tr>
<tr><td>VA_01</td><td>IAM
NoAS_VALUE: "national (significant) number"(NDC+SN)</td><td>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</td></tr>
<tr><td>VA_02</td><td>IAM
NoAS_VALUE: "international number" ("+"CC+NDC+SN)</td><td>INVITE
FHf_Addr_SPEC_ID: the complete GenericNumber Address Signals is mapped to the user portion of URI scheme used.</td></tr>
</table>

TP402013	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.1.3**
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria	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, 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 values	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value "id" is present				
ISUP parameter values	**IAM** message with the **Calling party number parameter** coded Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: "international number"				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP402014	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.1.3**
TSS reference:	ISUP-SIP-ISUP/SS/CLIR				
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, 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 values	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value "id" is present				
ISUP parameter values	**IAM** message with the **Calling party number parameter** coded Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: "national (significant) number"				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP402015	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria	PICS 1/7				
Test purpose	Ensure when no additional calling party number is included in the encapsulated IAM or the additional calling party number in the in the encapsulated IAM is not identical to the From header field, Privacy value "id" received. Send an IAM the additional calling party number is derived from From header field. The Address Presentation Restricted Indicator is set to Presentation restricted.				
SIP parameter values	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value "id" is present				
ISUP parameter values	**IAM** message with the **Additional Calling party number parameter** coded Address signals = number derived from SIP From header Screening indicator = User provided, not verified" Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: "international number"				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP402016	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria	NOT PICS 1/7				
Test purpose	Ensure when no additional calling party number is included in the encapsulated IAM or the additional calling party number in the in the encapsulated IAM is not identical to the From header field, Privacy value "id" received. Send an IAM the additional calling party number is derived from From header field. The Address Presentation Restricted Indicator is set to Presentation restricted.				
SIP parameter values	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value "id" is present				
ISUP parameter values	**IAM** message with the **Additional Calling party number parameter** coded Address signals = number derived from SIP From header Screening indicator = User provided, not verified" Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: "national (significant) number"				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

5.3.3 Connected line identification presentation (COLP)

TP403001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 5.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/COLP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Initiate COLP request** Ensure that the exchange can initiate successfully a call requesting the COLP service in the **optional forward call indicators.**				
SIP parameter values					
ISUP parameter values	**IAM;** **optional forward call indicators** Connected line identity request indicator = requested				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP403002</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 5.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/COLP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Connected number (user provided, verified and passed) with connected sub-address
Ensure that the SUT passes transparently a default connected number with the screening indicator set to "verified and passed" and an access transport parameter containing the connected sub-address.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM;
optional forward call indicators
Connected line identity request indicator: requested
a)
ANM;
Connected number parameter
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '01'B
Address signals = PIXIT
and an access transport parameter containing the connected sub-address.
b)
CON;
Connected number parameter
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '01'B
Address signals = PIXIT
and an access transport parameter containing the connected sub-address</td></tr>
<tr><td rowspan="10">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td colspan="5">CASE A</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td colspan="5">CASE B</td></tr>
<tr><td>200 OK INVITE(CON)</td><td>←</td><td></td><td>←</td><td>CON</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP403003</th><th>SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 5.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="2">ISUP-SIP-ISUP/SS/COLP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="2"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="2"></td></tr>
<tr><td>Test purpose</td><td colspan="2">Connected number (user provided, not verified) without connected sub-address
Ensure that the SUT passes transparently a default connected number with the screening indicator set to "network provided", a generic number containing the additional connected number with the screening indicator set to "user provided, not verified" without an access transport parameter containing the connected sub-address.</td></tr>
<tr><td>SIP parameter values</td><td colspan="2"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="2">IAM;
optional forward call indicators
Connected line identity request indicator: requested
a)
ANM;
Connected number parameter
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '11'B
Address signals = PIXIT
Additional connected number present
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '00'B
Address signals = PIXIT

b)
CON;
Connected number parameter
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '11'B
Address signals = PIXIT
Additional connected number present
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '00'B
Address signals = PIXIT</td></tr>
<tr><td>Comments</td><td colspan="2">
<table>
<tr><th>SIP-I</th><th></th><th>SUT</th><th></th><th>ISUP</th></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td colspan="5">CASE A</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td colspan="5">CASE B</td></tr>
<tr><td>200 OK INVITE(CON)</td><td>←</td><td></td><td>←</td><td>CON</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>
</td></tr>
</table>

<table>
<tr><td>TP403004</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 5.5.2.1.1</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/COLP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/7</td></tr>
<tr><td>Test purpose</td><td colspan="5">Converting the connected number to national format, if necessary
Ensure that the country code in the address signals of the connected number is removed if it is the network's own country code. The nature of address indicator shall be set to "national (significant) number", the address presentation restricted indicator and the screening indicator shall be transferred transparently.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">200 OK: 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 = ISUP_SI
Address signals = PIXIT</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM;
optional forward call indicators
Connected line identity request indicator: requested
a)
ANM;
Connected number parameter
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000100'B
Numbering plan indicator = '001'B
Screening indicator = ISUP_SI
Address signals = CC+PIXIT
b)
CON;
Connected number parameter
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000100'B
Numbering plan indicator = '001'B
Screening indicator = ISUP_SI
Address signals = CC+PIXIT
Generic number parameter not present</td></tr>
<tr><td rowspan="10">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td colspan="5">CASE A</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td colspan="5">CASE B</td></tr>
<tr><td>200 OK INVITE(CON)</td><td>←</td><td></td><td>←</td><td>CON</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><td>TP403005</td><td colspan="2">SIP reference: RFC 3261 [4]</td><td colspan="3">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 5.5.2.1.1</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/COLP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/7</td></tr>
<tr><td>Test purpose</td><td colspan="5">Converting the additional connected number to national format, if necessary
Ensure that the country code in the address signals of the generic number coded as an "additional connected number", if the numbering plan indicator is "ISDN Telephony" is removed if it is the network's own country code. The nature of address indicator shall be set to "national (significant) number", the address presentation restricted indicator and the screening indicator shall be transferred transparently.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">200 OK: encapsulated ANM or CON
additional connected number
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '01'B
Address signals = PIXIT</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM;
optional forward call indicators
Connected line identity request indicator: requested
a)
ANM;
Connected number parameter present
additional connected number
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000100'B
Numbering plan indicator = '001'B
Screening indicator = '01'B
Address signals = CC+PIXIT

b)
CON;
Connected number parameter present
additional connected number
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000100'B
Numbering plan indicator = '001'B
Screening indicator = '01'B
Address signals = CC+PIXIT</td></tr>
<tr><td rowspan="10">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td colspan="5">CASE A</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td colspan="5">CASE B</td></tr>
<tr><td>200 OK INVITE(CON)</td><td>←</td><td></td><td>←</td><td>CON</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP403006</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 5.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/COLP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/8 AND PICS 7/5</td></tr>
<tr><td>Test purpose</td><td colspan="5">Adding a prefix to an international connected number
Ensure that a prefix is added to the connected number and the nature of address indicator is set to "unknown" (see note).</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">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</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">ANM/CON:
Connected number parameter
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000010'B
Numbering plan indicator = '001'B
Screening indicator = '11'B
Address signals = Prefix+PIXIT</td></tr>
<tr><td rowspan="10">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td colspan="5">CASE A</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td colspan="5">CASE B</td></tr>
<tr><td>CON</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(CON)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
<tr><td colspan="6">NOTE: The coding "unknown" is a national option (@).</td></tr>
</table>

<table>
<tr><th>TP403007</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 5.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/COLP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/8 AND PICS 7/3</td></tr>
<tr><td>Test purpose</td><td colspan="5">Discarding the connected number in case of bilateral agreements
Ensure that the connected number is discarded in case of bilateral agreements, if the address presentation restricted indicator is set to "presentation allowed" (see note).</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">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</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM
optional forward call indicators
Connected line identity request indicator: requested
a)
ANM
No Connected number parameter
b)
CON;
No Connected number parameter</td></tr>
<tr><td rowspan="10">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td colspan="5">CASE A</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td colspan="5">CASE B</td></tr>
<tr><td>CON</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(CON)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
<tr><td colspan="6">NOTE: This bilateral agreement prohibits the transferral of the connected number in any case. The test with the address presentation restricted indicator set to "presentation restricted" is a COLR test.</td></tr>
</table>

<table>
<tr><td>TP403008</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 5.5.2.1.1</td></tr>
<tr><td>TSS reference:</td><td colspan="5">ISUP-SIP-ISUP/SS/COLP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/8 AND PICS 7/4</td></tr>
<tr><td>Test purpose</td><td colspan="5">Discarding the additional connected number in case of bilateral agreements
Ensure that the additional connected number in the generic number is discarded in case of bilateral agreements, if the address presentation restricted indicator is set to "presentation allowed" (see note).</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">200 OK INVITE with encapsulated ANM or CON
Additional Connected number parameter
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '00'B
Address signals = PIXIT</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM;
optional forward call indicators
Connected line identity request indicator: requested
a)
ANM;

No Connected number parameter
No Additional connected number present
b)
CON;
No Connected number parameter
No Additional connected number present</td></tr>
<tr><td rowspan="10">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td colspan="5">CASE A</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td colspan="5">CASE B</td></tr>
<tr><td>CON</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(CON)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
<tr><td colspan="6">NOTE: This bilateral agreement prohibits the transferral of the additional connected number in the generic number in any case.</td></tr>
</table>

TP403009	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 5.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/COLP	
SIP selection criteria		
ISUP selection criteria	PICS 1/8	
Test purpose	**Converting the connected number to international format** Ensure that the exchange can convert the **connected number** into an international number, setting the nature of address indicator to "international number" and can pass on the address presentation restricted indicator and the screening indicator transparently.	
SIP parameter values	200 OK INVITE with encapsulated ANM or CON **Connected number parameter** Address presentation restricted parameter = '00'B Nature of address indicator = '0000100'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = CC+PIXIT	
ISUP parameter values	**IAM;** **optional forward call indicators** Connected line identity request indicator: requested a) **ANM** **Connected number parameter** Address presentation restricted parameter = '00'B Nature of address indicator = '0000100'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = PIXIT Presentation restricted indicator = '00'B **additional connected number** present b) **CON;** **Connected number parameter** Address presentation restricted parameter = '00'B Nature of address indicator = '0000100'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = PIXIT Presentation restricted indicator = '00'B **additional connected number** present	
Comments		

SIP-I		SUT		ISUP
INVITE(IAM)	→		→	IAM
CASE A				
180 Ringing(ACM)	←		←	ACM
200 OK INVITE(ANM)	←		←	ANM
CASE B				
200 OK INVITE(CON)	←		←	CON
		Conversation		
BYE(REL)	→		→	REL
200 OK BYE(RLC)	←		←	RLC

TP403010	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 5.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/COLP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Handling unrequested COL** Ensure that the call can be successfully set up if the SUT receives an unsolicited COL.				
SIP parameter values	200 OK INVITE with encapsulated ANM or CON **Connected number parameter** Address presentation restricted parameter = '00'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = PIXIT				
ISUP parameter values	**IAM;** **optional forward call indicators** Connected line identity request indicator: **not requested** a) **ANM;** **Connected number parameter** Address presentation restricted parameter = '00'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = PIXIT **additional connected number** present b) **CON;** **Connected number parameter** Address presentation restricted parameter = '00'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = PIXIT **additional connected number** present				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	CASE A				
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	CASE B				
	200 OK INVITE(CON)	←		←	CON
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><td>TP403012</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 5.5.2.1.1</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/7</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that an ANM or CON encapsulated in a 200 OK INVITE is sent on the ISUP side without changing. The connected number is unchanged. The ATP contained the connected sub address is included.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">200 OK INVITE: encapsulated ANM or CON included</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">a)
ANM;
Connected number parameter
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '11'B
Address signals = PIXIT
and an access transport parameter containing the connected sub-address.
b)
CON;
Connected number parameter
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '11'B
Address signals = PIXIT
and an access transport parameter containing the connected sub-address.</td></tr>
<tr><td rowspan="10">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td colspan="5">CASE A</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td colspan="5">CASE B</td></tr>
<tr><td>CON</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(CON)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP403013	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 5.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/CLIR	
SIP selection criteria		
ISUP selection criteria		
Test purpose	*O-MGCF: connected number and additional connected number transferred transparently* Ensure that an ANM or CON encapsulated in a 200 OK INVITE is sent on the ISUP side without changing. The connected number is unchanged. The ATP contained the connected sub address is included.	
SIP parameter values	200 OK INVITE: encapsulated ANM or CON included	
ISUP parameter values	a) **ANM;** **Connected number parameter** Address presentation restricted parameter = '00'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = PIXIT **Additional connected number** present Address presentation restricted parameter = '00'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '00'B Address signals = PIXIT and an **access transport** parameter containing the connected sub-address. b) **CON;** **Connected number parameter** Address presentation restricted parameter = '00'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = PIXIT **Additional connected number** present Address presentation restricted parameter = '00'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '00'B Address signals = PIXIT and an **access transport** parameter containing the connected sub-address.	

Comments

ISUP		**SUT**		**SIP-I**
IAM	→		→	INVITE(IAM)
CASE A				
ACM	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
CASE B				
CON	←		←	200 OK INVITE(CON)
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

5.3.4 Connected Line Identification Restriction (COLR)

TP404001	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 6.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/COLR	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Passing on information relating to COLR** Ensure that the SUT shall pass transparently all information related to the COLR supplementary service in the address presentation restricted indicator of the connected number.	
SIP parameter values		
ISUP parameter values	IAM; **optional forward call indicators** Connected line identity request indicator: requested a) **ANM;** **Connected number parameter** Address presentation restricted parameter = '01' B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '01'B Address signals = PIXIT b) **CON;** **Connected number parameter** Address presentation restricted parameter = '01' B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '01'B Address signals = PIXIT	
Comments		

SIP-I		**SUT**		**ISUP**
INVITE(IAM)	→		→	IAM
CASE A				
180 Ringing(ACM)	←		←	ACM
200 OK INVITE(ANM)	←		←	ANM
CASE B				
200 OK INVITE(CON)	←		←	CON
		Conversation		
BYE(REL)	→		→	REL
200 OK BYE(RLC)	←		←	RLC

TP404002	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 6.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/COLR	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Passing on information relating to COLR** Ensure that the SUT shall pass transparently all information related to the COLR supplementary service in the address presentation restricted indicator of the **connected number** and the additional connect number in the **generic number.**	
SIP parameter values		
ISUP parameter values	IAM; **optional forward call indicators** Connected line identity request indicator: requested a) **ANM;** **Connected number parameter** Address presentation restricted parameter = '01' B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = PIXIT **Additional connected number present** Address presentation restricted parameter = '01' B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '00'B Address signals = PIXIT b) **CON;** **Connected number parameter** Address presentation restricted parameter = '01' B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = PIXIT **Additional connected number present** Address presentation restricted parameter = '01' B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '00'B Address signals = PIXIT	
Comments		

SIP-I		SUT		ISUP
INVITE(IAM)	→		→	IAM
CASE A				
180 Ringing(ACM)	←		←	ACM
200 OK INVITE(ANM)	←		←	ANM
CASE B				
200 OK INVITE(CON)	←		←	CON
		Conversation		
BYE(REL)	→		→	REL
200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP404003</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 6.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/COLR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Restricted connected number (user provided, verified and passed) with connected sub-address
Ensure that the SUT can pass transparently a connected number with the screening indicator set to "user provided, verified and passed" and with the address presentation restricted indicator set to "presentation restricted", if the user provided COL is valid. Additionally, an access transport parameter containing the connected sub-address shall also be provided.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM;
optional forward call indicators
Connected line identity request indicator: requested
a)
ANM;
Connected number parameter
Address presentation restricted parameter = '01' B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '01'B
Address signals = PIXIT
access transport parameter containing the connected sub-address
b)
CON;
Connected number parameter
Address presentation restricted parameter = '01' B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '01'B
Address signals = PIXIT
access transport parameter containing the connected sub-address</td></tr>
<tr><td rowspan="10">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td colspan="5">CASE A</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td colspan="5">CASE B</td></tr>
<tr><td>200 OK INVITE(CON)</td><td>←</td><td></td><td>←</td><td>CON</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP404004	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 6.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/COLR				
SIP selection criteria					
ISUP selection criteria	PICS 7/1				
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 restricted indicator is set to "presentation restricted".				
SIP parameter values	200 INVITE: encapsulated ANM or CON No Connected number parameter included				
ISUP parameter values	**IAM;** **optional forward call indicators** Connected line identity request indicator: requested a) **ANM;** **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 b) **CON;** **Connected number parameter** Address presentation restricted parameter = '01'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = PIXIT				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	CASE A				
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	CASE B				
	200 OK INVITE(CON)	←		←	CON
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><td>TP404005</td><td colspan="5">SIP reference: RFC 3261 [4]</td><td>ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 6.5.2.1.1</td></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP-ISUP/SS/COLR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6">PICS 7/2</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">Discarding the additional connected number in the generic number if the presentation is restricted
Ensure that the additional connected number in the generic number is discarded in case of bilateral agreements, if the address presentation restricted indicator is set to "presentation restricted".</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">200 INVITE: encapsulated ANM or CON
No Additional Connected number parameter included</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">IAM;
optional forward call indicators
Connected line identity request indicator: requested
a)
ANM;
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

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</td></tr>
<tr><td rowspan="11">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td colspan="2">ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td colspan="2">IAM</td></tr>
<tr><td colspan="6">CASE A</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td colspan="2">ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td colspan="2">ANM</td></tr>
<tr><td colspan="6">CASE B</td></tr>
<tr><td>200 OK INVITE(CON)</td><td>←</td><td></td><td>←</td><td colspan="2">CON</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="2"></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td colspan="2">REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td colspan="2">RLC</td></tr>
</table>

TP404007	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 6.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/COLR	
SIP selection criteria		
ISUP selection criteria		
Test purpose	*O-MGCF: Connected number, additional connected number and connected subaddress transferred* Ensure that an ANM or CON encapsulated in a 200 OK INVITE is sent on the ISUP side without changing. The connected number is unchanged. The ATP contained the connected sub address is included.	
SIP parameter values	200 OK INVITE: encapsulated ANM or CON included	
ISUP parameter values	**ANM;** **Connected number parameter** Address presentation restricted parameter = '01'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = PIXIT **Additional connected number** present Address presentation restricted parameter = '01'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '00'B Address signals = PIXIT and an **access transport** parameter containing the connected sub-address. b) **CON;** **Connected number parameter** Address presentation restricted parameter = '01'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = PIXIT **Additional connected number** present Address presentation restricted parameter = '01'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '00'B Address signals = PIXIT and an **access transport** parameter containing the connected sub-address.	

Comments

ISUP		SUT		SIP-I
IAM	→		→	INVITE(IAM)
CASE A				
ACM	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
CASE B				
CON	←		←	200 OK INVITE(CON)
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

5.3.5 Terminal Portability (TP)

TP405001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.733 [i.6], clause 4.5.2.1**
TSS reference:	ISUP-SIP-ISUP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Terminal portability, requested by the calling party** Ensure that SUT informs the called party that a suspend and a resume have been requested by the calling party upon receipt of user initiated **SUS** and **RES** messages.				
SIP parameter values	INFO: Content-Type: application/ISUP ; SUS and RES encapsulated in the MIME body				
ISUP parameter values					
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	SUS	→		→	INFO(SUS)
				←	200 OK INFO
	RES	→		→	INFO(RES)
				←	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP405002	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.733 [i.6], clause 4.5.2.1**
TSS reference:	ISUP-SIP-ISUP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Terminal portability, 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 values	INFO: Content-Type: application/ISUP ; SUS and RES encapsulated in the MIME body				
ISUP parameter values					
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	SUS	←		←	INFO(SUS)
				→	200 OK INFO
	RES	←		←	INFO(RES)
				→	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP405003	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], Q.733 [i.6], clause 4.5.2.1**	
TSS reference	ISUP-SIP-ISUP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Terminal portability, 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: application/ISUP ; SUS encapsulated in the MIME body BYE : Content-Type: application/ISUP ; REL encapsulated in the MIME body				
ISUP parameter values					
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	SUS	→		→	INFO(SUS)
				←	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP405004	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], Q.733 [i.6], clause 4.5.2.1**	
TSS reference	ISUP-SIP-ISUP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Terminal portability, release suspended call** Ensure that a suspended call can be released, if the remote user releases the call.				
SIP parameter values	INFO: Content-Type: application/ISUP ; SUS encapsulated in the MIME body BYE : Content-Type: application/ISUP ; REL encapsulated in the MIME body				
ISUP parameter values					
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	SUS	→		→	INFO(SUS)
					200 OK INFO
	REL	←		←	BYE(REL)
	RLC	→		→	200 OK BYE(RLC)

5.3.6 SUB-addressing (SUB)

<table>
<tr><th>TP406001</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 8.5.2.1.1/</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISUP-SIP-ISUP/SS/SUB</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: application/ISUP ; IAM encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="7">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP406002</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 8.5.2.1.1/</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/SUB</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="7">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP406003	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 8.5.2.1.1/**
TSS reference	ISUP-SIP-ISUP/SS/SUB				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Sending the calling sub-address in the access transport parameter* Ensure that the SUT can include the calling sub-address in the **access transport** parameter in the encapsulated IAM.				
SIP parameter values	INVITE: Content-Type: application/ISUP ; IAM encapsulated in the MIME body				
ISUP parameter values					
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP406004	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 8.5.2.1.1/**
TSS reference	ISUP-SIP-ISUP/SS/SUB				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Receiving the calling sub-address in the access transport parameter* Ensure that the SUT can include the calling sub-address in the **access transport** parameter in the ISUP IAM.				
SIP parameter values					
ISUP parameter values					
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

5.3.7 Malicious Call Identification (MCID)

TP407001	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], Q.731.7 [i.3], clause 7.5.2.1.1**
TSS reference:	ISUP-SIP-ISUP/SS/MCID	
SIP selection criteria		
ISUP selection criteria		
Test purpose	*Successful MCID request O-MGCF* Ensure that the SUT can successfully pass on a 183 Session Progress containing an encapsulated **IDR** having the **MCID request indicator** set to "MCID request" and pass on an **IRS** with **MCID response indicator** set to "MCID included" and the **calling party number** included. ISUP to SIP-I interworking.	
SIP parameter values	183 Session Progress: Content-Type: application/ISUP; IDR encapsulated in the MIME body INFO: Content-Type: application/ISUP; IRS encapsulated in the MIME body	
ISUP parameter values		
Comments		

ISUP		**SUT**		**SIP-I**
IAM	→		→	INVITE(IAM)
IDR	←		←	183 Session Progress(IDR)
IRS	→		→	INFO(IRS)
			←	200 OK INFO
ACM	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

TP407002	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], Q.731.7 [i.3], clause 7.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/MCID	
SIP selection criteria		
ISUP selection criteria		
Test purpose	*Successful MCID request I-MGCF* Ensure that the SUT can successfully pass on an **IDR** having the **MCID request indicator** set to "MCID request" and pass on an **IRS** with **MCID response indicator** set to "MCID included" and the **calling party number** included. SIP-I to ISUP interworking.	
SIP parameter values	183 Session Progress: Content-Type: application/ISUP; IDR encapsulated in the MIME body INFO: Content-Type: application/ISUP; IRS encapsulated in the MIME body	
ISUP parameter values		
Comments		

SIP-I		**SUT**		**ISUP**
INVITE(IAM)	→		→	IAM
183 Session Progress(IDR)	←		←	IDR
INFO(IRS)	→		→	IRS
200 OK INFO	←			
180 Ringing(ACM)	←		←	ACM
200 OK INVITE(ANM)	←		←	ANM
		Conversation		
BYE(REL)	→		→	REL
200 OK BYE(RLC)	←		←	RLC

TP407003	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.731.7 [i.3], clause 7.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/MCID				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Successful MCID request - after ACM** Ensure that the SUT will accept and pass on correctly an MCID request after ACM has been received. The SUT should pass on an **IDR** having the **MCID request indicator** set to "MCID request" and pass on an **IRS** with **MCID response indicator** set to "MCID included" and the **calling party number** included (see note).				
SIP parameter values	INFO: Content-Type: application/ISUP; IDR encapsulated in the MIME body INFO: Content-Type: application/ISUP; IRS encapsulated in the MIME body				
ISUP parameter values	IRS containing the calling party number parameter				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	CASE A				
	180 Ringing(ACM)	←		←	ACM
	183 Session Progress(IDR)	←		←	IDR
	INFO(IRS)	→		→	IRS
	200 OK INFO	←			
	200 OK INVITE(ANM)	←		←	ANM
	CASE B				
	183 Session Progress(ACM)	←		←	ACM(early)
	183 Session Progress(IDR)	←		←	IDR
	INFO(IRS)	→		→	IRS
	200 OK INFO	←			
	180 Ringing(CPG)	←		←	CPG(alerting)
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC
NOTE: This situation may occur e.g. if the call has been forwarded before reaching the destination.					

TP407004	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.731.7 [i.3], clause 7.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/MCID				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**MCID request - MCID not supported by the OLE** O-MGCF Ensure that the SUT rejects a MCID request by sending an **IRS** with the **MCID response indicator** set to "MCID not included". ISUP to SIP-I interworking.				
SIP parameter values	183 Session Progress: Content-Type: application/ISUP; IDR encapsulated in the MIME body INFO: Content-Type: application/ISUP; IRS encapsulated in the MIME body				
ISUP parameter values					
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	IDR	←		←	183 Session Progress(IDR)
	IRS	→		→	INFO(IRS)
				←	200 OK INFO
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP407005	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.731.7 [i.3], clause 7.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/MCID				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**MCID request - MCID not supported by the OLE** I-MGCF Ensure that the SUT rejects a MCID request by sending an **IRS** with the **MCID response indicator** set to "MCID not included". SIP-I to ISUP interworking.				
SIP parameter values	183 Session Progress: Content-Type: application/ISUP; IDR encapsulated in the MIME body INFO: Content-Type: application/ISUP; IRS encapsulated in the MIME body				
ISUP parameter values					
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	183 Session Progress(IDR)	←		←	IDR
	INFO(IRS)	→		→	IRS
	200 OK INFO	←			
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP407006	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.731.7 [i.3], clause 7.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/MCID				
SIP selection criteria					
ISUP selection criteria	PICS 1/7				
Test purpose	**MCID information passed and set correctly - outgoing** Ensure that a received **IDR** is transferred transparently into the national network, the subsequent **IRS** being transferred into the international network so that the country code in the address signals of the **calling party number** is added and the nature of address indicator is set to "international number": • the IDR request is transferred into the national network; • The IRS is received from the national network having the calling party number coded as an "international number". Calling party sub-address in ATP.				
SIP parameter values	183 Session Progress: Content-Type: application/ISUP; IDR encapsulated in the MIME body INFO: Content-Type: application/ISUP; IRS encapsulated in the MIME body				
ISUP parameter values					
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	183 Session Progress(IDR)	←		←	IDR
	INFO(IRS)	→		→	IRS
	200 OK INFO	←			
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

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

TP407008	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], Q.731.7 [i.3], clause 7.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/MCID			
SIP selection criteria				
ISUP selection criteria				
Test purpose	**Successful MCID request with calling sub-address I-MGCF** Ensure that the SUT can successfully reply to an **IDR** having the **MCID request indicator** set to "MCID request" by sending an **IRS** with **MCID response indicator** set to "MCID included", the **calling party number** and a calling sub-address in the **access transport** parameter. SIP-I to ISUP interworking.			
SIP parameter values	183 Session Progress: Content-Type: application/ISUP; IDR encapsulated in the MIME body INFO: Content-Type: application/ISUP; IRS encapsulated in the MIME body			
ISUP parameter values				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	183 Session Progress(IDR)	←		←	IDR
	INFO(IRS)	→		→	IRS
	200 OK INFO	←			
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP407009</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1],
Q.731.7 [i.3], clause 7.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/MCID</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">MCID timer (T39) expiry O-MGCF
Ensure that call setup is continued (user is alerted) if no IRS is received within timer T39 expiry, after having sent the IDR with MCID request indicator set to "MCID requested".
ISUP to SIP-I interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">183 Session Progress: Content-Type: application/ISUP; IDR encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="9">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>IDR</td><td>←</td><td></td><td>←</td><td>183 Session Progress(IDR)</td></tr>
<tr><td></td><td></td><td></td><td colspan="2">T39 expiry</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP407010</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1],
Q.731.7 [i.3], clause 7.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/MCID</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">MCID timer (T39) expiry O-MGCF
Ensure that call setup is continued (user is alerted) if no IRS is received within timer T39 expiry, after having sent the IDR with MCID request indicator set to "MCID requested".
SIP-I to ISUP interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">183 Session Progress: Content-Type: application/ISUP; IDR encapsulated in the MIME body
INFO: Content-Type: application/ISUP; IRS encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="9">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>183 Session Progress(IDR)</td><td>←</td><td></td><td>←</td><td>IDR</td></tr>
<tr><td></td><td></td><td></td><td colspan="2">T39 expiry</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

5.3.8 Call hold (HOLD)

<table>
<tr><th>TP408001</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
Q.733 [i.6], clauses 2.5.2.1.1.1; 2.5.2.1.1.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/HOLD</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Call hold after answer, requested by the originating user

Ensure that the notifications that a call is placed on hold and retrieved are sent with CPG messages having the event indicator set to "progress". O-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="15">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>CPG(progress, hold)</td><td>→</td><td></td><td>→</td><td>INVITE(CPG, sendonly)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>CPG(progress, retrieve)</td><td>→</td><td></td><td>→</td><td>INVITE(CPG, sendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP408002	SIP reference: RFC 3261 [4]		ISUP reference: Q.1912.5 [1], Q.733 [i.6], clauses 2.5.2.1.1.1; 2.5.2.1.1.2	
TSS reference	ISUP-SIP-ISUP/SS/HOLD			
SIP selection criteria				
ISUP selection criteria				
Test purpose	*Call hold after answer, requested by the originating 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: application/ISUP; CPG encapsulated in the MIME body			
ISUP parameter values				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	INVITE(CPG, sendonly)	→		→	CPG(progress, hold)
	200 OK INVITE	←			
	ACK	→			
	INVITE(CPG, sendrecv)	→		→	CPG(progress, retrieve)
	200 OK INVITE	←			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP408003	SIP reference: RFC 3261 [4]		ISUP reference: Q.1912.5 [1], Q.733 [i.6], clauses 2.5.2.1.1.1; 2.5.2.1.1.2		
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Call hold after answer, requested by the terminating user* Ensure that the notifications that a call is placed on hold and retrieved are sent with **CPG** messages having the **event indicator** set to "progress". O-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: application/ISUP; CPG encapsulated in the MIME body				
ISUP parameter values					
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	CPG(progress, hold)	←		←	INVITE(CPG, sendonly)
				→	200 OK INVITE
				←	ACK
	CPG(progress, retrieve)	←		←	INVITE(CPG, sendrecv)
				→	200 OK INVITE
				←	ACK
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP408004	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.733 [i.6], clauses 2.5.2.1.1.1; 2.5.2.1.1.2
TSS reference	ISUP-SIP-ISUP/SS/HOLD			
SIP selection criteria				
ISUP selection criteria				
Test purpose	*Call hold after answer, requested by the terminating user* Ensure that the notifications that a call is placed on hold and retrieved are sent with **CPG** messages having the **event indicator** set to "progress". I-MGCF interworking.			
SIP parameter values	INVITE: Content-Type: application/ISUP; CPG encapsulated in the MIME body			
ISUP parameter values				

Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	INVITE(CPG, sendonly)	←		←	CPG(progress, hold)
	200 OK INVITE	→			
	ACK	←			
	INVITE(CPG, sendrecv)	←		←	CPG(progress, retrieve)
	200 OK INVITE	→			
	ACK	←			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP408005	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.733 [i.6], clauses 2.2.1; 2.5.2.1.1.1; 2.5.2.1.1.2
TSS reference	ISUP-SIP-ISUP/SS/HOLD			
SIP selection criteria				
ISUP selection criteria	PICS 8/1			
Test purpose	**Call hold after alerting, requested by the calling user** Ensure that when an outgoing call is placed on hold and retrieved after alerting the notifications are sent with **CPG** messages. 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	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	CPG(progress, hold)	→		→	INVITE(CPG, sendonly)
				←	200 OK INVITE
				→	ACK
	CPG(progress, retrieve)	→		→	INVITE(CPG, sendrecv)
				←	200 OK INVITE
				→	ACK
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP408006	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], Q.733 [i.6], clauses 2.2.1; 2.5.2.1.1.1; 2.5.2.1.1.2**		
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria	PICS 8/1				
Test purpose	**Call hold after alerting, requested by the calling use***r* Ensure that when an outgoing call is placed on hold and retrieved after alerting the notifications are sent with **CPG** messages. I-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: application/ISUP; CPG encapsulated in the MIME body				
ISUP parameter values					
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	INVITE(CPG, sendonly)	→		→	CPG(progress, hold)
	200 OK INVITE	←			
	ACK	→			
	INVITE(CPG, sendrecv)	→		→	CPG(progress, retrieve)
	200 OK INVITE	←			
	ACK	→			
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP408007	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], Q.764 [i.12], clause 2.3**		
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call hold after answer, release of the call by the calling served user** Ensure that a call in the held state can be released by the user who activated the Call hold service. 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	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	CPG(progress, hold)	→		→	INVITE(CPG, sendonly)
				←	200 OK INVITE
				→	ACK
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP408008	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], Q.764 [i.12], clause 2.3**
TSS reference	ISUP-SIP-ISUP/SS/HOLD			
SIP selection criteria				
ISUP selection criteria				
Test purpose	**Call hold after answer, release of the call by the calling served user** Ensure that a call in the held state can be released by the user who activated the Call hold service. I-MGCF interworking.			
SIP parameter values	INVITE: Content-Type: application/ISUP; CPG encapsulated in the MIME body			
ISUP parameter values				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	INVITE(CPG, sendonly)	→		→	CPG(progress, hold)
	200 OK INVITE	←			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP408009	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], Q.764 [i.12], clause 2.3**
TSS reference	ISUP-SIP-ISUP/SS/HOLD			
SIP selection criteria				
ISUP selection criteria				
Test purpose	**Call hold after answer, release of the call by the terminating user** Ensure that a call in the held state can be released by the user who did not activate the Call hold service. 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	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	CPG(progress, hold)	←		←	INVITE(CPG, sendonly)
				→	200 OK INVITE
				←	ACK
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP408010	SIP reference: RFC 3261 [4]		ISUP reference: Q.1912.5 [1], Q.764 [i.12], clause 2.3		
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call hold after answer, release of the call by the terminating user** Ensure that a call in the held state can be released by the user who did not activate the Call hold service. I-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: application/ISUP; CPG encapsulated in the MIME body				
ISUP parameter values					
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	INVITE(CPG, sendonly)	←		←	CPG(progress, hold)
	200 OK INVITE	→			
	ACK	←			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP408011	SIP reference: RFC 3261 [4]		ISUP reference: Q.1912.5 [1], Q.764 [i.12], clause 2.3		
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call hold after alerting, release of the call by the calling user** Ensure that a held call can be released by the user who activated the Call hold service without retrieving the call. 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	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
			Ringing		
	CPG(progress, hold)	→		→	INVITE(CPG, sendonly)
				←	200 OK INVITE
				→	ACK
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP408012	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], Q.764 [i.12], clause 2.3**
TSS reference	ISUP-SIP-ISUP/SS/HOLD			
SIP selection criteria				
ISUP selection criteria				
Test purpose	**Call hold after alerting, release of the call by the calling user** Ensure that a held call can be released by the user who activated the Call hold service without retrieving the call. I-MGCF interworking.			
SIP parameter values	INVITE: Content-Type: application/ISUP; CPG encapsulated in the MIME body			
ISUP parameter values				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
			Ringing		
	INVITE(CPG, sendonly)	→		→	CPG(progress, hold)
	200 OK INVITE	←			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

5.3.9 Call Waiting (CW)

TP409001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.733 [i.6], clause 1.5.2.1.1**
TSS reference:	ISUP-SIP-ISUP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Call waiting indication in 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: application/ISUP; ACM encapsulated in the MIME body				
ISUP parameter values	ACM: Generic notification indicator "Call is a waiting call"				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM(waiting)	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP409002	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.733 [i.6], clause 1.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Call waiting indication in ACM* Ensure that a call can be successfully established if the **ACM** indicates that this call is a waiting call. I-MGCF interworking.				
SIP parameter values	180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body				
ISUP parameter values	ACM: Generic notification indicator "Call is a waiting call"				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM(waiting)
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP409003	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.733 [i.6], clause 1.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/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**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	183 Session Progress(ACM)
	CPG(waiting)	←		←	180 Ringing(CPG)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP409004	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.733 [i.6], clause 1.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Call waiting indication in CPG* Ensure that a call can be successfully established if the **CPG** indicates that this call is a waiting call. I-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	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	183 Session Progress ACM)	←		←	ACM
	180 Ringing(CPG)	←		←	CPG(waiting)
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP409005	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.733 [i.6], clause 1.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**User rejects the waiting call** Ensure that the SUT pass on a **REL** with cause #21 (call rejected) if a busy user rejects the waiting call. O-MGCF interworking.				
SIP parameter values	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	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM(waiting)	←		←	180 Ringing(ACM)
	REL(#21)	←		←	480 Temporarily Unavailable(REL)
	RLC	→		→	ACK

TP409006	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.733 [i.6], clause 1.5.2.1.1	
TSS reference	ISUP-SIP-ISUP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**User rejects the waiting 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)	←		←	ACM(waiting)
	480 Temporarily Unavailable(REL)	←		←	REL(#21)
	ACK	→		→	RLC

TP409007	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.733 [i.6], clause 1.5.2.1.1	
TSS reference	ISUP-SIP-ISUP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call waiting ignored (expiry of call waiting supervision timer)** Ensure that the SUT pass on a **REL** with cause #19 (no answer from user, user alerted) if a busy user does not answer the waiting call. O-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 #19 (no answer from user, user alerted)				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM(waiting)	←		←	180 Ringing(ACM)
	T9 expiry				
	CASE A				
	REL(#19)	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)
				←	487 Request Terminated
				→	ACK
	CASE B				
	REL(#19)	→		→	CANCEL
	RLC	←		←	200 OK CANCEL
				←	487 Request Terminated
				→	ACK

TP409008	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.733 [i.6], clause 1.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call waiting ignored (expiry of call waiting supervision timer)** Ensure that the SUT pass on a **REL** with cause #19 (no answer from user, user alerted) if a busy user does not answer the waiting call. I-MGCF interworking.				
SIP parameter 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 #19 (no answer from user, user alerted)				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM(waiting)
	T9 expiry				
	BYE(REL)	→		→	REL(#19)
	200 OK BYE(RLC)	←		←	RLC
	487 Request Terminated	←			
	ACK	→			

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

TP410001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.732 [i.4], clause 2.5**
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*"Call is diverting" indication received in 180 Ringing* Verify that a call can be successfully established, if diversion occurs. The **ACM** contains the **generic notification indicator** set to "call is diverting", the **call diversion information** and the **redirection number.** The Redirection reason is set to **CV_redirection_reason**. CPG (alerting) is coded as if it has been mapped from the 180 Ringing (CPG). O-MCGF interworking.				
SIP parameter values	183 Session Progress: Content-Type: application/ISUP; ACM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; CPG encapsulated in the MIME body				
ISUP parameter values	ACM: BCI Called party status indicator "No indication" Generic notification Call diversion information Redirection number CPG: Event indicator=alerting				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM(no indication)	←		←	183 Session Progress(ACM)
	CPG(alerting)	←		←	180 Ringing(CPG)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

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

CV_redirection_reason, TP410001, TP410002	
VA_1	User busy
VA_2	Unconditional
VA_3	Deflection immediate response

<table>
<tr><th>TP410003</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], Q.732 [i.4], clause 2.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/Call Diversion</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">"Call diversion may occur" received in 180 Ringing(ACM)
Verify that a call can be successfully established, if diversion may occur. The encapsulated ACM indicates that "call diversion may occur" in the optional backward call indicators. The following CPG contains the generic notification indicator set to "call is diverting", the call diversion information and the redirection number, if diversion occurs. The CPG (progress) contains CV_redirection_reason in call diversion information and also Redirection number. The CPG (alerting) is coded as if it has been mapped from ACM, with RnNbRes parameter (optional).
O-MCGF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body
183 Session Progress: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">ACM: BCI Called party status indicator "subscriber free", Optional backward call indicator: "Call diversion may occur"
CPG: Event information=progress, Call diversion information; Generic notification; Redirection number
CPG: Event information=alerting</td></tr>
<tr><td rowspan="9">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM(free)</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>CPG</td><td>←</td><td></td><td>←</td><td>183 Session Progress(CPG)</td></tr>
<tr><td>CPG(alerting)</td><td>←</td><td></td><td>←</td><td>183 Session Progress(CPG)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP410004</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], Q.732 [i.4], clause 2.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP-ISUP/SS/Call Diversion</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">"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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body
183 Session Progress: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">ACM: BCI Called party status indicator "subscriber free", Optional backward call indicator: "Call diversion may occur"
CPG: Event information=progress, Call diversion information; Generic notification; Redirection number
CPG: Event information=alerting</td></tr>
<tr><td rowspan="9">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td colspan="2">ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td colspan="2">IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td colspan="2">ACM(free)</td></tr>
<tr><td>183 Session Progress(CPG)</td><td>←</td><td></td><td>←</td><td colspan="2">CPG</td></tr>
<tr><td>183 Session Progress(CPG)</td><td>←</td><td></td><td>←</td><td colspan="2">CPG(alerting)</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td colspan="2">ANM</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="2"></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td colspan="2">REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td colspan="2">RLC</td></tr>
</table>

<table>
<tr><th colspan="2">CV_redirection_reason TP410003, TP410004</th></tr>
<tr><td>VA_1</td><td>No reply</td></tr>
<tr><td>VA_2</td><td>Deflection during alerting</td></tr>
</table>

TP410005	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.732 [i.4], clause 2.4.2
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Multiple diversions -Verify that a call can be successfully established, if multiple diversion occur** Several messages each containing the **call diversion information** are received, as if multiple forwardings have occurred. The **CV_redirection_reason** is used as redirection reason. The Redirection number restriction parameter is passed on. O-MCGF interworking.				
SIP parameter values	183 Session Progress: Content-Type: application/ISUP; ACM encapsulated in the MIME body 183 Session Progress: Content-Type: application/ISUP; CPG encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; CPG encapsulated in the MIME body				
ISUP parameter values	ACM: BCI Called party status indicator "No indication" Generic notification Call diversion information Redirection reason unconditional Redirection number CPG1: Event information=progress Generic notification Call diversion information Redirection reason **CV_redirection_reason** Redirection number Redirection number restriction CPG2: Event information=alerting, Redirection number restriction				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM(no indication)	←		←	183 Session Progress(ACM)
	CPG1	←		←	183 Session Progress(CPG)
	CPG2(alerting)	←		←	180 Ringing(CPG)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP410006	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.732 [i.4], clause 2.4.2
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Multiple diversions -Verify that a call can be successfully established, if multiple diversion occur** Several messages each containing the **call diversion information** are received, as if multiple forwardings have occurred. The **CV_redirection_reason** is used as redirection reason. The Redirection number restriction parameter is passed on. I-MCGF interworking.				
SIP parameter values	183 Session Progress: Content-Type: application/ISUP; ACM encapsulated in the MIME body 183 Session Progress: Content-Type: application/ISUP; CPG encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; CPG encapsulated in the MIME body				
ISUP parameter values	ACM: BCI Called party status indicator "No indication" Generic notification Call diversion information Redirection reason unconditional Redirection number CPG: Event information=progress Generic notification Call diversion information Redirection reason **CV_redirection_reason** Redirection number Redirection number restriction CPG: Event information=alerting, Redirection number restriction				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	183 Session Progress(ACM)	←		←	ACM(no indication)
	183 Session Progress(CPG)	←		←	CPG1
	180 Ringing(CPG)	←		←	CPG2(alerting)
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

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

<table>
<tr><th>TP410007</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], Q.732 [i.4], clause 2.5.2.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/Call Diversion</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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).
O-MCGF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body
200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: Redirecting number, Original called number, Redirection information
ANM: Redirection address restriction</td></tr>
<tr><td rowspan="7">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
<tr><td colspan="6">NOTE: Altered in Gateways.</td></tr>
</table>

TP410008	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.732 [i.4], 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 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). I-MCGF interworking.				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body				
ISUP parameter values	IAM: Redirecting number, Original called number, Redirection information ANM: Redirection address restriction				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC
NOTE: Altered in Gateways.					

TP410009	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 3.5.2.4.1
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion				
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: application/ISUP; IAM containing an Original called number encapsulated in the MIME body				
ISUP parameter values	IAM: No original called number present				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP410010	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 4.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion				
SIP selection 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 values	INVITE: Content-Type: application/ISUP; IAM containing an Original called number called number encapsulated in the MIME body				
ISUP parameter values	IAM: Original called number "International number"				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP410011	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 4.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion				
SIP selection 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: Discarding the **original called number**, if the address is marked not available. The PTC will send an IAM with an "address not available" OriCdNb.				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM containing an Original called number called number encapsulated in the MIME body				
ISUP parameter values	IAM: No original called number present				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

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

TP410013	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 4.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/Call 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 numbe**r according to the procedures as defined for CLIP: Discarding the **redirecting number** if case of bilateral agreements.				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM containing a Redirecting number encapsulated in the MIME body				
ISUP parameter values	IAM: No Redirecting number present				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP410014	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 4.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion			
SIP selection criteria				
ISUP selection criteria	PICS 1/7			
Test purpose	**Redirecting number in the outgoing international gateway** Verify that the outgoing international gateway checks and manipulates the **redirecting numbe**r according to the procedures as defined for CLIP: Discarding the **redirecting number**, if the address is marked not available. The PTC will send an IAM with an "address not available" RgNb.			
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM containing a Redirecting number encapsulated in the MIME body			
ISUP parameter values	IAM: No Redirecting number present			
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP410015	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.732 [i.4], clause 2.5.2.3, Q.731 [i.2], clause 3.5.2.3**
TSS reference:	ISUP-SIP-ISUP/SS/Call Diversion				
SIP selection criteria					
ISUP selection criteria	PICS 1/7				
Test purpose	**Redirecting number in the outgoing international gateway** Verify that the outgoing international gateway checks and manipulates the **redirecting numbe**r according to the procedures as defined for CLIP: Converting the **redirecting number** to international format with transparent transferral of address presentation restriction indicator. The PTC will send an IAM with a national significant RgNb.				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM containing a Redirecting number "National number" encapsulated in the MIME body				
ISUP parameter values	IAM: Redirecting number "International number"				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP410016</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1],
Q.732 [i.4], clause 2.5.2.3,
Q.731 [i.2], clause 3.5.2.3</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/Call Diversion</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/8</td></tr>
<tr><td>Test purpose</td><td colspan="5">Redirecting number in the incoming international gateway
Verify that the incoming international gateway checks and manipulates the redirecting number according to the procedures as defined for CLIP:
Converting the redirecting number to national format, if necessary (own country code).
The PTC will send an IAM with RgNb.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: application/ISUP; IAM containing a Redirecting number "International number" encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: Redirecting number "national number"</td></tr>
<tr><td rowspan="7">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP410017</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1],
Q.732 [i.4], clause 2.5.2.3,
Q.731 [i.2], clause 3.5.2.3</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/Call Diversion</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/8 AND 10/4</td></tr>
<tr><td>Test purpose</td><td colspan="5">Redirecting number in the incoming international gateway
Verify that the incoming international gateway checks and manipulates the redirecting number according to the procedures as defined for CLIP:
Adding a prefix to an international redirecting number.
The PTC will send an IAM with RgNb.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: application/ISUP; IAM containing a Redirecting number encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: Redirecting number</td></tr>
<tr><td rowspan="7">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP410018</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
Q.732 [i.4], clause 2.5.2.4,
Q.731 [i.2], clause 3.5.2.4</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/Call Diversion</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 10/5 AND PICS 1/8</td></tr>
<tr><td>Test purpose</td><td colspan="5">Redirection number in the incoming international gateway
Verify that the incoming international gateway checks and manipulates the redirection number according to the procedures defined for COLP:
discarding the redirection number in case of bilateral agreements;
removes the redirection number restriction parameter.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">183 Session Progress: Content-Type: application/ISUP; ACM containing a Redirection number encapsulated in the MIME body
200 OK INVITE: Content-Type: application/ISUP; ANM containing a Redirection address restriction parameter encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">ACM: Called party status=no indication
Generic notification
Call diversion information Redirection reason unconditional
No Redirection number
ANM: No Redirection number restriction parameter</td></tr>
<tr><td rowspan="8">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM(no indication)</td><td>←</td><td></td><td>←</td><td>183 Session Progress(ACM)</td></tr>
<tr><td>CPG</td><td>←</td><td></td><td>←</td><td>180 Ringing(CPG)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP410019</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
Q.732 [i.4], clause 2.5.2.3,
Q.731 [i.2], clause 3.5.2.3</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/Call Diversion</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/7</td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">183 Session Progress: Content-Type: application/ISUP; ACM containing a Redirection number "International number" encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">ACM: Called party status=no indication
Generic notification
Call diversion information Redirection reason unconditional
Redirection number "National number"</td></tr>
<tr><td rowspan="8">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM(no indication)</td><td>←</td><td></td><td>←</td><td>183 Session Progress(ACM)</td></tr>
<tr><td>CPG</td><td>←</td><td></td><td>←</td><td>180 Ringing(CPG)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP410020	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], Q.732 [i.4], clause 2.5.2.3, Q.731 [i.2], clause 3.5.2.3**
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion			
SIP selection criteria				
ISUP selection criteria	PICS 1/8			
Test purpose	**Redirection number in the incoming international gateway** Verify that the incoming international gateway checks and manipulates the **redirection numbe**r according to the procedures defined for COLP: Converting the **redirection number** to international format.			
SIP parameter values	183 Session Progress: Content-Type: application/ISUP; ACM containing a Redirection number "National number" encapsulated in the MIME body			
ISUP parameter values	ACM: Called party status=no indication Generic notification Call diversion information Redirection reason unconditional Redirection number "International number"			
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM(no indication)	←		←	183 Session Progress(ACM)
	CPG	←		←	180 Ringing(CPG)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP410021	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 5.5.2.3.1**
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion				
SIP selection criteria					
ISUP selection criteria	PICS 1/8 AND PICS 10/6				
Test purpose	**Redirection number in the outgoing international gateway** Verify that the outgoing international gateway checks and manipulates the **redirection numbe**r according to the procedures defined for COLP: Adding a prefix to an international **redirection number.** The PTC will provide the necessary stimulus.ACM with CDInf, GenNot = "call is diverting" and an international RnNb with foreign country code.				
SIP parameter values	183 Session Progress: Content-Type: application/ISUP; ACM containing a Redirection number "International number" encapsulated in the MIME body				
ISUP parameter values	ACM: Called party status=no indication Generic notification Call diversion information Redirection reason unconditional Redirection number Number with Prefix				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM(no indication)	←		←	183 Session Progress(ACM)
	CPG	←		←	180 Ringing(CPG)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

5.3.11 CONF

TP411001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15**
TSS reference	ISUP-SIP-ISUP/SS/CONF				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Generic notification transfer "conference established" and "other party added" To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message: 1. assist a call set up from ISUP to SIP-I; 2. check that the notification "conference established" is received in the CPG from conferee at SIP-I; 3. check the notification "other party added" in the CPG. O-MGCF interworking.				
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body				
ISUP parameter values	CPG: Generic notification: conference established CPG: Generic notification: other party added				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	CPG(conference established)	→		→	INFO(CPG)
				←	200 OK INFO
	CPG(other party added)	→		→	INFO(CPG)
				←	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP411002</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
Q.734 [i.7], clause 1.6.15</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CONF</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Generic notification transfer "conference established" and "other party added"

To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message:
1. Assist a call set up from SIP-I to ISUP;
2. Check that the notification "conference established" is received in the CPG from conferee at the ISUP;
3. Check the notification "other party added" in the CPG.
I-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">CPG: Generic notification: conference established
CPG: Generic notification: other party added</td></tr>
<tr><td rowspan="13">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>INFO(CPG)</td><td>→</td><td></td><td>→</td><td>CPG(conference established)</td></tr>
<tr><td>200 OK INFO</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>INFO(CPG)</td><td>→</td><td></td><td>→</td><td>CPG(other party added)</td></tr>
<tr><td>200 OK INFO</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><td>TP411003</td><td colspan="5">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1],
Q.734 [i.7], clause 1.6.15</td></tr>
<tr><td>TSS reference</td><td colspan="7">ISUP-SIP-ISUP/SS/CONF</td></tr>
<tr><td>SIP selection criteria</td><td colspan="7"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="7"></td></tr>
<tr><td>Test purpose</td><td colspan="7">Generic notification transfer "conference established" and "isolated"

To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message:
1. Assist a call set up from ISUP to SIP-I;
2. Check that the notification "conference established" is received in the CPG from conferee at the SIP-I;
3. Check the notification "isolated" in the CPG.
O-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="7">INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="7">CPG: Generic notification: conference established
CPG: Generic notification: isolated</td></tr>
<tr><td rowspan="14">Comments</td><td>ISUP</td><td></td><td colspan="2">SUT</td><td></td><td colspan="2">SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td colspan="2"></td><td>→</td><td colspan="2">INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td colspan="2"></td><td>←</td><td colspan="2">180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td colspan="2"></td><td>←</td><td colspan="2">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="4">Conversation</td><td colspan="2"></td></tr>
<tr><td>CPG(conference established)</td><td>→</td><td colspan="2"></td><td>→</td><td colspan="2">INFO(CPG)</td></tr>
<tr><td></td><td></td><td colspan="2"></td><td>←</td><td colspan="2">200 OK INFO</td></tr>
<tr><td></td><td></td><td colspan="2"></td><td></td><td colspan="2"></td></tr>
<tr><td>CPG(isolated)</td><td>→</td><td colspan="2"></td><td>→</td><td colspan="2">INFO(CPG)</td></tr>
<tr><td></td><td></td><td colspan="2"></td><td>←</td><td colspan="2">200 OK INFO</td></tr>
<tr><td></td><td></td><td colspan="2"></td><td></td><td colspan="2"></td></tr>
<tr><td>REL</td><td>→</td><td colspan="2"></td><td>→</td><td colspan="2">BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td colspan="2"></td><td>←</td><td colspan="2">200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP411004</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
Q.734 [i.7], clause 1.6.15</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CONF</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Generic notification transfer "conference established" and "isolated"

To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message:
1. assist a call set up from SIP-I to ISUP;
2. check that the notification "conference established" is received in the CPG from conferee at SIP-I;
3. check the notification "isolated" in the CPG.
I-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">CPG: Generic notification: conference established
CPG: Generic notification: isolated</td></tr>
<tr><td rowspan="14">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>INFO(CPG)</td><td>→</td><td></td><td>→</td><td>CPG(conference established)</td></tr>
<tr><td>200 OK INFO</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>INFO(CPG)</td><td>→</td><td></td><td>→</td><td>CPG(isolated)</td></tr>
<tr><td>200 OK INFO</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP411005</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1],
Q.734 [i.7], clause 1.6.15</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP-ISUP/SS/CONF</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">Generic notification transfer "conference established", "isolated" and "reattached"

To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message:
1. Assist a call set up from ISUP to SIP-I;
2. Check that the notification "conference established" is received in the CPG from conferee at SIP-I;
3. Check the notification "reattached" in the CPG.
O-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">CPG: Generic notification: conference established
CPG: Generic notification: isolated
CPG: Generic notification: reattached</td></tr>
<tr><td rowspan="16">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td colspan="2">SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="2"></td></tr>
<tr><td>CPG(conference established)</td><td>→</td><td></td><td>→</td><td colspan="2">INFO(CPG)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>CPG(isolated)</td><td>→</td><td></td><td>→</td><td colspan="2">INFO(CPG)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>CPG(reattached)</td><td>→</td><td></td><td>→</td><td colspan="2">INFO(CPG)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td colspan="2">BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK BYE(RLC)</td></tr>
</table>

TP411006	**SIP reference: RFC 3261 [4]**			**ISUP reference:** **Q.1912.5 [1],** **Q.734 [i.7], clause 1.6.15**	
TSS reference	ISUP-SIP-ISUP/SS/CONF				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Generic notification transfer "conference established", "isolated" and "reattached" To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message: 1. Assist a call set up from SIP-I to ISUP; 2. Check that the notification "conference established" is received in the CPG from conferee at SIP-I; 3. Check the notification "reattached" in the CPG. I-MGCF interworking.				
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body				
ISUP parameter values	CPG: Generic notification: conference established CPG: Generic notification: isolated CPG: Generic notification: reattached				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	INFO(CPG)	→		→	CPG(conference established)
	200 OK INFO	←			
	INFO(CPG)	→		→	CPG(isolated)
	200 OK INFO	←			
	INFO(CPG)	→		→	CPG(reattached)
	200 OK INFO	←			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP411007</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
Q.734 [i.7], clause 1.6.15</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CONF</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Generic notification transfer "conference established", "other party added" and "other party disconnected"

To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message:
1. assist a call set up from ISUP to SIP-I;
2. check the notification "other party disconnected" in the CPG.
O-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">CPG: Generic notification: conference established
CPG: Generic notification: other party added
CPG: Generic notification: other party disconnected</td></tr>
<tr><td rowspan="16">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>CPG(conference established)</td><td>→</td><td></td><td>→</td><td>INFO(CPG)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>CPG(other party added)</td><td>→</td><td></td><td>→</td><td>INFO(CPG)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>CPG(other party disconnected)</td><td>→</td><td></td><td>→</td><td>INFO(CPG)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP411008</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1],
Q.734 [i.7], clause 1.6.15</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP-ISUP/SS/CONF</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">Generic notification transfer "conference established", "other party added" and "other party disconnected"

To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message:
1. Assist a call set up from SIP-I to ISUP;
2. Check the notification "other party disconnected" in the CPG.
I-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">CPG: Generic notification: conference established
CPG: Generic notification: other party added
CPG: Generic notification: other party disconnected</td></tr>
<tr><td rowspan="16">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td colspan="2">ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td colspan="2">IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td colspan="2">ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td colspan="2">ANM</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="2"></td></tr>
<tr><td>INFO(CPG)</td><td>→</td><td></td><td>→</td><td colspan="2">CPG(conference established)</td></tr>
<tr><td>200 OK INFO</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>INFO(CPG)</td><td>→</td><td></td><td>→</td><td colspan="2">CPG(other party added)</td></tr>
<tr><td>200 OK INFO</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>INFO(CPG)</td><td>→</td><td></td><td>→</td><td colspan="2">CPG(other party disconnected)</td></tr>
<tr><td>200 OK INFO</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td colspan="2">REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td colspan="2">RLC</td></tr>
</table>

TP411009	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/CONF	
SIP selection criteria		
ISUP selection criteria		
Test purpose	Generic notification transfer "conference established", and disconnect the conference To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message: 1. Assist a call set up from ISUP to SIP-I; 2. Check that the notification "conference established" is received in the CPG from conferee at ISUP; 3. Release the conference. O-MGCF interworking.	
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body	
ISUP parameter values	CPG: Generic notification: conference established	
Comments		

ISUP		SUT		SIP-I
IAM	→		→	INVITE(IAM)
ACM	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
		Conversation		
CPG(conference established)	→		→	INFO(CPG)
			←	200 OK INFO
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

TP411010	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/CONF	
SIP selection criteria		
ISUP selection criteria		
Test purpose	Generic notification transfer "conference established", and disconnect the conference To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message: 1. Assist a call set up from SIP-I to ISUP; 2. Check that the notification "conference established" is received in the INFO(CPG) from conferee at SIP-I; 3. Release the conference. I-MGCF interworking.	
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body	
ISUP parameter values	CPG: Generic notification: conference established	
Comments		

SIP-I		SUT		ISUP
INVITE(IAM)	→		→	IAM
180 Ringing(ACM)	←		←	ACM
200 OK INVITE(ANM)	←		←	ANM
		Conversation		
INFO(CPG)	→		→	CPG(conference established)
200 OK INFO	←			
BYE(REL)	→		→	REL
200 OK BYE(RLC)	←		←	RLC

5.3.12 ECT

TP412001	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], Q.732.7 [i.5], clause 7.5.2.1.1.1 a)
TSS reference	ISUP-SIP-ISUP/SS/ECT	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Capability of sending a call transfer number for the active user** Verify that the IUT is able to send the Generic notification parameter "Call transfer active", the service activation parameter "call transfer" and the call transfer number, received in the ISUP FAC, in an INFO request for the active user. O-MGCF interworking.	
SIP parameter values	INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body	
ISUP parameter values	FAC: Generic notification=call transfer active, Call transfer number (PIXIT)	
Comments		

ISUP		**SUT**		**SIP-I**
IAM	→		→	INVITE(IAM)
ACM	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
		Conversation		
FAC(call transfer active, CTNb)	→		→	INFO(FAC)
			←	200 OK INFO
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

TP412002	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/ECT	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Capability of sending the call transfer number for the active user** Verify that the IUT is able to send the Generic notification parameter "Call transfer active", the service activation parameter "call transfer" and the call transfer number, received in the INFO request containing the encapsulated FAC, in a ISUP FAC for the active user. I-MGCF interworking.	
SIP parameter values	INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body	
ISUP parameter values	FAC: Generic notification=call transfer active, Call transfer number (PIXIT	
Comments		

SIP-I		**SUT**		**ISUP**
INVITE(IAM)	→		→	IAM
180 Ringing(ACM)	←		←	ACM
200 OK INVITE(ANM)	←		←	ANM
		Conversation		
INFO(FAC)	→		→	FAC(call transfer active, CTNb)
200 OK INFO	←			
BYE(REL)	→		→	REL
200 OK BYE(RLC)	←		←	RLC

TP412005	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.732.7 [i.5], clause 7.5.2.1.1.1 a)**
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Capability of sending the call transfer number for the held user** Verify that the IUT is able to transfer the call transfer number received in an ISUP FAC and sent in INFO request containing the FAC for the held user. O-MGCF interworking.				
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body				
ISUP parameter values	CPG: Event indicator=progress, Generic notification=hold FAC: Generic notification=call transfer active, Call transfer number(PIXIT)				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	CPG(hold)	→		→	INVITE(CPG, sendonly)
				←	200 OK INVITE(recvonly)
				→	ACK
	FAC(call transfer active, CTNb)	→		→	INFO(FAC)
				←	200 OK INFO
				→	INVITE(sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

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

TP412009	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.732.7 [i.5], clause 7.5.2.1.1.1 a)**
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Loop prevention procedure - initiation** Verify that the SUT is able to transfer the loop request received in an ISUP LOP in an INFO request containing the LOP message. SUT is able to transfer the loop response received in an ISUP LOP in an SIP INFO request containing the ISUP LOP message. O-MGCF interworking.				
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body INFO: Content-Type: application/ISUP; LOP encapsulated in the MIME body				
ISUP parameter values	CPG: Event indicator=progress, Generic notification=hold LOP: request: Call transfer reference LOP: response: Call transfer reference FAC: Generic notification=call transfer active, Call transfer number(PIXIT)				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	CPG(hold)	→		→	INVITE(CPG, sendonly)
				←	200 OK INVITE(recvonly)
				→	ACK
	LOP(request)	→		→	INFO(LOP)
				←	200 OK INFO
	LOP(response)	←		←	INFO(LOP)
				→	200 OK INFO
	FAC(call transfer active, CTNb)	→		→	INFO(FAC)
				←	200 OK INFO
				→	INVITE(sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	REL	→		→	BYE(REL)
	RLC(RLC)	←		←	200 OK BYE

TP412010	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15**
TSS reference	ISUP-SIP-ISUP/SS/ECT	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Loop prevention procedure - initiation** Verify that the SUT is able to transfer the loop request received in an INFO request containing the ISUP LOP message. Verify that the SUT is able to transfer the loop response received in an ISUP LOP message in the SIP INFO request containing the ISUP LOP message. I-MGCF interworking.	
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body INFO: Content-Type: application/ISUP; LOP encapsulated in the MIME body	
ISUP parameter values	CPG: Event indicator=progress, Generic notification=hold LOP: request: Call transfer reference LOP: response: Call transfer reference FAC: Generic notification=call transfer active, Call transfer number(PIXIT)	

Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	INVITE(CPG, sendonly)	→		→	CPG(hold)
	200 OK INVITE(recvonly)	←			
	ACK	→			
	INFO(LOP)	→		→	LOP(request)
	200 OK INFO	←			
	INFO(LOP)	←		←	LOP(response)
	200 OK INFO	→			
	INFO(FAC)	→		→	FAC(call transfer active, CTNb)
	200 OK INFO	←			
	INVITE(sendrecv)	→			
	200 OK INVITE(sendrecv)	←			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP412011	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.732.7 [i.5], clause 7.5.2.1.1.1 a)**
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Loop prevention procedure - unsuccessful on timer expiry** To verify that the SUT is able to transfer the loop request received in an ISUP LOP in an INFO request containing the LOP message. Verify that the connection is unsuccessful if the loop detection procedure is unsuccessful. The connection is released from the remote end. O-MGCF interworking.				
SIP parameter values	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, Generic notification=hold LOP: request: Call transfer reference				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	CPG(hold)	→		→	INVITE(CPG, sendonly)
				←	200 OK INVITE(recvonly)
				→	ACK
	LOP(request)	→		→	INFO(LOP)
				←	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP412012	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/CONF				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Loop prevention procedure - unsuccessful on timer expiry** To verify that SUT is able to transfer the loop request received in an INFO request containing the LOP message in an ISUP LOP message. Verify that the connection is unsuccessful if the loop detection procedure is unsuccessful. The connection is released from the remote end. I-MGCF interworking.				
SIP parameter values	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, Generic notification=hold LOP: request: Call transfer reference				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	INVITE(CPG, sendonly)	→		→	CPG(hold)
	200 OK INVITE(recvonly)	←			
	ACK	→			
	INFO(LOP)	→		→	LOP(request)
	200 OK INFO	←			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP412013	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.732.7 [i.5], clause 7.5.2.1.1.1 a)
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Loop prevention procedure - successful on timer expiry** Verify that the SUT is able to transfer the loop request received in an ISUP LOP in an INFO request containing the LOP message. Verify that the connection is successful if the loop detection procedure is unsuccessful. O-MGCF interworking.				
SIP parameter values	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 values	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	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	CPG(hold)	→		→	INVITE(CPG, sendonly)
				←	200 OK INVITE(recvonly)
				→	ACK
	LOP(request)	→		→	INFO(LOP)
				←	200 OK INFO
	FAC(call transfer active, CTNb)	→		→	INFO(FAC)
				←	200 OK INFO
				→	INVITE(sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP412014</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
Q.734 [i.7], clause 1.6.15</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/ECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Loop prevention procedure - successful on timer expiry
Verify that the SUT is able to transfer the loop request received in an INFO request containing the LOP message in an ISUP LOP message. Verify that the connection is successful if the loop detection procedure is unsuccessful.
I-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">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</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">CPG: Event indicator=progress, Generic notification=hold
LOP: request: Call transfer reference
FAC: Generic notification=call transfer active, Call transfer number(PIXIT)</td></tr>
<tr><td rowspan="22">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>INVITE(CPG, sendonly)</td><td>→</td><td></td><td>→</td><td>CPG(hold)</td></tr>
<tr><td>200 OK INVITE(recvonly)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>INFO(LOP)</td><td>→</td><td></td><td>→</td><td>LOP(request)</td></tr>
<tr><td>200 OK INFO</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>INFO(FAC)</td><td>→</td><td></td><td>→</td><td>FAC(call transfer active, CTNb)</td></tr>
<tr><td>200 OK INFO</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>INVITE(sendrecv)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK INVITE(sendrecv)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP412015	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.732.7 [i.5], clause 7.5.2.1.1.1 a)
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Facility message with generic notification sent to the remote user** Verify that the SUT is able to transfer the generic notification "call transfer, active" or "call transfer, alerting" and the **service activation** parameter set to "call transfer" received in an ISUP FAC in a SIP INFO request containing the ISUP FAC. O-MGCF interworking.				
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body				
ISUP parameter values	CPG: Event indicator=progress, Generic notification=hold FAC: Generic notification=call transfer active, Call transfer number(PIXIT)				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	CPG(hold)	→		→	INVITE(CPG, sendonly)
				←	200 OK INVITE(recvonly)
				→	ACK
	FAC(call transfer active, CTNb)	→		→	INFO(FAC)
				←	200 OK INFO
				→	INVITE(sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP412016</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1],
Q.734 [i.7], clause 1.6.15</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP-ISUP/SS/ECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">Facility message with generic notification sent to the remote user
Verify that the SUT is able to transfer the generic notification generic notification set to "call transfer, active" or "call transfer, alerting" and the service activation parameter set to "call transfer" received in a SIP-I INFO request containing the ISUP FAC message in an ISUP FAC message. O-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body
INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">CPG: Event indicator=progress, Generic notification=hold
FAC: Generic notification=call transfer active, Call transfer number(PIXIT)</td></tr>
<tr><td rowspan="18">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td colspan="2">ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td colspan="2">IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td colspan="2">ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td colspan="2">ANM</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="2"></td></tr>
<tr><td>INVITE(CPG, sendonly)</td><td>→</td><td></td><td>→</td><td colspan="2">CPG(hold)</td></tr>
<tr><td>200 OK INVITE(recvonly)</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>INFO(FAC)</td><td>→</td><td></td><td>→</td><td colspan="2">FAC(call transfer active, CTNb)</td></tr>
<tr><td>200 OK INFO</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>INVITE(sendrecv)</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>200 OK INVITE(sendrecv)</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td colspan="2">REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td colspan="2">RLC</td></tr>
</table>

<table>
<tr><th>TP412017</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1],
Q.732.7 [i.5], clause 7.5.2.1.1.1 a)</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP-ISUP/SS/ECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">Call progress message with generic notification sent to the remote user
Verify that the transfer the CPG with the generic notification set to "call transfer, active" and the service activation parameter set to "call transfer" in a SIP-I INFO request containing the ISUP CPG message.
O-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">CPG: Generic notification=call transfer active, Call transfer number (PIXIT)</td></tr>
<tr><td rowspan="11">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td colspan="2">SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing(ACM)</td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>CPG(call transfer active, CTNb)</td><td>→</td><td></td><td>→</td><td colspan="2">INFO(CPG)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="2"></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td colspan="2">BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP412018</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1],
Q.734 [i.7], clause 1.6.15</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/ECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Call progress message with generic notification sent to the remote user
Verify that the SUT is able to transfer the ISUP CPG with the generic notification set to "call transfer, active" and the service activation parameter set to "call transfer" contained in SIP-I INFO request in an ISUP CPG. The held user is retrieved by receiving a re-INVITE sendrecv.
I-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">CPG: Event indicator=progress, Generic notification=hold
CPG: Generic notification=call transfer active, Call transfer number(PIXIT)</td></tr>
<tr><td rowspan="18">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>INVITE(CPG, sendonly)</td><td>→</td><td></td><td>→</td><td>CPG(hold)</td></tr>
<tr><td>200 OK INVITE(recvonly)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>INFO(CPG)</td><td>→</td><td></td><td>→</td><td>CPG(call transfer active, CTNb)</td></tr>
<tr><td>200 OK INFO</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>INVITE(sendrecv)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK INVITE(sendrecv)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP412019</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1],
Q.734 [i.7], clause 1.6.15</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/ECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Call transfer number - removal of number
Verify that the exchange removes the call transfer number in the SIP-I INFO request containing a FAC or CPG before sending it to the next exchange, if its indicator is set to "presentation restricted" and there is no bilateral agreement to transfer the number.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body
INFO: Content-Type: application/ISUP; FAC(CTNb=restricted) encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">CPG: Event indicator=progress, Generic notification=hold
FAC: Generic notification=call transfer active, no Call transfer number(PIXIT)</td></tr>
<tr><td rowspan="18">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>INVITE(CPG, sendonly)</td><td>→</td><td></td><td>→</td><td>CPG(hold)</td></tr>
<tr><td>200 OK INVITE(recvonly)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>INFO(FAC)</td><td>→</td><td></td><td>→</td><td>FAC(call transfer active)</td></tr>
<tr><td>200 OK INFO</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>INVITE(sendrecv)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK INVITE(sendrecv)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

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

TP412021	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.732.7 [i.5], clause 7.5.2.1.1.1 a)
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call transfer number - removal of own country code** Verify that the IUT removes the country code in the address signals of the **call transfer number** if it is the network's own country code contained in the ISUP FAC message. The nature of address indicator shall be set to "national (significant) number"				
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC(CTNb=international) encapsulated in the MIME body				
ISUP parameter values	CPG: Event indicator=progress, Generic notification=hold FAC: Generic notification=call transfer active, Call transfer number=national(PIXIT)				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	CPG(hold)	→		→	INVITE(CPG, sendonly)
				←	200 OK INVITE(recvonly)
				→	ACK
	FAC(call transfer active, CTNb)	→		→	INFO(FAC)
				←	200 OK INFO
				→	INVITE(sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP412022</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1],
Q.732.7 [i.5], clause 7.5.2.1.1.1 a)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/ECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">ECT - Interaction with SUB
Verify that if the IUT is able to transfer the sub-address in the access transport parameter in the ISUP FAC message contained in the SIP-I INFO request in ISUP FAC message and vice versa received in an ISUP FAC message in a SIP-I INFO request containing the ISUP FAC message. These are the calling sub-address for incoming calls and the connected sub-address for outgoing calls.
O-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">FAC: Generic notification=call transfer active, Call transfer number(PIXIT)
FAC: ATP contained the connected sub address</td></tr>
<tr><td rowspan="16">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>FAC(call transfer active, CTNb)</td><td>→</td><td></td><td>→</td><td>INFO(FAC)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>FAC(ATP=SUB)</td><td>←</td><td></td><td>←</td><td>INFO(FAC)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>FAC(ATP=SUB)</td><td>→</td><td></td><td>→</td><td>INFO(FAC)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

5.3.13 3PTY

TP413001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.734.2 [i.8], clause 2.4; 2.2.1**
TSS reference	ISUP-SIP-ISUP/SS/3PTY				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Served user initiates 3PTY** Verify that the served user with two active calls is located, can successfully join this call (remote held user) to a three-way conversation, and notify the implied remote party accordingly. The IUT should transfer an ISUP **CPG** message with the **generic notification indicator** set to "conference established" in a SIP-I INFO request containing the ISUP CPG message. The **event indicator** in the **CPG** should be set to "progress": 1. setup a call to user B; 2. put this call on hold; 3. join this call to a conference. O-MGCF interworking.				
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body				
ISUP parameter values	CPG: Event indicator=progress, Generic notification=hold CPG: Event indicator=progress, Generic notification=conference established				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	CPG(hold)	→		→	INVITE(CPG, sendonly)
				←	200 OK INVITE(recvonly)
				→	ACK
	CPG(conference established)	→		→	INVITE(CPG, sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP413002</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1],
Q.734.2 [i.8], clause 2.4; 2.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP-ISUP/SS/3PTY</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">Served user initiates 3PTY
Verify that the served user with two active calls is located, can successfully join this call (remote held user) to a three-way conversation, and notify the implied remote party accordingly.
The IUT should send a CPG message with the generic notification indicator set to "conference established" to both implied parties. The event indicator in the CPG should be set to "progress":
1. setup a call to user B;
2. put this call on hold;
3. join this call to a conference.
I-MGCF interworking</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">CPG: Event indicator=progress, Generic notification=hold
CPG: Event indicator=progress, Generic notification=conference established</td></tr>
<tr><td rowspan="16">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td colspan="2">ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td colspan="2">IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td colspan="2">ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td colspan="2">ANM</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="2"></td></tr>
<tr><td>INVITE(CPG, sendonly)</td><td>→</td><td></td><td>→</td><td colspan="2">CPG(hold)</td></tr>
<tr><td>200 OK INVITE(recvonly)</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>INVITE(CPG, sendrecv)</td><td>→</td><td></td><td>→</td><td colspan="2">CPG(conference established)</td></tr>
<tr><td>200 OK INVITE(sendrecv)</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="2"></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td colspan="2">REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td colspan="2">RLC</td></tr>
</table>

TP413003	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], Q.734.2 [i.8], clause 2.4; 2.2.1**
TSS reference	ISUP-SIP-ISUP/SS/3PTY	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Served user initiates 3PTY** Verify that the served user with two active calls is located, can successfully join this call (remote 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 encapsulated in the MIME body	
ISUP parameter values	CPG: Event indicator=progress, Generic notification=conference established	
Comments		

ISUP		**SUT**		**SIP-I**
IAM	→		→	INVITE(IAM)
ACM	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
		Conversation		
CPG(conference established)	→		→	INFO(CPG)
			←	200 OK INFO
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

TP413004	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], Q.734.2 [i.8], clause 2.4; 2.2.1**
TSS reference	ISUP-SIP-ISUP/SS/3PTY	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Served user initiates 3PTY** Verify that the served user with two active calls is located, can successfully join this call (remote active user) to a three-way conversation, and notify the implied remote party accordingly. The IUT should send a **CPG** message with the **generic notification indicator** set to "conference established" to both implied parties. The **event indicator** in the **CPG** should be set to "progress": 1. setup a call to user B; 2. establish a conference. I-MGCF interworking.	
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body	
ISUP parameter values	CPG: Event indicator=progress, Generic notification=conference established	
Comments		

SIP-I		**SUT**		**ISUP**
INVITE(IAM)	→		→	IAM
180 Ringing(ACM)	←		←	ACM
200 OK INVITE(ANM)	←		←	ANM
		Conversation		
INFO(CPG)	→		→	CPG(conference established)
200 OK INFO	←			
		Conversation		
BYE(REL)	→		→	REL
200 OK BYE(RLC)	←		←	RLC

<table>
<tr><td>TP413005</td><td colspan="4">SIP reference: RFC 3261 [4]</td><td>ISUP reference:
Q.1912.5 [1],
Q.734.2 [i.8], clause 2.5.2.1.1.3 a</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/3PTY</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Served user creates a private communication with a remote user
Verify that a 3PTY call can successfully create private communication with the active-held user. The appropriate notification received in a ISUP CPG and is sent in INVITE/INFO (CPG) messages to the SIP-I.
O-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">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</td></tr>
<tr><td rowspan="30">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>CPG 1(hold)</td><td>→</td><td></td><td>→</td><td>INVITE(CPG, sendonly)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(recvonly)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>CPG 2(conference established)</td><td>→</td><td></td><td>→</td><td>INVITE(CPG, sendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(sendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>CPG 3(conference disconnected)</td><td>→</td><td></td><td>→</td><td>INFO(CPG)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>CPG 4(hold)</td><td>→</td><td></td><td>→</td><td>INVITE(CPG, sendonly)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(recvonly)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>CPG 5(retrieve)</td><td>→</td><td></td><td>→</td><td>INVITE(CPG, sendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(sendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>CPG 6(conference established)</td><td>→</td><td></td><td>→</td><td>INFO(CPG)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP413006</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
Q.734.2 [i.8], clause 2.5.2.1.1.3 a</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/3PTY</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Served user creates a private communication with a remote user

Verify that a 3PTY call can successfully create private communication with the active-held user. The appropriate notification received in a INVITE/INFO (CPG) and is sent in CPG messages to the ISUP.
I-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">CPG: Event indicator=progress, Generic notification=hold
CPG: Event indicator=progress, Generic notification=retrieve
CPG: Event indicator=progress, Generic notification=conference established
CPG: Event indicator=progress, Generic notification=conference disconnected</td></tr>
<tr><td rowspan="28">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>INVITE(CPG, sendonly)</td><td>→</td><td></td><td>→</td><td>CPG(hold)</td></tr>
<tr><td>200 OK INVITE(recvonly)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>INVITE(CPG, sendrecv)</td><td>→</td><td></td><td>→</td><td>CPG(conference established)</td></tr>
<tr><td>200 OK INVITE(sendrecv)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>INFO(CPG)</td><td>→</td><td></td><td>→</td><td>CPG(conference disconnected)</td></tr>
<tr><td>200 OK INFO</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>INVITE(CPG, sendonly)</td><td>→</td><td></td><td>→</td><td>CPG(hold)</td></tr>
<tr><td>200 OK INVITE(recvonly)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>INVITE(CPG, sendrecv)</td><td>→</td><td></td><td>→</td><td>CPG(retrieve)</td></tr>
<tr><td>200 OK INVITE(sendrecv)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>INFO(CPG)</td><td>→</td><td></td><td>→</td><td>CPG(conference established)</td></tr>
<tr><td>200 OK INFO</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td></td><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP413007</th><th colspan="4">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1],
Q.734.2 [i.8], clause 2.5.2.1.1.3 a</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/3PTY</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Served user creates a private communication with a remote user
Verify that the IUT (controlling the conference) on a 3PTY call can successfully create private communication with the active-idle user. The appropriate notification is sent in CPG messages to the user.
O-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">CPG: Event indicator=progress, Generic notification=conference established
CPG: Event indicator=progress, Generic notification=conference disconnected</td></tr>
<tr><td rowspan="16">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>CPG(conference established)</td><td>→</td><td></td><td>→</td><td>INFO(CPG)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>CPG(conference disconnected)</td><td>→</td><td></td><td>→</td><td>INFO(CPG)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>CPG(conference established)</td><td>→</td><td></td><td>→</td><td>INFO(CPG)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP413008	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.734.2 [i.8], clause 2.5.2.1.1.3 a**
TSS reference	ISUP-SIP-ISUP/SS/3PTY				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Served user creates a private communication with a remote user** Verify that the IUT (controlling the conference) on a 3PTY call can successfully create private communication with the active-idle user. The appropriate notification is sent in **CPG** messages to the user. I-MGCF interworking.				
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body				
ISUP parameter values	CPG: Event indicator=progress, Generic notification=conference established CPG: Event indicator=progress, Generic notification=conference disconnected				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	INFO(CPG)	→		→	CPG(conference established)
	200 OK INFO	←			
	INFO(CPG)	→		→	CPG(conference disconnected)
	200 OK INFO	←			
			Conversation		
	INFO(CPG)	→		→	CPG(conference established)
	200 OK INFO	←			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP413009</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1],
Q.734.2 [i.8], clause 2.5.2.1.1.3 b</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP-ISUP/SS/3PTY</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">Served user disconnects one remote user and retains the other
Verify that the IUT (controlling the conference) on a 3PTY call can successfully disconnect the active-held user and retain and notify the other user appropriately using CPG messages.
The IUT should send to the appropriate remote users CPG messages with a generic notification indicator. The event indicator in the CPG should be set to "progress".
O-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">CPG: Event indicator=progress, Generic notification=conference established
CPG: Event indicator=progress, Generic notification=conference disconnected</td></tr>
<tr><td rowspan="14">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td colspan="2">SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="2"></td></tr>
<tr><td>CPG(conference established)</td><td>→</td><td></td><td>→</td><td colspan="2">INFO(CPG)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>CPG(conference disconnected)</td><td>→</td><td></td><td>→</td><td colspan="2">INFO(CPG)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK INFO</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="2"></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td colspan="2">BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK BYE(RLC)</td></tr>
</table>

TP413010	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], Q.734.2 [i.8], clause 2.5.2.1.1.3 b**
TSS reference	ISUP-SIP-ISUP/SS/3PTY	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Served user disconnects one remote user and retains the other** Verify that the IUT (controlling the conference) on a 3PTY call can successfully disconnect the active-held user and retain and notify the other user appropriately using **CPG** messages. The IUT should send to the appropriate remote users **CPG** messages with a **generic notification indicator**. The **event indicator** in the **CPG** should be set to "progress" I-MGCF interworking.	
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body	
ISUP parameter values	CPG: Event indicator=progress, Generic notification=conference established CPG: Event indicator=progress, Generic notification=conference disconnected	
Comments		

SIP-I		**SUT**		**ISUP**
INVITE(IAM)	→		→	IAM
180 Ringing(ACM)	←		←	ACM
200 OK INVITE(ANM)	←		←	ANM
		Conversation		
INFO(CPG)	→		→	CPG(conference established)
200 OK INFO	←			
INFO(CPG)	→		→	CPG(conference disconnected)
200 OK INFO	←			
		Conversation		
BYE(REL)	→		→	REL
200 OK BYE(RLC)	←		←	RLC

TP413011	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], Q.734.2 [i.8], clause 2.5.2.1.1.3 b**
TSS reference	ISUP-SIP-ISUP/SS/3PTY	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Served user disconnects one remote user and retains the other** Verify that the IUT (controlling the conference) on a 3PTY call can successfully disconnect the active-idle user and retain and notify the other user appropriately using **CPG** messages. The IUT should send to the appropriate remote users **CPG** messages with a **generic notification indicator**. The **event indicator** in the **CPG** should be set to "progress". O-MGCF interworking.	
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body	
ISUP parameter values	CPG: Event indicator=progress, Generic notification=hold CPG: Event indicator=progress, Generic notification=conference established CPG: Event indicator=progress, Generic notification=conference disconnected	
Comments		

ISUP		**SUT**		**SIP-I**
IAM	→		→	INVITE(IAM)
ACM	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
		Conversation		
CPG(hold)	→		→	INVITE(CPG, sendonly)
			←	200 OK INVITE(recvonly)
			→	ACK
CPG(conference established)	→		→	INVITE(CPG, sendrecv)
			←	200 OK INVITE(sendrecv)
			→	ACK
CPG(conference disconnected)	→		→	INFO(CPG)
			←	200 OK INFO
CPG(hold)	→		→	INVITE(CPG, sendonly)
			←	200 OK INVITE(recvonly)
			→	ACK
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

TP413012	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.734.2 [i.8], clause 2.5.2.1.1.3 b**
TSS reference	ISUP-SIP-ISUP/SS/3PTY				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Served user disconnects one remote user and retains the other** Verify that the IUT (controlling the conference) on a 3PTY call can successfully disconnect the active-idle user and retain and notify the other user appropriately using **CPG** messages. The IUT should send to the appropriate remote users **CPG** messages with a **generic notification indicator**. The **event indicator** in the **CPG** should be set to "progress". O-MGCF interworking.				
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body				
ISUP parameter values	CPG: Event indicator=progress, Generic notification=hold CPG: Event indicator=progress, Generic notification=conference established CPG: Event indicator=progress, Generic notification=conference disconnected				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	INVITE(CPG, sendonly)	→		→	CPG(hold)
	200 OK INVITE(recvonly)	←			
	ACK	→			
	INVITE(CPG, sendrecv)	→		→	CPG(conference established)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
	INFO(CPG)	→		→	CPG(conference disconnected)
	200 OK INFO	←			
	INVITE(CPG, sendonly)	→		→	CPG(hold)
	200 OK INVITE(recvonly)	←			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

5.3.14 User-to-user service

5.3.14.1 User-to-user service 1

TP414001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.1.5.2.3 and 4, Q.737 [i.10]**
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Service 1 implicit request: User-to-user information in the IAM* Ensure that the SUT can successfully transfer the User-to-user service 1 implicit request in the encapsulated IAM. O-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM containing the user-to-user information parameter encapsulated in the MIME body				
ISUP parameter values	IAM: User-to-user information parameter				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP414002	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses A.1.1 1.1.5.2.3 and 4, Q.737 [i.10]**
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Service 1 implicit request: User-to-user information in the INVITE* Ensure that the SUT can successfully transfer the User-to-user service 1 implicit request in the encapsulated IAM. I-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM containing the user-to-user information parameter encapsulated in the MIME body				
ISUP parameter values	IAM: User-to-user information parameter				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP414003	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.1.5.2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS1			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that the SUT can successfully transfer the User-to-user service 1 explicit request not essential in the encapsulated IAM. O-MGCF interworking.			
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM containing the user-to-user indicator parameter encapsulated in the MIME body			
ISUP parameter values	IAM: User-to-user information parameter, User-to-user indicator = service 1 explicit request			
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP414004	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.1.5.2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS1			
SIP selection criteria				
ISUP selection criteria				
Test purpose	*Service 1 explicit request: User-to-user indicator in the INVITE* Ensure that the SUT can successfully transfer the User-to-user service 1 explicit request essential received in the IAM. O-MGCF interworking.			
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM containing the user-to-user indicator parameter encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM containing the user-to-user indicator parameter encapsulated in the MIME body			
ISUP parameter values	IAM: User-to-user information parameter, User-to-user indicator ACM: User-to-user indicator set to service 1 supported response			
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP414005	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.1.5.2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS1			
SIP selection criteria				
ISUP selection criteria				
Test purpose	*Service 1 explicit request: User-to-user indicator in the INVITE* Ensure that the SUT can successfully transfer the User-to-user service 1 explicit request essential received in the encapsulated IAM. I-MGCF interworking.			
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM containing the user-to-user indicator parameter encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM containing the user-to-user indicator parameter encapsulated in the MIME body			
ISUP parameter values	IAM: User-to-user information parameter, User-to-user indicator ACM: User-to-user indicator set to service 1 supported response			
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP414006	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.1.5.2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS1			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that the SUT can successfully transfer the User-to-user service 1 implicit response in the encapsulated ACM. O-MGCF interworking.			
SIP parameter values	Service 1 implicit response: User-to-user information in the 180 Ringing INVITE: Content-Type: application/ISUP; IAM containing the user-to-user information parameter encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM containing the user-to-user information parameter encapsulated in the MIME body			
ISUP parameter values	IAM: User-to-user information parameter ACM: User-to-user information parameter			
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP414007	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.1.5.2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Service 1 implicit response: User-to-user information in the ACM Ensure that the SUT can successfully transfer the User-to-user service 1 implicit response in the encapsulated ACM. I-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM containing the user-to-user information parameter encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM containing the user-to-user information parameter encapsulated in the MIME body				
ISUP parameter values	IAM: User-to-user information parameter ACM: User-to-user information parameter				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP414008	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.1.5.2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Service 1 explicit response service 1 not supported in the 180 Ringing* Ensure that the SUT can successfully transfer the User-to-user service 1 explicit response not supported in the encapsulated ACM. O-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM containing the user-to-user information parameter encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM containing the user-to-user indicator parameter encapsulated in the MIME body				
ISUP parameter values	IAM: User-to-user information parameter, User-to-user indicator set to service 1 request ACM: User-to-user indicator set to service 1 not supported response				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP414009	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.1.5.2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS1			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Service 1 explicit response service 1 not supported in the ACM Ensure that the SUT can successfully transfer the User-to-user service 1 explicit response not supported in the ACM. I-MGCF interworking.			
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM containing the user-to-user information parameter encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM containing the user-to-user indicator parameter encapsulated in the MIME body			
ISUP parameter values	IAM: User-to-user information parameter, User-to-user indicator set to service 1 request ACM: User-to-user indicator set to service 1 not supported response			
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP414010	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.1.5.2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS1			
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 values	INVITE: Content-Type: application/ISUP; IAM containing the user-to-user information parameter encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM containing the User-to-user indicator parameter encapsulated in the MIME body			
ISUP parameter values	IAM: User-to-user information parameter ACM: User-to-user indicator set to discarded by the network response			
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP414011</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clauses A.1.1, 1.1.5.2.3 and 4, Q.737 [i.10]</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/UUS1</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT can successfully transfer the User-to-user service 1 discarded by the network in the encapsulated ACM.
I-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: application/ISUP; IAM containing the user-to-user information parameter encapsulated in the MIME body
180 Ringing: Content-Type: application/ISUP; ACM containing the User-to-user indicator parameter encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: User-to-user information parameter
ACM: User-to-user indicator set to discarded by the network response</td></tr>
<tr><td rowspan="7">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

5.3.14.2 User-to-user service 2

<table>
<tr><th>TP414101</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clauses A.1.1, 1.2.5.2.3 and 4, Q.737 [i.10]</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/UUS2</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Service 2 request not essential transferred in the INVITE

Ensure that the SUT can successfully transfer the User-to-user service 2 explicit request and User-to user information in the encapsulated IAM. An additional User-to-user information is sent in a USR message encapsulated in an INFO request.
O-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: application/ISUP; IAM containing the user-to-user indicator and User-to-user information encapsulated in the MIME body
INFO: Content-Type: application/ISUP; USR containing the User-to-user information parameter encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: User-to-user information parameter, User-to-user indicator
USR: User-to-user information</td></tr>
<tr><td rowspan="13">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>USR</td><td>→</td><td></td><td>→</td><td>INFO(USR)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>USR</td><td>←</td><td></td><td>←</td><td>INFO(USR)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP414102	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.2.5.2.3 and 4, Q.737 [i.10]**
TSS reference	ISUP-SIP-ISUP/SS/UUS2				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Service 2 request not essential transferred in the IAM Ensure that the SUT can successfully transfer the User-to-user service 2 explicit request in the encapsulated IAM. An additional User-to-user information is sent in a USR message encapsulated in an INFO request. O-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM containing the user-to-user indicator encapsulated in the MIME body INFO: Content-Type: application/ISUP; USR containing the User-to-user information parameter encapsulated in the MIME body				
ISUP parameter values	IAM: User-to-user information parameter, User-to-user indicator USR: User-to-user information				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	INFO(USR)	→		→	USR
	200 OK INFO	←			
	INFO(USR)	←		←	USR
	200 OK INFO	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP414103	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.2.5.2.3 and 4, Q.737 [i.10]**
TSS reference	ISUP-SIP-ISUP/SS/UUS2				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Service 2 response not provided transferred in the INVITE Ensure that the SUT can successfully transfer the User-to-user service 2 explicit response not supported in the encapsulated ACM. I-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM containing the user-to-user information parameter encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM containing the user-to-user indicator parameter encapsulated in the MIME body				
ISUP parameter values	IAM: User-to-user information parameter, User-to-user indicator set to service 2 request ACM: User-to-user indicator set to service 2 not supported response				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP414104	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.2.5.2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS2				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Service 2 response not provided transferred in the IAM Ensure that the SUT can successfully transfer the User-to-user service 2 explicit response not provided in the encapsulated ACM. I-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM containing the user-to-user indicator parameter encapsulated in the MIME body				
ISUP parameter values	IAM: User-to-user information parameter, User-to-user indicator				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

5.3.14.3 User-to-user service 3

TP414201	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.3.5.2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can successfully transfer the User-to-user service 3 explicit request in the encapsulated IAM. Additional User-to-user information is sent in several USR message encapsulated in an INFO request. O-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM containing the user-to-user indicator encapsulated in the MIME body INFO: Content-Type: application/ISUP; USR containing the User-to-user information parameter encapsulated in the MIME body				
ISUP parameter values	IAM: User-to-user information parameter, User-to-user indicator USR: User-to-user information				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	USR	→		→	INFO(USR)
				←	200 OK INFO
	USR	←		←	INFO(USR)
				→	200 OK INFO
	USR	←		←	INFO(USR)
				→	200 OK INFO
	USR	→		→	INFO(USR)
				←	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP414202	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.3.5.2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can successfully transfer the User-to-user service 3 explicit request in the encapsulated IAM. Additional User-to-user information is sent in several USR message encapsulated in an INFO request. I-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM containing the user-to-user indicator encapsulated in the MIME body INFO: Content-Type: application/ISUP; USR containing the User-to-user information parameter encapsulated in the MIME body				
ISUP parameter values	IAM: User-to-user information parameter, User-to-user indicator USR: User-to-user information				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
		Conversation			
	INFO(USR)	→		→	USR
	200 OK INFO	←			
	INFO(USR)	←		←	USR
	200 OK INFO	→			
	INFO(USR)	←		←	USR
	200 OK INFO	→			
	INFO(USR)	→		→	USR
	200 OK INFO	←			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><td>TP414203</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1], clauses A.1.1,
1.3.5.2.3 and 4, Q.737 [i.10]</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/UUS3</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that a User-to-user request service 3 encapsulated in an INFO request during the confirmed state can successful proceeded. The User-to-user information is passed on in several encapsulated USR messages.
O-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INFO: Content-Type: application/ISUP; FAR containing the user-to-user indicator encapsulated in the MIME body
INFO: Content-Type: application/ISUP; FAA containing the user-to-user indicator encapsulated in the MIME body
INFO: Content-Type: application/ISUP; USR containing the User-to-user information parameter encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">FAR: User-to-user indicator service 3 request not essential
FAA: User-to-user indicator service 3 response provided
USR: User-to-user information</td></tr>
<tr><td rowspan="25">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>FAR</td><td>→</td><td></td><td>→</td><td>INFO(FAR)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>FAA</td><td>←</td><td></td><td>←</td><td>INFO(FAA)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>USR</td><td>→</td><td></td><td>→</td><td>INFO(USR)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>USR</td><td>←</td><td></td><td>←</td><td>INFO(USR)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>USR</td><td>←</td><td></td><td>←</td><td>INFO(USR)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>USR</td><td>→</td><td></td><td>→</td><td>INFO(USR)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP414204	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.3.5.2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that a User-to-user request service 3 encapsulated in an INFO request during the confirmed state can successful proceeded. The User-to-user information is passed on in several encapsulated USR messages. I-MGCF interworking.				
SIP parameter values	INFO: Content-Type: application/ISUP; FAR containing the user-to-user indicator encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAA containing the user-to-user indicator encapsulated in the MIME body INFO: Content-Type: application/ISUP; USR containing the User-to-user information parameter encapsulated in the MIME body				
ISUP parameter values	FAR: User-to-user indicator service 3 request not essential FAA: User-to-user indicator service 3 response provided USR: User-to-user information				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	INFO(FAR)	→		→	FAR
	200 OK INFO	←			
	INFO(FAA)	←		←	FAA
	200 OK INFO	→			
	INFO(USR)	→		→	USR
	200 OK INFO	←			
	INFO(USR)	←		←	USR
	200 OK INFO	→			
	INFO(USR)	←		←	USR
	200 OK INFO	→			
	INFO(USR)	→		→	USR
	200 OK INFO	←			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP414205</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses A.1.1,
1.3.5.2.3 and 4, Q.737 [i.10]</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/UUS3</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT can successfully transfer the User-to-user service 3 explicit response in the encapsulated ANM.
O-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: application/ISUP; IAM containing the user-to-user indicator parameter encapsulated in the MIME body
200 OK INVITE: Content-Type: application/ISUP; ANM containing the user-to-user indicator parameter encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: User-to-user indicator set to service 3 request
ANM: User-to-user indicator set to service 3 provided response</td></tr>
<tr><td rowspan="7">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP414206</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses A.1.1,
1.3.5.2.3 and 4, Q.737 [i.10]</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/UUS3</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT can successfully transfer the User-to-user service 3 explicit response in the encapsulated ANM.
O-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: application/ISUP; IAM containing the user-to-user indicator parameter encapsulated in the MIME body
200 OK INVITE: Content-Type: application/ISUP; ANM containing the user-to-user indicator parameter encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: User-to-user indicator set to service 3 request
ANM: User-to-user indicator set to service 3 provided response</td></tr>
<tr><td rowspan="7">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP414207	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.3.5.2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that a User-to-user request service 3 encapsulated in an INFO request during the confirmed state can successful proceeded. The user to user request is rejected.				
SIP parameter values	INFO: Content-Type: application/ISUP; FAR containing the user-to-user indicator encapsulated in the MIME body INFO: Content-Type: application/ISUP; FRJ containing the user-to-user indicator encapsulated in the MIME body				
ISUP parameter values	FAR: User-to-user indicator service 3 request not essential FRJ: User-to-user indicator service 3 response not provided				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	FAR	→		→	INFO(FAR)
				←	200 OK INFO
	FRJ	←		←	INFO(FRJ)
				→	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP414208	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.3.5.2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that a User-to-user request service 3 encapsulated in an INFO request during the confirmed state can successful proceeded. The user to user request is rejected				
SIP parameter values	INFO: Content-Type: application/ISUP; FAR containing the user-to-user indicator encapsulated in the MIME body INFO: Content-Type: application/ISUP; FRJ containing the user-to-user indicator encapsulated in the MIME body				
ISUP parameter values	FAR: User-to-user indicator service 3 request not essential FRJ: User-to-user indicator service 3 response not provided				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	INFO(FAR)	→		→	FAR
	200 OK INFO	←			
	INFO(FRJ)	←		←	FRJ
	200 OK INFO	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

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

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

A.1.1 Test purposes for ISDN/SIP Basic call

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

A.1.1.1.1 Sending of the SETUP Message

<table>
<tr><th>TP501001</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN reference:
Q.1912.5 [1], clause 6.1.2 (i,1)</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/4 AND NOT PICS 4/5</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5">NOT PICS 1/6</td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">SIP INVITE: Audio RTP/AVP 0 8
200 OK: Audio RTP/AVP 8</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">SETUP BC: A-law</td></tr>
<tr><td rowspan="9">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ALERTING</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>CONNECT</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>DISCONNECT</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RELEASE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>RELEASE COMPLETE</td></tr>
</table>

TP501002	**SIP reference: RFC 3261 [4]**		**ISDN reference: Q.1912.5 [1], clause 6.1.2 (i,2b**		
TSS reference:	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection criteria	PICS 4/4 AND PICS 4/5				
ISDN selection criteria	NOT PICS 1/6				
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, **with an SDP offer** and 100rel extensions and preconditions extensions in the SIP Supported header: including a SDP for **A-law** (PCMA) and **µ-law** (PCMU): • the SUT shall delete µ-law (PCMU), if present, from the media description that it will send back in the SDP answer; • the SETUP shall be deferred until all preconditions have been met. The BC is constructed from the ISUP TMR or USI. The bearer path shall be connected in both directions when both of the following conditions are satisfied: • the I-IWU determines (BCng the procedures defined in RFC 3312 [5]) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable). In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "*notification not required*", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312 [5]) that sufficient preconditions have been met for the session to proceed.				
SIP parameter values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 8				
ISDN parameter values	SETUP BC: A-law				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→			
	183 session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	SETUP
	200 OK UPDATE	←			
	180 Ringing(ACM)	←		←	ALERTING
	200 OK INVITE(ANM)	←		←	CONNECT
	ACK	→			
			Conversation		
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

TP501003	SIP reference: RFC 3261 [4]	ISDN reference: Q.1912.5 [1], clause 6.1.2 (i,1)
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message	
SIP selection criteria	NOT PICS 4/4 AND NOT PICS 4/5	
ISDN selection criteria	NOT PICS 1/6	
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, **with an SDP offer** including a SDP for **A-law** (PCMA) and **µ-law** (PCMU): • the SUT shall delete µ-law (PCMU) from the media description that it will send back in the SDP answer; • the SUT shall immediately send out the SETUP. The BC is constructed from the ISUP TMR or USI.	
SIP parameter values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 8	
ISDN parameter values	SETUP BC: A-law	
Comments		

SIP-I		SUT		ISDN
INVITE(IAM)	→		→	SETUP
180 Ringing(ACM)	←		←	ALERTING
200 OK INVITE(ANM)	←		←	CONNECT
ACK	→			
		Conversation		
BYE(REL)	→		→	DISCONNECT
200 OK BYE(RLC)	←		←	RELEASE
			→	RELEASE COMPLETE

TP501004	SIP reference: RFC 3261 [4]	ISDN reference: Q.1912.5 [1], clause 6.1.2 (i,2b)
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message	
SIP selection criteria	PICS 4/4 AND PICS 4/5	
ISDN selection criteria	NOT PICS 1/6	
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, **with an SDP offer** and 100rel extensions and preconditions extensions in the SIP Require header including a SDP for **A-law** (PCMA) and **µ-law** (PCMU): • the SUT shall delete µ-law (PCMU), if present, from the media description that it will send back in the SDP answer; • the SETUP shall be deferred until all preconditions have been met. The BC is constructed from the ISUP TMR or USI.	
SIP parameter values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 8	
ISDN parameter values	SETUP BC: A-law	
Comments		

SIP-I		SUT		ISDN
INVITE(IAM)	→			
183 session Progress	←			
PRACK	→			
200 OK PRACK	←			
UPDATE	→		→	SETUP
200 OK UPDATE	←			
180 Ringing(ACM)	←		←	ALERTING
200 OK INVITE(ANM)	←		←	CONNECT
ACK	→			
		Conversation		
BYE(REL)	→		→	DISCONNECT
200 OK BYE(RLC)	←		←	RELEASE
			→	RELEASE COMPLETE

TP501005	SIP reference: RFC 3261 [4]	ISDN reference: Q.1912.5 [1], clause 6.1.2 (i,1)
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message	
SIP selection criteria	NOT PICS 4/4 AND NOT PICS 4/5	
ISDN selection criteria	PICS 1/6	
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, **with an SDP offer** including a SDP for **A-law** (PCMA) and **µ-law** (PCMU): • the SUT shall delete A-law (PCMA) from the media description that it will send back in the SDP answer; • the SUT shall immediately send out the SETUP. The BC is constructed from the ISUP TMR or USI.	
SIP parameter values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 8	
ISDN parameter values	SETUP BC: A-law	

Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	180 Ringing(ACM)	←		←	ALERTING
	200 OK INVITE(ANM)	←		←	CONNECT
	ACK	→			
			Conversation		
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

TP501006	SIP reference: RFC 3261 [4]	ISDN reference: Q.1912.5 [1], clause 6.1.2 (i,2b)
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message	
SIP selection criteria	PICS 4/4 AND PICS 4/5	
ISDN selection criteria	PICS 1/6	
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, **with an SDP offer** and 100rel extensions and preconditions extensions in the SIP Supported header: including a SDP for **A-law** (PCMA) and **µ-law** (PCMU): • the SUT shall delete A-law (PCMA), if present, from the media description that it will send back in the SDP answer; • the SETUP shall be deferred until all preconditions have been met. The BC is constructed from the ISUP TMR or USI.	
SIP parameter values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 8	
ISDN parameter values	SETUP BC: A-law	

Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→			
	183 session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	SETUP
	200 OK UPDATE	←			
	180 Ringing(ACM)	←		←	ALERTING
	200 OK INVITE(ANM)	←		←	CONNECT
	ACK	→			
			Conversation		
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

TP501007	SIP reference: RFC 3261 [4]			ISDN reference: Q.1912.5 [1], clause 6.1.2 (i,1)	
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection criteria	NOT PICS 4/4 AND NOT PICS 4/5				
ISDN selection criteria	PICS 1/6				
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, **with an SDP offer** including a SDP for **A-law** (PCMA) and **µ-law** (PCMU): • the SUT shall delete A-law (PCMA) from the media description that it will send back in the SDP answer; • the SUT shall immediately send out the SETUP. The BC is constructed from the ISUP TMR or USI.				
SIP parameter values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 8				
ISDN parameter values	SETUP BC: A-law				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	180 Ringing(ACM)	←		←	ALERTING
	200 OK INVITE(ANM)	←		←	CONNECT
	ACK	→			
			Conversation		
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

TP501008	SIP reference: RFC 3261 [4]			ISDN reference: Q.1912.5 [1], clause 6.1.2 (i,2b)	
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection criteria	PICS 4/4 AND PICS 4/5				
ISDN selection criteria	PICS 1/6				
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, **with an SDP offer** and 100rel extensions and preconditions extensions in the SIP Require header including a SDP for **A-law** (PCMA) and **µ-law** (PCMU): • the SUT shall delete A-law (PCMA), if present, from the media description that it will send back in the SDP answer; • the SETUP shall be deferred until all preconditions have been met. The BC is constructed from the ISUP TMR or USI. The BC is constructed from the ISUP TMR or USI.				
SIP parameter values	INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 8				
ISDN parameter values	SETUP: BC: A-law				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→			
	183 session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	SETUP
	200 OK UPDATE	←			
	180 Ringing(ACM)	←		←	ALERTING
	200 OK INVITE(ANM)	←		←	CONNECT
	ACK	→			
			Conversation		
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

<table>
<tr><th>TP501009</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/4 AND NOT PICS 4/5 AND PICS 1/9</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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 shall be returned in the SDP answer as preferred codec.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">Offer: m=audio 4711 RTP/AVP 0 8
Answer: m=audio 4712 RTP/AVP 8 0</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="9">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ALERTING</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>CONNECT</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>DISCONNECT</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RELEASE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>RELEASE COMPLETE</td></tr>
</table>

<table>
<tr><th>TP501010</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 1/9</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT on receipt of an INVITE message with a SDP offer for µ-Law and A-Law and 100rel extensions and preconditions extensions in the SIP Supported header, then independent from the received order of preference:
• the SETUP shall be deferred until all preconditions have been met;
• the G.711 a-law codec shall be returned in the SDP answer as preferred codec.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">Offer: m=audio 4711 RTP/AVP 0 8
Answer: m=audio 4712 RTP/AVP 8 0</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="14">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>183 session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ALERTING</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>CONNECT</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>DISCONNECT</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RELEASE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>RELEASE COMPLETE</td></tr>
</table>

<table>
<tr><th>TP501011</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 1/9</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT on receipt of an INVITE message with a SDP offer for µ-Law and A-Law and 100rel extensions and preconditions extensions in the SIP Require header, then independent from the received order of preference:
• the SETUP shall be deferred until all preconditions have been met;
• the G.711 a-law codec shall be returned in the SDP answer as preferred codec.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">Offer: m=audio 4711 RTP/AVP 0 8
Answer: m=audio 4712 RTP/AVP 8 0</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="14">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>183 session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ALERTING</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>CONNECT</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>DISCONNECT</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RELEASE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>RELEASE COMPLETE</td></tr>
</table>

<table>
<tr><th>TP501012</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN reference:
Q.1912.5 [1], clause 6.1.2 (i,1)</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/4 AND NOT PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">Offer: m=audio 4711 RTP/AVP 8
m= video 4712 RTP/AVP 31

Answer: m=audio 4711 RTP/AVP 8
m=video 0 RTP/AVP 31</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="9">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ALERTING</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>CONNECT</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>DISCONNECT</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RELEASE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>RELEASE COMPLETE</td></tr>
</table>

TP501013	**SIP reference: RFC 3261 [4]**			**ISDN reference: Q.1912.5 [1], clause 6.1.2 (i,1)**	
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND NOT 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 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 values	Offer: m=audio 4711 RTP/AVP 8 m= video 4712 RTP/AVP 31 Answer: m=audio 4711 RTP/AVP 8 m=video 0 RTP/AVP 31				
ISDN parameter values					
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→			
	183 session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	SETUP
	200 OK UPDATE	←			
	180 Ringing(ACM)	←		←	ALERTING
	200 OK INVITE(ANM)	←		←	CONNECT
	ACK	→			
			Conversation		
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

TP501014	**SIP reference: RFC 3261 [4]**				**ISDN reference: Q.1912.5 [1], clause 6.1.2 (i,1)**
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND NOT 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: 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 RTP/AVP 8 m= video 4712 RTP/AVP 31 Answer: m=audio 4711 RTP/AVP 8 m=video 0 RTP/AVP 31				
ISDN parameter values					
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→			
	183 session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	SETUP
	200 OK UPDATE	←			
	180 Ringing(ACM)	←		←	ALERTING
	200 OK INVITE(ANM)	←		←	CONNECT
	ACK	→			
			Conversation		
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

TP501015	**SIP reference: RFC 3261 [4]**				**ISDN reference: Q.1912.5 [1], clause 6.1.2 (i,1)**
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection criteria	NOT PICS 4/4 AND NOT 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 and based on operator policy then:** the call is refused with a 415 Unsupported media type response.				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 8 m= video 4712 RTP/AVP 31				
ISDN parameter values					
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→			
	415 Unsupported media type	←			
	ACK	→			

TP501016	**SIP reference: RFC 3261 [4]**				**ISDN reference: Q.1912.5 [1], 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 Supported header: **and based on operator policy then:** the call is refused with a 415 Unsupported media type response.				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 8 m= video 4712 RTP/AVP 31				
ISDN parameter values					
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→			
	415 Unsupported media type	←			
	ACK	→			

TP501017	**SIP reference: RFC 3261 [4]**				**ISDN reference: Q.1912.5 [1], clause 6.1.2 (i,1)**
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection 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 values	Offer: m=audio 4711 RTP/AVP 8 m= video 4712 RTP/AVP 31				
ISDN parameter values					
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→			
	415 Unsupported media type	←			
	ACK	→			

<table>
<tr><th>TP501018</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN reference:
Q.1912.5 [1], clause 6.1.3</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 1/2</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5">PICS 1/9</td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">SETUP; BC Coding standard: CCITT standardized coding
Information transfer capability: Constructed from the USI or from the TMR
transfer mode: circuit mode
information transfer rate: 64 kbits/s</td></tr>
<tr><td rowspan="9">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ALERTING</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>CONNECT</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>DISCONNECT</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RELEASE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>RELEASE COMPLETE</td></tr>
</table>

Values for test purposes TP501018							
	a_b_m_LINE_VALUE						
m= line				b= line	a= line	BC_VALUE	
test purposes	**<media>**	**<transport>**	**<fmt-list>**	**<modifier>:<bandwidth-value>**	**rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>**	**Information Transport Capability**	**User Information Layer 1 Protocol Indicator**
				NOTE: <bandwidth value> for <modifier> of AS is evaluated to be B kbit/s.			
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	**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	**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	**"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 [4]**				**ISDN reference: Q.1912.5 [1], clause 6.1.3.5**
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message containing an encapsulated IAM, sends an SETUP message with the **HLC** information element constructed from the encapsulated ATP (HLC).				
SIP parameter values					
ISDN parameter values					
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	180 Ringing(ACM)	←		←	ALERTING
	200 OK INVITE(ANM)	←		←	CONNECT
	ACK	→			
		Conversation			
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

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

TP501021	**SIP reference: RFC 3261 [4]**				**ISDN reference:**
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT on receipt of an INVITE message with a Called party number **+CC NDC SN** where CC is the country code of the network in which the next hop terminates, component of the Request-URI send a SETUP message. Type of number: "**Subscriber number**", remove "+CC NDC" and use the remaining digits to fill the Address signals contained in the user info. **Numbering plan Indicator ISDN (Telephony) numbering plan.**				
SIP parameter values					
ISDN parameter values	SETUP : Called party number				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	180 Ringing(ACM)	←		←	ALERTING
	200 OK INVITE(ANM)	←		←	CONNECT
	ACK	→			
			Conversation		
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

TP501020	**SIP reference: RFC 3261 [4]**				**ISDN reference:**
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT on receipt of an INVITE message with a Called party number **+CC NDC SN** where CC is the country code of the network in which the next hop terminates, component of the Request-URI send a SETUP message. Type of number: "**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				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	180 Ringing(ACM)	←		←	ALERTING
	200 OK INVITE(ANM)	←		←	CONNECT
	ACK	→			
			Conversation		
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

A.1.1.1.1.1 Void

A.1.1.1.1.2 Sending of the INFO

TP502001	**SIP reference: RFC 3261 [4]**		**ISDN reference:**	
TSS reference	SIP-I-ISDN/Basic_call/Sending_of INFO_message			
SIP selection 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 which whereby the number of digits in the Request-URI is **greater** than the number of digits already accumulated for the call: • sends a INFO and pass it to outgoing ISDN procedures; • the INFO shall contain in its Subsequent Number parameter only the additional digits received in this Request-URI compared with the digits already accumulated for the call.			
SIP parameter values				
ISDN parameter values				

Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	INVITE(SAM)	→		→	INFO
	INVITE(SAM)	→		→	INFO
	180 Ringing(ACM)	←		←	ALERTING
	200 OK INVITE(ANM)	←		←	CONNECT
	ACK	→			
			Conversation		
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

TP502002	**SIP reference: RFC 3261 [4]**		**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 **fewer** than the number of digits already accumulated for the call: • then the SUT shall immediately send a **484 Address Incomplete** response for this INVITE; • in this case no INFO is sent to ISDN.			
SIP parameter values				
ISDN parameter values				

Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→			
	INVITE(SAM)	→			
	INVITE(SAM)	→			
	484 Address incomplete	←			
	ACK	→			

A.1.1.1.1.3 Receipt of the ALERTING - CALL PROCEEDING - PROGRESS message

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

TP503002	**SIP reference: RFC 3261 [4]**				**ISDN reference: Q.1912.5 [1], clause 6.5 1)**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in call state N6, on receipt the ALERTING message: • a 180 Ringing SIP response is sent. Ensure that the in-band information can be transmitted to the calling user.				
SIP parameter values					
ISDN parameter values					
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	180 Ringing(ACM)	←		←	ALERTING
	Inband Info				
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

TP503003	**SIP reference: RFC 3261 [4]**				**ISDN reference: Q.1912.5 [1], clause 6.5 2)**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in call state N9, on receipt the ALERTING message; • a **180 Ringing** SIP response is sent. Ensure that the in-band information can be transmitted to the calling user.				
SIP parameter values					
ISDN parameter values					
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROCEEDING
	180 Ringing(CPG)	←		←	ALERTING
	Inband Info				
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

TP503004	**SIP reference: RFC 3261 [4]**				**ISDN reference: Q.1912.5 [1], clause 6.5 2)**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection criteria	PICS 3/8 AND PICS 1/6				
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 values	183 Session Progress encapsulated ACM: BCi Called party status = no indication				
ISDN parameter values	CALL PROCEEDING				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROCEEDING
	CANCEL(REL)	→		→	RELEASE
	200 OK CANCEL(RLC)	←		←	RELEASE COMPLETE

TP503005	**SIP reference: RFC 3261 [4]**				**ISDN reference: Q.1912.5 [1], clause 6.5 2)**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in call state N6, on receipt of the CALL PROCEEDING message containing a **progress indicator** set to PI_VALUE: • a 183 Session Progress with an encapsulated ACM is sent to the previous entity.				
SIP parameter values	183 Session Progress encapsulated ACM: BCi Called party status = no indication, ATP with Progress indicator				
ISDN parameter values	CALL PROCEEDING				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROCEEDING(PI)
	CANCEL(REL)	→		→	RELEASE
	200 OK CANCEL(RLC)	←		←	RELEASE COMPLETE

<table>
<tr><th>TP503006</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISDN reference:
Q.1912.5 [1], clause 6.5 2)</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in call state N9, on receipt of the PROGRESS message containing a progress indicator set to PI_VALUE:
• a 183 Session Progress with an encapsulated ACM is sent to the previous entity.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">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</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">CALL PROCEEDING</td></tr>
<tr><td rowspan="7">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>183 Session Progress(ACM)</td><td>←</td><td></td><td>←</td><td>CALL PROCEEDING</td></tr>
<tr><td>183 Session Progress(CPG)</td><td>←</td><td></td><td>←</td><td>PROGRESS(PI)</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>CANCEL(REL)</td><td>→</td><td></td><td>→</td><td>RELEASE</td></tr>
<tr><td>200 OK CANCEL(RLC)</td><td>←</td><td></td><td>←</td><td>RELEASE COMPLETE</td></tr>
</table>

<table>
<tr><th>TP503007</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISDN reference:
Q.1912.5 [1], clause 6.5 2)</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5">PICS 1/6</td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">180 Ringing encapsulated ACM: BCi called party status=subscriber free, ATP with Progress indicator</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">ALERTING(PI)</td></tr>
<tr><td rowspan="7">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ALERTING(PI)</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>DISCONNECT</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RELEASE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>RELEASE COMPLETE</td></tr>
</table>

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

TP503009	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.5 2)
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection criteria	PICS 1/6				
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message sends out a SETUP message in state N6, where the end of address signalling is determined by analysis of the called party number to indicate that **a sufficient number of digits has been received** to route the call to the called party, on receipt of a CALL PROCEEDING: a 183 Session Progress with an encapsulated ACM is sent to the previous entity.				
SIP parameter values					
ISDN parameter values					
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROCEEDING
	CANCEL(REL)	→		→	RELEASE
	200 OK CANCEL(RLC)	←		←	RELEASE COMPLETE

TP503010	SIP reference: RFC 3261 [4]		ISDN reference: Q.1912.5 [1], clause 6.5 2)		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of 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)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROCEEDING
				←	PROGRESS
	CANCEL(REL)	→		→	RELEASE
	200 OK CANCEL(RLC)	←		←	RELEASE COMPLETE

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

TP503012	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.6
TSS reference	SIP-I-ISDN/Basic_call/Receipt_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, receives a CALL PROCEEDING message, on receipt of a PROGRESS message: • no message is sent.				
SIP parameter values	183 Session Progress: Encapsulated ACM, called party status indicator=no indication				
ISDN parameter values					
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROCEEDING
				←	PROGRESS
	CANCEL(REL)	→		→	RELEASE
	200 OK CANCEL(RLC)	←		←	RELEASE COMPLETE

TP503013	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.6
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection criteria	PICS 1/6				
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 an ALERTING: • sends a 180 Ringing with encapsulated CPG Alerting.				
SIP parameter values	183 Session Progress with encapsulated ACM: called party status indicator=no indication 180 Ringing encapsulated CPG: Event indicator=Alerting				
ISDN parameter values					
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROCEEDING
	180 Ringing(CPG)	←		←	ALERTING
	Inband Info				
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

TP503014	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.6
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection criteria	PICS 1/6				
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** where the progress description value is set to PI_VALUE, on receipt of a ALERTING Message: • sent a 180 Ringing message.				
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 Progress indicator 180 Ringing encapsulated CPG: Event indicator=Alerting				
ISDN parameter values					
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROCEEDING
	183 Session Progress(CPG)			←	PROGRESS(PI)
	180 Ringing(CPG)	←		←	ALERTING
	Inband Info				
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

TP503015	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.6
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection criteria	PICS 1/6				
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 **progress indicator** where the progress description value is set to PI_VALUE: • sent a 183 Session Progress message containing a encapsulated CPG.				
SIP parameter values	180 Ringing encapsulated ACM: BCi called party status=subscriber free 183 Session Progress with encapsulated CPG: event indicator=Progress, ATP with Progress indicator				
ISDN parameter values					
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	180 Ringing(ACM)	←		←	ALERTING
	183 Session Progress(CPG)	←		←	PROGRESS(PI)
	Inband Info				
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

TP503016	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.6
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection criteria	PICS 1/6				
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message sends out a SETUP message, receives a CALL PROCEEDING, receives a PROGRESS message with a progress indicator where the progress description value is set to PI_VALUE: • no message is sent.				
SIP parameter values	183 Session Progress with encapsulated ACM: called party status indicator=no indication 183 Session Progress with encapsulated CPG event indicator=Progress, ATP with Progress indicator				
ISDN parameter values					
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROCEEDING
	183 Session Progress(CPG)			←	PROGRESS(PI)
	Inband Info				
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

A.1.1.1.1.4 Receipt of the CONNECT Message

TP504001	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], 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, (see ITU-T Recommendation Q.1902.4 [**Error! Reference source not found.**]) is successfully completed. In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "*notification not required*", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312 [5]) that sufficient preconditions have been met for the session to proceed.				
SIP parameter values					
ISDN parameter values					
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	180 Ringing(ACM)	←		←	ALERTING
	200 OK INVITE(ANM)	←		←	CONNECT
	ACK	→			
		Conversation			
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

TP504002	**SIP reference: RFC 3261 [4]**	**ISDN reference: Q.1912.5 [1], 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 a ALERTING message, on receipt of a CONNECT message sends a 200 OK INVITE to the previous entity.	
SIP parameter values		
ISDN parameter values		
Comments		

SIP-I		**SUT**		**ISDN**
INVITE(IAM)	→			
183 session Progress	←			
PRACK	→			
200 OK PRACK	←			
UPDATE	→		→	SETUP
200 OK UPDATE	←			
180 Ringing(ACM)	←		←	ALERTING
200 OK INVITE(ANM)	←		←	CONNECT
ACK	→			
		Conversation		
BYE(REL)	→		→	DISCONNECT
200 OK BYE(RLC)	←		←	RELEASE
			→	RELEASE COMPLETE

TP504003	**SIP reference: RFC 3261 [4]**	**ISDN reference: Q.1912.5 [1], clause 6.7**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of CONNECT_message	
SIP selection criteria		
ISDN selection criteria		
Test purpose	SDP offer was not received in the initial INVITE. Ensure that the SUT, having received the ALERTING message, on receipt of an CONNECT message: • sends a 200 OK INVITE to the UAC. The 200 OK INVITE shall include an SDP offer consistent with the BC used.	
SIP parameter values	200 OK SDP offer ACK SDP answer	
ISDN parameter values		
Comments		

SIP-I		**SUT**		**ISDN**
INVITE(IAM)	→		→	SETUP
180 Ringing(ACM)	←		←	ALERTING
200 OK INVITE(ANM; SDP1)	←		←	CONNECT
ACK(SDP2)	→			
		Conversation		
BYE(REL)	→		→	DISCONNECT
200 OK BYE(RLC)	←		←	RELEASE
			→	RELEASE COMPLETE

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

A.1.1.1.1.5 Initiation of the release procedure from the ISDN side

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

<table>
<tr><td>TP505002</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISDN reference:
Q.1912.5 [1], clause 6.11.2</td></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, on receipt of a RELEASE with the Cause value CV_ISDN, location LOC_ISDN:
• the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN REL_COMP is returned to the ISDN side;
• the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">SIP Statue-Code: SIP_FAILURE_VA (PIXIT)</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">RELEASE; cause value: CV_ISDN (PIXIT)</td></tr>
<tr><td rowspan="4">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>SIP_FAILURE_VA(REL)</td><td>←</td><td></td><td>←</td><td>RELEASE</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>RELEASE COMPLETE</td></tr>
</table>

<table>
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Q.1912.5 [1], clause 6.11.2</td></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">SIP Statue-Code: SIP_FAILURE_VA (PIXIT)</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">DISC; cause value: CV_ISDN (PIXIT)</td></tr>
<tr><td rowspan="5">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>SIP_FAILURE_VA(REL)</td><td>←</td><td></td><td>←</td><td>DISCONNECT</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>RELEASE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>RELEASE COMPLETE</td></tr>
</table>

<table>
<tr><td>TP505004</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISDN reference:
Q.1912.5 [1], clause 6.11.2</td></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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 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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">SIP Statue-Code: SIP_FAILURE_VA (PIXIT)</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">REL_COMP: cause value: CV_ISDN (PIXIT)</td></tr>
<tr><td rowspan="5">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>SETUP ACK</td></tr>
<tr><td>SIP_FAILURE_VA(REL)</td><td>←</td><td></td><td>←</td><td>RELEASE COMPLETE</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><td>TP505005</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISDN reference:
Q.1912.5 [1], clause 6.11.2</td></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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 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, a REL_COMP is returned to the ISDN side;
• the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">SIP Statue-Code: SIP_FAILURE_VA (PIXIT)</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">RELEASE; cause value: CV_ISDN (PIXIT)</td></tr>
<tr><td rowspan="5">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>SETUP ACK</td></tr>
<tr><td>SIP_FAILURE_VA(REL)</td><td>←</td><td></td><td>←</td><td>RELEASE</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>RELEASE COMPLETE</td></tr>
</table>

TP505006	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection 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 , and on receipt of a DISCONNECT message with the **Cause value** CV_ISDN, **location** LOC_ISDN: • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE message is returned to the ISDN side; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.				
SIP parameter values	SIP Statue-Code: SIP_FAILURE_VA (PIXIT				
ISDN parameter values	DISC: **cause value**: CV_ISDN (PIXIT)				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
				←	SETUP ACK
	SIP_FAILURE_VA(REL)	←		←	DISCONNECT
	ACK	→		→	RELEASE
				←	RELEASE COMPLETE

TP505007	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a SETUP ACKNOWLEDGE message, on receipt of a re-INVITE message sends an INFORMATION message and on receipt of a REL EASE COMPLETE message with the **Cause value** CV_ISDN, **location** LOC_ISDN • the SUT immediately requests the disconnection of the internal bearer path. • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA				
SIP parameter values	SIP Statue-Code: SIP_FAILURE_VA (PIXIT)				
ISDN parameter values	REL_COMP: **cause value**: CV_ISDN (PIXIT)				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
				←	SETUP ACK
	INVITE(IAM)	→		→	INFO
	SIP_FAILURE_VA(REL)	←		←	RELEASE COMPLETE
	ACK	→			

<table>
<tr><th>TP505008</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN reference:
Q.1912.5 [1], clause 6.11.2</th></tr>
<tr><td>TSS reference</td><td colspan="7">SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="7">NOT PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="7"></td></tr>
<tr><td>Test purpose</td><td colspan="7">Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a SETUP ACKNOWLEDGE message, on receipt of a re-INVITE message sends an INFORMATION message and on receipt of a REL EASE message with the Cause value CV_ISDN, location LOC_ISDN:
• the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side;
• the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.</td></tr>
<tr><td>SIP parameter values</td><td colspan="7">SIP Statue-Code: SIP_FAILURE_VA (PIXIT)</td></tr>
<tr><td>ISDN parameter values</td><td colspan="7">RELEASE; cause value: CV_ISDN (PIXIT)</td></tr>
<tr><td rowspan="6">Comments</td><td>SIP-I</td><td></td><td colspan="3">SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td colspan="3"></td><td>→</td><td>SETUP</td></tr>
<tr><td></td><td></td><td colspan="3"></td><td>←</td><td>SETUP ACK</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td colspan="3"></td><td>→</td><td>INFO</td></tr>
<tr><td>SIP_FAILURE_VA(REL)</td><td>←</td><td colspan="3"></td><td>←</td><td>RELEASE</td></tr>
<tr><td>ACK</td><td>→</td><td colspan="3"></td><td>→</td><td>RELEASE COMPLETE</td></tr>
</table>

<table>
<tr><th>TP505009</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN reference:
Q.1912.5 [1], clause 6.11.2</th></tr>
<tr><td>TSS reference</td><td colspan="7">SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="7">NOT PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="7"></td></tr>
<tr><td>Test purpose</td><td colspan="7">Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a SETUP ACKNOWLEDGE message, on receipt of a re-INVITE message sends an INFORMATION message and on receipt of a DISCONNECT message with the Cause value CV_ISDN, location LOC_ISDN:
• the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE message is returned to the ISDN side;
• the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.</td></tr>
<tr><td>SIP parameter values</td><td colspan="7">SIP Statue-Code: SIP_FAILURE_VA (PIXIT</td></tr>
<tr><td>ISDN parameter values</td><td colspan="7">DISC: cause value: CV_ISDN (PIXIT)</td></tr>
<tr><td rowspan="7">Comments</td><td>SIP-I</td><td></td><td colspan="3">SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td colspan="3"></td><td>→</td><td>SETUP</td></tr>
<tr><td></td><td></td><td colspan="3"></td><td>←</td><td>SETUP ACK</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td colspan="3"></td><td>→</td><td>INFO</td></tr>
<tr><td>SIP_FAILURE_VA(REL)</td><td>←</td><td colspan="3"></td><td>←</td><td>DISCONNECT</td></tr>
<tr><td>ACK</td><td>→</td><td colspan="3"></td><td>→</td><td>RELEASE</td></tr>
<tr><td></td><td></td><td colspan="3"></td><td>←</td><td>RELEASE COMPLETE</td></tr>
</table>

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) else 480 Temporarily unavailable	Cause Value in the Class 010 (No circuit/channel available, Cause Value No. 34)
VA_20	500 Server Internal Error	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38 to 47) (47 is class default)
VA_21	500 Server Internal Error	Cause Value No. 50 ("requested facility not subscribed")
VA_22	500 Server Internal Error (SIP-I only)	Cause Value No. 55 ("incoming calls barred within CUG")
VA_23	500 Server Internal Error	Cause Value No. 57 ("bearer capability not authorized")
VA_24	500 Server Internal Error	Cause Value No. 58 ("bearer capability not presently")
VA_25	500 Server Internal Error	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_26	500 Server Internal Error	Cause Value in the Class 100 (service or option not implemented Cause Value No. 65 to 79) (79 is class default)
VA_27	500 Server Internal Error	Cause Value No. 87 ("user not member of CUG")
VA_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)

<table>
<tr><td>TP505010</td><td colspan="4">SIP reference: RFC 3261 [4]</td><td colspan="2">ISDN reference:
Q.1912.5 [1], clause 6.11.2</td></tr>
<tr><td>TSS reference</td><td colspan="6">SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6">NOT PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, and receives a CALL PROCEEDING message, on receipt of a RELEASE COMPLETE message with the Cause value CV_ISDN, location LOC_ISDN:
• the SUT immediately requests the disconnection of the internal bearer path;
• the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">183 Session Progress encapsulated ACM: BCi Called party status = no indication
SIP Statue-Code: SIP_FAILURE_VA (PIXIT)</td></tr>
<tr><td>ISDN parameter values</td><td colspan="6">REL_COMP: cause value: CV_ISDN (PIXIT)</td></tr>
<tr><td rowspan="5">Comments</td><td>SIP-I</td><td></td><td colspan="2">SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td colspan="2"></td><td>→</td><td>SETUP</td></tr>
<tr><td>183 Session Progress(ACM)</td><td>←</td><td colspan="2"></td><td>←</td><td>CALL PROC</td></tr>
<tr><td>SIP_FAILURE_VA(REL)</td><td>←</td><td colspan="2"></td><td>←</td><td>RELEASE COMPLETE</td></tr>
<tr><td>ACK</td><td>→</td><td colspan="2"></td><td></td><td></td></tr>
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Q.1912.5 [1], clause 6.11.2</td></tr>
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<tr><td>SIP selection criteria</td><td colspan="6">NOT PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, and receives a CALL PROCEEDING message, on receipt of a RELEASE message with the Cause value CV_ISDN, location LOC_ISDN:
• the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side;
• the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">183 Session Progress encapsulated ACM: BCi Called party status = no indication
SIP Statue-Code: SIP_FAILURE_VA (PIXIT)</td></tr>
<tr><td>ISDN parameter values</td><td colspan="6">RELEASE; cause value: CV_ISDN (PIXIT)</td></tr>
<tr><td rowspan="5">Comments</td><td>SIP-I</td><td></td><td colspan="2">SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td colspan="2"></td><td>→</td><td>SETUP</td></tr>
<tr><td>183 Session Progress(ACM)</td><td>←</td><td colspan="2"></td><td>←</td><td>CALL PROC</td></tr>
<tr><td>SIP_FAILURE_VA(REL)</td><td>←</td><td colspan="2"></td><td>←</td><td>RELEASE</td></tr>
<tr><td>ACK</td><td>→</td><td colspan="2"></td><td>→</td><td>RELEASE COMPLETE</td></tr>
</table>

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

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

<table>
<tr><th>TP505014</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN reference:
Q.1912.5 [1], clause 6.11.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">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)</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">RELEASE; cause value: CV_ISDN (PIXIT)</td></tr>
<tr><td rowspan="6">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>183 Session Progress(ACM)</td><td>←</td><td></td><td>←</td><td>CALL PROC</td></tr>
<tr><td>183 Session Progress(CPG)</td><td>←</td><td></td><td>←</td><td>PROGRESS(PI)</td></tr>
<tr><td>SIP_FAILURE_VA(REL)</td><td>←</td><td></td><td>←</td><td>RELEASE</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>RELEASE COMPLETE</td></tr>
</table>

<table>
<tr><th>TP505015</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN reference:
Q.1912.5 [1], clause 6.11.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">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)</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">DISC; cause value: CV_ISDN (PIXIT)</td></tr>
<tr><td rowspan="7">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>183 Session Progress(ACM)</td><td>←</td><td></td><td>←</td><td>CALL PROC</td></tr>
<tr><td>183 Session Progress(CPG)</td><td>←</td><td></td><td>←</td><td>PROGRESS(PI)</td></tr>
<tr><td>SIP_FAILURE_VA(REL)</td><td>←</td><td></td><td>←</td><td>DISCONNECT</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>RELEASE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>RELEASE COMPLETE</td></tr>
</table>

Table 21

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

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

<table>
<tr><th>TP505017</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN reference:
Q.1912.5 [1], clause 6.11.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">SIP Statue-Code: SIP_FAILURE_VA (PIXIT)</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">RELEASE; cause value: CV_ISDN (PIXIT)</td></tr>
<tr><td rowspan="6">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>SETUP ACK</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ALERTING</td></tr>
<tr><td>SIP_FAILURE_VA(REL)</td><td>←</td><td></td><td>←</td><td>RELEASE</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>RELEASE COMPLETE</td></tr>
</table>

<table>
<tr><th>TP505018</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN reference:
Q.1912.5 [1], clause 6.11.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">SIP Statue-Code: SIP_FAILURE_VA (PIXIT</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">DISC: cause value: CV_ISDN (PIXIT)</td></tr>
<tr><td rowspan="7">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>SETUP ACK</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ALERTING</td></tr>
<tr><td>SIP_FAILURE_VA(REL)</td><td>←</td><td></td><td>←</td><td>DISCONNECT</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>RELEASE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>RELEASE COMPLETE</td></tr>
</table>

TP505019	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message followed by a ALERTING message, having sent a 180 Ringing message on receipt of an ISDN RELEASE 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 encapsulated ACM: BCi Called party status = no indication SIP Statue-Code: SIP_FAILURE_VA (PIXIT)				
ISDN parameter values	REL_COMP: **cause value**: CV_ISDN (PIXIT)				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROC
	180 Ringing(CPG)	←		←	ALERTING
	SIP_FAILURE_VA(REL)	←		←	RELEASE COMPLETE
	ACK	→			

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

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

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

TP505023	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message, receives a PROGRESS message with a **progress indicator** where the progress description value is set to PI_VALUE, on receipt of a ALERTING Message, having sent a 180 Ringing message on receipt of an RELEASE message with the **Cause value** CV_ISDN, **location** LOC_ISDN: • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.				
SIP parameter values	183 Session Progress 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	RELEASE; **cause value**: CV_ISDN (PIXIT)				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROC
	183 Session Progress(CPG)	←		←	PROGRESS(PI)
	180 Ringing(CPG)	←		←	ALERTING
	SIP_FAILURE_VA(REL)	←		←	RELEASE
	ACK	→		→	RELEASE COMPLETE

TP505024	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message, receives a PROGRESS message with a **progress indicator** where the progress description value is set to PI_VALUE, on receipt of a ALERTING Message, having sent a 180 Ringing message on receipt of an DISCONNECT message with the **Cause value** CV_ISDN, **location** LOC_ISDN: • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.				
SIP parameter values	183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with Progress indicator SIP Statue-Code: SIP_FAILURE_VA (PIXIT)				
ISDN parameter values	DISC; **cause value**: CV_ISDN (PIXIT)				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROC
	183 Session Progress(CPG)	←		←	PROGRESS(PI)
	180 Ringing(CPG)	←		←	ALERTING
	SIP_FAILURE_VA(REL)	←		←	DISCONNECT
	ACK	→		→	RELEASE
				←	RELEASE COMPLETE

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

TP505026	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message, 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.				
SIP parameter values	183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with Progress indicator SIP Statue-Code: SIP_FAILURE_VA (PIXIT)				
ISDN parameter values	RELEASE; **cause value**: CV_ISDN (PIXIT)				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROC
	180 Ringing(CPG)	←		←	ALERTING
	183 Session Progress(CPG)	←		←	PROGRESS(PI)
	SIP_FAILURE_VA(REL)	←		←	RELEASE
	ACK	→		→	RELEASE COMPLETE

TP505027	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message, on receipt of a ALERTING Message, having sent a 180 Ringing message, receives a PROGRESS message with a **progress indicator** where the progress description value is set to PI_VALUE, on receipt of an DISCONNECT message with the **Cause value** CV_ISDN, **location** LOC_ISDN: • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.				
SIP parameter values	183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with Progress indicator SIP Statue-Code: SIP_FAILURE_VA (PIXIT)				
ISDN parameter values	DISC; **cause value**: CV_ISDN (PIXIT)				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROC
	180 Ringing(CPG)	←		←	ALERTING
	183 Session Progress(CPG)	←		←	PROGRESS(PI)
	SIP_FAILURE_VA(REL)	←		←	DISCONNECT
	ACK	→		→	RELEASE
				←	RELEASE COMPLETE

Table 22

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

TP505028	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	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 **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.				
SIP parameter values	183 Session Progress encapsulated ACM: BCi Called party status = no indication				
ISDN parameter values	REL_COMP: **cause value**: CV_ISDN (PIXIT)				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROC
	200 OK INVITE(ANM)	←		←	CONNECT
			Communication		
	BYE(REL)	←		←	RELEASE COMPLETE
	200 OK BYE(RLC)	→			

TP505029	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a CALL PROCEEDING message, 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.				
SIP parameter values	183 Session Progress encapsulated ACM: BCi Called party status = no indication				
ISDN parameter values	RELEASE; **cause value**: CV_ISDN (PIXIT)				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROC
	200 OK INVITE(ANM)	←		←	CONNECT
			Communication		
	BYE(REL)	←		←	RELEASE
	200 OK BYE(RLC)	→		→	RELEASE COMPLETE

TP505030	**SIP reference: RFC 3261 [4]**	**ISDN reference: Q.1912.5 [1], clause 6.11.2**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE	
SIP selection criteria	NOT PICS 4/10	
ISDN selection criteria		
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a CALL PROCEEDING message, 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 encapsulated ACM: BCi Called party status = no indication	
ISDN parameter values	DISC; **cause value**: CV_ISDN (PIXIT)	
Comments		

SIP-I		**SUT**		**ISDN**
INVITE(IAM)	→		→	SETUP
183 Session Progress(ACM)	←		←	CALL PROC
200 OK INVITE(ANM)	←		←	CONNECT
		Communication		
BYE(REL)	←		←	DISCONNECT
200 OK BYE(RLC)	→		→	RELEASE
			←	RELEASE COMPLETE

TP505031	**SIP reference: RFC 3261 [4]**	**ISDN reference: Q.1912.5 [1], clause 6.11.2**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE	
SIP selection criteria	NOT PICS 4/10	
ISDN selection criteria		
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sending out a SETUP message, 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.	
SIP parameter values		
ISDN parameter values	REL_COMP: **cause value**: CV_ISDN (PIXIT)	
Comments		

SIP-I		**SUT**		**ISDN**
INVITE(IAM)	→		→	SETUP
200 OK INVITE(ANM)	←		←	CONNECT
		Communication		
BYE(REL)	←		←	RELEASE COMPLETE
200 OK BYE(RLC)	→			

TP505032	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
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 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.				
SIP parameter values					
ISDN parameter values	RELEASE; **cause value**: CV_ISDN (PIXIT)				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	200 OK INVITE(ANM)	←		←	CONNECT
			Communication		
	BYE(REL)	←		←	RELEASE
	200 OK BYE(RLC)	→		→	RELEASE COMPLETE

TP505033	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out an SETUP message, receives a CONNECT message, a 200 OK message is sent, on receipt of a DISCONNECT message with the **Cause value** CV_ISDN, **location** LOC_ISDN: • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE message is returned to the ISDN side; • the SUT shall send a BYE message.				
SIP parameter values					
ISDN parameter values	DISC; **cause value**: CV_ISDN (PIXIT)				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	200 OK INVITE(ANM)	←		←	CONNECT
			Communication		
	BYE(REL)	←		←	DISCONNECT
	200 OK BYE(RLC)	→		→	RELEASE
				←	RELEASE COMPLETE

TP505034	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, via a broadcast data link, after time-out of **T303**: • the SUT shall send a 480 Temporarily unavailable final response.				
SIP parameter values	480 Temporarily unavailable: Encapsulated REL with cause 18				
ISDN parameter values					
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
				→	SETUP
			T303 expiry		
	480 Temporarily unavailable(REL)	←			
	ACK	→			

Table 23

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

TP505035	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends 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**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	SIP_FAILURE_VA(REL)	←		←	RELEASE COMPLETE
	ACK	→			

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

Table 24

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

Values for test purposes TP108036, TP108037		
←SIP Message SIP_FAILURE_VA CV_SIP		← REL Cause Indicators parameter CV_ISDN
VA_18	480 Temporarily unavailable Cause Value No. 31	Cause Value No. 31 ("normal unspecified") (Class default)
VA_19	486 Busy here if Diagnostics indicator includes the (CCBS indicator = CCBS possible) else 480 Temporarily unavailable Cause Value No. 34	Cause Value in the Class 010 (resource unavailable, Cause Value No. 34)
VA_20	500 Server Internal Error Cause Value No. 47	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38 to 47) (47 is class default)
VA_21	500 Server Internal Error Cause Value No. 50	Cause Value No. 50 ("requested facility not subscribed")
VA_22	500 Server Internal Error Cause Value No. 55	Cause Value No. 55 ("incoming calls barred within CUG")
VA_23	500 Server Internal Error Cause Value No. 57	Cause Value No. 57 ("bearer capability not authorized")
VA_24	500 Server Internal Error Cause Value No. 58	Cause Value No. 58 ("bearer capability not presently")
VA_25	500 Server Internal Error Cause Value No. 63	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_26	500 Server Internal Error Cause Value No. 65 to 79	Cause Value in the Class 100 (service or option not implemented Cause Value No. 65 to 79) (79 is class default)
VA_27	500 Server Internal Error Cause Value No. 87	Cause Value No. 87 ("user not member of CUG")
VA_28	500 Server Internal Error Cause Value No. 88	Cause Value No. 88 ("incompatible destination")
VA_29	500 Server Internal Error Cause Value No. 90	Cause Value No. 90 ("Non-existent CUG")
VA_30	404 Not Found Cause Value No. 91	Cause Value No. 91 ("invalid transit network selection")
VA_31	500 Server Internal Error Cause Value No. 95	Cause Value No. 95 ("invalid message") (Class default)
VA_32	500 Server Internal Error Cause Value No. 97	Cause Value No. 97 ("Message type non-existent or not implemented")
VA_33	500 Server Internal Error Cause Value No. 99	Cause Value No. 99 ("information element/parameter non-existent or not implemented")
VA_34	480 Temporarily unavailable Cause Value No. 102	Cause Value No. 102 ("recovery on timer expiry")
VA_35	500 Server Internal Error Cause Value No. 103	Cause Value No. 103 ("Parameter non-existent or not implemented, pass on")
VA_36	500 Server Internal Error Cause Value No. 110	Cause Value No. 110 ("Message with unrecognized Parameter, discarded")
VA_37	500 Server Internal Error Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)
VA_38	480 Temporarily unavailable Cause Value No. 127	Cause Value No. 127 ("interworking unspecified") (Class default)

TP505037	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, and receives a CALL PROCEEDING message, on receipt of a RELEASE COMPLETE message with the **Cause value** CV_ISDN, **location** LOC_ISDN: • the SUT immediately requests the disconnection of the internal bearer path; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA; • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.				
SIP parameter values	183 Session Progress encapsulated ACM: BCi Called party status = no indication 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**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROC
	SIP_FAILURE_VA(REL)	←		←	RELEASE COMPLETE
	ACK	→			

TP505038	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection 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 message with the **Cause value** CV_ISDN, **location** LOC_ISDN: • the SUT immediately requests the disconnection of the internal bearer path When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA; • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.				
SIP parameter values	183 Session Progress encapsulated ACM: BCi Called party status = no indication SIP Statue-Code: SIP_FAILURE_VA (PIXIT), **Reason header value**: CV_SIP (PIXIT)				
ISDN parameter values	RELEASE; **cause value**: CV_ISDN (PIXIT)				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROC
	SIP_FAILURE_VA(REL)	←		←	RELEASE
	ACK	→		→	RELEASE COMPLETE

TP505039	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, 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 encapsulated ACM: BCi Called party status = no indication SIP Statue-Code: SIP_FAILURE_VA (PIXIT), **Reason header value**: CV_SIP (PIXIT)				
ISDN parameter values	DISC; **cause value**: CV_ISDN (PIXIT)				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROC
	SIP_FAILURE_VA(REL)	←		←	DISCONNECT
	ACK	→		→	RELEASE
				←	RELEASE COMPLETE

TP505040	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message followed by a PROGRESS message with a **progress indicator** PI_VALUE, on receipt of a RELEASE COMPLETE message with the **Cause value** CV_ISDN, **location** LOC_ISDN: • the SUT immediately requests the disconnection of the internal bearer path; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA; • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.				
SIP parameter values	183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with Progress indicator SIP Statue-Code: SIP_FAILURE_VA (PIXIT), **Reason header value**: CV_SIP (PIXIT)				
ISDN parameter values	REL_COMP: **cause value**: CV_ISDN (PIXIT)				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROC
	183 Session Progress(CPG)	←		←	PROGRESS(PI)
	SIP_FAILURE_VA(REL)	←		←	RELEASE COMPLETE
	ACK	→			

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

TP505042	**SIP reference: RFC 3261 [4]**			**ISDN reference: Q.1912.5 [1], clause 6.11.2**	
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message followed by a PROGRESS message with a **progress indicator** PI_VALUE, on receipt of a DISCONNECT message with the **Cause value** CV_ISDN, **location** LOC_ISDN: • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE message is returned to the ISDN side; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA; • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.				
SIP parameter values	183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with Progress indicator SIP Statue-Code: SIP_FAILURE_VA (PIXIT), **Reason header value**: CV_SIP (PIXIT)				
ISDN parameter values	DISC; **cause value**: CV_ISDN (PIXIT)				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROC
	183 Session Progress(CPG)	←		←	PROGRESS(PI)
	SIP_FAILURE_VA(REL)	←		←	DISCONNECT
	ACK	→		→	RELEASE
				←	RELEASE COMPLETE

Table 25

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

<table>
<tr><th>TP505043</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN reference:
Q.1912.5 [1], clause 6.11.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">SIP Statue-Code: SIP_FAILURE_VA (PIXIT), Reason header value: CV_SIP (PIXIT)</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">REL_COMP: cause value: CV_ISDN (PIXIT)</td></tr>
<tr><td rowspan="6">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>SETUP ACK</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ALERTING</td></tr>
<tr><td>SIP_FAILURE_VA(REL)</td><td>←</td><td></td><td>←</td><td>RELEASE COMPLETE</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
</table>

TP505044	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends a SETUP message, receives a 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 (PIXIT), **Reason header value**: CV_SIP (PIXIT)				
ISDN parameter values	RELEASE; **cause value**: CV_ISDN (PIXIT)				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
				←	SETUP ACK
	180 Ringing(ACM)	←		←	ALERTING
	SIP_FAILURE_VA(REL)	←		←	RELEASE
	ACK	→		→	RELEASE COMPLETE

TP505045	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends a SETUP message, receives a 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 values	SIP Statue-Code: SIP_FAILURE_VA (PIXIT), **Reason header value**: CV_SIP (PIXIT)				
ISDN parameter values	DISC: **cause value**: CV_ISDN (PIXIT)				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
				←	SETUP ACK
	180 Ringing(ACM)	←		←	ALERTING
	SIP_FAILURE_VA(REL)	←		←	DISCONNECT
	ACK	→		→	RELEASE
				←	RELEASE COMPLETE

Table 26

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

<table>
<tr><td>TP505046</td><td colspan="4">SIP reference: RFC 3261 [4]</td><td colspan="2">ISDN reference:
Q.1912.5 [1], clause 6.11.2</td></tr>
<tr><td>TSS reference</td><td colspan="6">SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6">PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">183 Session Progress encapsulated ACM: BCi Called party status = no indication
SIP Statue-Code: SIP_FAILURE_VA (PIXIT), Reason header value: CV_SIP (PIXIT)</td></tr>
<tr><td>ISDN parameter values</td><td colspan="6">REL_COMP: cause value: CV_ISDN (PIXIT)</td></tr>
<tr><td rowspan="7">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td colspan="2">ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td colspan="2">SETUP</td></tr>
<tr><td>183 Session Progress(ACM)</td><td>←</td><td></td><td>←</td><td colspan="2">CALL PROC</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td colspan="2">CONNECT</td></tr>
<tr><td></td><td colspan="3">Communication</td><td colspan="2"></td></tr>
<tr><td>BYE(REL)</td><td>←</td><td></td><td>←</td><td colspan="2">RELEASE COMPLETE</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
</table>

<table>
<tr><th>TP505047</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN reference:
Q.1912.5 [1], clause 6.11.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">183 Session Progress encapsulated ACM: BCi Called party status = no indication
SIP Statue-Code: SIP_FAILURE_VA (PIXIT), Reason header value: CV_SIP (PIXIT)</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">RELEASE: cause value: CV_ISDN (PIXIT)</td></tr>
<tr><td rowspan="7">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>183 Session Progress(ACM)</td><td>←</td><td></td><td>←</td><td>CALL PROC</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>CONNECT</td></tr>
<tr><td></td><td colspan="3">Communication</td><td></td></tr>
<tr><td>BYE(REL)</td><td>←</td><td></td><td>←</td><td>RELEASE</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>→</td><td></td><td>→</td><td>RELEASE COMPLETE</td></tr>
</table>

<table>
<tr><th>TP505048</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN reference:
Q.1912.5 [1], clause 6.11.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message, receives a CONNECT message, a 200 OK message is sent, on receipt of a DISCONNECT message with the Cause value CV_ISDN, location LOC_ISDN:
• the SUT immediately requests the disconnection of the internal bearer path When the bearer channel is available for re-selection, an ISDN RELEASE is returned to the ISDN side;
• the SUT shall send a BYE message;
• the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">183 Session Progress encapsulated ACM: BCi Called party status = no indication
SIP Statue-Code: SIP_FAILURE_VA (PIXIT), Reason header value: CV_SIP (PIXIT)</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">DISC: cause value: CV_ISDN (PIXIT)</td></tr>
<tr><td rowspan="8">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>183 Session Progress(ACM)</td><td>←</td><td></td><td>←</td><td>CALL PROC</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>CONNECT</td></tr>
<tr><td></td><td colspan="3">Communication</td><td></td></tr>
<tr><td>BYE(REL)</td><td>←</td><td></td><td>←</td><td>DISCONNECT</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>→</td><td></td><td>→</td><td>RELEASE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>RELEASE COMPLETE</td></tr>
</table>

<table>
<tr><th>TP505049</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN reference:
Q.1912.5 [1], clause 6.11.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">SIP Statue-Code: SIP_FAILURE_VA (PIXIT), Reason header value: CV_SIP (PIXIT)</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">REL_COMP: cause value: CV_ISDN (PIXIT)</td></tr>
<tr><td rowspan="6">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>CONNECT</td></tr>
<tr><td></td><td colspan="3">Communication</td><td></td></tr>
<tr><td>BYE(REL)</td><td>←</td><td></td><td>←</td><td>RELEASE COMPLETE</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>→</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><th>TP505050</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN reference:
Q.1912.5 [1], clause 6.11.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CONNECT message, a 200 OK message is sent, on receipt of a RELEASE message with the Cause value CV_ISDN, location LOC_ISDN:
• the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RLC is returned to the ISDN side;
• the SUT shall send a BYE message;
• the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">SIP Statue-Code: SIP_FAILURE_VA (PIXIT), Reason header value: CV_SIP (PIXIT)</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">RELEASE: cause value: CV_ISDN (PIXIT)</td></tr>
<tr><td rowspan="6">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>CONNECT</td></tr>
<tr><td></td><td colspan="3">Communication</td><td></td></tr>
<tr><td>BYE(REL)</td><td>←</td><td></td><td>←</td><td>RELEASE</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>→</td><td></td><td>→</td><td>RELEASE COMPLETE</td></tr>
</table>

TP505051	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CONNECT message, a 200 OK message is sent, on receipt of a DISCONNECT message with the **Cause value** CV_ISDN, **location** LOC_ISDN: • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE is returned to the ISDN side; • the SUT shall send a BYE message; • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.				
SIP parameter values	SIP Statue-Code: SIP_FAILURE_VA (PIXIT), **Reason header value**: CV_SIP (PIXIT)				
ISDN parameter values	DISC: **cause value**: CV_ISDN (PIXIT)				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	200 OK INVITE(ANM)	←		←	CONNECT
		Communication			
	BYE(REL)	←		←	DISCONNECT
	200 OK BYE(RLC)	→		→	RELEASE
				←	RELEASE COMPLETE

Table 27

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

A.1.1.1.1.6 Receipt of BYE / CANCEL messages

TP506001	**SIP reference: RFC 3261 [4]**				**ISDN reference: Q.1912.5 [1], clause 6.11.1**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sending out a SETUP message, receives an ALERTING and CONNECT message. On receipt of SIP **BYE**, the SUT shall send an ISDN DISCONNECT with the cause and location mapped from the encapsulated REL in the received BYE to the ISDN side.				
SIP parameter values					
ISDN parameter values	DISC: Cause value and location mapped from the encapsulated REL in the received BYE				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	180 Ringing(ACM)	←		←	ALERTING
	200 OK INVITE(ANM)	←		←	CONNECT
	ACK	→			
			Conversation		
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

TP506002	**SIP reference: RFC 3261 [4]**				**ISDN reference: Q.1912.5 [1], clause 6.11.1**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out a SETUP message, receives an ALERTING message, the SUT on receipt of SIP **CANCEL**, the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received CANCEL to the ISDN side.				
SIP parameter values					
ISDN parameter values	DISC: Cause value and location mapped from the encapsulated REL in the received CANCEL				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	180 Ringing(ACM)	←		←	ALERTING
	CANCEL(REL)	→		→	DISCONNECT
	200 OK CANCEL	←		←	RELEASE
	487 Request Terminated	←		→	RELEASE COMPLETE
	ACK	→			

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

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

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

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

TP506007	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.1
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message followed by a PROGRESS message, on receipt of SIP **CANCEL**, the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received CANCEL to the ISDN side.				
SIP parameter values	183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with Progress indicator				
ISDN parameter values	DISC: Cause value and location mapped from the encapsulated REL in the received CANCEL				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROC
	183 Session Progress(CPG)	←		←	PROGRESS(PI)
	CANCEL(REL)	→		→	DISCONNECT
	200 OK CANCEL	←		←	RELEASE
	487 Request Terminated	←		→	RELEASE COMPLETE
	ACK	→			

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

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

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

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

TP506012	**SIP reference: RFC 3261 [4]**				**ISDN reference: Q.1912.5 [1], clause 6.11.1**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message, on receipt of SIP **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 mapped from the encapsulated REL in the received BYE				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROC
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
	487 Request Terminated	←		→	RELEASE COMPLETE
	ACK	→			

TP506013	**SIP reference: RFC 3261 [4]**				**ISDN reference: Q.1912.5 [1], clause 6.11.1**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message followed by a PROGRESS message, on receipt of SIP **BYE**, the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received BYE to the ISDN side.				
SIP parameter values	183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with Progress indicator				
ISDN parameter values	DISC: Cause value and location mapped from the encapsulated REL in the received BYE				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROC
	183 Session Progress(CPG)	←		←	PROGRESS(PI)
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
	487 Request Terminated	←		→	RELEASE COMPLETE
	ACK	→			

TP506014	**SIP reference: RFC 3261 [4]**				**ISDN reference: Q.1912.5 [1], clause 6.11.1**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a SETUP ACKNOWLEDGE message followed by a ALERTING message, on receipt of SIP **BYE**, the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received BYE to the ISDN side.				
SIP parameter values					
ISDN parameter values	DISC: Cause value and location mapped from the encapsulated REL in the received BYE				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
				←	SETUP ACK
	180 Ringing(ACM)	←		←	ALERTING
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
	487 Request Terminated	←		→	RELEASE COMPLETE
	ACK	→			

TP506015	**SIP reference: RFC 3261 [4]**				**ISDN reference: Q.1912.5 [1], clause 6.11.1**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message, on receipt of a ALERTING Message, on receipt of SIP **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 mapped from the encapsulated REL in the received BYE				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROC
	180 Ringing(CPG)	←		←	ALERTING
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
	487 Request Terminated	←		→	RELEASE COMPLETE
	ACK	→			

TP506016	**SIP reference: RFC 3261 [4]**				**ISDN reference: Q.1912.5 [1], clause 6.11.1**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message, a ALERTING Message, a PROGRESS message, on receipt of SIP **BYE**, the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received BYE to the ISDN side				
SIP parameter values	183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with Progress indicator				
ISDN parameter values	DISC: Cause value and location mapped from the encapsulated REL in the received BYE				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROC
	180 Ringing(CPG)	←		←	ALERTING
	183 Session Progress(CPG)	←		←	PROGRESS(PI)
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
	487 Request Terminated	←		→	RELEASE COMPLETE
	ACK	→			

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

A.1.1.1.2.1 Sending of the INVITE message

TP601001	**SIP reference: RFC 3261 [4]**			**ISDN/ISDN reference: Q.1912.5 [1], clause 7.1 1 a)**	
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in Idle state, on receipt of a SETUP message containing the complete **called party number** and the **sending complete** indication: • sends the INVITE message.				
SIP parameter values					
ISDN parameter values	SETUP; **Called party number**: with send complete indication				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONNECT	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	DISCONNECT	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE(RLC)
	RELEASE COMPLETE	→			

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

TP601003	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.1 1 c)
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the complete **called party number** where the end of address signalling is determined by analysis of the called party number to indicate that **a sufficient number of digits has been received** to route the call to the called party: • sends the INVITE message.				
SIP parameter values					
ISDN parameter values	SETUP; **Called party number**: sufficient number of digits to route to the called party				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONNECT	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	DISCONNECT	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE(RLC)
	RELEASE COMPLETE	→			

TP601004	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.1 1 d)
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in Idle state, on receipt of a SETUP message containing the complete **called party number** where the end of address signalling is determined by the expiration timer T302 after the receipt of the latest address message: • sends the INVITE message.				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONNECT	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	DISCONNECT	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE(RLC)
	RELEASE COMPLETE	→			

<table>
<tr><th>TP601005</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISDN/ISDN reference:
Q.1912.5 [1], clause 7.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISDN-SIP /Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISDN selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="6">INVITE: a_b_m_LINE_VALUE, IAM encapsulated in a MIME-body</td></tr>
<tr><td rowspan="10">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td colspan="2">SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(IAM)</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing(ACM)</td></tr>
<tr><td>CONNECT</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="2"></td></tr>
<tr><td>DISCONNECT</td><td>→</td><td></td><td>→</td><td colspan="2">BYE(REL)</td></tr>
<tr><td>RELEASE</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK BYE(RLC)</td></tr>
<tr><td>RELEASE COMPLETE</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
</table>

Table 28

Values for test purpose TP301005									
VA	**ISDN**				**SDP - a_b_m_LINE_VALUE**				
		BC parameter		**HLC**	**m= line**			**b= line**	**a= line**
	BC	**Information Transport Capability**	**User Information Layer 1 Protocol Indicator**	**High Layer Characteristics Identification**	**<media>**	**<transport>**	**<fmt-list>**	**<modifier>:<bandwidth-value>**	**rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>**
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)
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
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

TP601006	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter 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 values	INVITE: To: sip: ….; user=phone				
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONNECT	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	DISCONNECT	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE(RLC)
	RELEASE COMPLETE	→			

TP601007	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter of the SETUP and the and the followed INFO: • to the addr-spec component of the **To header field.**				
SIP parameter values	INVITE: To:				
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→			
	INFO	→			
	INFO	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONNECT	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	DISCONNECT	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE(RLC)
	RELEASE COMPLETE	→			

TP601008	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT is mapping in the Called Party Number parameter contained in the Called Party address information of the SETUP and followed INFO: • to the addr-spec component of the **To header field** which shall include the "user=phone" URI parameter if the To header field contains a sip: URI.				
SIP parameter values	INVITE: To: sip: ….; user=phone				
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→			
	INFO	→			
	INFO	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONNECT	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	DISCONNECT	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE(RLC)
	RELEASE COMPLETE	→			

TP601009	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, TON **= "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	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONNECT	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	DISCONNECT	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE(RLC)
	RELEASE COMPLETE	→			

TP601010	**SIP reference: RFC 3261 [4]**				**ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2**
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, TON **= "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	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONNECT	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	DISCONNECT	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE(RLC)
	RELEASE COMPLETE	→			

TP601011	**SIP reference: RFC 3261 [4]**				**ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2**
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, TON **= "unknown"** of the SETUP: • 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	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONNECT	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	DISCONNECT	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE(RLC)
	RELEASE COMPLETE	→			

TP601012	**SIP reference: RFC 3261 [4]**	**ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2**
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria		
ISDN selection criteria		
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, TON **= "International number"** of the SETUP and the and the followed INFO: • to the addr-spec component of the **To header field**; • the format of the To header field is "+CC+NDC+SN"; • the forward address information is derived from the user info component of the INVITE Request-URI.	
SIP parameter values	INVITE: To: …	
ISDN parameter values		
Comments		

ISDN		**SUT**		**SIP-I**
SETUP	→			
INFO	→			
INFO	→		→	INVITE(IAM)
ALERTING	←		←	180 Ringing(ACM)
CONNECT	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
DISCONNECT	→		→	BYE(REL)
RELEASE	←		←	200 OK BYE(RLC)
RELEASE COMPLETE	→			

TP601013	**SIP reference: RFC 3261 [4]**	**ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2**
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria		
ISDN selection criteria		
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, TON **= "National (significant) number"** of the SETUP and the followed INFO: • to the addr-spec component of the **To header field**; • the format of the To header field is "+CC+NDC+SN"; • the forward address information is derived from the user info component of the INVITE Request-URI.	
SIP parameter values	INVITE: To: …	
ISDN parameter values		
Comments		

ISDN		**SUT**		**SIP-I**
SETUP	→			
INFO	→			
INFO	→		→	INVITE(IAM)
ALERTING	←		←	180 Ringing(ACM)
CONNECT	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
DISCONNECT	→		→	BYE(REL)
RELEASE	←		←	200 OK BYE(RLC)
RELEASE COMPLETE	→			

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

A.1.1.1.2.2 Overlap sending

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

<table>
<tr><th>TP602002</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN/ISDN reference:
Q.1912.5 [1], clause 7.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Overlap sending</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 3/2</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in Idle state, on receipt of an SETUP message sends an INVITE message. On receipt of a INFO from the ISDN access the SUT shall:
1) Stop timer TOIW3 (if it is running);
2) TOIW2 shall be restarted and the SUT shall invoke the following procedures:
a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call;
b) A new INVITE with the INFOe Call-ID and From header (including tag) as the previous INVITE is sent;
c) The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question;
d) All other contents of the new INVITE are interworked from the parameters of the original SETUP.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="12">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>INFO</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>INFO</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>CONNECT</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>DISCONNECT</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RELEASE</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
<tr><td>RELEASE COMPLETE</td><td>→</td><td></td><td></td><td></td></tr>
</table>

TP602003	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.2.1
TSS reference	ISDN-SIP /Basic call/Overlap sending				
SIP selection criteria	PICS 3/2				
ISDN selection criteria					
Test purpose	The SUT in Idle state, on receipt of an SETUP message sends a INVITE message On receipt of a INFO from the ISDN access the SUT shall: • TOIW2 shall be restarted and the SUT shall invoke the following procedures: Ensure that if timer TOIW2 has expired, subsequent INFOs received after the SUT has sent the INVITE are ignored.				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	INFO	→		→	INVITE(IAM)
			T_{oiw2} expired		
	INFO	→			
	ALERTING	←		←	180 Ringing(ACM)
	CONNECT	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	DISCONNECT	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE(RLC)
	RELEASE COMPLETE	→			

TP602004	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.2.1
TSS reference	ISDN-SIP /Basic call/Overlap sending				
SIP selection criteria	PICS 3/1				
ISDN selection criteria					
Test purpose	The SUT in Idle state, on receipt of a SETUP message. On receipt of a INFO from the BICC/ISDN the SUT shall: • sends an INVITE message if the minimum number of digits for routing the call has been received in the SETUP and the INFO TOIW1 and TIOW2 shall be started and the SUT shall invoke the following procedures; • ensure that if timer TOIW2 has expired, subsequent INFOs received after the SUT has sent the INVITE are ignored.				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→			
	INFO	→		→	INVITE(IAM)
			T_{oiw2} expired		
	INFO	→			
	ALERTING	←		←	180 Ringing(ACM)
	CONNECT	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	DISCONNECT	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE(RLC)
	RELEASE COMPLETE	→			

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

TP602006	**SIP reference: RFC 3261 [4]**			**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6**	
TSS reference	ISDN-SIP /Basic call/Overlap sending				
SIP selection criteria	NOT PICS 3/2				
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT before having received an backward message, on receipt of a Failure message (4xx, 5xx, 6xx) defined as SIP_Failure_VA: • sends a DISCONNECT or RELEASE message cause value 28.				
SIP parameter values	SIP_Failure_VA: ISUP REL encapsulated in the MIME body				
ISDN parameter values	DISCONNECT/RELEASE: Cause value constructed from the encapsulated REL				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
				←	484 Address Incomplete
	CASE A			→	ACK
	RELEASE	←			
	RELEASE COMPLETE	→			
	CASE B				
	DISCONNECT	←			
	RELEASE	→			
	RELEASE COMPLETE	←			

A.1.1.1.2.3 Sending of the CALL PROCEEDING / ALERTING message

TP603001	**SIP reference: RFC 3261 [4]**			**ISDN/ISDN reference: Q.1912.5 [1], clauses 7.1 1) a), 7.3.1**	
TSS reference	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING				
SIP selection criteria	PICS 3/1				
ISDN selection criteria					
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the complete **called party number** and the **sending complete** indication: • sends the INVITE message to called user; • sends the CALL PROCEDING message; • sends the PROGRESS message, with the with progress description set to PI_VAL.				
SIP parameter values	183 Session Progress with encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with Progress indicator				
ISDN parameter values	CALL PROCEEDING PROGRESS(PI value=PI_VAL)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	CALL PROCEEDING	←		←	183 Session Progress(ACM)
	PROGRESS(PI)	←		←	183 Session Progress(CPG)
	ALERTING	←		←	180 Ringing(ACM
	CONNECT	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	DISCONNECT	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE(RLC)
	RELEASE COMPLETE	→			

<table>
<tr><th>TP603002</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN/ISDN reference:
Q.1912.5 [1], clauses 7.1 1) b), 7.3.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 3/1</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">183 Session Progress with encapsulated ACM: BCi Called party status = no indication, ATP with PI</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">CALL PROCEEDING(PI value=PI_VAL)</td></tr>
<tr><td rowspan="12">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>CALL PROCEEDING</td><td>←</td><td></td><td>←</td><td>183 Session Progress(ACM)</td></tr>
<tr><td></td><td colspan="3"></td><td></td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>CONNECT</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>DISCONNECT</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RELEASE</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
<tr><td>RELEASE COMPLETE</td><td>→</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><th>TP603003</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN/ISDN reference:
Q.1912.5 [1], clauses 7.1, 7.3.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 3/2</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT if overlap addressing is to be used toward the SIP network, on receipt of an SETUP message containing the minimum number of digits required for routing the call has been received (start timer TOIW2 and invoke the appropriate outgoing SIP signalling procedure)
• Sends an INVITE message to the called user
• after the expiration of T_{OIW2} sends the CALL PROCEEDING message</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">CALL PROCEEDING</td></tr>
<tr><td rowspan="13">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>INFO</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td colspan="2">T_{oiw2} expired</td><td></td><td></td></tr>
<tr><td>CALL PROCEEDING</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>CONNECT</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>DISCONNECT</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RELEASE</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
<tr><td>RELEASE COMPLETE</td><td>→</td><td></td><td></td><td></td></tr>
</table>

TP603004	SIP reference: RFC 3261 [4]			ISDN/ISDN reference: Q.1912.5 [1], clauses 7.1 1) a), 7.3.1	
TSS reference	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING				
SIP selection criteria	PICS 3/1				
ISDN selection criteria					
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the complete **called party number** and the **sending complete** indication, on receipt of a 180 Ringing message: • sdends the ALERTING message with the with the with progress description value PI_VAL.				
SIP parameter values	180 Ringing encapsulated ACM: BCi Called party status = subscriber free, ATP with Progress indicator PI_VAL				
ISDN parameter values	ALERTING: Progress indicator value PI_VAL included				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	CALL PROCEEDING	←			
	ALERTING	←		←	180 Ringing(ACM(PI))
	CONNECT	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	DISCONNECT	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE(RLC)
	RELEASE COMPLETE	→			

TP603005	SIP reference: RFC 3261 [4]			ISDN/ISDN reference: Q.1912.5 [1], clause 7.1 1 a)	
TSS reference	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING				
SIP selection criteria	PICS 3/2				
ISDN selection criteria					
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the complete **called party number** where the end of address signalling is determined by the expiration timer T_{OIW2} after the receipt of the latest address message on receipt of a 183 Session Progress with encapsulated ACM: • a PROGRESS is sent backward.				
SIP parameter values	183 Session Progress encapsulated ACM: BCi Called party status = no indication, ATP with Progress indicator				
ISDN parameter values	PROGRESS				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→			
	INFO	→		→	INVITE(IAM)
			T_{oiw2} expired		
	CALL PROCEEDING	←			
	PROGRESS	←		←	183 Session Progress(ACM(PI))
	ALERTING	←		←	180 Ringing(CPG)
	CONNECT	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	DISCONNECT	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE(RLC)
	RELEASE COMPLETE	→			

TP603006	SIP reference: RFC 3261 [4]	ISDN/ISDN reference: Q.1912.5 [1], clause 7.1 1 a)
TSS reference	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING	
SIP selection criteria	PICS 3/2	
ISDN selection criteria		
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the complete **called party number** where the end of address signalling is determined by the expiration timer T_{OIW2} after the receipt of the latest address message on receipt of a 183 Session Progress: • no information is sent backward.	
SIP parameter values	183 Session Progress encapsulated ACM: BCi Called party status = no indication, without ATP	
ISDN parameter values		
Comments		

ISDN		**SUT**		**SIP-I**
SETUP	→			
INFO	→		→	INVITE(IAM)
	T_{oiw2} expired			
			←	183 Session Progress(ACM)
ALERTING	←		←	180 Ringing(ACM
CONNECT	←		←	200 OK INVITE(ANM)
			→	ACK
	Conversation			
DISCONNECT	→		→	BYE(REL)
RELEASE	←		←	200 OK BYE(RLC)
RELEASE COMPLETE	→			

TP603007	SIP reference: RFC 3261 [4]	ISDN/ISDN reference: Q.1912.5 [1], clause 7.1 1 a)
TSS reference	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING	
SIP selection criteria	PICS 3/2	
ISDN selection criteria		
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the complete **called party number** where the end of address signalling is determined by the expiration timer T_{OIW2} after the receipt of the latest address message on receipt of a 183 Session Progress: • no information is sent backward.	
SIP parameter values	183 Session Progress without encapsulated ISUP message	
ISDN parameter values		
Comments		

ISDN		**SUT**		**SIP-I**
SETUP	→			
INFO	→		→	INVITE(IAM)
	T_{oiw2} expired			
			←	183 Session Progress
ALERTING	←		←	180 Ringing(ACM
CONNECT	←		←	200 OK INVITE(ANM)
			→	ACK
	Conversation			
DISCONNECT	→		→	BYE(REL)
RELEASE	←		←	200 OK BYE(RLC)
RELEASE COMPLETE	→			

A.1.1.1.2.4 Sending of the CONNECT message

TP604001	**SIP reference: RFC 3261 [4]**			**ISDN/ISDN reference: Q.1912.5 [1], 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 T_{OIW2} (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	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM
	CONNECT	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	DISCONNECT	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE(RLC)
	RELEASE COMPLETE	→			

TP604002	**SIP reference: RFC 3261 [4]**			**ISDN/ISDN reference: Q.1912.5 [1], 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 T_{OIW2} (if running): • send CON as determined by ISDN procedures; • Sstop any existing awaiting answer indication (e.g. ringing tone).				
SIP parameter values	200 OK INVITE: encapsulated CON in the MIME body				
ISDN parameter values	CONNECT				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	CONNECT	←		←	200 OK INVITE(CON)
				→	ACK
			Conversation		
	DISCONNECT	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE(RLC)
	RELEASE COMPLETE	→			

A.1.1.1.2.5 Receipt of the RELEASE or DISCONNECT

TP605001	**SIP reference: RFC 3261 [4]**			**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1, 1)**
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT			
SIP selection criteria				
ISDN selection criteria				
Test purpose	Ensure that the SUT after receiving the SETUP 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**		**SUT**		**SIP-I**
	SETUP	→			
	RELEASE COMPLETE	→			

TP605002	**SIP reference: RFC 3261 [4]**			**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1, 1)**
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT			
SIP selection criteria				
ISDN selection criteria				
Test purpose	Ensure that the SUT after receiving the SETUP but before an INVITE has been sent. On receipt of a 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	→			
	RELEASE	→			
	RELEASE COMPLETE	←			

TP605003	**SIP reference: RFC 3261 [4]**			**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1, 1)**
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT			
SIP selection criteria				
ISDN selection criteria				
Test purpose	Ensure that the SUT after receiving the SETUP but before an INVITE has been sent. On receipt of a DISCONNECT message: • no action is required on the SIP side other than to terminate local procedures if any are in progress.			
SIP parameter values				
ISDN parameter values	DISCONNECT; **cause value**: (PIXIT), **location** (PIXIT)			
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→			
	DISCONNECT	→			
	RELEASE	←			
	RELEASE COMPLETE	→			

<table>
<tr><th>TP605004</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN/ISDN reference:
Q.1912.5 [1], clause 7.7.1 2)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">RELEASE COMPLETE; cause value: (PIXIT), location (PIXIT)</td></tr>
<tr><td rowspan="7">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>RELEASE COMPLETE</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP605005</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN/ISDN reference:
Q.1912.5 [1], clause 7.7.1 2)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE ISDN message before a 200 OK (any) response message has been received which establishes a confirmed dialogue:
• the SUT shall hold the release procedure until a SIP 200 OK INVITE response has been received;
• the SUT shall send a BYE request.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">RELEASE; cause value: (PIXIT), location (PIXIT)</td></tr>
<tr><td rowspan="8">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>RELEASE</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>RELEASE COMPLETE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP605006</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN/ISDN reference:
Q.1912.5 [1], clause 7.7.1 2)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">DISCONNECT; cause value: (PIXIT), location (PIXIT)</td></tr>
<tr><td rowspan="8">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>DISCONNECT</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>RELEASE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>RELEASE COMPLETE</td><td>→</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP605007</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN/ISDN reference:
Q.1912.5 [1], clause 7.7.1 2) 3)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">RELEASE COMPLETE; cause value: (PIXIT), location (PIXIT)</td></tr>
<tr><td rowspan="10">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>100 Trying</td></tr>
<tr><td>RELEASE COMPLETE</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>CANCEL(REL)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK CANCEL</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP605008	**SIP reference: RFC 3261 [4]**			**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 2) 3)**	
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE 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	RELEASE; **cause value**: (PIXIT), **location** (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
				←	100 Trying
	RELEASE	→			
	RELEASE COMPLETE	←		→	CANCEL(REL)
				←	200 OK INVITE(ANM)
				→	ACK
				←	200 OK CANCEL
				→	BYE(REL)
		→		←	200 OK BYE(RLC)

TP605009	**SIP reference: RFC 3261 [4]**			**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 2) 3)**	
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a DISCONNECT message ISDN **before** 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 value**: (PIXIT), **location** (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
				←	100 Trying
	DISCONNECT	→			
	RELEASE	←		→	CANCEL(REL)
	RELEASE COMPLETE	→		←	200 OK INVITE(ANM)
				→	ACK
				←	200 OK CANCEL
				→	BYE(REL)
				←	200 OK BYE(RLC)

<table>
<tr><th>TP605010</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN/ISDN reference:
Q.1912.5 [1], clause 7.7.1 2) 3)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">RELEASE COMPLETE; cause value: (PIXIT), location (PIXIT)</td></tr>
<tr><td rowspan="15">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>100 Trying</td></tr>
<tr><td>RELEASE COMPLETE</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>SIP_MESSAGE_VA</td></tr>
<tr><td></td><td></td><td></td><td colspan="2">CASE A</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>CANCEL(REL)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK CANCEL</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>487 Request Terminated</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td colspan="2">CASE B</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>487 Request Terminated</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
</table>

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

TP605012	SIP reference: RFC 3261 [4]	ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 2) 3)
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT	
SIP selection criteria	NOT PICS 4/10	
ISDN selection criteria		
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a DISCONNECT message ISDN **before** an early dialogue with the message defined as SIP_MESSAGE_VA has been established: • the SUT shall hold the release procedure until a **SIP_MESSAGE_VA** response has been received; • the SUT shall send a CANCEL or BYE request.	
SIP parameter values		
ISDN parameter values	DISCONNECT; **cause value**: (PIXIT), **location** (PIXIT)	
Comments	(see flow below)	

ISDN		SUT		SIP-I
SETUP	→		→	INVITE(IAM)
			←	100 Trying
DISCONNECT	→			
RELEASE	←			
RELEASE COMPLETE	→			
			←	SIP_MESSAGE_VA
CASE A				
			→	CANCEL(REL)
			←	200 OK CANCEL
			←	487 Request Terminated
			→	ACK
CASE B				
			→	BYE(REL)
			←	200 OK BYE(RLC)
			←	487 Request Terminated
			→	ACK

TP605013	SIP reference: RFC 3261 [4]	ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 4)
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT	
SIP selection criteria	NOT PICS 4/10	
ISDN selection criteria		
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE message ISDN message **after** 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; **cause value**: (PIXIT), **location** (PIXIT)	
Comments	(see flow below)	

ISDN		SUT		SIP-I
SETUP	→		→	INVITE(IAM)
ALERTING	←		←	180 Ringing(ACM
CONNECT	←		←	200 OK INVITE(ANM)
RELEASE COMPLETE	→			
			→	ACK
			→	BYE(REL)
			←	200 OK BYE(RLC)

TP605014	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 4)
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 **after** a 200 OK response message has been received • The SUT shall hold the release procedure until an ACK has been sent • The SUT shall send a BYE request				
SIP parameter values					
ISDN parameter values	RELEASE; **cause value**: (PIXIT), **location** (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM
	CONNECT	←		←	200 OK INVITE(ANM)
	RELEASE	→			
	RELEASE COMPLETE	←		→	ACK
				→	BYE(REL)
				←	200 OK BYE(RLC)

TP605015	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 4)
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a DISCONNECT ISDN message **after** a 200 OK response message has been received: • the SUT shall hold the release procedure until an ACK has been sent; • the SUT shall send a BYE request.				
SIP parameter values					
ISDN parameter values	DISCONNECT; **cause value:** (PIXIT), **location** (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM
	CONNECT	←		←	200 OK INVITE(ANM)
	DISCONNECT	→			
	RELEASE	←		→	ACK
	RELEASE COMPLETE	→		→	BYE(REL)
				←	200 OK BYE(RLC)

TP605016	**SIP reference: RFC 3261 [4]**				**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 3)**
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE message ISDN message **after** an early dialogue with the SIP message defined with the **SIP_MESSAGE_VA** has been established: • the SUT shall send a CANCEL or BYE request.				
SIP parameter values					
ISDN parameter values	RELEASE COMPLETE; **cause value**: (PIXIT), **location** (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
				←	SIP_MESSAGE_VA
	RELEASE COMPLETE	→			
	CASE A				
				→	CANCEL(REL)
				←	200 OK CANCEL
				←	487 Request Terminated
				→	ACK
	CASE B				
				→	BYE(REL)
				←	200 OK BYE(RLC)
				←	487 Request Terminated
				→	ACK

TP605017	**SIP reference: RFC 3261 [4]**				**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 3)**
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection 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 message ISDN message **after** an early dialogue with the SIP message defined with the **SIP_MESSAGE_VA** has been established: • the SUT shall send a CANCEL or BYE request;				
SIP parameter values					
ISDN parameter values	RELEASE; **cause value**: (PIXIT), **location** (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
				←	SIP_MESSAGE_VA
	RELEASE	→			
	RELEASE COMPLETE	←			
	CASE A				
				→	CANCEL(REL)
				←	200 OK CANCEL
				←	487 Request Terminated
				→	ACK
	CASE B				
				→	BYE(REL)
				←	200 OK BYE(RLC)
				←	487 Request Terminated
				→	ACK

TP605018	**SIP reference: RFC 3261 [4]**		**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 3)**		
TSS reference	ISDN-SIP /Basic call/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 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 value**: (PIXIT), **location** (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
				←	SIP_MESSAGE_VA
	DISCONNECT	→			
	RELEASE	←			
	RELEASE COMPLETE	→			
	CASE A				
				→	CANCEL(REL)
				←	200 OK CANCEL
				←	487 Request Terminated
				→	ACK
	CASE B				
				→	BYE(REL)
				←	200 OK BYE(RLC)
				←	487 Request Terminated
				→	ACK

TP605019	**SIP reference: RFC 3261 [4]**		**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 2) 4)**		
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message on receipt RELEASE 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 CV_SIP, encapsulated REL constructed from the ISDN RELEASE COMPLETE				
ISDN parameter values	RELEASE COMPLETE; **cause value:** (PIXIT), **location** (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	RELEASE COMPLETE	→			
				←	200 OK INVITE(ANM)
				→	ACK
				→	BYE(REL)
				←	200 OK BYE(RLC)

TP605020	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 2) 4)
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message on receipt RELEASE message with **Cause value** CV_ISDN, **location** LOC_ISDN **before** a 200 OK response (any) message has been received which establishes a confirmed dialogue: • the SUT shall hold the release procedure until a SIP 200 OK INVITE response has been received; • the SUT shall send a BYE request. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP.				
SIP parameter values	BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN RELEASE				
ISDN parameter values	RELEASE; **cause value:** (PIXIT), **location** (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	RELEASE	→			
	RELEASE COMPLETE	←			
				←	200 OK INVITE(ANM)
				→	ACK
				→	BYE(REL)
				←	200 OK BYE(RLC)

TP605021	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 2) 4)
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message on receipt DISCONNECT message with **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 CV_SIP, encapsulated REL constructed from the ISDN DISCONNECT				
ISDN parameter values	DISCONNECT; **cause value:** (PIXIT), **location** (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM
	CONNECT	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	DISCONNECT	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE(RLC)
	RELEASE COMPLETE	→			

<table>
<tr><th>TP605022</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN/ISDN reference:
Q.1912.5 [1], clause 7.7.1 2) 3</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE message with Cause value CV_ISDN, location LOC_ISDN before a 200 OK response message has been received:
the SUT shall hold the REL message a CANCEL is sent when any SIP response was been received;
on subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN RELEASE COMPLETE</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">RELEASE COMPLETE; cause value: (PIXIT), location (PIXIT)</td></tr>
<tr><td rowspan="10">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>100 Trying</td></tr>
<tr><td>RELEASE COMPLETE</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>CANCEL(REL)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK CANCEL</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP605023</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN/ISDN reference:
Q.1912.5 [1], clause 7.7.1 2) 3</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN RELEASE</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">RELEASE; cause value: (PIXIT), location (PIXIT)</td></tr>
<tr><td rowspan="10">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>100 Trying</td></tr>
<tr><td>RELEASE</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>RELEASE COMPLETE</td><td>←</td><td></td><td>→</td><td>CANCEL(REL)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK CANCEL</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td></td><td>→</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP605024	**SIP reference: RFC 3261 [4]**	**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 2) 3**
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT	
SIP selection criteria	PICS 4/10	
ISDN selection criteria		
Test purpose	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.	
SIP parameter values	BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN DISCONNECT	
ISDN parameter values	DISCONNECT; **cause value:** (PIXIT), **location** (PIXIT)	
Comments		

ISDN		**SUT**		**SIP-I**
SETUP	→		→	INVITE(IAM)
			←	100 Trying
DISCONNECT	→			
RELEASE	←		→	CANCEL(REL)
RELEASE COMPLETE	→		←	200 OK INVITE(ANM)
			→	ACK
			←	200 OK CANCEL
			→	BYE(REL)
			←	200 OK BYE(RLC)

<table>
<tr><th>TP605025</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN/ISDN reference:
Q.1912.5 [1], clause 7.7.1 3)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN RELEASE COMPLETE</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">RELEASE COMPLETE; cause value: (PIXIT), location (PIXIT)</td></tr>
<tr><td rowspan="15">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>100 Trying</td></tr>
<tr><td>RELEASE COMPLETE</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>SIP_MESSAGE_VA</td></tr>
<tr><td>CASE A</td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>CANCEL(REL)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK CANCEL</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>487 Request Terminated</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>CASE B</td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>487 Request Terminated</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
</table>

TP605026	**SIP reference: RFC 3261 [4]**	**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 3)**
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT	
SIP selection criteria	PICS 4/10	
ISDN selection criteria		
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE message with **Cause value** CV_ISDN, **location** LOC_ISDN **before** an early dialogue with the message defined as SIP_MESSAGE has been established: • the SUT shall hold the release procedure until a **SIP_MESSAGE_VA** response has been received; • the SUT shall send a **CANCEL** request or a BYE request. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP.	
SIP parameter values	BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN RELEASE	
ISDN parameter values	RELEASE; **cause value:** (PIXIT), **location** (PIXIT)	
Comments		

ISDN		**SUT**		**SIP-I**
SETUP	→		→	INVITE(IAM)
			←	100 Trying
RELEASE	→			
RELEASE COMPLETE	←			
			←	SIP_MESSAGE_VA
CASE A				
			→	CANCEL(REL)
			←	200 OK CANCEL
			←	487 Request Terminated
			→	ACK
CASE B				
			→	BYE(REL)
			←	200 OK BYE(RLC)
			←	487 Request Terminated
			→	ACK

TP605027	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 3)
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a DISCONNECT message with **Cause value** CV_ISDN, **location** LOC_ISDN **before** an early dialogue with the message defined as SIP_MESSAGE has been established: • the SUT shall hold the release procedure until a **SIP_MESSAGE_VA** response has been received; • the SUT shall send a **CANCEL** request or a BYE request. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP.				
SIP parameter values	BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN DISCONNECT				
ISDN parameter values	DISCONNECT; **cause value:** (PIXIT), **location** (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
				←	100 Trying
	DISCONNECT	→			
	RELEASE	←			
	RELEASE COMPLETE	→			
				←	SIP_MESSAGE_VA
	CASE A				
				→	CANCEL(REL)
				←	200 OK CANCEL
				←	487 Request Terminated
				→	ACK
	CASE B				
				→	BYE(REL)
				←	200 OK BYE(RLC)
				←	487 Request Terminated
				→	ACK

TP605028	**SIP reference: RFC 3261 [4]**				**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 3)**
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE message with **Cause value** CV_ISDN, **location** LOC_ISDN **after** a 200 OK response message has been received: • the SUT shall send a BYE request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP.				
SIP parameter values	BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN RELEASE COMPLETE				
ISDN parameter values	RELEASE COMPLETE; **cause value:** (PIXIT), **location** (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM
	CONNECT	←		←	200 OK INVITE(ANM)
	RELEASE COMPLETE	→			
				→	ACK
				→	BYE(REL)
				←	200 OK BYE(RLC)

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

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

TP605031	**SIP reference: RFC 3261 [4]**				**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 3)**
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection 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; **cause value:** (PIXIT), **location** (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
				←	SIP_MESSAGE_VA
	RELEASE COMPLETE	→			
	CASE A				
				→	CANCEL(REL)
				←	200 OK CANCEL
				←	487 Request Terminated
				→	ACK
	CASE B				
				→	BYE(REL)
				←	200 OK BYE(RLC)
				←	487 Request Terminated
				→	ACK

<table>
<tr><td>TP605032</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 3)</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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;</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN RELEASE</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">RELEASE; cause value: (PIXIT), location (PIXIT)</td></tr>
<tr><td rowspan="15">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>SIP_MESSAGE_VA</td></tr>
<tr><td>RELEASE</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>RELEASE COMPLETE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CASE A</td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>CANCEL(REL)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK CANCEL</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>487 Request Terminated</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>CASE B</td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>487 Request Terminated</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
</table>

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

Table 29

Values for 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

Table 30

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

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

Cause indications parameter field	Value of parameter field	component of SIP Reason header field	Component value
-	-	Protocol	"**Q.850**"
Cause Value	"**XX**" (see note 1)	Protocol-cause	"**cause= XX**" (see note 1)
-	-	Reason-text	Should be filled with the definition text as stated in ITU-T Recommendation Q.850 [3] (see note 2)
NOTE 1: "XX" is the Cause Value as defined in ITU-T Recommendation Q.850 [3]. NOTE 2: Due to the fact that the Cause Indications parameter does not include the definition text as defined in table 1/ITU-T Recommendation Q.850 [3] this is based on provisioning in the O-IWU.			

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

TP606001	**SIP reference: RFC 3261 [4]**				**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.2**
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message and on receipt of a BYE message where a Reason header field with Q.850 Cause Value is **not** included: • sends a DISCONNECT message constructed from the encapsulated REL in the received BYE MIME body.				
SIP parameter values	BYE: ISUP REL encapsulated in the MIME body				
ISDN parameter values	DISCONNECT: Cause value constructed from the encapsulated REL				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM
	CONNECT	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	DISCONNECT	←		←	BYE(REL)
	RELEASE	→		→	200 OK BYE(RLC)
	RELEASE COMPLETE	←			

<table>
<tr><th>TP606002</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN/ISDN reference:
Q.1912.5 [1], clause 7.7.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">BYE: ISUP REL encapsulated in the MIME body, Reason header value = (PIXIT)</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">DISCONNECT: Cause value constructed from the encapsulated REL</td></tr>
<tr><td rowspan="10">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM</td></tr>
<tr><td>CONNECT</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>DISCONNECT</td><td>←</td><td></td><td>←</td><td>BYE(REL)</td></tr>
<tr><td>RELEASE</td><td>→</td><td></td><td>→</td><td>200 OK BYE(RLC)</td></tr>
<tr><td>RELEASE COMPLETE</td><td>←</td><td></td><td></td><td></td></tr>
</table>

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

component of SIP Reason header field	value	ISDN Cause indicator I.E.	value
Protocol	"**Q.850**"	Cause Indication parameter	-
protocol-cause	"**cause = XX**" (see note 1)	Cause Value	constructed from the encapsulated REL
-	-	Location	constructed from the encapsulated REL
NOTE: "XX" is the Cause Value as defined in ITU-T Recommendation Q.850 [3].			

TP606003	**SIP reference: RFC 3261 [4]**			**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6**
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE			
SIP selection criteria				
ISDN selection criteria				
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT in state N2, before having received an backward message, on receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with Q.850 Cause Value is **not** included defined as SIP_Failure_VA: • sends a DISCONNECT or RELEASE message with the Cause value set to the value of the encapsulated REL.			
SIP parameter values	SIP_Failure_VA:ISUP REL encapsulated in the MIME body			
ISDN parameter values	DISCONNECT/RELEASE: **cause value:** mapped from the encapsulated REL			

Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	SETUP ACK	←			
	CASE A			←	SIP_Failure_VA(REL)
	RELEASE	←		→	ACK
	RELEASE COMPLETE	→			
	CASE B				
	DISCONNECT	←			
	RELEASE	→			
	RELEASE COMPLETE	←			

TP606004	**SIP reference: RFC 3261 [4]**			**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6**
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE			
SIP selection criteria				
ISDN selection criteria				
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT in state N3, before having received an backward message, on receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with Q.850 Cause Value is **not** included defined as SIP_Failure_VA: • sends a DISCONNECT or RELEASE message with the Cause value set to the value of the encapsulated REL.			
SIP parameter values	SIP_Failure_VA: ISUP REL encapsulated in the MIME body			
ISDN parameter values	DISCONNECT/RELEASE: **cause value**: mapped from the encapsulated REL			

Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	CALL PROCEEDING	←			
	CASE A			←	SIP_Failure_VA(REL)
	RELEASE	←		→	ACK
	RELEASE COMPLETE	→			
	CASE B				
	DISCONNECT	←			
	RELEASE	→			
	RELEASE COMPLETE	←			

TP606005	**SIP reference: RFC 3261 [4]**				**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6**
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP message containing the complete **called party number** where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party: • sends out an INVITE message. Ensure that the SUT in state N3 on receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with Q.850 Cause Value is **not** included defined as SIP_Failure_VA; • sends a DISCONNECT or RELEASE message with the Cause value set to the value of the encapsulated REL.				
SIP parameter values	SIP_Failure_VA: no ISUP REL encapsulated in the MIME body				
ISDN parameter values	DISCONNECT/RELEASE: **cause value:** mapped from the encapsulated REL				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	CALL PROCEEDING	←			
	CASE A			←	SIP_Failure_VA(REL)
	DISCONNECT	←		→	ACK
	RELEASE	→			
	RELEASE COMPLETE	←			
	CASE B				
	RELEASE	←			
	RELEASE COMPLETE	→			

Table 33

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

TP606006	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria	PICS 4/12				
ISDN selection criteria					
Test purpose	Ensure that the SUT if the SIP Failure response is interworked to ISDN after receiving an SETUP message sends out an INVITE message. Ensure that the SUT in state N2, before having received an backward message, on receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with Q.850 Cause Value is **not** included defined as SIP_Failure_VA: • sends a DISCONNECT or RELEASE message with the Cause value set to CV_ ISDN.				
SIP parameter values					
ISDN parameter values	DISCONNECT/RELEASE; **cause value:** CV_ISDN (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	SETUP ACK	←			
	CASE A			←	SIP_Failure_VA
	RELEASE	←		→	ACK
	RELEASE COMPLETE	→			
	CASE B				
	DISCONNECT	←			
	RELEASE	→			
	RELEASE COMPLETE	←			

TP606007	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria	PICS 4/12				
ISDN selection criteria					
Test purpose	Ensure that the SUT if the SIP Failure response is interworked to ISDN after receiving an SETUP message sends out an INVITE message. Ensure that the SUT in state N3, before having received an backward message, on receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with Q.850 Cause Value is **not** included defined as SIP_Failure_VA: • sends a DISCONNECT or RELEASE message with the Cause value set to CV_ ISDN.				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	CALL PROCEEDING	←			
	CASE A			←	SIP_Failure_VA
	RELEASE	←		→	ACK
	RELEASE COMPLETE	→			
	CASE B				
	DISCONNECT	←			
	RELEASE	→			
	RELEASE COMPLETE	←			

Table 34

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

TP606008	**SIP reference: RFC 3261 [4]**			**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6**
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE			
SIP selection criteria	NOT PICS 4/12			
ISDN selection criteria				
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. On receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with Q.850 Cause Value is **not** included defined as SIP_Failure_VA: • no action is taken on the ISDN.			
SIP parameter values				
ISDN parameter values				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
				←	SIP_Failure_VA
				→	ACK
	Further SIP procedures apply				

Table 35

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 [4]**				**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6**
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria					
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**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
				←	100 Trying
	RELEASE	→		→	CANCEL(REL)
	RELEASE COMPLETE	←		←	200 OK CANCEL
				←	487 Request terminated
				→	ACK

TP606010	**SIP reference: RFC 3261 [4]**				**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6**
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message, on receipt of a Failure message **491 Request Pending** where a Reason header field with Q.850 Cause Value is **not** included: • no action is taken on the ISDN.				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
				←	100 Trying
				←	491 Request Pending
				→	ACK

TP606011	**SIP reference: RFC 3261 [4]**				**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6**
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria	NOT PICS 4/11				
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message, a SIP message defined as **SIP MESSAGE_VA** has been received, on receipt of a Failure message (4xx, 5xx, 6xx) defined as **SIP_Failure_VA** where a Reason header field with Q.850 Cause Value is **not** included: • sends a RELEASE or DISCONNECT message with the Cause value set to CV_ISDN.				
SIP parameter values					
ISDN parameter values	RELEASE/DISCONNECT; **cause value**: CV_ISDN				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
				←	SIP MESSAGE_VA
	CASE A			←	SIP_Failure_VA
	RELEASE	←		→	ACK
	RELEASE COMPLETE	→			
	CASE B				
	DISCONNECT	←			
	RELEASE	→			
	RELEASE COMPLETE	←			

Table 36

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

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

Table 37

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

TP606013	**SIP reference: RFC 3261 [4]**				**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6**
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria	PICS 4/11				
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT in state N2, before having received an backward message, a SIP message defined as **SIP_MESSAGE_VA** has been received on receipt of a Failure message (4xx, 5xx, 6xx) defined as **SIP_Failure_VA** where a Reason header field with Q.850 Cause Value is included: • sends a DISC message. The Cause Value in the header field set to CV_SIP is mapped to the ISDN Cause Value field in the ISDN REL message with the Cause value set to CV_ ISDN.				
SIP parameter values	SIP_Failure_VA Reason header CV_ SIP (PIXIT)				
ISDN parameter values	DISCONNECT/RELEASE cause value CV_ ISDN (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	SETUP ACK	←			
				←	SIP MESSAGE_VA
	CASE A			←	SIP_Failure_VA
	RELEASE	←		→	ACK
	RELEASE COMPLETE	→			
	CASE B				
	DISCONNECT	←			
	RELEASE	→			
	RELEASE COMPLETE	←			

TP606014	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria	PICS 4/11				
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 header CV_ SIP (PIXIT)				
ISDN parameter values	DISCONNECT/RELEASE cause value CV_ ISDN (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	CALL PROCEEDING	←			
				←	SIP MESSAGE_VA
	CASE A			←	SIP_Failure_VA
	RELEASE	←		→	ACK
	RELEASE COMPLETE	→			
	CASE B				
	DISCONNECT	←			
	RELEASE	→			
	RELEASE COMPLETE	←			

TP606015	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria	PICS 4/11				
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 header CV_ SIP (PIXIT)				
ISDN parameter values	DISCONNECT/RELEASE cause value CV_ ISDN (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	CALL PROCEEDING	←			
				←	SIP MESSAGE_VA
	CASE A			←	SIP_Failure_VA
	RELEASE	←		→	ACK
	RELEASE COMPLETE	→			
	CASE B				
	DISCONNECT	←			
	RELEASE	→			
	RELEASE COMPLETE	←			

Table 38

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

Table 39

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

<table>
<tr><th>TP606016</th><th>SIP reference: RFC 3261 [4]</th><th>ISDN/ISDN reference:
Q.1912.5 [1], clause 7.7.6</th></tr>
<tr><td>TSS reference</td><td colspan="2">ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="2">NOT PICS 4/17</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="2"></td></tr>
<tr><td>Test purpose</td><td colspan="2">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="2"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="2">REL; cause value: CV_ISDN</td></tr>
<tr><td>Comments</td><td colspan="2">
<table>
<tr><th>ISDN</th><th></th><th>SUT</th><th></th><th>SIP-I</th></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>SETUP ACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CASE A</td><td></td><td></td><td>←</td><td>SIP_Response_VA</td></tr>
<tr><td>RELEASE</td><td>←</td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>RELEASE COMPLETE</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>CASE B</td><td></td><td></td><td></td><td></td></tr>
<tr><td>DISCONNECT</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>RELEASE</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>RELEASE COMPLETE</td><td>←</td><td></td><td></td><td></td></tr>
</table>
</td></tr>
</table>

<table>
<tr><th>TP606017</th><th>SIP reference: RFC 3261 [4]</th><th>ISDN/ISDN reference:
Q.1912.5 [1], clause 7.7.6</th></tr>
<tr><td>TSS reference</td><td colspan="2">ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="2">NOT PICS 4/17</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="2"></td></tr>
<tr><td>Test purpose</td><td colspan="2">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="2"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="2">REL; cause value: CV_ISDN</td></tr>
<tr><td>Comments</td><td colspan="2">
<table>
<tr><th>ISDN</th><th></th><th>SUT</th><th></th><th>SIP-I</th></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>CALL PROCEEDING</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CASE A</td><td></td><td></td><td>←</td><td>SIP_Response_VA</td></tr>
<tr><td>RELEASE</td><td>←</td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>RELEASE COMPLETE</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>CASE B</td><td></td><td></td><td></td><td></td></tr>
<tr><td>DISCONNECT</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>RELEASE</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>RELEASE COMPLETE</td><td>←</td><td></td><td></td><td></td></tr>
</table>
</td></tr>
</table>

TP606018	SIP reference: RFC 3261 [4]	ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE	
SIP selection criteria	NOT PICS 4/17	
ISDN selection criteria		
Test purpose	Ensure that the SUT after receiving the SETUP message containing the complete **called party number** where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party: • sends out an INVITE message. Ensure that the SUT in state N3, on receipt of a response message (3xx) defined as **SIP_Response_VA,** the SUT; • sends a DISC message with the **Cause value** 127 Interworking.	
SIP parameter values		
ISDN parameter values	DISC; **cause value**: CV_ISDN	
Comments		

ISDN		SUT		SIP-I
SETUP	→		→	INVITE(IAM)
CALL PROCEEDING	←			
CASE A			←	SIP_Response_VA
RELEASE	←		→	ACK
RELEASE COMPLETE	→			
CASE B				
DISCONNECT	←			
RELEASE	→			
RELEASE COMPLETE	←			

TP606019	SIP reference: RFC 3261 [4]	ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE	
SIP selection criteria	PICS 4/17	
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 values	SIP_Response_VA Contact: URI of new destination INVITE: Request URI of new destination	
ISDN parameter values		
Comments		

ISDN		SUT		SIP-I
SETUP	→		→	INVITE(IAM)
SETUP ACK	←			
			←	SIP_Response_VA
			→	ACK
			→	INVITE(IAM)
ALERTING	←		←	180 Ringing(ACM
CONNECT	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
DISCONNECT	←		←	BYE(REL)
RELEASE	→		→	200 OK BYE(RLC)
RELEASE COMPLETE	←			

TP606020	**SIP reference: RFC 3261 [4]**	**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6**
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE	
SIP selection criteria	PICS 4/17	
ISDN selection criteria		
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT in state N3, before having received an backward message on receipt of a response message (3xx) defined as **SIP_Response_VA**, the SUT: • sends an INVITE using the value of the Contact header field in the received **SIP_Response_VA** in the Request URI.	
SIP parameter values	SIP_Response_VA Contact: URI of new destination INVITE: Request URI of new destination	
ISDN parameter values		
Comments		

ISDN		**SUT**		**SIP-I**
SETUP	→		→	INVITE(IAM)
CALL PROCEEDING	←			
			←	SIP_Response_VA
			→	ACK
			→	INVITE(IAM)
ALERTING	←		←	180 Ringing(ACM
CONNECT	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
DISCONNECT	←		←	BYE(REL)
RELEASE	→		→	200 OK BYE(RLC)
RELEASE COMPLETE	←			

<table>
<tr><th>TP606021</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISDN/ISDN reference:
Q.1912.5 [1], clause 7.7.6</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/17</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT after receiving the SETUP message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party:
• sends out an INVITE message. Ensure that the SUT in state N3, on receipt of a response message (3xx) defined as SIP_Response_VA, the SUT;
• sends an INVITE using the value of the Contact header field in the received SIP_Response_VA in the Request URI.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">SIP_Response_VA Contact: URI of new destination
INVITE: Request URI of new destination</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="14">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>CALL PROCEEDING</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>SIP_Response_VA</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM</td></tr>
<tr><td>CONNECT</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>DISCONNECT</td><td>←</td><td></td><td>←</td><td>BYE(REL)</td></tr>
<tr><td>RELEASE</td><td>→</td><td></td><td>→</td><td>200 OK BYE(RLC)</td></tr>
<tr><td>RELEASE COMPLETE</td><td>←</td><td></td><td></td><td></td></tr>
</table>

Table 40

<table>
<tr><th colspan="2">Values for test purpose
TP606016, TP606017, TP606018
TP606019, TP606020, TP606021</th></tr>
<tr><th>VA</th><th>SIP_Response_VA</th></tr>
<tr><td>VA_1</td><td>300 Multiple Choices</td></tr>
<tr><td>VA_2</td><td>301 Moved Permanently</td></tr>
<tr><td>VA_3</td><td>302 Move Temporarily</td></tr>
<tr><td>VA_4</td><td>305 Use Proxy</td></tr>
<tr><td>VA_5</td><td>380 Alternative Service</td></tr>
</table>

A.1.1.1.2.7 Autonomous release at the MGC

TP607001	**SIP reference: RFC 3261 [4]**			**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6.1**	
TSS reference	ISDN-SIP /Basic call/Autonomous_release				
SIP selection criteria	PICS 3/2				
ISDN selection criteria					
Test purpose	Ensure that the SUT a On receipt of a 484 Address Incomplete response for the current INVITE (i.e. there are no other pending INVITE transactions for this call), if the SUT is configured to propagate overlap signalling into the SIP network, the SUT: • shall not clear immediately the bearer channel and shall instead start timer T_{OIW3}. The RELEASE or DISCONNECT message shall only be sent if T_{OIW3} expires.				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
			Start timer T_{OIW3}	←	484 Address incomplete
				→	ACK
			Timeout T_{OIW3}		
	CASE A				
	RELEASE	←			
	RELEASE COMPLETE	→			
	CASE B				
	DISCONNECT	←			
	RELEASE	→			
	RELEASE COMPLETE	←			

TP607002	**SIP reference: RFC 3261 [4]**			**ISDN/ISDN reference: Q.1912.5 [1], 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	→		→	INVITE(IAM)
				←	484 Address incomplete
	CASE A			→	ACK
	RELEASE	←			
	RELEASE COMPLETE	→			
	CASE B				
	DISCONNECT	←			
	RELEASE	→			
	RELEASE COMPLETE	←			

TP607003	**SIP reference: RFC 3261 [4]**				**ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.3**
TSS reference	ISDN-SIP /Basic call/Autonomous_release				
SIP selection criteria	PICS 4/4 AND 4/5				
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	→		→	INVITE(IAM)
					183 Session Progress
					PRACK
					200 OK PRACK
		Internal resource reservation unsuccessful			
	CASE A				
	RELEASE	←		→	CANCEL(REL)
	RELEASE COMPLETE	→		←	200 OK CANCEL
				←	487 Request Terminated
				→	ACK
	CASE B				
	RELEASE	←		→	BYE(REL)
	RELEASE COMPLETE	→		←	200 OK BYE(RLC)
				←	487 Request Terminated
				→	ACK

A.1.1.2 Test purposes for ISDN/SIP Supplementary services

A.1.1.2.1 Calling Line Identification Presentation (CLIP)

TP701001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 3.5.2.1.1**
TSS reference	ISDN-(ISUP)-SIP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Calling party number (network provided)** Verify that the IUT can successfully originate a call having a **calling party number** with the screening indicator set to "network provided" and the presentation restricted indicator set to "presentation allowed".				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body				
ISDN parameter values					
Comments	ISDN		SUT		SIP-I
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
		Communication			
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

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

TP701003	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 3.5.2.1.1**
TSS reference	ISDN-(ISUP)-SIP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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 values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body				
ISDN parameter values					
Comments	ISDN		SUT		SIP-I
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
		Communication			
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

TP701004	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 3.5.2.1.1**
TSS reference	ISDN-(ISUP)-SIP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Calling party number (user 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	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
		Communication			
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

TP701005	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 3.5.2.1.1**
TSS reference	ISDN-(ISUP)-SIP/SS/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		SUT		SIP-I
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
		Communication			
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

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

A.1.1.2.2 Calling Line Identification Restriction (CLIR)

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

TP702002	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 4.5.2.1.1**		
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Restricted calling party number (network provided) with calling sub-address** *V*erify 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 values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body				
ISDN parameter values					
Comments	ISDN		SUT		SIP-I
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

TP702003	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 4.5.2.1.1**		
ISDN parameter values	ISDN-(ISUP)-SIP/SS/CLIR				
ISDN parameter values					
ISDN parameter values					
ISDN parameter values	**Restricted calling party number (user provided, verified and passed)** Verify that the IUT can successfully originate a call having the **calling party number** with the screening indicator set to "user provided, verified and passed" and the address presentation restricted indicator set to "presentation restricted.				
ISDN parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body				
ISDN parameter values					
Comments	ISDN		SUT		SIP-I
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

<table>
<tr><td>TP702004</td><td colspan="5">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 4.5.2.1.1</td></tr>
<tr><td>TSS reference</td><td colspan="7">ISDN-(ISUP)-SIP/SS/CLIR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="7"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="7"></td></tr>
<tr><td>Test purpose</td><td colspan="7">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="7">INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body
180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body</td></tr>
<tr><td>ISDN parameter values</td><td colspan="7"></td></tr>
<tr><td rowspan="9">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td colspan="3">SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td colspan="3">INVITE(IAM)</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td colspan="3">180 Ringing(ACM)</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td colspan="3">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="3">ACK</td></tr>
<tr><td></td><td colspan="3">Communication</td><td colspan="3"></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td colspan="3">BYE(REL)</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td colspan="3">200 OK BYE(RLC)</td></tr>
<tr><td>REL_COM</td><td>→</td><td></td><td></td><td colspan="3"></td></tr>
</table>

<table>
<tr><td>TP702005</td><td colspan="5">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 4.5.2.1.1</td></tr>
<tr><td>TSS reference</td><td colspan="7">ISDN-(ISUP)-SIP/SS/CLIR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="7"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="7"></td></tr>
<tr><td>Test purpose</td><td colspan="7">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".</td></tr>
<tr><td>SIP parameter values</td><td colspan="7">INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body
180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body</td></tr>
<tr><td>ISDN parameter values</td><td colspan="7"></td></tr>
<tr><td rowspan="9">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td colspan="3">SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td colspan="3">INVITE(IAM)</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td colspan="3">180 Ringing(ACM)</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td colspan="3">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="3">ACK</td></tr>
<tr><td></td><td colspan="3">Communication</td><td colspan="3"></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td colspan="3">BYE(REL)</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td colspan="3">200 OK BYE(RLC)</td></tr>
<tr><td>REL_COM</td><td>→</td><td></td><td></td><td colspan="3"></td></tr>
</table>

TP702006	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 4.5.2.1.1**
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR			
SIP selection criteria				
ISUP selection criteria				
Test purpose	**Restricted calling party number (user provided, not verified) with calling sub-address** Verify that the IUT can successfully originate a call having a default **calling party number** with the screening indicator set to "network provided", a **generic number** containing the additional calling party number with the screening indicator set to "user provided, not verified", both having the address presentation restricted indicator set to "presentation restricted" and an **access transport** parameter containing the calling sub-address.			
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body			
ISDN parameter values				

Comments	ISDN		SUT		SIP-I
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

TP702007	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 4.2.1**
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR			
SIP selection criteria				
ISUP selection criteria	DLE			
Test purpose	**Presentation of the address - interaction with MCID** Verify that the information conveyed in an incoming call (especially the **calling party number** and the additional calling party number in the **generic number**) is registered in the network regardless of whether the calling user has activated the CLIR service or not, if the called user has MCID activated.			
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body			
ISDN parameter values				

Comments	ISDN		SUT		SIP-I
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

TP702008	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 4.2.1**	
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria	DLE				
Test purpose	**Presentation of the address - called party has override category** Verify that the **calling party number** and the additional calling party number in the **generic number** are passed to the access regardless of whether the calling user has activated the CLIR service or not if the called user has the override category.				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body				
ISDN parameter values					
Comments	ISDN		SUT		SIP-I
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

TP702009	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 4.2.1**	
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria	DLE				
Test purpose	**Presentation of the address - called party has not override category** Verify that the **calling party number** is **not** passed to the access.				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body				
ISDN parameter values					
Comments	ISDN		SUT		SIP-I
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

TP7020010	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 4.2.1
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria	DLE				
Test purpose	**Presentation of the address - called party has not override category** Verify that the **calling party number** and the additional calling party number in the **generic number** are **not** passed to the access.				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body				
ISDN parameter values					
Comments	ISDN		SUT		SIP-I
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

A.1.1.2.3 Connected Line Identification Presentation (COLP)

TP703001	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 5.5.2.5.1 i)
TSS reference	ISDN-(ISUP)-SIP/SS/COLP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Initiate COLP request** Verify that the exchange can initiate successfully a call requesting the COLP service in the **optional forward call indicators.**				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body				
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

TP703002	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 5.5.2.5.1 i)
TSS reference	ISDN-(ISUP)-SIP/SS/COLP	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**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 values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body	
ISDN parameter values		
Comments		

ISDN		**SUT**		**SIP-I**
SETUP	→		→	INVITE(IAM)
CASE A				
ALERTING	←		←	180 Ringing(ACM)
CONN	←		←	200 OK INVITE(ANM)
			→	ACK
CASE B				
CONN	←		←	200 OK INVITE(CON)
			→	ACK
		Communication		
DISC	→		→	BYE(REL)
REL	←		←	200 OK BYE(RLC)
REL_COM	→			

TP703003	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 5.5.2.5.1 i)
TSS reference	ISDN-(ISUP)-SIP/SS/COLP	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Connected number (user provided, verified and passed) with connected sub-address** Verify that the IUT can provide a **connected number** with the screening indicator set to "user provided, verified and passed", if the user provided COL is valid and an **access transport** parameter containing the connected sub-address.	
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body	
ISDN parameter values		
Comments		

ISDN		**SUT**		**SIP-I**
SETUP	→		→	INVITE(IAM)
CASE A				
ALERTING	←		←	180 Ringing(ACM)
CONN	←		←	200 OK INVITE(ANM)
			→	ACK
CASE B				
CONN	←		←	200 OK INVITE(CON)
			→	ACK
		Communication		
DISC	→		→	BYE(REL)
REL	←		←	200 OK BYE(RLC)
REL_COM	→			

TP703004	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 5.5.2.5.1 i)**
TSS reference	ISDN-(ISUP)-SIP/SS/COLP	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Connected number (network provided)** Verify that the IUT can provide a default **connected number** with the screening indicator set to "network provided", if the user provided COL is not valid.	
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body	
ISDN parameter values		

Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	CASE A				
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
	CASE B				
	CONN	←		←	200 OK INVITE(CON)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

TP703005	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 5.5.2.5.1 i)**
TSS reference	ISDN-(ISUP)-SIP/SS/COLP	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Connected number (network provided) with connected sub-address** Verify that the IUT can provide a default **connected number** with the screening indicator set to "network provided", if the user provided COL is not valid and an **access transport** parameter containing the connected sub-address.	
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body	
ISDN parameter values		

Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	CASE A				
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
	CASE B				
	CONN	←		←	200 OK INVITE(CON)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

TP703006	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 5.5.2.5.1 i)**	
TSS reference	ISDN-(ISUP)-SIP/SS/COLP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Connected number (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 values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body				
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	CASE A				
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
	CASE B				
	CONN	←		←	200 OK INVITE(CON)
				→	ACK
		Communication			
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

TP703007	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 5.5.2.5.1 i)**	
TSS reference	ISDN-(ISUP)-SIP/SS/COLP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Connected number (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: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body				
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	CASE A				
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
	CASE B				
	CONN	←		←	200 OK INVITE(CON)
				→	ACK
		Communication			
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

TP703008	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 5.5.2.5.1 i)**	
TSS reference	ISDN-(ISUP)-SIP/SS/COLP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body				
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	CASE A				
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
	CASE B				
	CONN	←		←	200 OK INVITE(CON)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

A.1.1.2.4 Connected Line Identification Restriction (COLR)

TP704001	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 6.5.2.1.2**	
TSS reference	ISDN-(ISUP)-SIP/SS/COLR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Presentation of restricted COL** Verify that a local exchange will not pass the information on to the access signalling system when a **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	→		→	INVITE(IAM)
	CASE A				
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
	CASE B				
	CONN	←		←	200 OK INVITE(CON)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

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

TP704003	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 6.5.2.1.2**
TSS reference	ISDN-(ISUP)-SIP/SS/COLR				
SIP selection criteria					
ISUP selection criteria	DLE				
Test purpose	**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 values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body				
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	CASE A				
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
	CASE B				
	CONN	←		←	200 OK INVITE(CON)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

TP704004	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], Q.731 [i.2], 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	→		→	INVITE(IAM)
	CASE A				
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
	CASE B				
	CONN	←		←	200 OK INVITE(CON)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

TP704005	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 6.5.2.5**	
TSS reference	ISDN-(ISUP)-SIP/SS/COLR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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 values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body				
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	CASE A				
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
	CASE B				
	CONN	←		←	200 OK INVITE(CON)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

<table>
<tr><th>TP704006</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 6.5.2.1.2</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISDN-(ISUP)-SIP/SS/COLR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">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".</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">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</td></tr>
<tr><td>ISDN parameter values</td><td colspan="6"></td></tr>
<tr><td rowspan="13">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td colspan="2">SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(IAM)</td></tr>
<tr><td>CASE A</td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing(ACM)</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td>CASE B</td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE(CON)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td></td><td colspan="3">Communication</td><td colspan="2"></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td colspan="2">BYE(REL)</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK BYE(RLC)</td></tr>
<tr><td>REL_COM</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
</table>

<table>
<tr><td>TP704007</td><td colspan="5">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1],
Q.731 [i.2], clause 6.5.2.1.2</td></tr>
<tr><td>TSS reference</td><td colspan="7">ISDN-(ISUP)-SIP/SS/COLR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="7"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="7"></td></tr>
<tr><td>Test purpose</td><td colspan="7">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="7">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</td></tr>
<tr><td>ISDN parameter values</td><td colspan="7"></td></tr>
<tr><td rowspan="14">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td colspan="3">SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td colspan="3">INVITE(IAM)</td></tr>
<tr><td>CASE A</td><td></td><td></td><td></td><td colspan="3"></td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td colspan="3">180 Ringing(ACM)</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td colspan="3">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="3">ACK</td></tr>
<tr><td>CASE B</td><td></td><td></td><td></td><td colspan="3"></td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td colspan="3">200 OK INVITE(CON)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="3">ACK</td></tr>
<tr><td></td><td colspan="3">Communication</td><td colspan="3"></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td colspan="3">BYE(REL)</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td colspan="3">200 OK BYE(RLC)</td></tr>
<tr><td>REL_COM</td><td>→</td><td></td><td></td><td colspan="3"></td></tr>
</table>

A.1.1.2.5 Terminal Portability (TP)

TP705001	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], 4.5.2.1.1 a)/Q.733 [i.6]**	
TSS reference	ISDN-(ISUP)-SIP/SS/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	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
			Communication		
	SUSPEND	→		→	INFO(SUS)
				←	200 OK INFO
	RESUME	→		→	INFO(RES)
				←	200 OK INFO
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE
	REL_COM	→			

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

<table>
<tr><td>TP705003</td><td colspan="5">SIP reference: RFC 3261 [4]</td><td>ISUP reference:
Q.1912.5 [1],
4.5.2.1.2/Q.733 [i.6]</td></tr>
<tr><td>TSS reference</td><td colspan="6">ISDN-(ISUP)-SIP/SS/TP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">Terminal portability, requested by local served user, no Resume after Suspend
To verify that the call is released with cause #102 (recovery on timer expiry) by the IUT if timer T2 expires because the local served user does not resume the call.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="6"></td></tr>
<tr><td rowspan="10">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td colspan="2">SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(IAM)</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing(ACM)</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Communication</td><td colspan="2"></td></tr>
<tr><td>SUSPEND</td><td>→</td><td></td><td>→</td><td colspan="2">INFO(SUS)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK INFO</td></tr>
<tr><td></td><td colspan="3">T2 expiry</td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">BYE(REL)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK BYE</td></tr>
</table>

<table>
<tr><td>TP705004</td><td colspan="5">SIP reference: RFC 3261 [4]</td><td>ISUP reference:
Q.1912.5 [1],
4.5.2.1.1/Q.733 [i.6]</td></tr>
<tr><td>TSS reference</td><td colspan="6">ISDN-(ISUP)-SIP/SS/TP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">Terminal portability, release suspended call
To verify that a suspended call can be released by the IUT, if the local user or the remote user releases the call.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="6"></td></tr>
<tr><td rowspan="9">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td colspan="2">SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(IAM)</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing(ACM)</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td colspan="3">Communication</td><td colspan="2"></td></tr>
<tr><td>SUSPEND</td><td>→</td><td></td><td>→</td><td colspan="2">INFO(SUS)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">BYE(REL)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">200 OK BYE</td></tr>
</table>

TP705005	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], 4.5.2.5.1 a)/Q.733 [i.6]**	
TSS reference	ISDN-(ISUP)-SIP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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	→		→	180 Ringing(ACM)
	CONN	→		→	200 OK INVITE(ANM)
			Communication		
	NOTIFY(suspend)	←		←	INFO(SUS)
				→	200 OK INFO
	NOTIFY(resume)	←		←	INFO(RES)
				→	200 OK INFO
			Communication		
	DISC	←		←	BYE(REL)
	REL	→		→	200 OK BYE
	REL_COM	←			

TP705006	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], 4.5.2.5.1 b)/Q.733 [i.6]**	
TSS reference	ISDN-(ISUP)-SIP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Terminal portability, requested by the called part*y*** To verify that the called party can suspend and resume an incoming call and that user initiated **SUS** and **RES** messages are sent to the preceding exchange.				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP	←		←	INVITE(IAM)
	ALERTING	→		→	180 Ringing(ACM)
	CONN	→		→	200 OK INVITE(ANM)
			Communication		
	NOTIFY(suspend)	→		→	INFO(SUS)
				←	200 OK INFO
	NOTIFY(resume)	→		→	INFO(RES)
				←	200 OK INFO
			Communication		
	DISC	←		←	BYE(REL)
	REL	→		→	200 OK BYE
	REL_COM	←			

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

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

TP706001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 1.1.5.2.1.1.1; 1.1.5.2.1.1.3; 1.1.5.2.2-4.1/Q.737 [i.10]**
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS1 implicit - request** To verify that the IUT can successfully initiate/transit a call with an UUS 1 implicit request, having the **user-to-user information** parameter in the **IAM**, without the **user-to-user indicators** parameter.				
SIP parameter values					
ISDN parameter values	SETUP: User-to-user information				
Comments	**ISDN**		**SUT**		**SIP**
	SETUP(UUInf)	→		→	INVITE(IAM UUInf)
	ALERTING	←		←	180 Ringing(ACM)
	CONN(UUInf)	←		←	200 OK INVITE(ANM)
				→	ACK
			Communication		
	DISC	←		←	BYE(REL)
	REL	→		→	200 OK BYE
	REL_COM	←			

TP706002	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 1.1.5.2.5.2.3; 1.1.5.2.2-4.2/ Q.737 [i.10]**
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS1 implicit - discarded with indication received** To verify that the IUT can, after successfully initiating/transiting a call with an UUS1 implicit request, continue normal call set up if the first backward message is received with the **user-to-user indicators** set to "user-to-user information discarded by the network" (see note).				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP(UUInf)	→		→	INVITE(IAM UUInf)
	ALERTING	←		←	180 Ringing(ACM UUInd)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
			Communication		
	DISC	←		←	BYE(REL)
	REL	→		→	200 OK BYE
	REL_COM	←			
NOTE: The user-to-user information is discarded because the following network does not support it.					

<table>
<tr><th>TP706003</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1],
clauses 1.1.5.2.5.2.3; 1.1.5.2.3-5.2/
Q.737 [i.10]</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISDN-(ISUP)-SIP/SS/UUS1</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">UUS1 implicit - discarded but no indication received
To verify that the IUT can successfully initiate/transit a call with an UUS1 implicit request, and complete the call if no indication is provided in the backward direction (see note).</td></tr>
<tr><td>SIP parameter values</td><td colspan="6"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="6"></td></tr>
<tr><td rowspan="9">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td colspan="2">SIP</td></tr>
<tr><td>SETUP(UUInf)</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(IAM UUInf)</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing(ACM)</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td></td><td colspan="3">Communication</td><td colspan="2"></td></tr>
<tr><td>DISC</td><td>←</td><td></td><td>←</td><td colspan="2">BYE(REL)</td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td colspan="2">200 OK BYE</td></tr>
<tr><td>REL_COM</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td colspan="7">NOTE: The user-to-user information is discarded because:
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.</td></tr>
</table>

<table>
<tr><td>TP706004</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1],
clauses 1.1.5.2.1.1.1; 1.1.5.2.1.1.3;
1.1.5.2.3-5.1/Q.737 [i.10]</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-(ISUP)-SIP/SS/UUS1</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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).</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="18">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP(UUInf)</td><td>→</td><td></td><td>→</td><td>INVITE(IAM UUInf)</td></tr>
<tr><td>CASE A</td><td></td><td></td><td></td><td></td></tr>
<tr><td>ALERTING(UUInf)</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM UUInf)</td></tr>
<tr><td>CASE B</td><td></td><td></td><td></td><td></td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>NOTIFY(UUInf)</td><td></td><td></td><td>←</td><td>183 Session Progress(CPG UUInf)</td></tr>
<tr><td>CASE C</td><td></td><td></td><td></td><td></td></tr>
<tr><td>CONN(UUInf)</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(CON UUInf)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>CASE D</td><td></td><td></td><td></td><td></td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>CONN(UUInf)</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM UUInf)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Communication</td><td></td></tr>
<tr><td>DISC</td><td>←</td><td></td><td>←</td><td>BYE(REL)</td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>200 OK BYE</td></tr>
<tr><td>REL_COM</td><td>←</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><td>TP706005</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1],
clauses 1.1.5.2.5.2.3; 1.1.5.2.3-5.2/
Q.737 [i.10]</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-(ISUP)-SIP/SS/UUS1</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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).</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="9">Comments</td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE(IAM UUInf)</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing(ACM UUInd)</td><td>←</td><td></td><td>←</td><td>ALERTING</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Communication</td><td></td></tr>
<tr><td>BYE(REL)</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>REL_COM</td></tr>
<tr><td colspan="6">NOTE: The user-to-user information is discarded because the network does not support it.</td></tr>
</table>

<table>
<tr><td>TP706006</td><td colspan="3">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1],
clauses 1.1.5.2.1.1.2; 1.1.5.2.2-4.1/
Q.737 [i.10]</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-(ISUP)-SIP/SS/UUS1</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">UUS1 explicit non-essential - request
To verify that the IUT can successfully initiate/transit a call with an UUS1 explicit non-essential request, by including/transferring the user-to-user information parameter and the user-to-user indicators in the IAM set to "request, not essential".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="10">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP(FAC uus1reqness)</td><td></td><td></td><td></td><td>INVITE(IAM UUInd, UUInf)</td></tr>
<tr><td>ALERTING(FAC uus1rr)</td><td></td><td></td><td></td><td>180 Ringing(ACM)</td></tr>
<tr><td>CONN</td><td></td><td></td><td></td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td></td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Communication</td><td></td></tr>
<tr><td>BYE(REL)</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>REL_COM</td></tr>
</table>

TP706007	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 1.1.5.2.5.2.3, 1.1.5.2.2-4.2/ Q.737 [i.10]
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS1 explicit non-essential - explicit rejection received** To verify that the IUT can successfully initiate/transit a call with an UUS1 explicit non-essential request, and continue normal call set up if the UUS1 service is explicitly rejected (the **user-to-user indicators** parameter is received as "service not provided" in the **ACM** or **CPG** or **ANM** or **CON**) (see note).				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP(FAC uus1reqness)	→		→	INVITE(IAM UUInd, UUInf)
	CASE A				
	ALERTING(FAC uus1rr)	←		←	180 Ringing(ACM UUInd s1 prov)
	CONN	←		←	200 OK INVITE
	CASE B				
				←	183 Session Progress(ACM)
	ALERTING(FAC uus1rr)	←		←	180 Ringing(CPG UUInd s1 prov)
	CONN	←		←	200 OK INVITE
	CASE C				
	ALERTING	←		←	180 Ringing(ACM)
	CONN(FAC uus1rr)	←		←	200 OK INVITE(ANM UUInd s1 prov)
	CASE D				
	CONN(FAC uus1rr)	←		←	200 OK INVITE(ANM UUInd s1 prov)
		Communication			
	BYE(REL)	←		←	DISC
	200 OK BYE	→		→	REL
				←	REL_COM
NOTE: The user-to-user information is discarded because: 1) the network is unable to pass the explicit service 1 in any message; 2) the remote user may not be able to interpret incoming UUS information.					

<table>
<tr><th>TP706008</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
clauses1.1.5.2.5.2.3, 1.1.5.2.2-4.2/
Q.737 [i.10]</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-(ISUP)-SIP/SS/UUS1</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">UUS1 explicit non-essential - implicit (no explicit) rejection received
To verify that the IUT can successfully initiate/transit a call with an UUS1 explicit non-essential request, and continue normal call set up if no indication is provided in the backward direction (see note).</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="9">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP(FAC uus1reqness)</td><td>→</td><td></td><td>→</td><td>INVITE(IAM UUInd, UUInf)</td></tr>
<tr><td>ALERTING(FAC uus1rr)</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>CONN(FAC uus1reterr)</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Communication</td><td></td></tr>
<tr><td>BYE(REL)</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>REL_COM</td></tr>
<tr><td colspan="6">NOTE: The user-to-user information is discarded because:
1) the network is unable to pass the explicit service 1 in any message;
2) the remote user may not be able to interpret incoming UUS information.</td></tr>
</table>

TP706009	**SIP reference: RFC 3261 [4]**				**ISUP reference:** **Q.1912.5 [1],** **clauses 1.1.5.2.1.1.2, 1.1.5.2.3-5.1/** **Q.737 [i.10]**
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS1 explicit non-essential - acceptance** To verify that the IUT can successfully transit/accept a call with an UUS1 explicit non-essential request, by transferring/including the **user-to-user indicators** parameter in the **ACM, CPG, ANM** or **CON** set to "service provided".				
SIP parameter values					
ISDN parameter values					
Comments	**SIP**		**SUT**		**ISDN**
	INVITE(IAM UUInd, UUInf)	→		→	SETUP(FAC uus1reqness)
					CASE A
	180 Ringing(ACM UUInd s1 prov)	←		←	ALERTING(FAC uus1rr)
	200 OK INVITE	←		←	CONN
					CASE B
	183 Session Progress(ACM)	←			
	180 Ringing(CPG UUInd s1 prov)	←		←	ALERTING(FAC uus1rr)
	200 OK INVITE	←		←	CONN
					CASE C
	180 Ringing(ACM)	←		←	ALERTING
	200 OK INVITE(ANM UUInd s1 prov)	←		←	CONN(FAC uus1rr)
					CASE D
	200 OK INVITE(ANM UUInd s1 prov)	←		←	CONN(FAC uus1rr)
			Communication		
	DISC	←		←	BYE(REL)
	REL	→		→	200 OK BYE
	REL_COM			←	

TP706010	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 1.1.5.2.5.2.2, 1.1.5.2.2-5.2/ Q.737 [i.10]**
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS1 explicit non-essential - implicit (no explicit) rejection sent** To verify that the IUT can transfer/accept a call with an UUS1 explicit non-essential request, and reject the service by not providing any **user-to-user indicators** parameter in the **ACM**, **CPG**, **ANM** or **CON** (see note).				
SIP parameter values					
ISDN parameter values					
Comments	**SIP**		**SUT**		**ISDN**
	INVITE(IAM UUInd, UUInf)	→		→	SETUP(FAC uus1reqness)
	180 Ringing(ACM)	←		←	ALERTING
	200 OK INVITE(ANM)	←		←	CONN
	ACK	→			
			Communication		
	DISC	←		←	BYE(REL)
	REL	→		→	200 OK BYE
	REL_COM			←	
NOTE: The network or the user cannot support UUS1.					

TP706011	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 1.1.5.2.1.1.2, 1.1.5.2.2-5.1/ Q.737 [i.10]**
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS1 explicit essential - request** To verify that the IUT can successfully originate/transit a call having an UUS1 explicit essential request, by including/transferring in the **IAM** the **user-to-user information** parameter, the **user-to-user indicators** set to "request, essential" and the ISDN user part preference indicator in the **forward call indicators** set to "ISUP required all the way".				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP(FAC uus1reqess)				INVITE(IAM UUInd, UUInf)
	ALERTING(FAC uus1rr)				180 Ringing(ACM)
	CONN				200 OK INVITE(ANM)
					ACK
			Communication		
	BYE(REL)	←		←	DISC
	200 OK BYE	→		→	REL
				←	REL_COM

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

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

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

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

TP706101	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 1.2.5.2.1.1.2, 1.2.5.2.2-5.1/ Q.737 [i.10]**
TSS reference	ISDN-(ISUP)-SIP/SS/UUS2				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS2 explicit non-essential - request and acceptance** To verify that the IUT can successfully originate/transit a call with an UUS2 explicit non-essential request, having the **user-to-user indicators** in the **IAM** set to "request, not essential". To verify that the IUT can successfully complete a call with an UUS2 explicit non-essential request, having the **user-to-user indicators** parameter in the **ACM** or **CPG** set to "service provided".				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP	→		→	INVITE(IAM UU2 not ess)
	ALERTING	←		←	180 Ringing(ACM)
	USER INFO	→		→	INFO(USR)
				←	200 OK INFO
	USER INFO	←		←	INFO(USR)
				→	200 OK INFO
	CONN	←		←	200 OK INVITE(ANM)
			Communication		
	DISC	←		←	BYE(REL)
	RELEASE	→		→	200 OK BYE
	REL_COM	←			

TP706102	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 1.2.5.2.5.2.2, 1.2.5.2.2-5.2/ Q.737 [i.10]**
TSS reference	ISDN-(ISUP)-SIP/SS/UUS2				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS2 explicit non-essential - explicit rejection (service not provided)** To verify that the UUS2 service can be rejected and the **user-to-user indicators** in the **ACM** or **CPG** are set to "service 2 not provided" (see note).				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP(FAC uus2reqness)	→		→	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
		Communication			
	BYE(REL)	←		←	DISC
	200 OK BYE(RLC)	→		→	REL
				←	REL_COM
NOTE: The network or the user cannot support UUS2.					

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

TP706104	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 1.2.5.2.1.1.2, 1.2.5.2.2-5.1/ Q.737 [i.10]**
TSS reference	ISDN-(ISUP)-SIP/SS/UUS2				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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	→		→	INVITE(IAM UU2 ess)
	ALERTING	←		←	180 Ringing(ACM)
	USER INFO	→		→	INFO(USR)
				←	200 OK INFO
	USER INFO	←		←	INFO(USR)
				→	200 OK INFO
	CONN	←		←	200 OK INVITE(ANM)
			Communication		
	DISC	←		←	BYE(REL)
	RELEASE	→		→	200 OK BYE
	REL_COM	←			

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

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

TP706107	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 1.2 5.2.5.2.1, 1.2.5.2.2-5.2/ Q.737 [i.10]**
TSS reference	ISDN-(ISUP)-SIP/SS/UUS2				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS2 explicit essential - implicit rejection** To verify that the service can be rejected if no indication is received (no **user-to-user indicators** parameter) in the first backward message (implicit rejection of service 2) (see note).				
SIP parameter values	180 Ringing: the encapsulated ACM does not contain an user-to-user response indicator				
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP	→		→	INVITE(IAM UU2 ess)
				←	180 Ringing(ACM)
	RELEASE	←		→	CANCEL(REL)
	REL_COMP	→		←	200 OK CANCEL
				←	487 Request Terminated
				→	ACK
NOTE: The remote network does not understand the service 2 request or the remote user cannot support UUS2.					

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

<table>
<tr><th>TP706201</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
clauses 1.3.5.2.1.1.2, 1.3.5.2.2-5.1/
Q.737 [i.10]</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-(ISUP)-SIP/SS/UUS3</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">UUS3 explicit non-essential - request and acceptance
To verify that the IUT can successfully originate/transit a call with an UUS3 explicit non-essential request, having the user-to-user indicators in the IAM set to "request, not essential".
To verify that the IUT can successfully complete a call with an UUS3 explicit non-essential request, having the Service 3 field in the user-to-user indicators parameter in the ANM or CON set to "service provided".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="12">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP(UU3 req not ess)</td><td>→</td><td></td><td>→</td><td>INVITE(IAM UU3 not ess)</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>CONN(UU3 ret res)</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM UU3 prov)</td></tr>
<tr><td></td><td colspan="3">Communication</td><td></td></tr>
<tr><td>USER INFO</td><td>→</td><td></td><td>→</td><td>INFO(USR)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td>USER INFO</td><td>←</td><td></td><td>←</td><td>INFO(USR)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>200 OK INFO</td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RELEASE</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td>REL_COM</td><td>→</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><td>TP706202</td><td colspan="5">SIP reference: RFC 3261 [4]</td><td>ISUP reference:
Q.1912.5 [1],
clauses 1.3.5.2.1.1.2, 1.3.5.2.2-5.1/
Q.737 [i.10]</td></tr>
<tr><td>TSS reference</td><td colspan="6">ISDN-(ISUP)-SIP/SS/UUS3</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">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".</td></tr>
<tr><td>SIP parameter values</td><td colspan="6"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="6"></td></tr>
<tr><td rowspan="17">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td colspan="2">SIP</td></tr>
<tr><td>SETUP(UU3 req ess)</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(IAM UU3 ess)</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing(ACM)</td></tr>
<tr><td>CONN(UU3 ret res)</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE(ANM UU3 prov)</td></tr>
<tr><td></td><td colspan="3">Communication</td><td colspan="2"></td></tr>
<tr><td>USER INFO</td><td>→</td><td></td><td>→</td><td colspan="2">INFO(USR)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK INFO</td></tr>
<tr><td>USER INFO</td><td>←</td><td></td><td>←</td><td colspan="2">INFO(USR)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">200 OK INFO</td></tr>
<tr><td>USER INFO</td><td>→</td><td></td><td>→</td><td colspan="2">INFO(USR)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK INFO</td></tr>
<tr><td>USER INFO</td><td>←</td><td></td><td>←</td><td colspan="2">INFO(USR)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">200 OK INFO</td></tr>
<tr><td></td><td colspan="3">Communication</td><td colspan="2"></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td colspan="2">BYE(REL)</td></tr>
<tr><td>RELEASE</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK BYE</td></tr>
<tr><td>REL_COM</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
</table>

<table>
<tr><td>TP706203</td><td colspan="5">SIP reference: RFC 3261 [4]</td><td>ISUP reference:
Q.1912.5 [1],
clauses 1.3.5.2.5.2.2, 1.3.5.2.2-5.2/
Q.737 [i.10]</td></tr>
<tr><td>TSS reference</td><td colspan="6">ISDN-(ISUP)-SIP/SS/UUS3</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">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).</td></tr>
<tr><td>SIP parameter values</td><td colspan="6"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="6"></td></tr>
<tr><td rowspan="4">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td colspan="2">SIP</td></tr>
<tr><td>SETUP</td><td>←</td><td></td><td>←</td><td colspan="2">INVITE(IAM UU2 ess)</td></tr>
<tr><td>RELEASE</td><td>→</td><td></td><td>→</td><td colspan="2">500 Server Internal error(REL#29, 69)</td></tr>
<tr><td>REL_COM</td><td>←</td><td></td><td>←</td><td colspan="2">ACK</td></tr>
<tr><td colspan="7">NOTE: The network or the called user cannot support the service.</td></tr>
</table>

TP706204	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 1.3.5.2.1.1.2, 1.3.5.2.2-5.1/ Q.737 [i.10]**
TSS reference	ISDN-(ISUP)-SIP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS3 explicit non-essential - request and acceptance during the active phase of the call** To verify that the IUT can successfully generate/transit an UUS3 explicit non-essential request, with a **FAR** having the **facility indicator** parameter set to "user-to-user service" and the Service 3 field in the **user-to-user indicators** set to "request, not essential". To verify that the IUT can successfully reply to an UUS3 explicit non-essential request with a **FAA** having the **facility indicator** parameter set to "user-to-user service" and the Service 3 field in the **user-to-user indicators** parameter set to "service provided".				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE
			Communication		
	FAC(UU3 req not ess)	→		→	INFO(FAR req UU3 not ess)
				←	200 OK INFO
	FAC(UU3 ret res)	←		←	INFO(FAR resp UU3 prov)
	USER INFO	→		→	INFO(USR)
				←	200 OK INFO
	USER INFO	←		←	INFO(USR)
				→	200 OK INFO
	USER INFO	→		→	INFO(USR)
				←	200 OK INFO
	USER INFO	←		←	INFO(USR)
				→	200 OK INFO
			Communication		
	DISC	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE
	REL_COM	→			

<table>
<tr><td>TP706205</td><td colspan="5">SIP reference: RFC 3261 [4]</td><td>ISUP reference:
Q.1912.5 [1],
clause 1.3.5.2.5.2.2/Q.737 [i.10]</td></tr>
<tr><td>TSS reference</td><td colspan="6">ISDN-(ISUP)-SIP/SS/UUS3</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">UUS3 explicit non-essential - explicit rejection during call (service not provided - in FRJ)
To verify that the UUS3 explicit non-essential service can be rejected during the active phase of the call and the Service 3 field in the user-to-user indicators in the FRJ are set to "service 3 not provided".</td></tr>
<tr><td>SIP parameter values</td><td colspan="6"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="6"></td></tr>
<tr><td rowspan="13">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td colspan="2">SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(IAM)</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing(ACM)</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE</td></tr>
<tr><td></td><td colspan="3">Communication</td><td colspan="2"></td></tr>
<tr><td>FAC(UU3 req not ess)</td><td>→</td><td></td><td>→</td><td colspan="2">INFO(FAR req UU3 not ess)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK INFO</td></tr>
<tr><td>FAC(UU3 ret err)</td><td>←</td><td></td><td>←</td><td colspan="2">INFO(FRJ resp UU3 not prov)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">200 OK INFO</td></tr>
<tr><td></td><td colspan="3">Communication</td><td colspan="2"></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td colspan="2">BYE(REL)</td></tr>
<tr><td>RELEASE</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK BYE</td></tr>
<tr><td>REL_COM</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
</table>

<table>
<tr><td>TP706206</td><td colspan="5">SIP reference: RFC 3261 [4]</td><td>ISUP reference:
Q.1912.5 [1],
clause 1.3.5.2.5.2.2/Q.737 [i.10]</td></tr>
<tr><td>TSS reference</td><td colspan="6">ISDN-(ISUP)-SIP/SS/UUS3</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6">PICS 11/3</td></tr>
<tr><td>Test purpose</td><td colspan="6">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="6"></td></tr>
<tr><td rowspan="11">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td colspan="2">SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(IAM)</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing(ACM)</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE</td></tr>
<tr><td></td><td colspan="3">Communication</td><td colspan="2"></td></tr>
<tr><td>FAC(UU3 req not ess)</td><td>→</td><td></td><td>→</td><td colspan="2">INFO(FAR req UU3 not ess)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK INFO</td></tr>
<tr><td></td><td colspan="3">Communication</td><td colspan="2"></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td colspan="2">BYE(REL)</td></tr>
<tr><td>RELEASE</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK BYE</td></tr>
<tr><td>REL_COM</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
</table>

A.1.1.2.7 Closed User Group (CUG)

TP707001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 1.5.2.1.1 i) a)/Q.735 [i.9]**
TSS reference	ISDN-(ISUP)-SIP/SS/CUG				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**CUG without outgoing access in IAM** To verify that the IUT can successfully establish a CUG call by including the **CUG interlock code** together with an indication of "CUG call, outgoing access not allowed" in the **optional forward call indicators** in the **IAM**. The ISUP preference indicator of the **forward call indicators** in the **IAM** should be set to "ISUP required all the way".				
SIP parameter values					
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP	→		→	INVITE(IAM, CUG -OA)
	ALERTING	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
				→	ACK
			Communication		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	REL_COM	→			

TP707002	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 1.5.2.5.1; table 1-2/Q.735 [i.9]**
TSS reference	ISDN-(ISUP)-SIP/SS/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	→		→	180 Ringing(ACM)
	CONN	→		→	200 OK INVITE(ANM)
			Communication		
	DISC	←		←	BYE(REL)
	REL	→		→	200 OK BYE
	REL_COM	←			

TP707002	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 1.5.2.5.1; table 1-2/Q.735 [i.9]
TSS reference	ISDN-(ISUP)-SIP/SS/CUG				
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)
				→	500 Server Internal Error(REL#55)
				←	ACK

TP707003	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 1.5.2.5.1; table 1-2/Q.735 [i.9]
TSS reference	ISDN-(ISUP)-SIP/SS/CUG				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**CUG call without outgoing access; class of called user CUG with IA and no ICB activated** To verify that the IUT can successfully establish a CUG call.				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP	←		←	INVITE(IAM, CUG -OA)
	ALERTING	→		→	180 Ringing(ACM)
	CONN	→		→	200 OK INVITE(ANM)
			Communication		
	DISC	←		←	BYE(REL)
	REL	→		→	200 OK BYE
	REL_COM	←			

TP707004	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 1.5.2.5.1; table 1-2/Q.735 [i.9]**
TSS reference	ISDN-(ISUP)-SIP/SS/CUG				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**CUG call with outgoing access; class of called user CUG with IA and no ICB activated** To verify that the IUT can successfully establish a CUG call with outgoing access.				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP	←		←	INVITE(IAM, CUG +OA)
	ALERTING	→		→	180 Ringing(ACM)
	CONN	→		→	200 OK INVITE(ANM)
			Communication		
	DISC	←		←	BYE(REL)
	REL	→		→	200 OK BYE
	REL_COM	←			

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

TP707006	**SIP reference: RFC 3261 [4]**					**ISUP reference: Q.1912.5 [1], Q.730 [i.1]**
TSS reference	ISDN-(ISUP)-SIP/SS/CUG					
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)	
				→	500 Server Internal Error(REL#87)	
				←	ACK	

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

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

TP707009	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 1.5.2.5.1; table 1-2/Q.735 [i.9]**
TSS reference	ISDN-(ISUP)-SIP/SS/CUG				
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	→		→	180 Ringing(ACM)
	CONN	→		→	200 OK INVITE(ANM)
				←	ACK
		Communication			
	DISC	←		←	BYE(REL)
	REL	→		→	200 OK BYE(RLC)
	REL_COM	←			

TP707010	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 1.5.2.5.1; table 1-2/Q.735 [i.9]**
TSS reference	ISDN-(ISUP)-SIP/SS/CUG				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**CUG call 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)
				→	500 Server Internal Error(REL#87)
				←	ACK

TP707011	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 1.5.2.5.1; table 1-2/Q.735 [i.9]**
TSS reference	ISDN-(ISUP)-SIP/SS/CUG				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**CUG call with outgoing access; class of called user 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)
				→	500 Server Internal Error(REL#87)
				←	ACK

TP707012	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 1.5.2.5.1; table 1-2/Q.735 [i.9]**
TSS reference	ISDN-(ISUP)-SIP/SS/CUG				
SIP selection criteria					
ISUP selection criteria					
Test purpose	CUG call 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)
				→	500 Server Internal Error(REL#87)
				←	ACK

TP707013	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 1.5.2.5.1; table 1-2/Q.735 [i.9]**
TSS reference	ISDN-(ISUP)-SIP/SS/CUG				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**CUG call with outgoing access; class of called user other CUG with IA** To verify that the IUT can successfully establish a non-CUG call.				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN (CUG A +IA)**		**SUT**		**SIP-I**
	SETUP	←		←	INVITE(IAM, CUG B +OA)
	ALERTING	→		→	180 Ringing(ACM)
	CONN	→		→	200 OK INVITE(ANM)
			Communication		
	DISC	←		←	BYE(REL)
	REL	→		→	200 OK BYE
	REL_COM	←			

A.1.1.2.8 SUB-addressing (SUB)

TP708001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 8.5.2.1.1/Q.731 [i.2]**
TSS reference	ISDN-(ISUP)-SIP/SS/SUB				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Sending the called sub-address in the access transport parameter** To verify that the IUT can include the called sub-address in the **access transport** parameter in the IAM.				
SIP parameter values	INVITE: IAM encapsulated in the MIME body ATP with called sub-address included				
ISDN parameter values	SETUP: called sub-address included				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP(SUB)	→		→	INVITE(IAM, ATP(SUB))
	ALERTING	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
				→	ACK
			Communication		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	REL_COM	→			

TP708002	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 8.5.2.5.1/Q.731 [i.2]**
TSS reference	ISDN-(ISUP)-SIP/SS/SUB				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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 encapsulated in the MIME body ATP with called sub-address included				
ISDN parameter values	SETUP: called sub-address included				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP(SUB)	←		←	INVITE(IAM, ATP(SUB))
	ALERTING	→		→	180 Ringing(ACM)
	CONN	→		→	200 OK INVITE(ANM)
				←	ACK
			Communication		
	DISC	←		←	BYE(REL)
	REL	→		→	200 OK BYE(RLC)
	REL_COM	←			

A.1.1.2.9 Malicious Call Identification (MCID)

<table>
<tr><th>TP709001</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
clause 7.5.2.1.1/Q.731.7 [i.3]</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-(ISUP)-SIP/SS/MCID</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Successful MCID request
To verify that the IUT can successfully reply to an IDR having the MCID request indicator set to "MCID request" by sending an IRS with MCID response indicator set to "MCID included" and the calling party number included.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">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</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="13">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>INFO(IDR requested)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>INFO(IRS included)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Communication</td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td>REL_COM</td><td>→</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><th>TP709002</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
clause 7.5.2.1.1/Q.731.7 [i.3]</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-(ISUP)-SIP/SS/MCID</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Successful MCID request
To verify that the IUT can successfully reply to an IDR having the MCID request indicator set to "MCID request" by sending an IRS with MCID response indicator set to "MCID included" and the calling party number included.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">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</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="13">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP</td><td>←</td><td></td><td>←</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>INFO(IDR requested)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>INFO(IRS included)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>200 OK INFO</td></tr>
<tr><td>ALERTING</td><td>→</td><td></td><td>→</td><td>180 Ringing(ACM)</td></tr>
<tr><td>CONN</td><td>→</td><td></td><td>→</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Communication</td><td></td></tr>
<tr><td>DISC</td><td>←</td><td></td><td>←</td><td>BYE(REL)</td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>200 OK BYE</td></tr>
<tr><td>REL_COM</td><td>←</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><th>TP709003</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
clause 7.5.2.1.1/Q.731.7 [i.3]</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-(ISUP)-SIP/SS/MCID</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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).</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">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</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="14">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>INFO(IDR requested)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>INFO(IRS included)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Communication</td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td>REL_COM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td colspan="6">NOTE: This situation may occur e.g. if the call has been forwarded before reaching the destination.</td></tr>
</table>

<table>
<tr><th>TP709004</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
clause 7.5.2.1.1/Q.731.7 [i.3]</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-(ISUP)-SIP/SS/MCID</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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).</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">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</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="13">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP</td><td>←</td><td></td><td>←</td><td>INVITE(IAM)</td></tr>
<tr><td>ALERTING</td><td>→</td><td></td><td>→</td><td>180 Ringing(ACM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>INFO(IDR requested)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>INFO(IRS included)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>200 OK INFO</td></tr>
<tr><td>CONN</td><td>→</td><td></td><td>→</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Communication</td><td></td></tr>
<tr><td>DISC</td><td>←</td><td></td><td>←</td><td>BYE(REL)</td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>200 OK BYE</td></tr>
<tr><td>REL_COM</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td colspan="6">NOTE: This situation may occur e.g. if the call has been forwarded before reaching the destination.</td></tr>
</table>

TP709005	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.5.2.1.1/Q.731.7 [i.3]**	
TSS reference	ISDN-(ISUP)-SIP/SS/MCID				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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	→		→	INVITE(IAM)
				←	INFO(IDR requested)
				→	200 OK INFO
				→	INFO(IRS included)
				←	200 OK INFO
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE
	REL_COM	→			

TP70906	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.5.2.1.2/Q.731.7 [i.3]**	
TSS reference	ISDN-(ISUP)-SIP/SS/MCID				
SIP selection criteria					
ISUP selection criteria	PICS 9/1				
Test purpose	**MCID request - MCID not supported by the OLE** To verify that the IUT rejects a MCID request by sending a **IRS** with the **MCID response indicator** set to "MCID not included".				
SIP parameter values	INFO: The encapsulated IDR contains the MCID Request indicator "requested" INFO: The encapsulated IRS contains the MCID response indicator "not included"				
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP	→		→	INVITE(IAM)
				←	INFO(IDR requested)
				→	200 OK INFO
				→	INFO(IRS not included)
				←	200 OK INFO
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE
	REL_COM	→			

TP70907	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.5.2.1.2/Q.731.7 [i.3]**	
TSS reference	ISDN-(ISUP)-SIP/SS/MCID				
SIP selection criteria					
ISUP selection criteria	PICS 9/1				
Test purpose	**MCID request - MCID not supported by the OLE** To verify that the IUT rejects a MCID request by sending a **IRS** with the **MCID response indicator** set to "MCID not included".				
SIP parameter values	INFO: The encapsulated IDR contains the MCID Request indicator "requested" INFO: The encapsulated IRS contains the MCID response indicator "not included"				
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP	←		←	INVITE(IAM)
				→	INFO(IDR requested)
				←	200 OK INFO
				←	INFO(IRS not included)
				→	200 OK INFO
	ALERTING	→		→	180 Ringing(ACM)
	CONN	→		→	200 OK INVITE(ANM)
				←	ACK
			Communication		
	DISC	←		←	BYE(REL)
	REL	→		→	200 OK BYE
	REL_COM	←			

TP70908	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.5.2.5.2/Q.731.7 [i.3]**	
TSS reference	ISDN-(ISUP)-SIP/SS/MCID				
SIP selection criteria					
ISUP selection criteria	PICS 5/9				
Test purpose	**MCID timer (T39) expiry** To verify that call setup is continued (user is alerted) if no **IRS** is received within timer T39 expiry, after having sent the **IDR** with **MCID request indicator** set to "MCID requested".				
SIP parameter values	INFO: The encapsulated IDR contains the MCID Request indicator "requested"				
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP	→		→	INVITE(IAM)
				←	INFO(IDR requested)
				→	200 OK INFO
			T39 expiry		
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE
	REL_COM	→			

A.1.1.2.10 Conference call (CONF)

TP710001	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.5.2.1.1.2
TSS reference	ISDN-(ISUP)-SIP/SS/CONF	
SIP selection criteria		
ISUP selection criteria		
Test purpose	Verify that the IUT can successfully begin the conference **from an active call** and notify the implied parties correctly. Begin the conference and check that notification "conference established" is received in the (INFO/CPG). Release the call at the end terminal and check that all network resources are released (see note).	
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body	
ISDN parameter values	FAC: BeginCONF invoke component FAC: BeginCONF return result component	
Comments		

ISDN	**SUT**			**SIP 1**		**SIP 2**
SETUP(CRx)	→		→	INVITE		
ALERTING	←		←	180 Ringing		
CONN	←		←	200 OK INVITE		
FAC(**BeginCONF_inv)**	→		→	INFO(CPG conf est)		
FAC(**BeginCONF_rr)**	←		←	200 OK INFO		
			Conference communication			
DISC(CRx)	→		→	BYE		
RELEASE	←		←	200 OK BYE		
REL_COMP	→					

NOTE: The **generic notification indicator** set to "conference established" should be sent by the IUT in the **CPG**. The event indicator should be set to "progress".

TP710002	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.5.2.1.1.2**		
TSS reference	ISDN-(ISUP)-SIP/SS/CONF					
SIP selection criteria						
ISUP selection criteria						
Test purpose	**Verify that the IUT can successfully begin the conference from idle state and is able to add a conferee to a conference and notify the implied parties correctly.** Establish a conference from ISDN to SIP 1. Established a connection to SIP 2 and add this party to the conference. Notify subscriber at SIP 1 by sending him/her "other party added" in the CPG. The conference is released by call clearing by the served user at IADN (see note).					
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body					
ISDN parameter values	FAC: BeginCONF invoke component FAC: BeginCONF return result component FAC: addCONF invoke component DISC: addCONF return result component					
Comments						
ISDN	**SUT**			**SIP 1**		**SIP 2**
SETUP(CRx)	→		→	INVITE		
ALERTING	←		←	180 Ringing		
CONN	←		←	200 OK INVITE		
FAC(**BeginCONF_inv**)	→		→	INFO(CPG conf est)		
FAC(**BeginCONF_rr**)	←		←	200 OK INFO		
SETUP(CRy)	→				→	INVITE
ALERTING	←				←	180 Ringing
CONN	←				←	200 OK INVITE
FAC(**AddCONF_inv**)	→				→	INFO(CPG conf est)
DISC(**AddCONF_rr**,CRy)	←				←	200 OK INFO
RELEASE	→		→	INFO(CPG party add)		
REL_COMP	←		←	200 OK INFO		
Conference communication						
DISC(CRx)	→		→	BYE		
RELEASE	←		←	200 OK BYE		
REL_COMP	→					
					→	BYE
					←	200 OK BYE
NOTE: The **generic notification indicator** set to "conference established" should be sent by the IUT to the new affected conferee and the **generic notification indicator** set to "other party added" to the non-affected conferees. The event indicator in the **CPG** should be set to "progress".						

TP710003	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.5.2.1.1.3**
TSS reference	ISDN-(ISUP)-SIP/SS/CONF	
SIP selection criteria		
ISUP selection criteria		
Test purpose	Verify that the IUT can successfully isolate a conferee from the conference and notify the implied parties correctly. Establish a conference from ISDN to SIP 1. Add SIP 2 to the conference and notify subscriber at SIP 1 by sending him/her "other party added" in the CPG. Isolate a conferee and check that the notification "isolated" is received in the CPG. Reattach the conferee. The conference is released by call clearing by the served user at ISDN (see note).	
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body	
ISDN parameter values	FAC: BeginCONF invoke component FAC: BeginCONF return result component FAC: addCONF invoke component DISC: addCONF return result component FAC: isolateCONF invoke component FAC: isolateCONF return result component FAC reattachCONF invoke component FAC: reattachCONF return result component	
Comments		

ISDN	**SUT**			**SIP 1**		**SIP 2**
SETUP(CRx)	→		→	INVITE		
ALERTING	←		←	180 Ringing		
CONN	←		←	200 OK INVITE		
FAC(**BeginCONF**_inv**)**	→		→	INFO(CPG conf est)		
FAC(**BeginCONF**_rr**)**	←		←	200 OK INFO		
SETUP(CRy)	→				→	INVITE
ALERTING	←				←	180 Ringing
CONN	←				←	200 OK INVITE
FAC(**AddCONF**_inv,CRy**)**	→				→	INFO(CPG conf est)
DISC(**AddCONF**_rr,CRy**)**	←				←	200 OK INFO
RELEASE	→		→	INFO(CPG party add)		
REL_COMP	←		←	200 OK INFO		
Conference communication						
FAC(**IsolConf**_inv,CRx)	→				→	INFO(CPG isol)
FAC(**IsolConf**_rr,CRx)	←				←	200 OK INFO
			→	INFO(CPG party isol)		
			←	200 OK INFO		
Private communication ISDN with SIP 1						
FAC(**ReattConf**_inv,CRx)	→				→	INFO(CPG reatt)
FAC(**ReattConf**_rr,CRx)	←				←	200 OK INFO
			→	INFO(CPG party reatt)		
			←	200 OK INFO		
Conference communication						
DISC(CRx)	→		→	BYE		
RELEASE	←		←	200 OK BYE		
REL_COMP	→					
					→	BYE
					←	200 OK BYE
NOTE: The **generic notification indicator** set to "isolated" within **call progress** should be sent by the IUT to the affected conferee and the **generic notification indicator** set to "other party isolated" should be sent to the non-affected conferees. The event indicator in the **CPG** should be set to "progress". The isolated conferee should not be able to communicate with the rest of the conference.						

TP710004	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.5.2.1.1.5**
TSS reference:	ISDN-(ISUP)-SIP/SS/CONF	
SIP selection criteria		
ISUP selection criteria		
Test purpose	Verify that the IUT can create a **private communication** between the served user and one of the conferees and notify the implied parties correctly (see notes).	
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body	
ISDN parameter values	FAC: BeginCONF invoke component FAC: BeginCONF return result component FAC: addCONF invoke component DISC: addCONF return result component SETUP: splitCONF invoke component CONNECT: splitCONF return result component	
Comments		

ISDN	**SUT**			**SIP 1**		**SIP 2**
SETUP(CRx)	→		→	INVITE		
ALERTING	←		←	180 Ringing		
CONN	←		←	200 OK INVITE		
FAC(**BeginCONF**_inv**)**	→		→	INFO(CPG conf est)		
FAC(**BeginCONF**_rr**)**	←		←	200 OK INFO		
SETUP(CRy)	→				→	INVITE
ALERTING	←				←	180 Ringing
CONN	←				←	200 OK INVITE
FAC(**AddCONF**_inv,CRy**)**	→				→	INFO(CPG conf est)
DISC(**AddCONF**_rr,CRy**)**	←				←	200 OK INFO
RELEASE	→		→	INFO(CPG party add)		
REL_COMP	←		←	200 OK INFO		
Conference communication						
SETUP(**SplitConf**_inv,CRy)	→				→	INFO(CPG conf disc)
CONN(**SplitlConf**_rr,CRy)	←				←	200 OK INFO
			→	INFO(CPG party split)		
			←	200 OK INFO		
Private communication ISDN with SIP 1						
FAC(**AddCONF**_inv,CRy**)**	→				→	INFO(CPG conf est)
DISC(**AddCONF**_rr,CRy**)**	←				←	200 OK INFO
RELEASE	→		→	INFO(CPG party add)		
REL_COMP	←		←	200 OK INFO		
Conference communication						
DISC(CRx)	→		→	BYE		
RELEASE	←		←	200 OK BYE		
REL_COMP	→					
					→	BYE
					←	200 OK BYE

NOTE 1: The **generic notification indicator** set to "conference disconnected" should be sent by the IUT to the affected conferee and the **generic notification indicator** set to "other party split" should be sent to the non-affected conferees. The event indicator in the **CPG** should be set to "progress". The non--affected conferees should not be able to participate in the communication of the private communication.

NOTE 2: See also figure 1-5/ITU-T Recommendation Q.734 [i.7].

TP710005	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.5.2.1.1.6
TSS reference	ISDN-(ISUP)-SIP/SS/CONF	
SIP selection criteria		
ISUP selection criteria		
Test purpose	To verify that IUT can successfully disconnect a conferee from the conference, if requested by the served user, and notify the implied parties correctly. Establish a conference from ISDN to SIP 1. Add SIP 2 to the conference and notify subscriber at SIP 1 by sending him/her "other party added" in the CPG. Release the dropped party at SIP 2. The conference is released by call clearing by the served user at ISDN (see note).	
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body	
ISDN parameter values	FAC: BeginCONF invoke component FAC: BeginCONF return result component FAC: addCONF invoke component DISC: addCONF return result component FAC: dropCONF invoke component FAC: dropCONF return result component	
Comments		

ISDN	**SUT**			**SIP 1**		**SIP 2**
SETUP(CRx)	→		→	INVITE		
ALERTING	←		←	180 Ringing		
CONN	←		←	200 OK INVITE		
FAC(**BeginCONF**_inv**)**	→		→	INFO(CPG conf est)		
FAC(**BeginCONF**_rr**)**	←		←	200 OK INFO		
SETUP(CRy)	→				→	INVITE
ALERTING	←				←	180 Ringing
CONN	←				←	200 OK INVITE
FAC(**AddCONF**_inv,CRy**)**	→				→	INFO(CPG conf est)
DISC(**AddCONF**_rr,CRy**)**	←				←	200 OK INFO
RELEASE	→		→	INFO(CPG party add)		
REL_COMP	←		←	200 OK INFO		
Conference communication						
FAC(**DropCONF**_inv,CRx**)**	→				→	BYE
FAC(**DropCONF**_rr,CRx**)**	←				←	200 OK BYE
			→	INFO(CPG party disc)		
			←	200 OK INFO		
Communication						
DISC(CRx)	→		→	BYE		
RELEASE	←		←	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 **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 reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.5.2.1.1.7**
TSS reference	ISDN-(ISUP)-SIP/SS/CONF	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**To verify that IUT can successfully disconnect a conferee from the conference, if requested by the conferee, and notify the implied parties correctly** Establish a conference from ISDN to SIP 1. Add SIP 2 to the conference and notify subscriber at SIP 1 by sending him/her "other party added" in the CPG. Release request by the conferee at SIP 1. The conference is released by call clearing by the served user at ISDN (see note).	
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body	
ISDN parameter values	FAC: BeginCONF invoke component FAC: BeginCONF return result component FAC: addCONF invoke component DISC: addCONF return result component FAC: partyDISC invoke component FAC: partyDISC return result component	
Comments		

ISDN	**SUT**			**SIP 1**		**SIP 2**
SETUP(CRx)	→		→	INVITE		
ALERTING	←		←	180 Ringing		
CONN	←		←	200 OK INVITE		
FAC(**BeginCONF**_inv**)**	→		→	INFO(CPG conf est)		
FAC(**BeginCONF**_rr**)**	←		←	200 OK INFO		
SETUP(CRy)	→				→	INVITE
ALERTING	←				←	180 Ringing
CONN	←				←	200 OK INVITE
FAC(**AddCONF**_inv,CRy**)**	→				→	INFO(CPG conf est)
DISC(**AddCONF**_rr,CRy**)**	←				←	200 OK INFO
RELEASE	→		→	INFO(CPG party add)		
REL_COMP	←		←	200 OK INFO		
Conference communication						
FAC(**PartyDisc**_inv,CRy**)**	←		←	BYE		
FAC(**PartyDisc**_rr,CRy**)**	→		→	200 OK BYE		
					→	INFO(CPG party disc)
					←	200 OK INFO
Communication						
DISC(CRx)	→				→	BYE
RELEASE	←				←	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 reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.5.2.1.1.8
TSS reference	ISDN-(ISUP)-SIP/SS/CONF	
SIP selection criteria		
ISUP selection criteria		
Test purpose	To verify that IUT can successfully disconnect all conferees from the conference, if requested by the served user, and initiate the normal call release procedure towards each conferee. Establish a conference from ISDN to SIP 1. Add SIP 2 to the conference and notify subscriber at SIP 1 by sending him/her "other party added" in the CPG. The conference is released by call clearing by the served user at ISDN (see note).	
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body	
ISDN parameter values	FAC: BeginCONF invoke component FAC: BeginCONF return result component FAC: addCONF invoke component DISC: addCONF return result component	
Comments		

ISDN	**SUT**			**SIP 1**		**SIP 2**
SETUP(CRx)	→		→	INVITE		
ALERTING	←		←	180 Ringing		
CONN	←		←	200 OK INVITE		
FAC(**BeginCONF**_inv**)**	→		→	INFO(CPG conf est)		
FAC(**BeginCONF**_rr**)**	←		←	200 OK INFO		
SETUP(CRy)	→				→	INVITE
ALERTING	←				←	180 Ringing
CONN	←				←	200 OK INVITE
FAC(**AddCONF**_inv,CRy**)**	→				→	INFO(CPG conf est)
DISC(**AddCONF**_rr,CRy**)**	←				←	200 OK INFO
RELEASE	→		→	INFO(CPG party add)		
REL_COMP	←		←	200 OK INFO		
Conference communication						
DISC(CRx)	→		→	BYE		
RELEASE	←		←	200 OK BYE		
REL_COMP	→				→	INFO(CPG party disc)
					←	200 OK INFO
					→	BYE
					←	200 OK BYE
NOTE: The IUT should send **REL** to all conferees connected to the conference.						

TP710008	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15
TSS reference	ISDN-(ISUP)-SIP/SS/CONF	
SIP selection criteria		
ISUP selection criteria		
Test purpose	Verify that no retrieve notification is sent to a user put on hold and subsequently added to a conference call, but that the IUT sends the "conference established" notification to the held user.	
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body	
ISDN parameter values	FAC: BeginCONF invoke component FAC: BeginCONF return result component FAC: addCONF invoke component DISC: addCONF return result component	
Comments		

ISDN	**SUT**			**SIP 1**		**SIP 2**
SETUP(CRx)	→		→	INVITE		
ALERTING	←		←	180 Ringing		
CONN	←		←	200 OK INVITE		
FAC(**BeginCONF**_inv**)**	→		→	INFO(CPG conf est)		
FAC(**BeginCONF**_rr**)**	←		←	200 OK INFO		
SETUP(CRy)	→				→	INVITE
ALERTING	←				←	180 Ringing
CONN	←				←	200 OK INVITE
HOLD	→				→	INFO(CPG hold)
					←	200 OK INFO
FAC(**AddCONF**_inv,CRy**)**	→				→	INFO(CPG conf est)
DISC(**AddCONF**_rr,CRy**)**	←				←	200 OK INFO
RELEASE	→		→	INFO(CPG party add)		
REL_COMP	←		←	200 OK INFO		
Conference communication						
DISC(CRx)	→		→	BYE		
RELEASE	←		←	200 OK BYE		
REL_COMP	→				→	INFO(CPG party disc)
					←	200 OK INFO
					→	BYE
					←	200 OK BYE

TP710009	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15**	
TSS reference	ISDN-(ISUP)-SIP/SS/CONF					
SIP selection criteria						
ISUP selection criteria						
Test purpose	To verify that no hold and no retrieve notification is sent to the conferees when the conference controller puts the conference on hold.					
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body					
ISDN parameter values	FAC: BeginCONF invoke component FAC: BeginCONF return result component FAC: addCONF invoke component DISC: addCONF return result component					
Comments						
ISDN	**SUT**			**SIP 1**		**SIP 2**
SETUP(CRx)	→		→	INVITE		
ALERTING	←		←	180 Ringing		
CONN	←		←	200 OK INVITE		
FAC(**BeginCONF**_inv**)**	→		→	INFO(CPG conf est)		
FAC(**BeginCONF**_rr**)**	←		←	200 OK INFO		
SETUP(CRy)	→				→	INVITE
ALERTING	←				←	180 Ringing
CONN	←				←	200 OK INVITE
FAC(**AddCONF**_inv,CRy**)**	→				→	INFO(CPG conf est)
DISC(**AddCONF**_rr,CRy**)**	←				←	200 OK INFO
RELEASE	→		→	INFO(CPG party add)		
REL_COMP	←		←	200 OK INFO		
Conference communication						
HOLD	→					
RETRIVE	→					
DISC(CRx)	→		→	BYE		
RELEASE	←		←	200 OK BYE		
REL_COMP	→				→	INFO(CPG party disc)
					←	200 OK INFO
					→	BYE
					←	200 OK BYE

A.1.1.2.11 Explicit Call Transfer (ECT)

TP711001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.5.2.1.1.1 a)/Q.732.7 [i.5]**		
TSS reference	ISDN-(ISUP)-SIP/SS/ECT						
SIP selection criteria							
ISUP selection criteria							
Test purpose	**Capability of storing and sending the additional calling party number in the call transfer number** To verify that the IUT is able to store the additional calling party number in the **generic number** when the **calling party number** and the **generic number** have been received from the remote user. This information is sent by the IUT to the other remote user in the **call transfer number** in either the **FAC** or **CPG** when the call transfer is activated.						
SIP parameter values	INVITE: encapsulated IAM contains the additional calling party number of user B INVITE B SDP sendonly, encapsulated CPG generic notification remote hold INFO C: encapsulated FAC contains generic notification call transfer active, call transfer number derived from the additional calling party number of user B (SIP-I 1)						
ISDN parameter values	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	→		→	180 Ringing(ACM)		
	CONN	→		→	200 OK INVITE(ANM)		
				←	ACK		
	HOLD	→		→	INVITE(CPG hold)		
				←	200 OK INVITE		
				→	ACK		
	SETUP	→				→	INVITE(IAM)
	ALERTING	←				←	180 Ringing(ACM)
	CONN	←				←	200 OK INVITE(ANM)
						→	ACK
	FAC(ECT invoke)	→					
	DISCONNECT(rr)	←		→	INFO (FAC ect active)		
	RELEASE	→		←	200 OK INFO		
	RELEASE COMPL	←				→	INFO (FAC ect active)
						←	200 OK INFO
					BYE(REL)	→	BYE(REL)
					200 OK BYE(RLC	←	200 OK BYE(RLC

<table>
<tr><td>TP711002</td><td colspan="6">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1],
clause 7.5.2.1.1.1 a)/Q.732.7 [i.5]</td></tr>
<tr><td>TSS reference</td><td colspan="8">ISDN-(ISUP)-SIP/SS/ECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="8"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="8"></td></tr>
<tr><td>Test purpose</td><td colspan="8">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 call transfer number in either the FAC or CPG when the call transfer is activated.</td></tr>
<tr><td>SIP parameter values</td><td colspan="8">INVITE: encapsulated IAM contains the calling party number of user B
INVITE B SDP sendonly, encapsulated CPG generic notification remote hold
INFO C: encapsulated FAC contains generic notification call transfer active, call transfer number derived from the calling party number of user B (SIP-I 1)</td></tr>
<tr><td>ISDN parameter values</td><td colspan="8">FAC: ECT invoke request component
DISCONNECT: ECT invoke return result component</td></tr>
<tr><td rowspan="22">Comments</td><td>ISDN 2</td><td></td><td>SUT</td><td></td><td>SIP-I 1</td><td></td><td colspan="2">SIP-I 3</td></tr>
<tr><td>SETUP</td><td>←</td><td></td><td>←</td><td>INVITE(IAM)</td><td></td><td colspan="2"></td></tr>
<tr><td>ALERTING</td><td>→</td><td></td><td>→</td><td>180 Ringing(ACM)</td><td></td><td colspan="2"></td></tr>
<tr><td>CONN</td><td>→</td><td></td><td>→</td><td>200 OK INVITE(ANM)</td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>ACK</td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>HOLD</td><td>→</td><td></td><td>→</td><td>INVITE(CPG hold)</td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE</td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td></td><td></td><td>→</td><td colspan="2">INVITE(IAM)</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td></td><td></td><td>←</td><td colspan="2">180 Ringing(ACM)</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td>FAC(ECT invoke)</td><td>→</td><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>DISCONNECT(rr)</td><td>←</td><td></td><td>→</td><td>INFO (FAC ect active)</td><td></td><td colspan="2"></td></tr>
<tr><td>RELEASE</td><td>→</td><td></td><td>←</td><td>200 OK INFO</td><td></td><td colspan="2"></td></tr>
<tr><td>RELEASE COMPL</td><td>←</td><td></td><td></td><td></td><td>→</td><td colspan="2">INFO (FAC ect active)</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK INFO</td></tr>
<tr><td colspan="8"></td></tr>
<tr><td></td><td></td><td></td><td></td><td>BYE(REL)</td><td>→</td><td colspan="2">BYE(REL)</td></tr>
<tr><td></td><td></td><td></td><td></td><td>200 OK BYE(RLC</td><td>←</td><td colspan="2">200 OK BYE(RLC</td></tr>
</table>

TP711003	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.5.2.1.1.1 b)/Q.732.7 [i.5]**		
TSS reference	ISDN-(ISUP)-SIP/SS/ECT						
SIP selection criteria							
ISUP selection criteria							
Test purpose	**Capability of storing and sending the additional connected number in the call transfer number** To verify that the IUT is able to store the additional connected number in the **generic number** when the **connected 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.						
SIP parameter values	INVITE B SDP sendonly, encapsulated CPG generic notification remote hold 200 OK INVITE: encapsulated ANM containing the additional connected number INFO 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 values	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	→		→	180 Ringing(ACM)		
	CONN	→		→	200 OK INVITE(ANM)		
				←	ACK		
	HOLD	→		→	INVITE(CPG hold)		
				←	200 OK INVITE		
				→	ACK		
	SETUP	→				→	INVITE(IAM)
	ALERTING	←				←	180 Ringing(ACM)
	CONN	←				←	200 OK INVITE(ANM)
						→	ACK
	FAC(ECT invoke)	→					
	DISCONNECT(rr)	←		→	INFO (FAC ect active)		
	RELEASE	→		←	200 OK INFO		
	RELEASE COMPL	←				→	INFO (FAC ect active)
						←	200 OK INFO
					BYE(REL)	→	BYE(REL)
					200 OK BYE(RLC	←	200 OK BYE(RLC

TP711004	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.5.2.1.1.1 b)/Q.732.7 [i.5]**		
TSS reference	ISDN-(ISUP)-SIP/SS/ECT						
SIP selection criteria							
ISUP selection criteria							
Test purpose	**Capability of storing and sending the connected number in call transfer number** To verify that the IUT is able to store **connected number** when only this COL has been received from the remote user. This information is sent by the IUT to the other remote user in the **call transfer number** in either the **FAC** or **CPG** when the call transfer is activated.						
SIP parameter values	INVITE B SDP sendonly, encapsulated CPG generic notification remote hold 200 OK INVITE: encapsulated ANM containing the connected number INFO B: encapsulated FAC contains generic notification call transfer active, call transfer number derived from the connected of user C (SIP-I 2)						
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	←		←	INVITE(IAM)		
	ALERTING	→		→	180 Ringing(ACM)		
	CONN	→		→	200 OK INVITE(ANM)		
				←	ACK		
	HOLD	→		→	INVITE(CPG hold)		
				←	200 OK INVITE		
				→	ACK		
	SETUP	→				→	INVITE(IAM)
	ALERTING	←				←	180 Ringing(ACM)
	CONN	←				←	200 OK INVITE(ANM)
						→	ACK
	FAC(ECT invoke)	→					
	DISCONNECT(rr)	←		→	INFO (FAC ect active)		
	RELEASE	→		←	200 OK INFO		
	RELEASE COMPL	←				→	INFO (FAC ect active)
						←	200 OK INFO
					BYE(REL)	→	BYE(REL)
					200 OK BYE(RLC	←	200 OK BYE(RLC

TP711005	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.5.2.1.1.2.1/Q.732.7 [i.5]**		
TSS reference	ISDN-(ISUP)-SIP/SS/ECT						
SIP selection criteria							
ISUP selection criteria							
Test purpose	**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 values	INFO: encapsulated LOP request, call transfer reference INFO: encapsulated LOP response, call transfer reference, response indicator: "no loop exists"						
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	←				←	180 Ringing(ACM)
	CONN	←				←	200 OK INVITE(ANM)
						→	ACK
				→	INFO(LOP request)		
				←	200 OK INFO		
						→	INFO(LOP request)
						←	200 OK INFO
				←	INFO(LOP response)		
				→	200 OK INFO		
						←	INFO(LOP response)
						→	200 OK INFO
	FAC(ECT invoke)	→					
	DISCONNECT(rr)	←		→	INFO (FAC ect active)		
	RELEASE	→		←	200 OK INFO		
	RELEASE COMPL	←				→	INFO (FAC ect active)
						←	200 OK INFO
					BYE(REL)	→	BYE(REL)
					200 OK BYE(RLC	←	200 OK BYE(RLC

TP711006	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.5.2.1.1.2.2 a)/Q.732.7 [i.5]**		
TSS reference	ISDN-(ISUP)-SIP/SS/ECT						
SIP selection criteria							
ISUP selection criteria							
Test purpose	**Facility message with generic notification sent to the remote user** To verify that the local exchange controlling the ECT can successfully initiate a call transfer by sending **FAC** with the **generic notification** set to "call transfer, active" or "call transfer, alerting" and the **service activation** parameter set to "call transfer".						
SIP parameter values	INFO B: encapsulated FAC contains generic notification call transfer active INFO C: encapsulated FAC contains generic notification call transfer active						
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	←		←	INVITE(IAM)		
	ALERTING	→		→	180 Ringing(ACM)		
	CONN	→		→	200 OK INVITE(ANM)		
				←	ACK		
	HOLD	→		→	INVITE(CPG hold)		
				←	200 OK INVITE		
				→	ACK		
	SETUP	→				→	INVITE(IAM)
	ALERTING	←				←	180 Ringing(ACM)
	CONN	←				←	200 OK INVITE(ANM)
						→	ACK
	FAC(ECT invoke)	→					
	DISCONNECT(rr)	←		→	INFO (FAC ect active)		
	RELEASE	→		←	200 OK INFO		
	RELEASE COMPL	←				→	INFO (FAC ect active)
						←	200 OK INFO
					BYE(REL)	→	BYE(REL)
					200 OK BYE(RLC	←	200 OK BYE(RLC

TP711007	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.5.2.1.1.2.2 a)/Q.732.7 [i.5]**		
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 **CPG** with the **generic notification** set to "call transfer, active" and the **service activation** parameter set to "call transfer".						
SIP parameter values	INFO C: encapsulated CPG contains generic notification call transfer active INFO B: encapsulated FAC contains generic notification call transfer alerting INFO B encapsulated FAC contains generic notification call transfer active						
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	←		←	INVITE(IAM)		
	ALERTING	→		→	180 Ringing(ACM)		
	CONN	→		→	200 OK INVITE(ANM)		
				←	ACK		
	HOLD	→		→	INVITE(CPG hold)		
				←	200 OK INVITE		
				→	ACK		
	SETUP	→				→	INVITE(IAM)
	ALERTING	←				←	180 Ringing(ACM)
	FAC(ECT invoke)	→					
	DISCONNECT(rr)	←		→	INFO (FAC ect alert)		
	RELEASE	→		←	200 OK INFO		
	RELEASE COMPL	←				→	INFO (CPG ect active)
						←	200 OK INFO
						←	200 OK INVITE(ANM)
						→	ACK
				→	INFO (FAC ect active)		
				←	200 OK INFO		
					BYE(REL)	→	BYE(REL)
					200 OK BYE(RLC	←	200 OK BYE(RLC

TP711008	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.5.2.1.1.2.2 b)/Q.732.7 [i.5]**		
TSS reference	ISDN-(ISUP)-SIP/SS/ECT						
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 **ANM**, 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".						
SIP parameter values	INFO B encapsulated FAC contains generic notification call transfer active						
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	←				←	180 Ringing(ACM)
	FAC(ECT invoke)	→					
	DISCONNECT(rr)	←		→	INFO (FAC ect alert)		
	RELEASE	→		←	200 OK INFO		
	RELEASE COMPL	←				→	INFO (CPG ect active)
						←	200 OK INFO
						←	200 OK INVITE(ANM)
						→	ACK
				→	INFO (FAC ect active)		
				←	200 OK INFO		
					BYE(REL)	→	BYE(REL)
					200 OK BYE(RLC	←	200 OK BYE(RLC

TP711009	**SIP reference: RFC 3261 [4]**					**ISUP reference: Q.1912.5 [1], clause 7.5.2.1.1.2.2 b)/Q.732.7 [i.5]**	
TSS reference	ISDN-(ISUP)-SIP/SS/ECT						
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 other remote user upon receipt of the **ANM** conveys the **call transfer number** parameter with the information received in the **generic number** parameter if both the **connected number** and an additional connected number in the **generic number** are received in the **ANM.**						
SIP parameter values	200 OK INVITE: encapsulated ANM contains the connected number and the additional connected number INFO B: encapsulated FAC contains generic notification call transfer active and call transfer number derived from the additional connected 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	←				←	180 Ringing(ACM)
	FAC(ECT invoke)	→					
	DISCONNECT(rr)	←		→	INFO (FAC ect alert)		
	RELEASE	→		←	200 OK INFO		
	RELEASE COMPL	←				→	INFO (CPG ect active)
						←	200 OK INFO
						←	200 OK INVITE(ANM)
						→	ACK
				→	INFO (FAC ect active)		
				←	200 OK INFO		
					BYE(REL)	→	BYE(REL)
					200 OK BYE(RLC	←	200 OK BYE(RLC

TP711010	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.5.2.1.1.2.2 b)/Q.732.7 [i.5]**		
TSS reference	ISDN-(ISUP)-SIP/SS/ECT						
SIP selection criteria							
ISUP selection criteria							
Test purpose	**Capability of sending the connected number in the call transfer number parameter when the ECT is invoked while one call is alerting** To verify that, in case the ECT is invoked while one call is alerting, the **FAC** sent to the other remote user upon receipt of the **ANM** conveys the **call transfer number** parameter with the information received in the **connected number** parameter if only the **connected number** is received in the **ANM.**						
SIP parameter values	200 OK INVITE: encapsulated ANM contains the connected number and the additional connected number INFO B: encapsulated FAC contains generic notification call transfer active and call transfer number derived from the connected number						
ISDN parameter values							
Comments	**ISDN 2**		**SUT**		**SIP-I 1**		**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	←				←	180 Ringing(ACM)
	FAC(ECT invoke)	→					
	DISCONNECT(rr)	←		→	INFO (FAC ect alert)		
	RELEASE	→		←	200 OK INFO		
	RELEASE COMPL	←				→	INFO (CPG ect active)
						←	200 OK INFO
						←	200 OK INVITE(ANM)
						→	ACK
				→	INFO (FAC ect active)		
				←	200 OK INFO		
					BYE(REL)	→	BYE(REL)
					200 OK BYE(RLC	←	200 OK BYE(RLC

TP711011	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 7.3; 7.5.2.3.1/Q.732.7 [i.5]**		
TSS reference	ISDN-(ISUP)-SIP/SS/ECT						
SIP selection criteria							
ISUP selection criteria							
Test purpose	**Call transfer number - conversion to international number** To verify that the IUT converts the **call transfer number** to international format. The nature of address indicator shall be set to "international number".						
SIP parameter values							
ISDN parameter values							
Comments	**ISDN 2**		**SUT**		**SIP-I 1**		**SIP-I 3**
	SETUP	←		←	INVITE(IAM)		
	ALERTING	→		→	180 Ringing(ACM)		
	CONN	→		→	200 OK INVITE(ANM)		
				←	ACK		
	HOLD	→		→	INVITE(CPG hold)		
				←	200 OK INVITE		
				→	ACK		
	SETUP	→				→	INVITE(IAM)
	ALERTING	←				←	180 Ringing(ACM)
	CONN	←				←	200 OK INVITE(ANM)
						→	ACK
	FAC(ECT invoke)	→					
	DISCONNECT(rr)	←		→	INFO (FAC ect active)		
	RELEASE	→		←	200 OK INFO		
	RELEASE COMPL	←				→	INFO (FAC ect active)
						←	200 OK INFO
					BYE(REL)	→	BYE(REL)
					200 OK BYE(RLC	←	200 OK BYE(RLC

TP711012	**SIP reference: RFC 3261 [4]**						**ISUP reference: Q.1912.5 [1], clauses 7.3; 7.5.2.4.1/Q.732.7 [i.5]**
TSS reference	ISDN-(ISUP)-SIP/SS/ECT						
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 transfer number** if it is the network's own country code. The nature of address indicator shall be set to "national (significant) number".						
SIP parameter 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 "international number with the networks own country code. INFO SIP-I 1: encapsulated FAC contains the call transfer number derived from connected number NoA "national number" INFO SIP-I 3: encapsulated FAC contains the call transfer number derived from calling 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	←				←	180 Ringing(ACM)
	CONN	←				←	200 OK INVITE(ANM)
						→	ACK
	FAC(ECT invoke)	→					
	DISCONNECT(rr)	←		→	INFO (FAC ect active)		
	RELEASE	→		←	200 OK INFO		
	RELEASE COMPL	←				→	INFO (FAC ect active)
						←	200 OK INFO
					BYE(REL)	→	BYE(REL)
					200 OK BYE(RLC	←	200 OK BYE(RLC

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

TP712001	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clause 2.5.2.1.1/Q.732 [i.4]**
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV	
SIP selection criteria		
ISUP selection criteria		
Test purpose	"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)	
SIP parameter values	183 Session Progress encapsulated ACM generic notification indicator "call is diverting"	
ISDN parameter values	NOTIFY: Notification indicator "call is diverting"	
Comments	(see message flow below)	

ISDN	SUT			SIP-I
SETUP	→		→	INVITE(IAM)
NOTIFY	←		←	183 Session Progress(ACM)
ALERTING	←		←	180 Ringing(CPG)
CONNECT	←		←	200 OK INVITE(ANM)
			→	ACK
DISCONNECT	→		→	BYE(REL)
RELEASE	←		←	200 OK BYE(RLC)
RELEASE COMPLETE	→			

TP712002	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 2.5.2.1.1/Q.732 [i.4]**	
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV				
SIP selection criteria					
ISUP selection criteria					
Test purpose	"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)				
SIP parameter values	180 Ringing: encapsulated ACM optional backward call indicator "call diversion may occur" 183 Session Progress: encapsulated CPG contains generic notification "call is diverting", call diversion information, redirection number				
ISDN parameter values					
Comments	ISDN	SUT			SIP-I
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
				←	183 Session Progress(CPG)
	CONNECT	←		←	200 OK INVITE(ANM)
				→	ACK
	DISCONNECT	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE(RLC)
	RELEASE COMPLETE	→			

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

<table>
<tr><th>TP712004</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
clause 2.4.2; table 2-1/Q.732 [i.4]</th></tr>
<tr><td>TSS reference</td><td colspan="4">ISDN-(ISUP)-SIP/SS/CDIV</td></tr>
<tr><td>SIP selection criteria</td><td colspan="4"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="4"></td></tr>
<tr><td>Test purpose</td><td colspan="4">Redirection number - presentation restricted - according to the notification subscription option
To verify that the originating exchange does not make the redirection number available to the calling access signalling system, if the notification subscription option of the call diversion information is coded "001 presentation not allowed", "011 presentation allowed without redirection number" or "000 unknown".</td></tr>
<tr><td>SIP parameter values</td><td colspan="4">183 Session Progress encapsulated ACM call diversion information notification subscription option "presentation allowed without redirection number"</td></tr>
<tr><td>ISDN parameter values</td><td colspan="4">NOTIFY: notification indicator "call is diverted"
ALERTING: no redirection number
CONNECT: no redirection number</td></tr>
<tr><td rowspan="11">Comments</td><td>ISDN</td><td colspan="2">SUT</td><td>SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>NOTIFY</td><td>←</td><td>←</td><td>183 Session Progress(ACM)</td></tr>
<tr><td>ALERTING</td><td>←</td><td>←</td><td>180 Ringing(CPG)</td></tr>
<tr><td>CONNECT</td><td>←</td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="2"></td><td></td></tr>
<tr><td>DISCONNECT</td><td>→</td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RELEASE</td><td>←</td><td>←</td><td>200 OK BYE(RLC)</td></tr>
<tr><td>RELEASE COMPLETE</td><td>→</td><td></td><td></td></tr>
</table>

<table>
<tr><th>TP712005</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
clause 2.4.2; table 2-1/Q.732 [i.4]</th></tr>
<tr><td>TSS reference</td><td colspan="4">ISDN-(ISUP)-SIP/SS/CDIV</td></tr>
<tr><td>SIP selection criteria</td><td colspan="4"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="4"></td></tr>
<tr><td>Test purpose</td><td colspan="4">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".</td></tr>
<tr><td>SIP parameter values</td><td colspan="4">183 Session Progress encapsulated ACM call diversion information notification subscription option "presentation allowed with redirection number"
200 OK INVITE: encapsulated ANM redirection number restriction "presentation restricted</td></tr>
<tr><td>ISDN parameter values</td><td colspan="4">CONNECT: no redirection number</td></tr>
<tr><td rowspan="11">Comments</td><td>ISDN</td><td colspan="2">SUT</td><td>SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>NOTIFY</td><td>←</td><td>←</td><td>183 Session Progress(ACM)</td></tr>
<tr><td>ALERTING</td><td>←</td><td>←</td><td>180 Ringing(CPG)</td></tr>
<tr><td>CONNECT</td><td>←</td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="2"></td><td></td></tr>
<tr><td>DISCONNECT</td><td>→</td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RELEASE</td><td>←</td><td>←</td><td>200 OK BYE(RLC)</td></tr>
<tr><td>RELEASE COMPLETE</td><td>→</td><td></td><td></td></tr>
</table>

<table>
<tr><th>TP712006</th><th colspan="4">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1],
clause 2.4.2; table 2-1/Q.732 [i.4]</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-(ISUP)-SIP/SS/CDIV</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">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</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">CONNECT: redirection number</td></tr>
<tr><td rowspan="10">Comments</td><td>ISDN</td><td colspan="3">SUT</td><td>SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>NOTIFY</td><td>←</td><td></td><td>←</td><td>183 Session Progress(ACM)</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing(CPG)</td></tr>
<tr><td>CONNECT</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3"></td><td></td></tr>
<tr><td>DISCONNECT</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RELEASE</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
<tr><td>RELEASE COMPLETE</td><td>→</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><th>TP712007</th><th colspan="4">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1],
clause 2.4.2/Q.732 [i.4]</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISDN-(ISUP)-SIP/SS/CDIV</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">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).</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">183 Session Progress encapsulated ACM call diversion information notification subscription option "presentation allowed with redirection number", redirection number
183 Session Progress encapsulated ACM call diversion information notification subscription option "presentation allowed with redirection number", no redirection number
200 OK INVITE: encapsulated ANM redirection number restriction "presentation allowed"</td></tr>
<tr><td>ISDN parameter values</td><td colspan="6">ALERTING: no redirection number
CONNECT: no redirection number</td></tr>
<tr><td rowspan="11">Comments</td><td>ISDN</td><td colspan="3">SUT</td><td colspan="2">SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(IAM)</td></tr>
<tr><td>NOTIFY</td><td>←</td><td></td><td>←</td><td colspan="2">183 Session Progress(ACM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">183 Session Progress(CPG)</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing(CPG)</td></tr>
<tr><td>CONNECT</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td></td><td colspan="3"></td><td colspan="2"></td></tr>
<tr><td>DISCONNECT</td><td>→</td><td></td><td>→</td><td colspan="2">BYE(REL)</td></tr>
<tr><td>RELEASE</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK BYE(RLC)</td></tr>
<tr><td>RELEASE COMPLETE</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td colspan="7">NOTE: The first diverting exchange sends the redirection number and allows for its presentation. The second (last) diversion allows for the presentation of the redirection number, but does not send it, i.e. only call diversion information is present in the message and the redirection number is missing. The redirection number restriction parameter is also received as "presentation allowed".</td></tr>
</table>

<table>
<tr><td>TP712008</td><td colspan="5">SIP reference: RFC 3261 [4]</td><td>ISUP reference:
Q.1912.5 [1],
clause 2.4.2/Q.732 [i.4]</td></tr>
<tr><td>TSS reference</td><td colspan="6">ISDN-(ISUP)-SIP/SS/CDIV</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">Multiple diversions - redirection number - presentation according to the most restrictive notification subscription option
To verify that the originating exchange handles the presentation of the redirection number according to the contents of the most restrictive notification subscription option of the call diversion information, if the forwarded-to user allows presentation of the number ("presentation allowed" in the redirection number restriction parameter) (see note).</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">183 Session Progress encapsulated ACM call diversion information notification subscription option "presentation allowed with redirection number", redirection number
183 Session Progress encapsulated ACM call diversion information notification subscription option "presentation allowed without redirection number", redirection number
200 OK INVITE: encapsulated ANM redirection number restriction "presentation allowed"</td></tr>
<tr><td>ISDN parameter values</td><td colspan="6">ALERTING: no redirection number
CONNECT: no redirection number</td></tr>
<tr><td rowspan="11">Comments</td><td>ISDN</td><td colspan="3">SUT</td><td colspan="2">SIP-I</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(IAM)</td></tr>
<tr><td>NOTIFY</td><td>←</td><td></td><td>←</td><td colspan="2">183 Session Progress(ACM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">183 Session Progress(CPG)</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing(CPG)</td></tr>
<tr><td>CONNECT</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td></td><td colspan="3"></td><td colspan="2"></td></tr>
<tr><td>DISCONNECT</td><td>→</td><td></td><td>→</td><td colspan="2">BYE(REL)</td></tr>
<tr><td>RELEASE</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK BYE(RLC)</td></tr>
<tr><td>RELEASE COMPLETE</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td colspan="7">NOTE: Several messages each containing the call diversion information are received, as if multiple forwardings have occurred (from option B - immediate release - diverting exchanges, so no collecting of information takes place).</td></tr>
</table>

TP712009	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 2.5.2.5.1.1/Q.732 [i.4]**		
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV						
SIP selection criteria							
ISUP selection criteria							
Test purpose	**Completion of diverted call by the diverted-to exchange** To verify that the IUT accepts and can successfully establish a diverted call.						
SIP parameter values	183 Session Progress: encapsulated ACM generic notification "call is diverted", redirection information, redirection number						
ISDN parameter values							
Comments	**ISDN 2**		**SUT**		**SIP-I 1**		**SIP-I 3**
				←	INVITE(IAM)		
			CDIV				
				→	183 Session Progress(ACM)		
						→	INVITE(IAM)
						←	180 Ringing(ACM)
				→	180 Ringing(ACM)		
						←	200 OK INVITE(ANM)
				→	200 OK INVITE(ANM)	→	ACK
				←	ACK		
				←	BYE(REL)	→	BYE(REL)
				→	200 OK BYE(RLC	←	200 OK BYE(RLC

TP712010	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 2.5.2.5.1.1/Q.732 [i.4]**	
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Setting of redirection number restriction parameter at the diverted-to exchange (pres. allowed)** To verify that the IUT includes the **redirection number restriction** indicator in the **ACM**, **CPG**, **ANM** or **CON** set to "presentation allowed" (COLR not activated).				
SIP parameter values	200 OK INVITE: encapsulated ANM redirection number restriction "presentation allowed"				
ISDN parameter values					
Comments	ISDN		SUT		SIP-I
	SETUP	←		←	INVITE(IAM)
	ALERTING	→		→	180 Ringing(ACM)
	CONNECT	→		→	200 OK INVITE(ANM)
				←	ACK
	DISCONNECT	←		←	BYE(REL)
	RELEASE	→		→	200 OK BYE(RLC)
	RELEASE COMPLETE	←			

TP712011	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause 2.5.2.5.1.1/Q.732 [i.4]
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Setting the redirection number restriction indicator at the diverted-to exchange (pres. restricted)** To verify that the IUT includes the **redirection number restriction** indicator in the **ACM**, **CPG**, **ANM** or **CON** set to "presentation restricted" (COLR activated).	
SIP parameter values	200 OK INVITE: encapsulated ANM redirection number restriction "presentation restricted"	
ISDN parameter values		
Comments	(see flow below)	

ISDN		SUT		SIP-I
SETUP	←		←	INVITE(IAM)
ALERTING	→		→	180 Ringing(ACM)
CONNECT	→		→	200 OK INVITE(ANM)
			←	ACK
DISCONNECT	←		←	BYE(REL)
RELEASE	→		→	200 OK BYE(RLC)
RELEASE COMPLETE	←			

TP712012	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause 2.5.2.5.1.2 b) 2)/Q.732 [i.4]
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Original called number generated by the diverting exchange** Verify that the IUT sets the address presentation restricted indicator of the **original called number** according to the "served user releases his/her number to the diverted-to user" option.	
SIP parameter values	INVITE SIP-I 3:encapsulated IAM original called number presentation allowed	
ISDN parameter values		
Comments	(see flow below)	

ISDN 2		**SUT**		**SIP-I 1**		**SIP-I 3**
			←	INVITE(IAM)		
		CDIV				
			→	183 Session Progress(ACM)		
					→	INVITE(IAM)
					←	180 Ringing(ACM)
			→	180 Ringing(ACM)		
					←	200 OK INVITE(ANM)
			→	200 OK INVITE(ANM)	→	ACK
			←	ACK		
			←	BYE(REL)	→	BYE(REL)
			→	200 OK BYE(RLC	←	200 OK BYE(RLC

TP712013	**SIP reference: RFC 3261 [4]**					**ISUP reference: Q.1912.5 [1]**	
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV						
SIP selection criteria							
ISUP selection criteria							
Test purpose	**Redirecting number generated by the diverting exchange** Verify that the IUT sets the address presentation restricted indicator of the **redirecting number** according to the "served user releases his/her number to the diverted-to user" option. The redirecting indicator in the **redirection information** shall be set to "011 Call diverted".						
SIP parameter values	INVITE SIP-I 3: redirecting number, redirection information						
ISDN parameter values							
Comments	**ISDN 2**		**SUT**		**SIP-I 1**		**SIP-I 3**
				←	INVITE(IAM)		
			CDIV				
				→	183 Session Progress(ACM)		
						→	INVITE(IAM)
						←	180 Ringing(ACM)
				→	180 Ringing(ACM)		
						←	200 OK INVITE(ANM)
				→	200 OK INVITE(ANM)	→	ACK
				←	ACK		
				←	BYE(REL)	→	BYE(REL)
				→	200 OK BYE(RLC	←	200 OK BYE(RLC

TP712014	**SIP reference: RFC 3261 [4]**					**ISUP reference: Q.1912.5 [1], clause 2.5.2.5.1.2 b) 5)/Q.732 [i.4]**	
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV						
SIP selection criteria							
ISUP selection criteria							
Test purpose	**ISDN user part preference indicator in the diverting exchange** To verify that the IUT can successfully divert a call and that ISDN user part preference indicator received in the **forward call indicators** with the value "ISDN user part: - not required all the way" shall be changed to "ISDN user part preferred all the way"; - preferred all the way" shall be left unchanged; - required all the way" shall be left unchanged.						
SIP parameter values	INVITE SIP-I 3 : encapsulated IAM forward call indicator ISDN user part required all the way						
ISDN parameter values							
Comments	**ISDN 2**		**SUT**		**SIP-I 1**		**SIP-I 3**
				←	INVITE(IAM)		
			CDIV				
				→	183 Session Progress(ACM)		
						→	INVITE(IAM)
						←	180 Ringing(ACM)
				→	180 Ringing(ACM)		
						←	200 OK INVITE(ANM)
				→	200 OK INVITE(ANM)	→	ACK
				←	ACK		
				←	BYE(REL)	→	BYE(REL)
				→	200 OK BYE(RLC	←	200 OK BYE(RLC

TP712015	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clause 2.5.2.5.1.2 c) ii); iii)/Q.732 [i.4]**
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Call diversion may occur in the diverting exchange** To verify that the IUT includes an **optional backward call indicator** with the indication "call diversion may occur" in the **ACM** in case of CFNR, CD(a), CFB(u,e) and CD(i,e)	
SIP parameter values	180 Ringing: encapsulated ACM called party status indicator "subscriber free" optional backward call indicator "call diversion may occur"	
ISDN parameter values	ALERTING: no mapping of optional backward call indicator value	
Comments		

ISDN		SUT		SIP-I
SETUP	→		→	INVITE(IAM)
ALERTING	←		←	180 Ringing(ACM)
CONNECT	←		←	200 OK INVITE(ANM)
			→	ACK
DISCONNECT	→		→	BYE(REL)
RELEASE	←		←	200 OK BYE(RLC)
RELEASE COMPLETE	←			

A.1.1.2.13 Call HOLD (HOLD)

TP713001	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clauses 2.5.2.1.1.1; 2.5.2.1.1.2/Q.733 [i.6]**
TSS reference	ISDN-(ISUP)-SIP/SS/HOLD	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Call hold after answer, requested by the local user** To verify that a call can be placed on hold and can be retrieved again by the local user and that notifications are sent with **CPG** messages having the **event indicator** set to "progress".	
SIP parameter values		
ISDN parameter values		
Comments		

ISDN		SUT		SIP
SETUP	→		→	INVITE(IAM)
ALERTING	←		←	180 Ringing(ACM)
CONN	←		←	200 OK INVITE(ANM)
			→	ACK
		Communication		
HOLD	→		→	INFO(CPG hold)
			←	200 OK INFO
RETRIVE	→		→	INFO(CPG retrieve)
			←	200 OK INFO
		Communication		
DISC	→		→	BYE(REL)
REL	←		←	200 OK BYE
REL_COM	→			

TP713002	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clauses 2.5.2.1.1.1; 2.5.2.1.1.2/Q.733 [i.6]**
TSS reference	ISDN-(ISUP)-SIP/SS/HOLD	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Call hold after answer, requested by the remote user** To verify that a call can be placed on hold and can be retrieved again by the remote user and that notifications are sent with **CPG** messages.	
SIP parameter values		
ISDN parameter values		

Comments	ISDN		SUT		SIP
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
			Communication		
	HOLD	←		←	INFO(CPG hold)
				→	200 OK INFO
	RETRIVE	←		←	INFO(CPG retrieve)
				→	200 OK INFO
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE
	REL_COM	→			

TP713003	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clauses 2.2.1; 2.5.2.1.1.1; 2.5.2.1.1.2/Q.733 [i.6]**
TSS reference	ISDN-(ISUP)-SIP/SS/HOLD	
SIP selection criteria		
ISUP selection criteria	PICS 8/1	
Test purpose	**Call hold after alerting, requested by the local user** To verify that an outgoing call can be placed on HOLD after alerting has commenced and can be retrieved afterwards by the local user and that notifications are sent with **CPG** messages.	
SIP parameter values		
ISDN parameter values		

Comments	ISDN		SUT		SIP
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	HOLD	→		→	INFO(CPG hold)
				←	200 OK INFO
	RETRIVE	→		→	INFO(CPG retrieve)
				←	200 OK INFO
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE
	REL_COM	→			

TP713004	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clauses 2.2.1; 2.5.2.5.1/Q.733 [i.6]**	
TSS reference	ISDN-(ISUP)-SIP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria	PICS 8/1				
Test purpose	**Call hold after alerting, requested by the remote user** To verify that an incoming call can be placed on hold and can be retrieved afterwards by the remote user.				
SIP parameter values					
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	HOLD	←		←	INFO(CPG hold)
				→	200 OK INFO
	RETRIVE	←		←	INFO(CPG retrieve)
				→	200 OK INFO
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE
	REL_COM	→			

TP713005	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 2.3/Q.764 [i.12]**	
TSS reference	ISDN-(ISUP)-SIP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call hold after answer, release of the call by the local served user** To verify that a call in the held state can be released by the user who activated the Call hold service.				
SIP parameter values					
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
			Communication		
	HOLD	→		→	INFO(CPG hold)
				←	200 OK INFO
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE
	REL_COM	→			

TP713006	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 2.3/Q.764 [i.12]**	
TSS reference	ISDN-(ISUP)-SIP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call hold after answer, release of the call by the non-served user** To verify that a call in the held state can be released by the user who did not activate the Call hold service.				
SIP parameter values					
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
			Communication		
	HOLD	→		→	INFO(CPG hold)
				←	200 OK INFO
	DISC	←		←	BYE(REL)
	REL	→		←	200 OK BYE
	REL_COM	←			

TP713007	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 2.3/Q.764 [i.12]**	
TSS reference	ISDN-(ISUP)-SIP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call hold after alerting, release of the call by the local served user** To verify that a held call can be released by the user who activated the Call hold service without retrieving the call.				
SIP parameter values					
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP	←		←	INVITE(IAM)
	ALERTING	→		→	180 Ringing(ACM)
	HOLD	→		→	INFO(CPG hold)
				←	200 OK INFO
	DISC	→		→	CANCEL/BYE
	RELEASE	←		←	200 OK CANCEL/BYE
	REL_COMP	→		←	487 Request Terminated
				→	ACK

TP713008	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 2.3/Q.764 [i.12]**	
TSS reference	ISDN-(ISUP)-SIP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call hold after answer, release of the call by the non-served user** To verify that a call in the held state can be released by the user who did not activate the Call hold service.				
SIP parameter values					
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP	←		←	INVITE(IAM)
	ALERTING	→		→	180 Ringing(ACM)
	HOLD	←		←	INFO(CPG hold)
				→	200 OK INFO
	DISC	←		←	CANCEL
	RELEASE	→		→	200 OK CANCEL/BYE
	REL_COMP	←		→	487 Request Terminated
				←	ACK

A.1.1.2.14 Call Waiting (CW)

TP714001	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 1.5.2.1.1/Q.733 [i.6]**	
TSS reference	ISDN-(ISUP)-SIP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call waiting indication in ACM** To verify that a call can be successfully established if the **ACM** indicates that it is a waiting call.				
SIP parameter values	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	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
		Communication			
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE
	REL_COM	→			

TP714002	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 1.5.2.1.1/Q.733 [i.6]**
TSS reference	ISDN-(ISUP)-SIP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call waiting indication in CPG** To verify that a call can be successfully established if the **CPG** indicates that it is a waiting call.				
SIP parameter values	180 Ringing: encapsulated ACM the called party status is set to "no indication" 183 Session Progress: encapsulated CPG Alerting contains the Generic notification parameter value "call is a waiting call"				
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP	→		→	INVITE(IAM)
				←	183 Session Progress(ACM)
	ALERTING	←		←	180 Ringing(CPG)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
		Communication			
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE
	REL_COM	→			

TP714003	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 1.5.2.5.1/Q.733 [i.6]**
TSS reference	ISDN-(ISUP)-SIP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call waiting indication in ACM or CPG** To verify that a call can be successfully established if the user has subscribed to the call waiting service (with notification) and if he is currently busy, but answers the waiting call. The indication shall be sent either in an **ACM** or a **CPG.**				
SIP parameter values	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	→		→	180 Ringing(ACM)
	CONN	→		→	200 OK INVITE(ANM)
				←	ACK
		Communication			
	DISC	←		←	BYE(REL)
	REL				200 OK BYE
	REL_COM	←			

TP714004	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 1.5.2.5.2/Q.733 [i.6]**	
TSS reference	ISDN-(ISUP)-SIP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call waiting rejected** To verify that the IUT sends a **REL** with cause #21 (call rejected) if a busy user rejects the waiting call.				
SIP parameter values	480 Temporarily unavailable: encapsulated REL cause 21				
ISDN parameter values	RELEASE COMPLETE: cause 21				
Comments	**ISDN**		**SUT**		**SIP**
	SETUP	←		←	INVITE(IAM)
	ALERTING	→		→	180 Ringing(ACM)
	CONN	→		→	200 OK INVITE(ANM)
				←	ACK
			Communication		
	SETUP	←		←	INVITE(IAM)
	ALERTING	→		→	180 Ringing(ACM waiting call)
	RELEASE COMPLETE	→		→	480 Temporarily unavailable(REL#21)
				←	ACK
			Communication		
	DISC	←		←	BYE(REL)
	REL				200 OK BYE
	REL_COM	←			

A.1.1.2.15 Three Party Service (3PTY)

TP715001	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clauses 2.4; 2.2.1/Q.734.2 [i.8]**
TSS reference	ISDN-(ISUP)-SIP/SS/3PTY	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Served user initiates 3PTY** To verify that the IUT, where the served user with two active calls is located, can successfully join these calls to form a three-way conversation, and notify the implied remote parties accordingly. The IUT should send **CPG** messages with the **generic notification indicator** set to "conference established" to both implied parties. The **event indicator** in the **CPG** should be set to "progress".	
SIP parameter values		
ISDN parameter values		
Comments		

ISDN		SUT		SIP-I 1		SIP-I 2
SETUP	→		→	INVITE(IAM)		
ALERTING	←		←	180 Ringing(ACM)		
CONN	←		←	200 OK INVITE(ANM)		
Communication						
HOLD	→		→	INVITE(CPG hold)		
			←	200 OK INVITE		
			→	ACK		
SETUP	→				→	INVITE(IAM)
ALERTING	←				←	180 Ringing(ACM)
CONN	←				←	200 OK INVITE(ANM)
FAC(est3pty)	→		→	INFO(CPG conf est)		
			←	200 OK INFO		
					→	INFO(CPG conf est)
					←	200 OK INFO
3 PTY communication						
DISC	←		←	BYE(REL)		
RELEASE	→		→	200 OK BYE		
REL_COM	←				→	INFO(CPG conf disc)
					←	200 OK INFO
DISC	→				→	BYE(REL)
RELEASE	←				←	200 OK BYE
REL_COM	→					

TP715002	SIP reference: RFC 3261 [4]						ISUP reference: Q.1912.5 [1], clause 2.5.2.1.1.3 a)/Q.734.2 [i.8]
TSS reference	ISDN-(ISUP)-SIP/SS/3PTY						
SIP selection criteria							
ISUP selection criteria							
Test purpose	**Served user creates a private communication with a remote user** To verify that the IUT (controlling the conference) on a 3PTY call can successfully create private communication with one of the remote users. The appropriate notification (depending on A-B active-held or A-C active-idle connection) is sent in **CPG** messages to the two users.						
SIP parameter values							
ISDN parameter values							
Comments	ISDN	SUT			SIP-I 1		SIP-I 2
	SETUP	→		→	INVITE(IAM)		
	ALERTING	←		←	180 Ringing(ACM)		
	CONN	←		←	200 OK INVITE(ANM)		
	Communication						
	HOLD	→		→	INVITE(CPG hold)		
				←	200 OK INVITE		
				→	ACK		
	SETUP	→				→	INVITE(IAM)
	ALERTING	←				←	180 Ringing(ACM)
	CONN	←				←	200 OK INVITE(ANM)
	FAC(est3pty)	→		→	INFO(CPG conf est)		
				←	200 OK INFO		
						→	INFO(CPG conf est)
						←	200 OK INFO
	3 PTY communication						
	FAC(end3pty)	→		→	INFO(CPG conf disc)		
	FAC(ret res)	←		←	200 OK INFO		
						→	INFO(CPG conf disc)
						←	200 OK INFO
			Communication ISDN - SIP-I 2				
	DISC	←		←	BYE(REL)		
	RELEASE	→		→	200 OK BYE		
	REL_COM	←					
	DISC	→				→	BYE(REL)
	RELEASE	←				←	200 OK BYE
	REL_COM	→					

TP715003	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 2.5.2.1.1.3 b)/Q.734.2 [i.8]**		
TSS reference	ISDN-(ISUP)-SIP/SS/3PTY						
SIP selection criteria							
ISUP 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							
Comments	ISDN		SUT		SIP-I 1		SIP-I 2
	SETUP	→		→	INVITE(IAM)		
	ALERTING	←		←	180 Ringing(ACM)		
	CONN	←		←	200 OK INVITE(ANM)		
	HOLD	→		→	INVITE(CPG hold)		
				←	200 OK INVITE		
				→	ACK		
	SETUP	→				→	INVITE(IAM)
	ALERTING	←				←	180 Ringing(ACM)
	CONN	←				←	200 OK INVITE(ANM)
	FAC(est3pty)	→		→	INFO(CPG conf est)		
				←	200 OK INFO		
						→	INFO(CPG conf est)
						←	200 OK INFO
	3 PTY communication						
	DISC	→				→	BYE(REL)
	RELEASE	←				←	200 OK BYE
	REL_COM	→		→	INFO(CPG conf disc)		
				←	200 OK INFO		
				→	INFO(CPG hold)		
				←	200 OK INFO		
	DISC	←		←	BYE(REL)		
	RELEASE	→		→	200 OK BYE		
	REL_COM	←					
NOTE: The "remote hold" notification should be sent in a **CPG** to the remaining remote user, followed by the "conference disconnected" notification in a separate **CPG.**							

TP715004	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 2.5.2.1.1.3/Q.734.2 [i.8]**		
TSS reference	ISDN-(ISUP)-SIP/SS/3PTY						
SIP selection criteria							
ISUP selection criteria							
Test purpose	**Served user disconnects both remote users and terminates the call** To verify that the IUT (controlling the conference) can send the appropriate notification to the two remote users when disconnecting both remote users on the 3PTY call. The IUT should send to the appropriate remote users a **CPG** with a **generic notification indicator** (depending on A-B active-held or A-C active-idle connection). The **event indicator** in the **CPG** is set to "progress".						
SIP parameter values							
ISDN parameter values							
Comments	ISDN		SUT		SIP-I 1		SIP-I 2
	SETUP(CRx)	→		→	INVITE(IAM)		
	ALERTING	←		←	180 Ringing(ACM)		
	CONN	←		←	200 OK INVITE(ANM)		
	HOLD	→		→	INVITE(CPG hold)		
				←	200 OK INVITE		
				→	ACK		
	SETUP(CRy)	→				→	INVITE(IAM)
	ALERTING	←				←	180 Ringing(ACM)
	CONN	←				←	200 OK INVITE(ANM)
	FAC(est3pty)	→		→	INFO(CPG conf est)		
				←	200 OK INFO		
						→	INFO(CPG conf est)
						←	200 OK INFO
	3 PTY communication						
	DISC(CRx)	→		→	BYE(REL)		
	RELEASE	←		←	200 OK BYE		
	REL_COM	→				→	INFO(CPG conf disc)
						←	200 OK INFO
	DISC(CRy)	→				→	BYE(REL)
	RELEASE	←				←	200 OK BYE
	REL_COM	→					

TP715005	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clause 2.2.1/Q.734.2 [i.8]**
TSS reference	ISDN-(ISUP)-SIP/SS/3PTY	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Remote user disconnects 3PTY call** To verify that the IUT (controlling the conference) can successfully continue the 3PTY call after receiving disconnection by one of the remote users, and send the appropriate notification to the remaining party. The IUT should send to the other remote user **CPG** with a **generic notification indicator** (depending on A-B active-held or A-C active-idle connection). The **event indicator** in the **CPG** is set to "progress" (see note).	
SIP parameter values		
ISDN parameter values		
Comments	(see message flow below)	

ISDN		SUT		SIP-I 1		SIP-I 2
SETUP	→		→	INVITE(IAM)		
ALERTING	←		←	180 Ringing(ACM)		
CONN	←		←	200 OK INVITE(ANM)		
HOLD	→		→	INVITE(CPG hold)		
			←	200 OK INVITE		
			→	ACK		
SETUP	→				→	INVITE(IAM)
ALERTING	←				←	180 Ringing(ACM)
CONN	←				←	200 OK INVITE(ANM)
FAC(est3pty)	→		→	INFO(CPG conf est)		
			←	200 OK INFO		
					→	INFO(CPG conf est)
					←	200 OK INFO
			3 PTY communication			
DISC	←				←	BYE(REL)
RELEASE	→				→	200 OK BYE
REL_COM	←		→	INFO(CGP (conf disc)		
			←	200 OK INFO		
			→	INFO(CGP (hold)		
			←	200 OK INFO		
DISC(CRx)	→		→	BYE(REL)		
RELEASE	←		←	200 OK BYE		
REL_COM	→					

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 [4]**			**ISUP reference: Q.1912.5 [1], clauses 2.4; 2.2.1/Q.734.2 [i.8]**	
TSS reference	ISDN-(ISUP)-SIP/SS/3PTY				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Remote user included in 3PTY** To verify that the IUT can receive the notification information related to 3PTY, and pass it on to the access signalling system. The IUT should be able to transparently transfer the **CPG** message with the following notifications in the **generic notification indicator** in both the forward and the backward direction: 1) "Conference established". 2) "Conference disconnected". 3) "Remote hold".				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP	←		←	INVITE(IAM)
	ALERTING	→		→	180 Ringing(ACM)
	CONN	→		→	200 OK INVITE(ANM)
			Communication		
				←	INVITE(CPG hold)
				→	200 OK INVITE
				←	ACK
	NOTIFY(conf est)	←		←	INFO(CPG conf est)
				→	200 OK INFO
			3 PTY communication		
	NOTIFY(conf disc)	←		←	INFO(CPG conf disc)
				→	200 OK INFO
	NOTIFY(hold)	←		←	INFO(CPG hold)
				→	200 OK INFO
	DISC	←		←	BYE(REL)
	REL	→		→	200 OK BYE
	REL_COM	←			

Annex B (informative): Bibliography

ITU-T Recommendations Q.761: "Signalling System No. 7 - ISDN User Part functional description".

ITU-T Recommendations Q.762: "Signalling System No. 7 - ISDN User Part general functions of messages and signals".

ITU-T Recommendations Q.763: "Signalling System No. 7 - ISDN User Part formats and codes".

ITU-T Recommendations Q.1902.1: " Bearer Independent Call Control protocol (Capability Set 2): Functional description".

ITU-T Recommendations Q.1902.2: "Bearer Independent Call Control protocol (Capability Set 2) and Signalling System No.7 ISDN User Part: General functions of messages and parameters".

ITU-T Recommendations Q.1902.3: "Bearer Independent Call Control protocol (Capability Set 2) and Signalling System No.7 ISDN User Part: Formats and codes".

ITU-T Recommendations Q.1902.4: "Bearer Independent Call Control protocol (Capability Set 2): Basic call procedures".

IETF RFC 3267: "Real-Time Transport Protocol (RTP) Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs".

ITU-T Recommendation Q.939: "Typical DSS 1 service indicator codings for ISDN telecommunications services".

Annex C (informative): Change history

Date	WG Doc.	CR	Rev	CAT	Title / Comment	Current Version	New Version
10-06-09	21PTD096r1	001		F	Update of test description and message flows	1.1.1	1.2.1
					Publication	1.2.1	1.2.1

History

Document history		
V1.1.1	June 2008	Publication
V1.2.1	November 2009	Publication