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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

DTS/TISPAN-06014-2-NGN

Keywords

SIP, ISUP, BICC, TSS&TP, testing

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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) Profile C";

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

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

1 Scope

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

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

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

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

2.1 Normative references

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

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

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

[3] ITU-T Recommendations Q.761 to Q.764 (2000): "Signalling System No.7 ISDN User Part (ISUP)".

[4] ITU-T Recommendations Q.1902.1 to Q.1902.4 (2001): "Bearer Independent Call Control Protocol (BICC)".

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

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

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

[8] ISO/IEC 9646-1 (1994): "Conformance testing methodology and framework - Part 1: General Concepts".

[9] ISO/IEC 9646-3 (1992): "Conformance testing methodology and framework - Part 3: The Tree and Tabular Combined Notation".

[10] ISO/IEC 9646-7 (1994): "Conformance testing methodology and framework - Part 7: Implementation Conformance Statements".

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

[12] IETF RFC 3267: "Real-Time Transport Protocol (RTP) Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs".

[13] ITU-T Recommendation Q.939: "Typical DSS 1 service indicator codings for ISDN telecommunications services".

2.2 Informative references

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

Not applicable.

3 Definitions and abbreviations

3.1 Definitions

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

- terms defined in SIP / ISUP interworking reference specification;
- terms defined in ISDN layer 3 reference specification;
- terms defined in ISDN User Part (ISUP) reference specification terms defined in ISO/IEC 9646-1 [8], ISO/IEC 9646-3 [9] and in ISO/IEC 9646-7 [10].

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

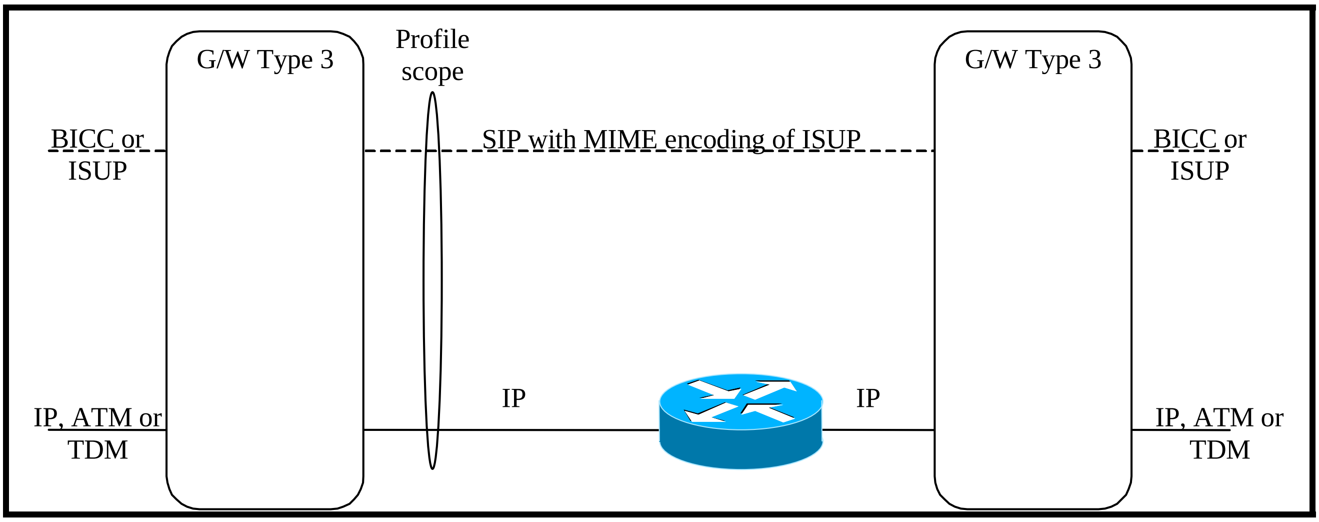

Figure 1: Profile scope for SIP with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateways

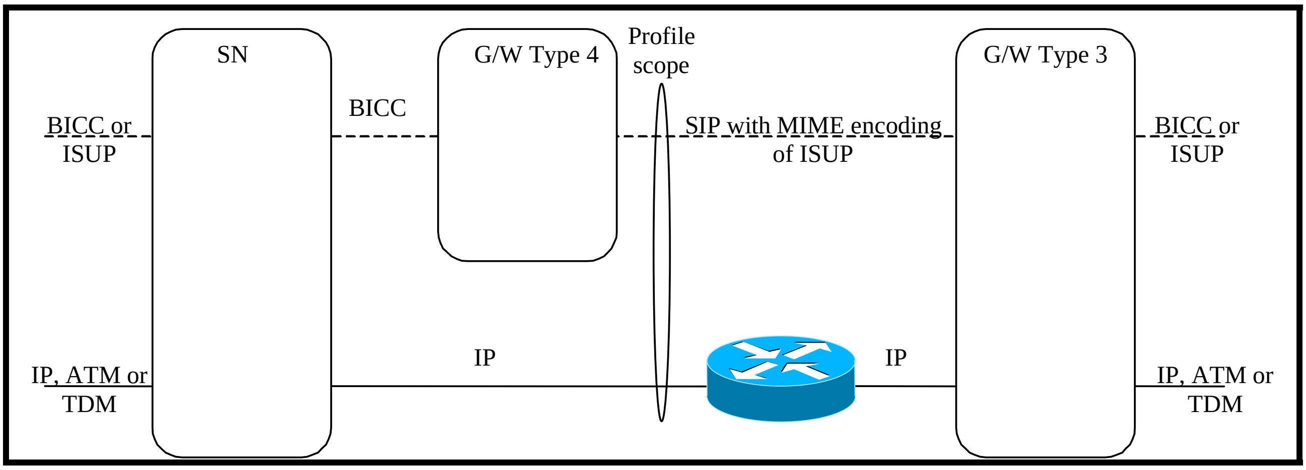

Figure 2: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 3 and 4 gateways

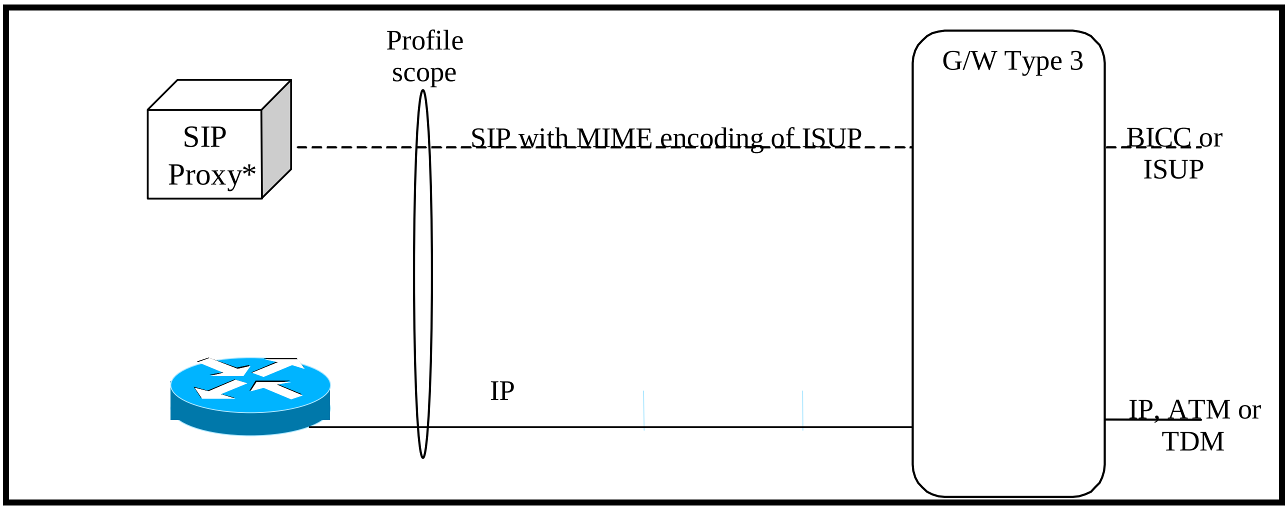

Figure 3: Profile scope for SIP with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateways

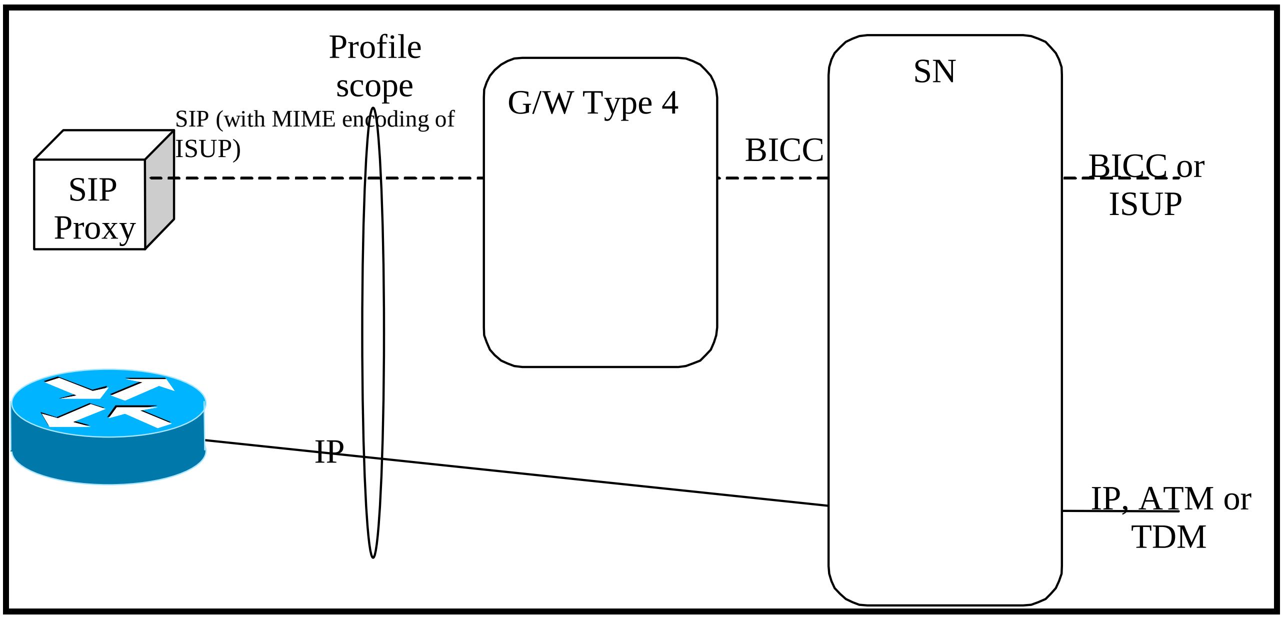

Figure 4: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 4 gateway

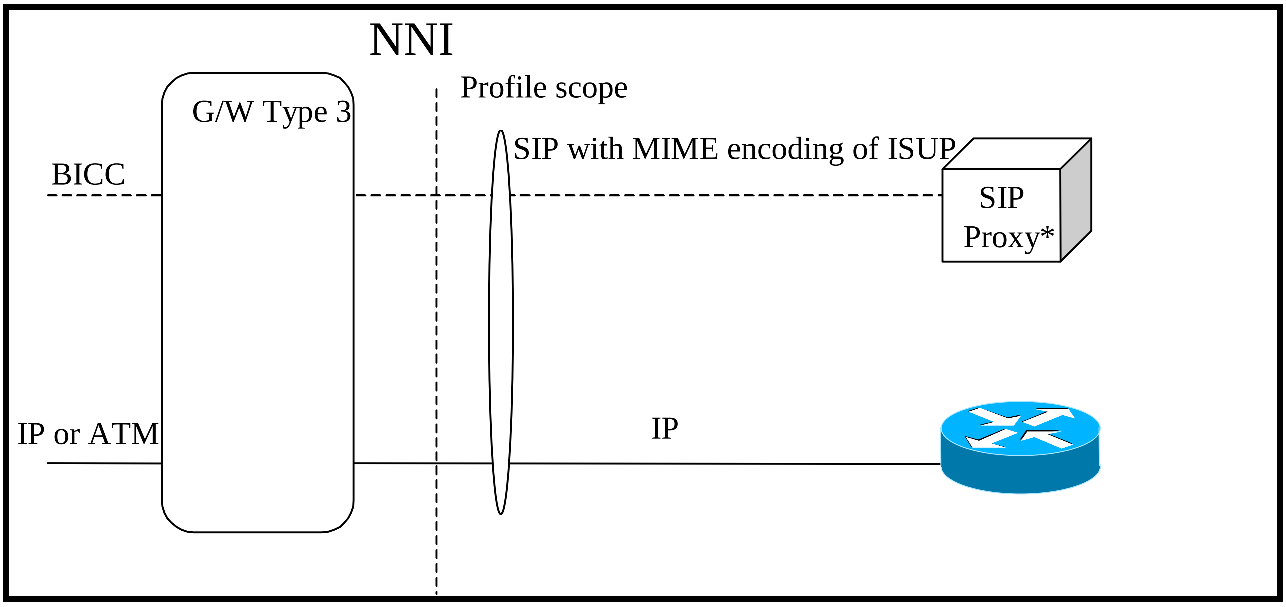

Figure 5: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateway.

3.2 Abbreviations

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

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

BICCP	Bearer Independent Call Control Protocol
CPS	Calling Party's Category
DLE	Destination Local Exchange
DSS1	Digital Subscriber System no. 1
FCI	Forward Call Indicators
HLC	High Layer Compatibility
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
MIME	Multi-purpose Internet Mail Extension
MOT	Means Of Testing
NCI	Nature of Connection Indicators
OBCI	Optional Backward Call Indicators
OLE	Originating Local Exchange
OSI	Open Systems Interconnection
PCO	Point of Control and Observation
PICS	Protocol Implementation Conformance Statement
PIXIT	Protocol Implementation eXtra Information for Testing
PTC	Parallel Test Component
SIP	Session Initiation Protocol
SP	Signalling Point
SUT	System Under Test
TMR	Transmission Medium Requirement
TP	Test Purpose
TSS	Test Suite Structure
UNI	User-Network Interface

4 Test Suite Structure (TSS)

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

SIP -ISUP basic call		
	Sending of the Initial Address Message (IAM)	TP101xxx
	Sending of the Subsequent Address Message (SAM)	TP102xxx
	Sending of COT	TP103xxx
	Receipt of the Address Complete Message (ACM)	TP104xxx
	Receipt of the Call Progress Message (CPG)	TP105xxx
	Receipt of the ANswer Mmessage (ANM)	TP106xxx
	Receipt of the CONnect message (CON)	TP107xxx
	Receipt of the RELease message (REL)	TP108xxx
	Autonomous release at I-IWU	TP109xxx
	Receipt of the BYE, CANCEL message / sending of a REL 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 (CIR)	TP402xxx
	COnnected Line identification Presentation (COLP)	TP403xxx
	COnnected Line identification Restriction (COLR)	TP404xxx
	Terminal Portability (TP)	TP405xxx
	SUBaddressing (SUB)	TP406xxx
	Malicious Call IDentification (MCID)	TP407xxx
	Call HOLD (HOLD)	TP408xxx
	Call Waiting (CW)	TP409xxx
	Call DIVersion (CDIV)	TP410xxx
	CONFerence calling (CONF)	TP411xxx
	Explicit Call transfer (ECT)	TP412xxx
	Three-Party (3PTY)	TP413xxx
	User to User Signalling (UUS)	
	User-to-user service 1	TP4140xx
	User-to-user service 2	TP4141xx
	User-to-user service 3	TP4142xx

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

SIP-I_ISDN basic call outgoing		
	Sending of the SETUP Message	TP501xxx
	Sending of the INFO	TP502xxx
	Receipt of the ALERTING - CALL PROCEEDING - 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 have been developed based on Recommendation Q.1912.5 [1].

5.1.3 Test purpose structure

The test purpose structure is according to the test suite structure (TSS).

5.2 Test purposes for the basic cal

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

5.2.1.1 Sending of the Initial Address Message (IAM)

<table>
<tr><th>TP101001</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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 PICS 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 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</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="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>

TP101002	**SIP reference: RFC 3261**			**ISUP reference: Q.1912.5 clause 6.1.2 (i,2ai)**	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial 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

TP101003	**SIP reference: RFC 3261**			**ISUP reference: Q.1912.5 clause 6.1.2 (i,2ai)**	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial 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 Require header • the SUT shall delete µ-law (PCMU), if present, from the media description that it will send back in the SDP answer • the IAM shall be sent out immediately on the BICC side with the coding of the Nature of Connection Indicators parameter: "**COT to be expected**"				
SIP parameter 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>TP101004</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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"</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, 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>
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<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>

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Q.1912.5 clause 6.1.2 (i,2aii)</th></tr>
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<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 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"</td></tr>
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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>TP101006</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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 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 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>TP101007</th><th colspan="5">SIP reference: RFC 3261</th><th>ISUP reference:
Q.1912.5 clause 6.1.2 (i,2aii)</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</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6">PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="6">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 performed on previous circuit"</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">SIP INVITE: Audio RTP/AVP 0 8
200 OK: Audio RTP/AVP 8</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">IAM Continuity Indicator: 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 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="3">Preconditions met</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>TP101008</th><th colspan="5">SIP reference: RFC 3261</th><th>ISUP reference:
Q.1912.5 clause 6.1.2 (i,2b)</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</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6">NOT PICS 1/6 AND PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="6">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</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">SIP INVITE: Audio RTP/AVP 0 8
200 OK: Audio RTP/AVP 8</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">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 colspan="2">ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td></td><td colspan="2"></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">IAM</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td colspan="3">Preconditions met</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>

TP101009	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 6.1.2 (i,2b)
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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 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				
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

TP101010	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 6.1.2 (i,1)
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection criteria	NOT PICS 4/4 AND NOT 4/5				
ISUP selection criteria	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 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				
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)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP101011</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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 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"</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>

<table>
<tr><th>TP101012</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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>

TP101013	**SIP reference: RFC 3261**			**ISUP reference: Q.1912.5 clause 6.1.2 (i,2aii)**	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial 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**"				
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**" 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

TP101014	**SIP reference: RFC 3261**			**ISUP reference: Q.1912.5 clause 6.1.2 (i,2aii)**	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial 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**"				
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**" 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

TP101015	**SIP reference: RFC 3261**			**ISUP reference: Q.1912.5 clause 6.1.2 (i,2aii)**	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial 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 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 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

TP101016	**SIP reference: RFC 3261**			**ISUP reference: Q.1912.5 clause 6.1.2 (i,2aii)**	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial 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 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 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

TP101017	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 6.1.2 (i,2b)
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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

TP101018	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 6.1.2 (i,2b)
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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

TP101019	**SIP reference: RFC 3261**				**ISUP reference:** **Q.1912.5 clause 6.1.3.2** **Q.1912.5 clause 6.1.3.3** **Q.1912.5 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

P101020	**SIP reference: RFC 3261**				**ISUP reference:** **Q.1912.5 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

TP101021	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 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 HLC shall be taken from the encapsulated ISUP • sends an IAM message, with the HLC taken from the encapsulated ISUP				
SIP parameter values	INVITE ;				
ISUP parameter values	IAM; **Access transport parameter HLC**: HLC_VALUE; USI				
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 TP1010021	
VA_01	HLC_VALUE = Telephony USI= speech
VA_02	HLC_VALUE = Facsimile Group 2/3 (Recommendation F.182) USI= 3.1 kHz audio
VA_03	HLC_VALUE == Facsimile Group 4 Class I (Recommendation F.184) USI= Unrestricted digital information
VA_04	HLC_VALUE == Teletex service, basic and mixed mode of operation (Recommendation F.230) and facsimile service Group 4, Classes II and III (Recommendation F.184) USI= Unrestricted digital information
VA_05	HLC_VALUE == Teletex service, basic and processable mode of operation (Recommendation F.220) USI= Unrestricted digital information
VA_06	HLC_VALUE = Teletex service, basic mode of operation (Recommendation F.200) USI= Unrestricted digital information
VA_07	HLC_VALUE = Syntax based Videotex (Recommendations F.300 and T.102) USI= Unrestricted digital information
VA_08	HLC_VALUE = International Videotex interworking via gateways or interworking units (Recommendations F.300 and T.101) USI= Unrestricted digital information
VA_09	HLC_VALUE = Telex service (Recommendation F.60) USI= Unrestricted digital information
VA_10	HLC_VALUE = Message Handling Systems (MHS) (X.400 - Series Recommendations) USI= Unrestricted digital information
VA_11	HLC_VALUE = OSI application (Note 2) (X.200 - Series Recommendations) USI= Unrestricted digital information
VA_12	HLC_VALUE = Audio visual (Recommendation F.721) USI= Unrestricted digital information

TP101022	SIP reference: RFC 3261				ISUP reference: Q.1912.5 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

TP101023	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 6.1.3.1
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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

TP101024	**SIP reference: RFC 3261**			**ISUP reference: Q.1912.5 clause 6.1.3.1**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/			
SIP selection 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

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

TP101025	**SIP reference: RFC 3261**	**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/	
SIP selection 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	(see flow below)	

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

TP101026	**SIP reference: RFC 3261**			**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/			
SIP selection 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

TP101027	**SIP reference: RFC 3261**			**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/			
SIP selection 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

TP101028	SIP reference: RFC 3261				ISUP reference: EN 383 001 clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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 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**" • 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: **continuity check required on this 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	←			
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101029	SIP reference: RFC 3261				ISUP reference: EN 383 001 clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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 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**". • 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: **continuity check required on this 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	←			
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101030	SIP reference: RFC 3261				ISUP reference: EN 383 001 clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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 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 performed on previous circuit**" • 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: **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	←			
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101031	SIP reference: RFC 3261				ISUP reference: EN 383 001 clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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 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 performed on previous circuit**" • 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: **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	←			
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101032	**SIP reference: RFC 3261**				**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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 Supported header:, **then independent from the received order of preference** • the shall be deferred until all preconditions have been met • the G.711 a-law codec shall be returned in the SDP answer as preferred codec				
SIP parameter 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

TP101033	**SIP reference: RFC 3261**				**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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

TP101034	**SIP reference: RFC 3261**			**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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

TP101035	**SIP reference: RFC 3261**			**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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

TP101036	**SIP reference: RFC 3261**			**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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

TP101037	**SIP reference: RFC 3261**			**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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 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**" • 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: **continuity check required on this 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

TP101038	SIP reference: RFC 3261			ISUP reference: EN 383 001 clause 6.1.3.5.2.2	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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 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**" • 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: **continuity check required on this 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

TP101039	SIP reference: RFC 3261			ISUP reference: EN 383 001 clause 6.1.3.5.2.2	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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 for a-Law and no µ-Law 100rel extensions and preconditions extensions in the SIP Supported header:, **then independent the normal offer answer procedures apply** • the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "**continuity check performed on previous circuit**" • the G.711 a-law codec shall be returned in the SDP answer				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 8 Answer: m=audio 4711 RTP/AVP 8				
ISUP parameter values	IAM: Continuity Indicator: **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**			**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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 for a-Law and no µ-Law 100rel extensions and preconditions extensions in the SIP Require header, **then independent the normal offer answer procedures apply** • the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "**continuity check performed on previous circuit** " • the G.711 a-law codec 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: **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

TP101041	**SIP reference: RFC 3261**			**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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 Supported header:, **then independent the normal offer answer procedures apply** • the IAM shall be deferred until all preconditions have been met • the G.711 a-law codec shall be returned in the SDP answer				
SIP parameter 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

TP101042	SIP reference: RFC 3261	ISUP reference: EN 383 001 clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/	
SIP selection 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

TP101043	SIP reference: RFC 3261	ISUP reference: EN 383 001 clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/	
SIP selection 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

TP101044	SIP reference: RFC 3261		ISUP reference: EN 383 001 clause 6.1.3.5.2.2		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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

TP101045	SIP reference: RFC 3261		ISUP reference: EN 383 001 clause 6.1.3.5.2.2		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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

TP101046	**SIP reference: RFC 3261**			**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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 Supported header: • the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "**continuity check required on this 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**, 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

TP101047	**SIP reference: RFC 3261**			**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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**" • **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**, 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

TP101048	SIP reference: RFC 3261			ISUP reference: EN 383 001 clause 6.1.3.5.2.2	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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 Supported header: • the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "**continuity check performed on previous circuit** " • **the u-law codec shall be rejected**				
SIP parameter 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 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**				**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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 **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 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 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

TP101050	**SIP reference: RFC 3261**				**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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

TP101051	**SIP reference: RFC 3261**				**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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

TP101052	**SIP reference: RFC 3261**				**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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	→			

TP101053	**SIP reference: RFC 3261**				**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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	→			

TP101054	**SIP reference: RFC 3261**				**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/				
SIP selection 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>TP101055</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
EN 383 001 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">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="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
• 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="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>TP101056</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
EN 383 001 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">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="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 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="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">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="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>TP101057</th><th colspan="2">SIP reference: RFC 3261</th><th colspan="3">ISUP reference:
EN 383 001 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">PICS 1/4 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 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="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">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="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>

TP101058	**SIP reference: RFC 3261**	**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/	
SIP selection 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**" • **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**, 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

TP101059	**SIP reference: RFC 3261**	**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/	
SIP selection 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**" • **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**, 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><td>TP101060</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISUP reference:
EN 383 001 clause 6.1.3.5.2.2</td></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">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 with more than one media streams 100rel extensions and preconditions extensions in the SIP Supported header: and based on operator policy then
• the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "continuity check performed on previous circuit"
• if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, 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">IAM: Continuity Indicator: 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="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>

TP101061	**SIP reference: RFC 3261**	**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/	
SIP selection 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 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 performed on previous circuit**, USI: A-law or absent COT: Continuity Indicator: **continuity check successful**	
Comments	(see message flow below)	

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><td>TP101062</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISUP reference:
EN 383 001 clause 6.1.3.5.2.2</td></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>

TP101063	**SIP reference: RFC 3261**	**ISUP reference: EN 383 001 clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/	
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19	
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 **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**	
SIP parameter values	Offer: m=audio 4711 RTP/AVP 8 m= audio 4712 RTP/AVP 8 m= video 4713 RTP/AVP 31 Answer: m=audio 4711 RTP/AVP 8 m=audio 0 RTP/AVP 8 m=video 0 RTP/AVP 31	
ISUP parameter values		
Comments	(see flow below)	

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

5.2.1.2 Sending of the Subsequent Address Message (SAM)

<table>
<tr><th>TP102001</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 clause 6.2 a)</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Subsequent Address Message (SAM)/</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">PICS 3/8</td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">SAM; subsequent number (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</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SAM</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SAM</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE</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>TP102002</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 clause 6.2 b)</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Subsequent Address Message (SAM)/</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">PICS 3/8</td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</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>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>REL</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>←</td><td>RLC</td></tr>
</table>

5.2.1.3 Sending of COT

TP103001	**SIP reference: RFC 3261**			**ISUP reference: Q.1912.5 clause 6.3**	
TSS reference	SIP-ISUP/Basic call/COT				
SIP selection 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**			**ISUP reference: Q.1912.5 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				ISUP reference: Q.1912.5 clause 6.5 2)
TSS reference	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT on receipt of an ACM message where the Called party status indicator is set to "no indication" • 183 Session Progress response is sent from the I-IWU. • The received ACM is encapsulated in the 183 Session Progress				
SIP parameter values					
ISUP parameter values	ACM Called party status: no 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				ISUP reference: Q.1912.5 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)

TP105001	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 clause 6.6**
TSS reference	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the **event information parameter event indicator** is set to "Alerting" • the 180 Ringing SIP response is sent • The received CPG is encapsulated in the 180 Ringing				
SIP parameter values					
ISUP parameter values	ACM: Called party status "no indication" CPG; **event information parameter 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
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP105002</th><th colspan="2">SIP reference: RFC 3261</th><th colspan="3">ISUP reference:
Q.1912.5 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</th><th colspan="3">ISUP reference:
Q.1912.5 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 colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 clause 6.7</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/ Receipt of the Answer message (ANM).</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 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, (Q.1902.4) is successfully completed, and;
• The I-IWU determines (using the procedures defined in RFC 3312) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable)
In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "notification not required", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312) that sufficient preconditions have been met for the session to proceed</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">200 OK INVITE with encapsulated ANM</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">ANM</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>

5.2.1.7 Receipt of the Connect message (CON)

TP107001	**SIP reference: RFC 3261**			**ISUP reference: Q.1912.5 clause 6.4, 6.7**	
TSS reference	SIP-ISUP/Basic call/ Receipt of the CONNECT message (CON).				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**SDP offer was received** in 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, (Q.1902.4) is successfully completed, and; • The I-IWU determines (using the procedures defined in RFC 3312) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable). In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "*notification not required*", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312) that sufficient preconditions have been met for the session to proceed.				
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	200 OK INVITE(CON)	←		←	CON
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

5.2.1.8 Receipt of the REL message

TP108001	**SIP reference: RFC 3261**			**ISUP reference: Q.1912.5 clause 6.11.2**	
TSS reference	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, 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
	SIP_FAILURE_VA(REL)	←		←	REL
	ACK	→		→	RLC

Table 1

Values for test purpose TP108001		
←SIP Message SIP_FAILURE_VA		← REL Cause Indicators parameter CV_ISUP,
VA_1	404 Not Found	Cause Value No. 1 ("unallocated (unassigned) number")
VA_2	500 Server Internal Error	Cause Value No. 2 ("no route to network")
VA_3	500 Server Internal Error	Cause Value No. 3 ("no route to destination")
VA_4	500 Server Internal Error	Cause Value No. 4 ("Send special information tone")
VA_5	404 Not Found	Cause Value No. 5 ("Misdialled trunk prefix")
VA_6	500 Server Internal Error	Cause Value No. 8 ("Pre-emption")
VA_7	500 Server Internal Error	Cause Value No. 9 ("Pre-emption-circuit reserved for reuse")
VA_8	486 Busy Here	Cause Value No. 17 ("user busy")
VA_9	480 Temporarily unavailable	Cause Value No. 18 ("no user responding")
VA_10	480 Temporarily unavailable	Cause Value No. 19 ("no answer from the user")
VA_11	480 Temporarily unavailable	Cause Value No. 20 ("subscriber absent")
VA_12	480 Temporarily unavailable	Cause Value No. 21 ("all rejected")
VA_13	410 Gone	Cause Value No. 22 ("number changed")
VA_14	480 Temporarily unavailable	Cause Value No. 25 ("Exchange routing error")
VA_15	502 Bad Gateway	Cause Value No. 27 ("destination out of order")
VA_16	484 Address Incomplete	Cause Value No. 28 ("invalid number format (address 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. 38-47) (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)

<table>
<tr><td>TP108002</td><td colspan="5">SIP reference: RFC 3261</td><td colspan="2">ISUP reference:
Q.1912.5 clause 6.11.2</td></tr>
<tr><td>TSS reference</td><td colspan="7">SIP-ISUP /Basic call/ Receipt of the Release message (REL)/</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">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</td></tr>
<tr><td>SIP parameter values</td><td colspan="7">SIP Statue-Code: SIP_FAILURE_VA (PIXIT)</td></tr>
<tr><td>ISUP parameter values</td><td colspan="7">REL; cause value: CV_ISUP (PIXIT)</td></tr>
<tr><td rowspan="5">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td colspan="3">ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td colspan="3">IAM</td></tr>
<tr><td>183 Session Progress(ACM)</td><td>←</td><td></td><td>←</td><td colspan="3">ACM(no indication)</td></tr>
<tr><td>SIP_FAILURE_VA(REL)</td><td>←</td><td></td><td>←</td><td colspan="3">REL</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td colspan="3">RLC</td></tr>
</table>

Table 2

<table>
<tr><th colspan="3">Values for test purpose TP108002</th></tr>
<tr><th colspan="2">←SIP Message
SIP_FAILURE_VA</th><th>← REL
Cause Indicators parameter
CV_ISUP,</th></tr>
<tr><td>VA_1</td><td>486 Busy Here
Cause Value No. 17</td><td>Cause Value No. 17 ("user busy")</td></tr>
<tr><td>VA_2</td><td>480 Temporarily unavailable</td><td>Cause Value No. 18 ("No user responding")</td></tr>
<tr><td>VA_3</td><td>480 Temporarily unavailable</td><td>Cause Value No. 21 ("all rejected")</td></tr>
<tr><td>VA_4</td><td>410 Gone</td><td>Cause Value No. 22 ("number changed")</td></tr>
<tr><td>VA_5</td><td>502 Bad Gateway</td><td>Cause Value No. 27 ("destination out of order")</td></tr>
<tr><td>VA_6</td><td>484 Address Incomplete</td><td>Cause Value No. 28 ("invalid number format (address incomplete")</td></tr>
<tr><td>VA_7</td><td>480 Temporarily unavailable</td><td>Cause Value No. 31 ("normal unspecified")
(Class default)</td></tr>
<tr><td>VA_8</td><td>500 Server Internal Error</td><td>Cause Value in the Class 010 (resource unavailable, Cause Value No. 38-47)
(47 is class default)</td></tr>
<tr><td>VA_9</td><td>500 Server Internal Error</td><td>Cause Value No. 63 ("service option not available, unspecified")
(Class default)</td></tr>
<tr><td>VA_10</td><td>500 Server Internal Error</td><td>Cause Value No. 88 ("incompatible destination")</td></tr>
<tr><td>VA_11</td><td>500 Server Internal Error</td><td>Cause Value No. 111 ("protocol error, unspecified")
(Class default)</td></tr>
</table>

TP108003	SIP reference: RFC 3261	ISUP reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria		
ISUP selection criteria		
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message where the CPS indicator is set to "**subscriber free**", having sent a 180 Ringing message on receipt of an ISUP REL • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side. • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA with the encapsulated REL message	
SIP parameter values	SIP Statue-Code: SIP_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	ISUP reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria		
ISUP selection criteria		
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message where the CPS indicator is set to "**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	
Comments	SIP SUT ISUP INVITE → → IAM 183 Session Progress ← ACM 180 Ringing ← ← CPG SIP_FAILURE_VA ← ← 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><th>TP108005</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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, 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="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></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>TP108006</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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, 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="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">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>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>

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**				**ISUP reference: Q.1912.5 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**				**ISUP reference: Q.1912.5 clause 6.11.3**
TSS reference	SIP-ISUP/Basic call/ Autonomous release at I-IWU				
SIP selection criteria					
ISUP selection criteria	PICS 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**				**ISUP reference: Q.1912.5 clause 6.11.3**
TSS reference	SIP-ISUP/Basic call/ Autonomous release at I-IWU				
SIP selection criteria					
ISUP selection criteria					
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**				**ISUP reference: Q.1912.5 clause 6.11.3**
TSS reference	SIP-ISUP/Basic call/ Autonomous release at I-IWU				
SIP selection criteria					
ISUP selection criteria	PICS 3/4				
Test purpose	Ensure that the SUT on receipt of insufficient digits received in an INVITE messages, • sends an 484 Address Incomplete message				
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	484 Address incomplete	←			
	ACK	→			

TP109004	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 clause 6.11.3**
TSS reference	SIP-ISUP/Basic call/ Autonomous release at I-IWU				
SIP selection criteria	PICS 3/4				
ISUP selection criteria					
Test purpose	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				
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	INVITE(IAM)	→			
	484 Address incomplete	←			
	ACK	→			

TP109005	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 clause 6.11.3**
TSS reference	SIP-ISUP/Basic call/ Autonomous release at I-IWU				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT in congestion on receipt of INVITE message sends an 480 Temporarily Unavailable message				
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	480 Temporarily unavailable	←			
	ACK	→			

TP109006	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause 6.11.3	
TSS reference	SIP-ISUP/Basic call/ Autonomous release at I-IWU				
SIP selection 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
Comments:	SIP INVITE → 500 Server internal error← ACK →		SUT → IAM ← ACM(???) → REL ← RLC		ISUP

TP109007	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause 6.11.3	
TSS reference	SIP-ISUP/Basic call/ Autonomous release at I-IWU				
SIP selection 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			ISUP reference: Q.1912.5 clause 6.11.3	
TSS reference	SIP-ISUP/Basic call/ Autonomous release at I-IWU				
SIP selection criteria					
ISUP selection criteria					
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**			**ISUP reference: Q.1912.5 clause 6.11.1**	
TSS reference	SIP-ISUP/Basic call/ Receipt of the BYE-CANCEL message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT on 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**			**ISUP reference: Q.1912.5 clause 6.11.1**	
TSS reference	SIP-ISUP/Basic call/ Receipt of the BYE-CANCEL message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT on 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**			**ISUP reference: Q.1912.5 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**				**ISUP reference: Q.1912.5 clause 6.11.4 and 5**
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a 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		ISUP reference: Q.1912.5 clause 6.11.4 and 5		
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a GRS message sends • 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		ISUP reference: Q.1912.5 clause 6.11.4		
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking 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**				**ISUP reference: Q.1912.5 clause 6.11.4 and 5**
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a 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**				**ISUP reference: Q.1912.5 clause 6.11.4 and 5**
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria:					
ISUP selection criteria:					
Test purpose:	Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a GRS message sends 200 OK INVITE if the SUT has not yet received an ACK for the 200 OK INVITE. • The SUT shall wait until it receives the ACK for the 200 OK INVITE before sending the 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				ISUP reference: Q.1912.5 clause 6.11.4
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking 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				ISUP reference: Q.1912.5 clause 6.11.4 and 5
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a RSC message sends • a 500 Server Internal Error on the SIP side.				
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	500 Server Internal Error(REL)	←		←	RSC
	ACK	→		→	RLC

TP111008	SIP reference: RFC 3261		ISUP reference: Q.1912.5 clause 6.11.4 and 5		
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a GRS message sends • 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	←		←	GRS
	ACK	→		→	GRA
Comments	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM 500 Server Internal Error← ← GRS ACK → → GRA				

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

TP111010	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 6.11.4 and 5
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking 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				ISUP reference: Q.1912.5 clause 6.11.4 and 5
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT 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				ISUP reference: Q.1912.5 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				ISUP reference: Q.1912.5 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**			**ISUP reference: Q.1912.5 clause 6.10**
TSS reference	SIP-ISUP/Basic call/			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that the SUT, on receipt of a RESUME message containing the suspend/resume indicator set to "network initiated" • The RES is transferred in an INFO message			
SIP parameter values				
ISUP parameter 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**				**ISUP reference: Q.1912.5 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		ISUP reference: Q.1912.5 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		ISUP reference: Q.1912.5 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)

TP301002	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause 7.1 1 b)
TSS reference	**ISUP-SIP /Basic call/Sending of the INVITE message**			
SIP selection criteria				
ISUP selection criteria				
Test purpose	**Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan sends the INVITE message**			
SIP parameter values				
ISUP parameter values	**IAM; Called party number complete number**			
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)

TP301003	SIP reference: RFC 3261			ISUP reference: Q.1912.5 clause 7.1 1 c)
TSS reference	ISUP-SIP /Basic call/Sending of the INVITE message			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete **called party number** where the end of address signalling is determined by analysis of the called party number to indicate that **a sufficient number of digits has been received** to route the call to the called party • sends the INVITE message			
SIP parameter values				
ISUP parameter values	IAM; **Called party number**: complete number			
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)

TP301004	**SIP reference: RFC 3261**			**ISUP reference: Q.1912.5 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**			**ISUP reference: Q.1912.5 clause 7.1 A)**	
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria					
ISUP selection criteria	PICS 1/5				
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message with the **complete called party number** containing the Continuity Check indicator in the Nature of Connection Indicators parameter is set to indicate **"continuity check not required"** • sends a INVITE message				
SIP parameter values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP301006	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 7.1 A)
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria	NOT PICS 4/4 AND NOT PICS 4/5 AND NOT PICS 4/15				
ISUP selection criteria	PICS 1/5				
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message with the **complete called party number** containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to **"continuity check required on this circuit"** • Sends the INVITE after the receipt of the Continuity message with the Continuity Indicators parameter "**continuity check successful**"				
SIP parameter values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	→			
	COT	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP301007	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 7.1 A)
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria	NOT PICS 4/4 AND NOT PICS 4/5 AND NOT PICS 4/15				
ISUP selection criteria	PICS 1/5				
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message with the **complete called party number** containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to **"continuity check performed on previous circuit"** • Sends the INVITE after the receipt of the Continuity message with the Continuity Indicators parameter "**continuity check successful**"				
SIP parameter values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	→			
	COT	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><td>TP301008</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISUP reference:
Q.1912.5 clause 7.1 A)</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">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 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"</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="3">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></td></tr>
</table>

<table>
<tr><td>TP301009</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISUP reference:
Q.1912.5 clause 7.1 A)</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">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 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</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>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>T8 expiry</td><td></td><td></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>
</table>

<table>
<tr><th>TP301010</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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 not required"
• sends an INVITE message without precondition using the SDP offer in the INVITE</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>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>TP301011</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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 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</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>

TP301012	SIP reference: RFC 3261				ISUP reference: Q.1912.5 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 performed on previous circuit"** sends an INVITE message with precondition using the SDP offer in the INVITE. The SDP offer or answer carrying the confirmation of a precondition being met is sent when the Continuity message with the Continuity Indicators parameter set to **"continuity check successful"** was received and the requested preconditions are met in the SIP network				
SIP parameter values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	183 Session Progress
				→	PRACK
				←	200 OK PRACK
	COT(successful)	→		→	UPDATE
				←	200 OK UPDATE
		Preconditions met			
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP301013	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 7.1 B)
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 4/15				
ISUP selection criteria	PICS 1/5 AND PICS 4/2				
Test purpose	The SUT in Idle state, receives an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to **"continuity check performed on previous circuit"** and sends an INVITE message with precondition using the SDP offer in the INVITE. The call has been cleared **before** an early dialogue has been established. Ensure that the SUT • sends CANCEL if on the SIP side the internal resource reservation was unsuccessful. • REL with Cause Value 47 (resource unavailable, unspecified) shall be sent on the ISUP side of the O-IWU				
SIP parameter values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	100 Trying
	internal resource reservation was unsuccessful				
	REL(#47)	←		→	CANCEL
	RLC	→		←	200 OK CANCEL
				←	487 Request Terminated
				→	ACK

<table>
<tr><th>TP301014</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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 performed on previous circuit" and sends an INVITE message with precondition using the SDP offer in the INVITE. The call has been cleared after an early dialogue with the message defined as SIP_MESSAGE_VA has been established. Ensure that the SUT
• sends CANCEL if on the SIP side the internal resource reservation was unsuccessful.
• REL with Cause Value 47 (resource unavailable, unspecified) shall be sent on the ISUP side of the O-IWU</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>SIP_MESSAGE_VA</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/BYE</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td>←</td><td>200 OK CANCEL/BYE</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</th><th colspan="2">ISUP reference:
Q.1912.5 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 call has been cleared before an early dialogue has been established. Ensure that the SUT
• sends CANCEL if on the SIP side the internal resource reservation was unsuccessful.
• REL with Cause Value 47 (resource unavailable, unspecified) shall be sent on the ISUP side of the O-IWU.</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">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>

<table>
<tr><th>TP301016</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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 call has been cleared after an early dialogue with the message defined as SIP_MESSAGE_VA has been established. Ensure that the SUT
• sends CANCEL if on the SIP side the internal resource reservation was unsuccessful.
• REL with Cause Value 47 (resource unavailable, unspecified) shall be sent on the ISUP side of the O-IWU</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>SIP_MESSAGE_VA</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/BYE</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td>←</td><td>200 OK CANCEL/BYE</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 5

<table>
<tr><td colspan="2">Values for test purpose
• TP301014
• TP301016
• TP301018
• TP301020
• TP301031
• TP301033</td></tr>
<tr><td>VA</td><td>SIP MESSAGE_VA</td></tr>
<tr><td>VA_1</td><td>180 Ringing</td></tr>
<tr><td>VA_2</td><td>181 Call Is Being Forwarded</td></tr>
<tr><td>VA_3</td><td>182 Queued</td></tr>
<tr><td>VA_4</td><td>183 Session Progress without SDP</td></tr>
</table>

TP301017	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 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	The SUT in Idle state, receives an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to **"continuity check required on this circuit"** and sends an INVITE message with precondition using the SDP offer in the INVITE. The Continuity message is received with the Continuity Indicators parameter set to **"continuity check failed"**. The call has been cleared **before** an early dialogue has been established. Ensure that the SUT • sends CANCEL on the SIP side				
SIP parameter values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	100 Trying
	COT(unsuccessful)	→		→	CANCEL
				←	200 OK CANCEL
				←	487 Request Terminated
				→	ACK

TP301018	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 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	The SUT in Idle state, receives an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to **"continuity check required on this circuit"** and sends an INVITE message with precondition using the SDP offer in the INVITE. The Continuity message is received with the Continuity Indicators parameter set to **"continuity check failed"**. The call has been cleared **after** an early dialogue with the message defined as SIP_MESSAGE_VA has been established. Ensure that the SUT • sends CANCEL or BYE on the SIP side				
SIP parameter values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	SIP_MESSAGE_VA
	COT(unsuccessful)	→		→	CANCEL
				←	200 OK CANCEL
				←	487 Request Terminated
				→	ACK

TP301019	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 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	The SUT in Idle state, receives an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to **"continuity check required on this circuit"** and sends an INVITE message with precondition using the SDP offer in the INVITE. The ISUP Timer **T8 expires**. The call has been cleared **before** an early dialogue has been established. Ensure that the SUT • sends CANCEL on the SIP side				
SIP parameter values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	100 Trying
	T8 expires				
	REL(#47)	←		→	CANCEL
	RLC	→		←	200 OK CANCEL
				←	487 Request Terminated
				→	ACK

TP301020	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 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	The SUT in Idle state, receives an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "**continuity check required on this circuit**" and sends an INVITE message with precondition using the SDP offer in the INVITE. The ISUP Timer **T8 expires**. The call has been cleared **after** an early dialogue with the message defined as SIP_MESSAGE_VA has been established. Ensure that the SUT • sends CANCEL or BYE on the SIP side				
SIP parameter values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	SIP_MESSAGE_VA
	T8 expires				
	REL(#47)	←		→	CANCEL/BYE
	RLC	→		←	200 OK CANCEL/BYE
				←	487 Request Terminated
				→	ACK

<table>
<tr><th>TP301021</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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>

<table>
<tr><th>TP301022</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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 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="10">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></td></tr>
<tr><td>APM</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>TP301023</th><th colspan="4">SIP reference: RFC 3261</th><th>ISUP reference:
Q.1912.5 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 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</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(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>

<table>
<tr><th>TP301024</th><th colspan="4">SIP reference: RFC 3261</th><th>ISUP reference:
Q.1912.5 clause 7.1 C) 2.4</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">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 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="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(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>

<table>
<tr><th>TP301025</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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"
• Sends not the INVITE if the Continuity message was not received, i.e. the BICC timer T8 expires.
o 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="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></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>
</table>

<table>
<tr><th>TP301026</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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>

TP301027	SIP reference: RFC 3261	ISUP reference: Q.1912.5 clause 7.1 D) 2.2
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 • APM with Action indicator set to "Connected" - for the forward bearer set-up cases (with, or without bearer control tunnelling) where the incoming Connect Type is "notification required", and for the fast set-up (backward) case	
SIP parameter values		
ISUP parameter values		
Comments		

ISUP/BICC		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><td>TP301028</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISUP reference:
Q.1912.5 clause 7.1 D) 2.3</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">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</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 Connect indication - for the backward bearer set-up case 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/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>

TP301029	SIP reference: RFC 3261				ISUP reference: Q.1912.5 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)

TP301030	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 7.1 D)
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 4/15				
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

TP301031	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 7.1 D)
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 4/15				
ISUP selection criteria	PICS 1/5 AND PICS 4/2				
Test purpose	The SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "**COT to be expected**", sends an INVITE message with precondition using the SDP offer in the INVITE • Ensure that the SUT sends CANCEL if the ISUP timer **T8 expires** if the call has been cleared **after** an early dialogue with the message defined as SIP_MESSAGE_VA has been established				
SIP parameter values					
ISUP parameter values					
Comments	BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	SIP_MESSAGE_VA
	T8 expires				
	REL(#47)	←		→	CANCEL/BYE
	RLC	→		←	200 OK CANCEL/BYE
				←	487 Request Terminated
				→	ACK

TP301032	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 7.1
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 containing Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "**COT to be expected**". Ensure that the SUT • sends CANCEL if on the SIP side the internal resource reservation was unsuccessful and if the call has been cleared **before** an early dialogue with the message has been established • A REL with Cause Value 47 (resource unavailable, unspecified) shall be sent on the ISUP side of the O-IWU				
SIP parameter values					
ISUP parameter values					
Comments	BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	100 Trying
	internal resource reservation was unsuccessful				
	REL(#47)	←		→	CANCEL
	RLC	→		←	200 OK CANCEL
				←	487 Request Terminated
				→	ACK

<table>
<tr><th>TP301033</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 clause 7.1</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 containing Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "COT to be expected". Ensure that the SUT
• sends CANCEL if on the SIP side the internal resource reservation was unsuccessful and if the call has been cleared after an early dialogue with the message defined as SIP_MESSAGE_VA has been established
• A REL with Cause Value 47 (resource unavailable, unspecified) shall be sent on the ISUP side of the O-IWU</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>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>SIP_MESSAGE_VA</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/BYE</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td>←</td><td>200 OK CANCEL/BYE</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>TP301034</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 clause 7.1.1</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">Based on table 6</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 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="5">INVITE : a_b_m_LINE_VALUE</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: TMR : ISUP_TMR</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><th>TP301035</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 clause 7.1.1</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">Based on table 7</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 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</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: a_b_m_LINE_VALUE</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: USI : ISUP_USI</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 6

Values for test purposes TP301053						
	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 TP301053, TP301054									
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) Note 1	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000) Note 1
	"speech"	"Speech"	"G.711 A-law"	Ignore	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000
VA_02	"3.1 KHz audio"	USI Absent		Ignore	audio	RTP/AVP	0 and/or 8 Note 1	AS:64	rtpmap:0 PCMU/8000 and/or rtpmap:8 PCMA/8000 Note 1
VA_03	"3.1 KHz audio"	"3.1 KHz audio"	"G.711 µ-law"		audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)
	"3.1 KHz audio"	"3.1 KHz audio"	"G.711 A-law"		audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000
VA_04	"3.1 KHz audio"	"3.1 KHz audio"	"G.711 A-law"	"Facsimile Group 2/3"	image	tcptl	t38	AS:64	Based on T.38.
	"3.1 KHz audio"	"3.1 KHz audio"	"G.711 A-law"	"Facsimile Group 2/3"	image	udptl	t38	AS:64	Based on T.38.
VA_05	"3.1 KHz audio"	"3.1 KHz audio"	"G.711 µ-law"	"Facsimile Group 2/3"	image	udptl	t38	AS:64	Based on T.38.
	"3.1 KHz audio"	"3.1 KHz audio"	"G.711 µ-law"	"Facsimile Group 2/3"	image	tcptl	t38	AS:64	Based on T.38.
VA_06	"64 kbit/s unrestricted"	"Unrestricted digital inf. W/tone/ann."	N/A	Ignore	audio	RTP/AVP	9	AS:64	Rtpmap:9 G722/8000
VA_07	"64 kbit/s unrestricted"	"Unrestricted digital information"	N/A	Ignore	Audio	RTP/AVP	Dynamic PT	AS:64	rtpmap:<dynamic-PT> CLEARMODE/8000 Note 2

<table>
<tr><th>TP301036</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 clause 7.1.2</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">NOT PICS 1/9</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 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</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: To: …</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>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>TP301037</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 clause 7.1.2</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">NOT PICS 1/9</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 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</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: To: sip: ….; user=phone</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>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>

TP301038	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 clause 7.1.2**
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message				
SIP selection 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)

TP301039	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 clause 7.1.2**
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message				
SIP selection 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)

TP301040	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 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)

TP301041	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 clause 7.1.2**
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message				
SIP selection 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)

TP301042	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 7.1.2
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message				
SIP selection 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)

TP301043	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 7.1.2
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message				
SIP selection 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)

TP301044	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 clause 7.1.2**
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message				
SIP selection 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**				**ISUP reference: Q.1912.5 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)

TP302002	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 7.2.1
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent				
SIP selection criteria	PICS 3/2				
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 indicate "**continuity check not required**" sends a INVITE message On receipt of a SAM from the ISUP the SUT shall: 1) Stop timer TOIW3 (if it is running) 2) TOIW2 shall be restarted and the SUT shall invoke the following procedures: a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent c) The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question d) All other contents of the new INVITE are interworked from the parameters of the original IAM				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	SAM	→		→	INVITE(IAM)
	SAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP302003</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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 1/5 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 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</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</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</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>

TP302004	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 clause 7.2.1**
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent				
SIP selection 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	ISUP/BICC		SUT		SIP-I
	IAM	→			
	SAM	→			
	COT	→		→	INVITE(IAM)
	SAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP302005	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 clause 7.2.1**
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent				
SIP selection 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	→			

<table>
<tr><th>TP302006</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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 1/5 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 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</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="6">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></td></tr>
<tr><td></td><td colspan="3">T8 expires</td><td></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>
</table>

TP302007	**SIP reference: RFC 3261**	**ISUP reference:** Q.1912.5 clause 7.2.1
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection 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)
ACM	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP302008</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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 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 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="5">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="5"></td></tr>
<tr><td rowspan="14">Comments:</td><td colspan="5">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.</td></tr>
<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>INVITE1(IAM)</td></tr>
<tr><td>SAM x</td><td>→</td><td></td><td></td><td></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>SAM</td><td>→</td><td></td><td>→</td><td>INVITE2 (IAM and digits from SAM X + SAM Y)</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>

TP302009	SIP reference: RFC 3261	ISUP reference: Q.1912.5 clause 7.2.1
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria	PICS 3/2 AND NOT PICS 4/15	
ISUP selection criteria	PICS 1/4 AND NOT PICS 4/2	
Test purpose	Ensure that the SUT in Idle state, on 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	
SIP parameter values		
ISUP parameter values		
Comments	(see flow below)	

ISUP/BICC		SUT		SIP-I
IAM	→			
SAM x	→			
COT	→		→	INVITE(IAM)
SAM y	→		→	INVITE(IAM)
ACM	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

<table>
<tr><td>TP302010</td><td colspan="4">SIP reference: RFC 3261</td><td colspan="2">ISUP reference:
Q.1912.5 clause 7.2.1</td></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 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="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>

TP302011	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 7.2.1
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent				
SIP selection criteria	PICS 3/2 AND NOT PICS 4/15				
ISUP selection criteria	PICS 1/4 AND PICS 4/2				
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "**COT to be expected**" The sending of the INVITE delays until all the following conditions are satisfied: • Continuity message, with the Continuity Indicators parameter set to "**continuity**" shall be received • Bearer Set-up Connect indication - for the backward bearer set-up case was received. On receipt of a SAM from the BICC the SUT shall: 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 x	→			
	COT	→		→	INVITE(IAM)
	SAM y	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP302012</th><th>SIP reference: RFC 3261</th><th>ISUP reference:
Q.1912.5 clause 7.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="2">ISUP-SIP/Basic call/Receipt of SAM after invite has been sent</td></tr>
<tr><td>SIP selection criteria</td><td colspan="2">PICS 3/2 AND NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="2">PICS 1/4 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 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="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/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>
</td></tr>
</table>

TP302013	**SIP reference: RFC 3261**	**ISUP reference: Q.1912.5 clause 7.2.1**
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection 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)
ACM	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

TP302014	**SIP reference: RFC 3261**	**ISUP reference: Q.1912.5 clause 7.2.1**
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection 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)
ACM	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

TP302015	**SIP reference: RFC 3261**	**ISUP reference: Q.1912.5 clause 7.2.1**
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection 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)
ACM	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

TP302016	**SIP reference: RFC 3261**	**ISUP reference: Q.1912.5 clause 7.2.1**
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection 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)
ACM	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

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

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

5.2.2.3 Sending of the ACM message

TP303001	SIP reference: RFC 3261	ISUP reference: Q.1912.5 clause 7.1, Q.764 clause 2.1.4.8
TSS reference	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria	PICS 1/3	
ISUP selection criteria	PICS 4/9	
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete **called party number** and the **sending complete** indication Sends the INVITE message to called user Sends the ACM message with • the **CPS indicator** set to " no indication (00)" • the **Called party's category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)" • the **interworking indicator** set to " INT_IND_VAL" • the **ISUP indicator** set to "ISUP_IND_ID" • the **ISDN access indicator** set to "ISDN_ACC_IND_VAL "	
SIP parameter values		
ISUP parameter values	IAM; **Called party number**: complete number ACM, **CPS indicator** no indication (00) **Called party's category indicator**: no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator**: INT_IND_VAL (PIXIT) **ISUP indicator**: ISUP_IND_ID (PIXIT) **ISDN access indicator** ISDN_ACC_IND_VAL (PIXIT)	

Comments

ISUP/BICC		SUT		SIP-I
IAM	→		→	INVITE(IAM)
ACM(no indication)	←			
CPG(Alerting)	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

TP303002	**SIP reference: RFC 3261**	**ISUP reference: Q.1912.5 clause 7.1, Q.764 clause 2.1.4.8**
TSS reference	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria	PICS 1/3	
ISUP selection criteria	PICS 4/9	
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the **maximum number of digits used in the national numbering plan** • Sends the INVITE message to called user • Sends the ACM message with • the **CPS indicator** set to " no indication (00)" • the **Called party's category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)" • the **interworking indicator** set to " INT_IND_VAL" • the **ISUP indicator** set to "ISUP_IND_ID" • the **ISDN access indicator** set to "ISDN_ACC_IND_VAL "	
SIP parameter values		
ISUP parameter values	IAM; **Called party number**: complete number ACM, Backward call indicator is set to the value in the encapsulated ACM**CPS indicator** no indication (00) **Called party's category indicator**: no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator**: INT_IND_VAL (PIXIT) **ISUP indicator**: ISUP_IND_ID (PIXIT) **ISDN access indicator** ISDN_ACC_IND_VAL (PIXIT)	
Comments		

ISUP/BICC		SUT		SIP-I
IAM	→		→	INVITE(IAM)
ACM(no indication)	←			
CPG(Alerting)	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP303003</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 clause 7.1,
Q.764 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>

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

<table>
<tr><th>TP303005</th><th colspan="5">SIP reference: RFC 3261</th><th>ISUP reference:
Q.1912.5 clause 7.1, 7.3.1</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP /Basic call/Sending of the ACM message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6">PICS 1/3</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6">NOT PICS 4/9</td></tr>
<tr><td>Test purpose</td><td colspan="6">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="6"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">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="12">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</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>SAM</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(IAM)</td></tr>
<tr><td></td><td colspan="3">T_{OIW2} expiry</td><td colspan="2"></td></tr>
<tr><td>ACM(no indication)</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>CPG(Alerting)</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>

<table>
<tr><th>TP303006</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 clause 7.1 1) a), 7.3.1</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, 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</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 ACM</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><th>TP303007</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 clause 7.1 1 a), 7.3.2</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 3/1</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 on receipt of a 183 Session Progress with encapsulated ACM
• Sends the INVITE message
• The encapsulated ACM message is sent unchanged backward</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="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>183 Session Progress(ACM)</td></tr>
<tr><td>CPG(Alerting)</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></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>TP303011</td><td colspan="5">SIP reference: RFC 3261</td><td>ISUP reference:
Q.1912.5 clause 7.1, 7.3.1, 7.4</td></tr>
<tr><td>TSS reference:</td><td colspan="6">ISUP-SIP /Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6">PICS 1/3</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6">PICS 4/2 AND NOT PICS 4/9</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 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="6"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">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 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></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></td><td colspan="3">T_{OIW1} expiry</td><td colspan="2"></td></tr>
<tr><td>ACM(no indication)</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>CPG(Alerting, BCi)</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>

TP303012	SIP reference: RFC 3261	ISUP reference: Q.1912.5 clause 7.1, 7.3.1, 7.4
TSS reference	ISUP-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria	PICS 1/3 AND PICS 3/2 AND PICS 4/5 AND PICS 4/4 AND PICS 4/15	
ISUP selection criteria	PICS 4/2 AND NOT PICS 4/9	
Test purpose	Ensure that the SUT if overlap addressing is to be used toward the SIP network, on receipt of an IAM message containing the **minimum number of digits required for routing the call** has been received (start timer TOIW2 and invoke the appropriate outgoing SIP signalling procedure) and the continuity check is performed (ISUP) or COT is expected (BICC). After the expiry of Toiw2 • sends the ACM message with the CPS indicator set to " no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "INT_IND_VAL", the ISUP indicator set to "ISUP_IND_ID", the ISDN access indicator set to "ISDN_ACC_IND_VAL"	
SIP parameter values		
ISUP parameter values	**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	

Comments

ISUP/BICC		SUT		SIP-I
IAM	→		→	INVITE(IAM)
			←	183 Session Progress without encapsulated ACM
COT	→		→	UPDATE
			←	200 OK UPDATE
		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)

<table>
<tr><th>TP303013</th><th colspan="5">SIP reference: RFC 3261</th><th>ISUP reference:
Q.1912.5 clause 7.1, 7.3.1, 7.4</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP /Basic call/Sending of the ACM message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6">PICS 1/3</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6">PICS 4/2 AND NOT PICS 4/9</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 complete called party number, the continuity check is performed (ISUP) or COT is expected (BICC) indication receipt of a 180 Ringing message
• Sends the ACM message with
• the CPS indicator set to the value in the encapsulated ACM
• the Called party's category indicator set to the value in the encapsulated ACM
• the interworking indicator set to the value in the encapsulated ACM
• the ISUP indicator set to the value in the encapsulated ACM
• the ISDN access indicator set to the value in the encapsulated ACM</td></tr>
<tr><td>SIP parameter values</td><td colspan="6"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">IAM; Called party number: complete number
ACM, Backward call indicator is set to the value in the encapsulated ACM</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">INVITE(IAM)</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>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>

<table>
<tr><td>TP303014</td><td colspan="5">SIP reference: RFC 3261</td><td>ISUP reference:
Q.1912.5 clause 7.1, 7.3.1, 7.4</td></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP /Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6">PICS 1/3 AND PICS 3/2 AND NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6">PICS 3/8 AND PICS 4/2 AND NOT PICS 4/9</td></tr>
<tr><td>Test purpose</td><td colspan="6">Ensure that the SUT if overlap addressing is to be used toward the SIP network, on receipt of an IAM message containing the minimum number of digits required for routing the call has been received (start timer TOIW2 and invoke the appropriate outgoing SIP signalling procedure) and the continuity check is performed (ISUP) or COT is expected (BICC). After the expiry of Toiw2
• sends the ACM message with the CPS indicator set to " no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to " INT_IND_VAL", the ISUP indicator set to "ISUP_IND_ID", the ISDN access indicator set to "ISDN_ACC_IND_VAL "</td></tr>
<tr><td>SIP parameter values</td><td colspan="6"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">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="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>COT</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(IAM)</td></tr>
<tr><td></td><td colspan="3">T_{OIW2} expiry</td><td colspan="2"></td></tr>
<tr><td>ACM(no indication)</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>CPG(Alerting)</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>

TP303015	**SIP reference: RFC 3261**				**ISUP reference:** **Q.1912.5 clause 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 • the **CPS indicator** set to the value in the encapsulated ACM • the **Called party's category indicator** set to the value in the encapsulated ACM • the **interworking indicator** set to the value in the encapsulated ACM • the **ISUP indicator** set to the value in the encapsulated ACM • the **ISDN access indicator** set to the value in the encapsulated ACM				
SIP parameter values					
ISUP parameter 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**				**ISUP reference:** **Q.1912.5 clause 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**			**ISUP reference: Q.1912.5 clause 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				
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**			**ISUP reference: Q.1912.5 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

<table>
<tr><th>TP306001</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 clause 7.5, 7.5.1</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISUP-SIP/Basic call/ Sending 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">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</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">200 OK INVITE;</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">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</td></tr>
<tr><td rowspan="7">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>CON</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 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>Comments</td><td colspan="5">ISUP SUT SIP
IAM → → INVITE(IAM)
CON ← ← 200 OK INVITE(CON)</td></tr>
</table>

5.2.2.7 Receipt of the Release message (REL)

<table>
<tr><th>TP307001</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 clause 7.7.1, 1)</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 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</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>REL</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>TP307002</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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</th><th colspan="2">ISUP reference:
Q.1912.5 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**				**ISUP reference: Q.1912.5 clause 7.7.1 2) 3)**
TSS reference	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/				
SIP selection criteria					
ISUP selection criteria					
Test purpose	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**				**ISUP reference: Q.1912.5 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				ISUP reference: Q.1912.5 clause 7.7.1 3)
TSS reference	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message **after** an early dialogue with the SIP message defined with the **SIP_MESSAGE_VA** has been established • The SUT shall send a CANCEL or BYE request. The received REL is encapsulated in the BYE or CANCEL				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		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**				**ISUP reference: Q.1912.5 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**				**ISUP reference: Q.1912.5 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 TP308003		
VA	**←REL (Cause Value) CV_ ISUP**	**←4XX/5XX/6XX SIP message SIP_Failure_VA**
VA_01	127 Interworking	400 Bad Request
VA_02	127 Interworking	402 Payment Required
VA_03	127 Interworking	403 Forbidden
VA_04	1 Unallocated number	404 Not Found
VA_05	127 Interworking	405 Method Not Allowed
VA_06	127 Interworking	406 Not Acceptable
VA_07	127 Interworking	408 Request Timeout
VA_08	22 Number changed (without diagnostic)	410 Gone
VA_9	127 Interworking	423 Interval Too Brief
VA_10	20 Subscriber absent	480 Temporarily Unavailable
VA_11	127 Interworking	481 Call/Transaction does not exist
VA_12	127 Interworking	482 Loop Detected
VA_13	127 Interworking	483 Too many hops
VA_14	127 Interworking	485 Ambiguous
VA_15	17 User busy	486 Busy Here
VA_16	127 Interworking	488 Not acceptable here
VA_17	127 Interworking	493 Undecipherable
VA_18	127 Interworking	500 Server Internal error
VA_19	127 Interworking	501 Not implemented
VA_20	127 Interworking	502 Bad Gateway
VA_21	127 Interworking	504 Server timeout
VA_22	17 User busy	600 Busy Everywhere
VA_23	21 Call rejected	603 Decline
VA_24	1 Unallocated number	604 Does not exist anywhere
VA_25	127 Interworking	606 Not acceptable

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

TP308004	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 7.7.6
TSS reference	ISUP-SIP /Basic call/ Sending of the Release message (REL)/				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM sends out an INVITE message, a SIP message defined as **SIP MESSAGE_VA** has been received, on receipt of a Failure message (4xx, 5xx, 6xx) defined as **SIP_Failure_VA** • sends a REL message constructed from the encapsulated REL.				
SIP parameter values					
ISUP parameter values	REL; **cause value**: CV_ISUP				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	XXX	←		←	SIP MESSAGE_VA
	REL	←		←	SIP_Failure_VA(REL)
	RLC	→		→	ACK

Table 10

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

Table 11

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

TP308005	SIP reference: RFC 3261				ISUP reference: Q.1912.5 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				ISUP reference: Q.1912.5 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*" (Note 1)	Protocol-cause	"*cause= XX*" (see note 1)
-	-	Reason-text	Should be filled with the definition text as stated in Q.850 (see note 2)
NOTE 1: "XX" is the Cause Value as defined in Q.850. NOTE 2: Due to the fact that the Cause Indications parameter does not include the definition text as defined in table1/Q.850 this is based on provisioning in the O-IWU.			

5.2.2.9 Autonomous release at O-IWU

5.2.2.9.1 Receipt of Reset Circuit message (RSC)

TP309001	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 clause 7.7.1, 1), 7.7.4, 7.7.5**
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM but before an INVITE has been sent on receipt of a RSC message • no action is required on the SIP side other than to terminate local procedures if any are in progress.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→			
	RSC	→			
	RLC	←			

TP309002	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 clause 7.7.1, 7.7.4, 7.7.5**
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt RSC message **before** a **SIP MESSAGE_VA** response message has been received • The SUT shall hold the RSC message until a SIP response has been received • The SUT shall send a CANCEL or BYE request. The RSC is not encapsulated				
SIP parameter values	CANCEL or BYE: A REL is encapsulated with cause 31				
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	RSC	→			
	RLC	←			
				←	SIP_MESSAGE_VA
	CASE A				
				→	CANCEL
				←	200 OK CANCEL
				←	487 Request terminated
				→	ACK
	CASE B				
				→	BYE(REL#31)
				←	200 OK BYE(RLC)
				←	487 Request terminated
				→	ACK

Table 15

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

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

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	ISUP reference: Q.1912.5 clause 7.7.1, 7.7.4, 7.7.5
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria		
ISUP selection criteria		
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message with the complete called party number, sending a BYE message on receipt RSC message **after** a 200 OK response message has been received • The SUT shall send a BYE request The RSC is not encapsulated	
SIP parameter values	BYE: A REL is encapsulated with cause 31	
ISUP parameter values		
Comments	(see flow below)	

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)

<table>
<tr><th>TP309006</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 clause 7.7.1, 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 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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">CANCEL or BYE: A REL is encapsulated with cause 31</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="15">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>SIP_MESSAGE_VA</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>
<tr><td colspan="3">CASE A</td><td></td><td></td></tr>
<tr><td></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>
<tr><td colspan="3">CASE B</td><td></td><td></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>
<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 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**			**ISUP reference: Q.1912.5 clause 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**			**ISUP reference: Q.1912.5 clause 7.7.1, 7.7.4, 7.7.5**	
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt 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

TP309009	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 clause 7.7.1 3), 7.7.4, 7.7.5**
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt GRS message **before** a 200 OK response message has been received • The SUT shall hold the GRS message until a response has been received. A CANCEL is sent The GRS is not encapsulated • On subsequently receiving 200 OK INVITE messages , the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	100 Trying
	GRS	→			
	GRA	←		→	CANCEL
				←	200 OK INVITE(CON)
				→	ACK
				←	200 OK CANCEL
				→	BYE(REL#31)
				←	200 OK BYE(RLC)

TP309011	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 7.7.1, 7.7.4, 7.7.5
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message with the complete called party number, sending a 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				ISUP reference: Q.1912.5 clause 7.7.1, 7.7.4, 7.7.5
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt 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**				**ISUP reference: Q.1912.5 clause 7.7.1, 7.7.4, 7.7.5**
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving more than one IAM's sending an INVITE message for each call association on receipt of a GRS message were the Range Parameter value is bigger than "**1**" • the SUT shall send a BYE requests for each call association The GRS is not encapsulated				
SIP parameter values	BYE1 contains the 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**				**ISUP reference: Q.1912.5 clause 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**				**ISUP reference:** Q.1912.5 clause 7.7.1, 7.7.4
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" **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 TP309014	
VA	**SIP MESSAGE_VA**
VA_1	100 Trying
VA_2	180 Ringing
VA_3	183 Session Progress

TP3090016	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 clause 7.7.1 3), 7.7.4**
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" **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	ISUP reference: Q.1912.5 clause 7.7.1, 7.7.4
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria		
ISUP selection criteria		
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" **after** a 200 OK response message has been received • The SUT shall send a BYE request The CGB is not encapsulated	
SIP parameter values		
ISUP parameter values	CGB(hardware failure oriented)	
Comments		

ISUP/BICC		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	ISUP reference: Q.1912.5 clause 7.7.1, 7.7.4
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria		
ISUP selection criteria		
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" **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 TP309114; TP309018	
VA	**SIP MESSAGE_VA**
VA_1	180 Ringing
VA_2	183 Session Progress

TP309019	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 clause 7.7.1, 7.7.4, 7.7.5**
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving more than one IAM's sending an INVITE message for each call association on receipt of a CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" were the Range and Status Parameter value is bigger than **"1"** • the SUT shall send a BYE requests for each call association The CGB is not encapsulated				
SIP parameter values	BYE1 contains the 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

TP310001	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause A.1.1.3
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Confusion message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number and contains an unknown parameter, sending a INVITE message with the complete called party number and encapsulated IAM as received Ensure that when the succeeding node discards an unknown parameter and send back a Confusion message if indicated in the parameter compatibility information and the sending of a Confusion message is requested, the CFN message encapsulated in a 183 Session Progress is sent Ensure ISUP message is transported through the SIP network encapsulated in the 183 Session Progress				
SIP parameter values	183 Session Progress with encapsulated CFN				
ISUP parameter values	CFN				
	ISUP				**SIP-I**
	IAM	→		→	INVITE(IAM with unknown parameter)
	CFN	←		←	183 Session Progress(CFN)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Communication		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

5.2.2.11 Receipt of "Suspend" or "Resume" message

TP311001	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 clause A.1.1.3**
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Receipt of **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**				**ISUP reference: Q.1912.5 clause A.1.1.3.1**
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Receipt of a Blocking message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the 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				ISUP reference: Q.1912.5 clause A.1.1.3.1
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Receipt of a Blocking message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the 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				ISUP reference: Q.1912.5 clause A.1.1.3.1
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Receipt of a Blocking message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**CGB and CGU sent** Ensure that the SUT is able to respond on a Circuit group blocking message with a with a CGBA and on a Circuit group unblocking message (both maintenance oriented) with a CGUA . Ensure the CGB / CGU messages are not encapsulated within SIP messages				
SIP parameter values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	CGB	→			
	CGBA	←			
	CGU	→			
	CGUA	←			

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

TP312005	**SIP reference: RFC 3261**			**ISUP reference:** **Q.1912.5 clause A.1.1.3.1** **Q.784 clause 1.3.2.4**	
TSS reference	ISUP-SIP/ ISUP 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**			**ISUP reference:** **Q.1912.5 clause A.1.1.3.1** **Q.784 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**			**ISUP reference:** **Q.1912.5 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**				**ISUP reference: Q.1912.5 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**				**ISUP reference: Q.1912.5 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)

TP401001	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 clause 7.1.3**
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can successfully transmit a call having a **calling party number** with the screening indicator set to "network provided" and the presentation restricted indicator set to "presentation allowed"				
SIP parameter values					
ISUP parameter values	IAM; **Calling party number parameter** Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '11'B presentation restricted indicator = presentation allowed, '00'B				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP401002	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 clause 7.1.3**
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can successfully transmit a call having a **calling party number** with the screening indicator set to "network provided" and an **access transport** parameter containing the **calling sub-address**				
SIP parameter values					
ISUP parameter values	IAM; **Calling party number parameter** Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '11'B presentation restricted indicator = presentation allowed, '00'B 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><th>TP401003</th><th colspan="2">SIP reference: RFC 3261</th><th colspan="3">ISUP reference:
Q.1912.5 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">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"</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
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>TP401004</th><th colspan="2">SIP reference: RFC 3261</th><th colspan="3">ISUP reference:
Q.1912.5 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">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</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
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>

<table>
<tr><th>TP401005</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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">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			ISUP reference: Q.1912.5 clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that 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			ISUP reference: Q.1912.5 clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP			
SIP selection criteria				
ISUP selection criteria	PICS 6/8			
Test purpose	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				ISUP reference: Q.1912.5 clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	PICS 6/7				
Test purpose	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				ISUP reference: Q.1912.5 clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	PICS 6/6				
Test purpose	Ensure that the **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			ISUP reference: Q.1912.5 clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that 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			ISUP reference: Q.1912.5 clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that 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			ISUP reference: Q.1912.5 clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that if the **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				ISUP reference: 3.5/Q.731
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	PICS 1/7				
Test purpose	**Convert the Calling party numbe3r into the international format** 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				ISUP reference: 3.5/Q.731
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** 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**				**ISUP reference: 3.5/Q.731**
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 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.					

TP401016	**SIP reference: RFC 3261**				**ISUP reference: 3.5/Q.731**
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	PICS 1/8				
Test purpose	**Converting the calling party number to national format, if necessary** Ensure that the country code in the address signals of the **calling party number** is removed if it is the network's own country code. The nature of address indicator shall be set to "national (significant) number". The address presentation restricted indicator shall be transferred transparently.				
SIP parameter values	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				
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**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP401017	SIP reference: RFC 3261				ISUP reference: 3.5/Q.731
TSS reference:	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	PICS 1/8				
Test purpose	**Converting the additional calling party number to national format, if necessary** Ensure that the country code in the address signals of the **generic number** coded as an "additional calling party number", if the numbering plan indicator is "ISDN Telephony" is removed if it is the network's own country code. The nature of address indicator shall be set to "national (significant) number". The address presentation restricted indicator shall be transferred transparently				
SIP parameter values	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				
ISUP parameter values	IAM; **Calling party number parameter** Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '11'B Presentation restricted indicator = presentation allowed, '00'B **Generic number 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				
Comments	**SIP-I**		**SUT**		**ISUP**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP401018	SIP reference: RFC 3261				ISUP reference: 3.5/Q.731
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	PICS 1/7				
Test purpose	**Adding a prefix to an international calling party number** Ensure that a prefix is added to the **calling party number** and the nature of address indicator is set to "unknown" (see note).				
SIP parameter values					
ISUP parameter values					
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	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				ISUP reference: 3.5/Q.731
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"** 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 (@).					

TP401020	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that 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				
SIP parameter values	INVITE: No P-Asserted Identity, From Header: anonymous@anonymous.inv				
ISUP parameter values	IAM; no Calling party number and no Additional calling party number 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)

<table>
<tr><th>TP401021</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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">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+NCD+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>

<table>
<tr><th>TP401022</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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">Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation allowed and the Generic Number is not applicable
Sends an INVITE message with the
• "P-Asserted-Identity header field" where the user portion of the addr-spec is set to value of the calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NCD+SN
• a "From header field" where the "addr-spec" is set to where the user portion of the addr-spec is set to value of the additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NCD+SN
• without "Privacy Header field" or "id" is not included</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: P-Asserted-Identity derived from the calling party number, Privacy=id, From header derived from the additional calling party number</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM; Calling party number is present and no Additional calling party number is 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>TP401023</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 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">Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation allowed and the Generic Number is applicable
Sends an INVITE message with the
• "P-Asserted-Identity header field" , " where the user portion of the addr-spec is set to value of the calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NCD+SN
• "From header field" " where the user portion of the addr-spec is set to value of the additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NCD+SN
• and without "Privacy Header field" or "id" is not included</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: P-Asserted-Identity derived from the calling party number, no Privacy header, From header derived from the additional calling party number</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM; Calling party number and Additional calling party number are 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 colspan="3">Values for test purpose TP401024</th></tr>
<tr><th>NoAS_VALUE</th><th>ISUP parameter values</th><th>SIP parameter values:</th></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>

TP401024	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 clause 7.1.3**
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
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, 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**				**ISUP reference: Q.1912.5 clause 7.1.3**
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	NOT PICS 1/7				
Test purpose	Ensure when no calling party number is included in the encapsulated IAM or the calling party number in the in the encapsulated IAM is not identical to the P-Asserted-Identity, no Privacy value "id" received Send an IAM the calling party number is derived from SIP P-Asserted-Identity. The Address Presentation Restricted Indicator is set to Presentation allowed				
SIP parameter 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**				**ISUP reference: Q.1912.5 clause 7.1.3**
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
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, 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**				**ISUP reference: Q.1912.5 clause 7.1.3**
TSS reference:	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	NOT PICS 1/7				
Test purpose	Ensure when no 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**				**ISUP reference:** **Q.1912.5** **Q.731 clause 4.5.2.1.1**
TSS reference:	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can successfully transmit a call having a **calling party number** with the screening indicator set to "network provided" and the 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)

TP402002	**SIP reference: RFC 3261**				**ISUP reference:** **Q.1912.5** **Q.731 clause 4.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Restricted calling party number (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**				
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** including 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)

TP402003	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.731 clause 4.5.2.1.1
TSS reference:	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Restricted calling party number (user provided, verified and passed)** Ensure that the SUT can pass transparently a call having the calling party number with the screening indicator set to "user provided, verified and passed" and the address presentation restricted indicator set to "presentation restricted"				
SIP parameter values					
ISUP parameter values	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				
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)

TP402004	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.731 clause 4.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Restricted calling party number (user provided, verified and passed) with calling sub-address** Ensure that the SUT can pass transparently a call having a **calling party number** with the screening indicator set to "user provided, verified and passed", the address presentation restricted indicator set to "presentation restricted" and an **access transport** parameter containing the **calling sub-address**				
SIP parameter values					
ISUP parameter values	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				
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)

TP402005	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.731 clause 4.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Restricted calling party number (user provided, not verified)** Ensure that the SUT can pass transparently a call having a default **calling party number** with the screening indicator set to "network provided" and a **generic number** containing the additional calling party number with the screening indicator set to "user provided, not verified", both having the address presentation restricted indicator set to "presentation restricted"				
SIP parameter values					
ISUP parameter values	IAM; **Calling party number parameter** Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '11'B Address presentation restricted parameter = '01'B **Generic number parameter** Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B				
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)

TP402006	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 Q.731 clause 4.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Restricted calling party number (user provided, not verified) with calling sub-address** Ensure that the SUT can pass transparently a call having a default **calling party number** with the screening indicator set to "network provided", a **generic number** containing the additional calling party number with the screening indicator set to "user provided, not verified", both having the address presentation restricted indicator set to "presentation restricted" and an **access transport** parameter containing the **calling sub-address**				
SIP parameter values					
ISUP parameter values	IAM; **Calling party number parameter** Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '11'B Address presentation restricted parameter = '01'B **Generic number parameter** Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B **Access transport parameter** including subaddress information				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP402007	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 Q.731 clause 4.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria	PICS 6/4				
Test purpose	**Discarding the calling 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"				
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

TP402008	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.731 clause 4.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria	PICS 6/4 AND PICS 6/5				
Test purpose	**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"				
SIP parameter values					
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

TP402009	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.731 clause 4.5.2.1.1
TSS reference:	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the calling 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

TP402010	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 Q.731 clause 4.5.2.1.1**
TSS reference:	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the additional 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**				**ISUP reference: Q.1912.5 clause 7.1.3**
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that when the SUT has received an IAM message, the **Calling Party Number is applicable** whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to **presentation restricted** and the **Generic Number is not applicable** Sends an INVITE message with the • "P-Asserted-Identity header field" where the user portion of the addr-spec is set to value of the additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NCD+SN • a "From header field" where the user portion of the addr-spec is set to value of the additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NCD+SN • and with "Privacy Header field" set to "id"				
SIP parameter values	INVITE: P-Asserted-Identity, From header field, 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)

TP402012	SIP reference: RFC 3261				ISUP reference: Q.1912.5 clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that when the SUT has received an IAM message, the **Calling Party Number is applicable** whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to **presentation restricted** and the **Generic Number is applicable** Sends an INVITE message with the • "P-Asserted-Identity header field", where the user portion of the addr-spec is set to value of the calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NCD+SN • "From header field" where the user portion of the addr-spec is set to value of the additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NCD+SN • and with "Privacy Header field" is set to "id"				
SIP parameter values	INVITE: P-Asserted-Identity, From header field, Privacy "id"				
ISUP parameter values	IAM: Calling party number. 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)

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

TP402013	SIP reference: RFC 3261				ISUP reference: Q.1912.5 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				ISUP reference: Q.1912.5 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

<table>
<tr><th>TP402015</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 clause 7.1.3</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">PICS 1/7</td></tr>
<tr><td>Test purpose</td><td colspan="5">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</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value "id" is present</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">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"</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>TP402016</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 clause 7.1.3</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">NOT PICS 1/7</td></tr>
<tr><td>Test purpose</td><td colspan="5">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</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value "id" is present</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">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"</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>

5.3.3 Connected line identification presentation (COLP)

TP403001	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 Q.731 clause 5.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/COLP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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

TP403002	SIP reference: RFC 3261	ISUP reference: Q.1912.5 Q.731 clause 5.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/COLP	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Connected number (user provided, not verified) with 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" and an **access transport** parameter containing the connected sub-address	
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 = '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	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

TP403003	**SIP reference: RFC 3261**	**ISUP reference: Q.1912.5 Q.731 clause 5.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/COLP	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Connected number (user provided, not verified) without connected sub-address** Ensure that the SUT passes transparently a default **connected number** with the screening indicator set to "network provided", a **generic number** containing the additional connected number with the screening indicator set to "user provided, not verified" without an **access transport** parameter containing the **connected sub-address**	
SIP parameter values		
ISUP parameter 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 = '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	

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>TP403004</td><td colspan="5">SIP reference: RFC 3261</td><td colspan="2">ISUP reference:
Q.1912.5
Q.731 clause 5.5.2.1.1</td></tr>
<tr><td>TSS reference</td><td colspan="7">ISUP-SIP-ISUP/SS/COLP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="7"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="7">PICS 1/7</td></tr>
<tr><td>Test purpose</td><td colspan="7">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="7">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="7">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 colspan="3">SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td colspan="3"></td><td>→</td><td>IAM</td></tr>
<tr><td colspan="7">CASE A</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td colspan="3"></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td colspan="3"></td><td>←</td><td>ANM</td></tr>
<tr><td colspan="7">CASE B</td></tr>
<tr><td>200 OK INVITE(CON)</td><td>←</td><td colspan="3"></td><td>←</td><td>CON</td></tr>
<tr><td></td><td colspan="5">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td colspan="3"></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td colspan="3"></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP403005</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5
Q.731 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 1/7</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="2">SIP reference: RFC 3261</th><th colspan="3">ISUP reference:
Q.1912.5
Q.731 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</th><th colspan="2">ISUP reference:
Q.1912.5
Q.731 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>SIP reference: RFC 3261</td><td colspan="4">ISUP reference:
Q.1912.5
Q.731 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>

<table>
<tr><td>TP403009</td><td colspan="5">SIP reference: RFC 3261</td><td colspan="2">ISUP reference:
Q.1912.5
Q.731 clause 5.5.2.1.1</td></tr>
<tr><td>TSS reference</td><td colspan="7">ISUP-SIP-ISUP/SS/COLP</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 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</td></tr>
<tr><td>SIP parameter values</td><td colspan="7">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</td></tr>
<tr><td>ISUP parameter values</td><td colspan="7">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</td></tr>
<tr><td rowspan="10">Comments</td><td>SIP-I</td><td></td><td colspan="3">SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td colspan="3"></td><td>→</td><td>IAM</td></tr>
<tr><td colspan="7">CASE A</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td colspan="3"></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td colspan="3"></td><td>←</td><td>ANM</td></tr>
<tr><td colspan="7">CASE B</td></tr>
<tr><td>200 OK INVITE(CON)</td><td>←</td><td colspan="3"></td><td>←</td><td>CON</td></tr>
<tr><td></td><td colspan="5">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td colspan="3"></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td colspan="3"></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP403010</th><th colspan="5">SIP reference: RFC 3261</th><th>ISUP reference:
Q.1912.5
Q.731 clause 5.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP-ISUP/SS/COLP</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">Handling unrequested COL
Ensure that the call can be successfully set up if the SUT receives an unsolicited COL</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">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="6">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</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>

TP403012	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.731 clause 5.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria	PICS 1/7				
Test purpose	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 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.				
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)

<table>
<tr><th>TP403013</th><th colspan="4">SIP reference: RFC 3261</th><th>ISUP reference:
Q.1912.5
Q.731 clause 5.5.2.1.1</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">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
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.</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>

5.3.4 Connected Line Identification Restriction (COLR)

TP404001	**SIP reference: RFC 3261**	**ISUP reference: Q.1912.5 Q.731 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 = '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><td>TP404002</td><td colspan="5">SIP reference: RFC 3261</td><td>ISUP reference:
Q.1912.5
Q.731 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"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">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</td></tr>
<tr><td>SIP parameter values</td><td colspan="6"></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
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</td></tr>
<tr><td rowspan="10">Comments</td><td>SIP-I</td><td></td><td colspan="2">SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td colspan="2"></td><td>→</td><td>IAM</td></tr>
<tr><td colspan="6">CASE A</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td colspan="2"></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td colspan="2"></td><td>←</td><td>ANM</td></tr>
<tr><td colspan="6">CASE B</td></tr>
<tr><td>200 OK INVITE(CON)</td><td>←</td><td colspan="2"></td><td>←</td><td>CON</td></tr>
<tr><td></td><td colspan="4">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td colspan="2"></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td colspan="2"></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP404003</th><th colspan="5">SIP reference: RFC 3261</th><th>ISUP reference:
Q.1912.5
Q.731 clause 6.5.2.1.1</th></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"></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, 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="6"></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
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 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>

TP404004	**SIP reference: RFC 3261**	**ISUP reference: Q.1912.5 Q.731 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

TP404005	SIP reference: RFC 3261	ISUP reference: Q.1912.5 Q.731 clause 6.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/COLR	
SIP selection criteria	PICS 7/2	
ISUP selection criteria		
Test purpose	**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"	
SIP parameter values	200 INVITE: encapsulated ANM or CON No Additional Connected number parameter included	
ISUP parameter values	**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	

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>TP404007</td><td colspan="5">SIP reference: RFC 3261</td><td>ISUP reference:
Q.1912.5
Q.731 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"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">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="6">200 OK INVITE: encapsulated ANM or CON included</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">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.</td></tr>
<tr><td rowspan="10">Comments</td><td>ISUP</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 colspan="6">CASE A</td></tr>
<tr><td>ACM</td><td>←</td><td colspan="2"></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td colspan="2"></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td colspan="6">CASE B</td></tr>
<tr><td>CON</td><td>←</td><td colspan="2"></td><td>←</td><td>200 OK INVITE(CON)</td></tr>
<tr><td></td><td colspan="4">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td colspan="2"></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td colspan="2"></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

5.3.5 Terminal Portability (TP)

TP405001	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 Q.733clause 4.5.2.1**
TSS reference:	ISUP-SIP-ISUP/SS/TP				
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**				**ISUP reference: Q.1912.5 Q.733clause 4.5.2.1**
TSS reference:	ISUP-SIP-ISUP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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**				**ISUP reference: Q.1912.5 Q.733clause 4.5.2.1**
TSS reference	ISUP-SIP-ISUP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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**				**ISUP reference: Q.1912.5 Q.733clause 4.5.2.1**
TSS reference	ISUP-SIP-ISUP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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><td>TP406001</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISUP reference:
Q.1912.5
Q.731 clause 8.5.2.1.1/</td></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><td>TP406002</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISUP reference:
Q.1912.5
Q.731 clause 8.5.2.1.1/</td></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				ISUP reference: Q.1912.5 Q.731 clause 8.5.2.1.1/
TSS reference	ISUP-SIP-ISUP/SS/SUB				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Sending the calling sub-address in the access transport parameter** Ensure that the SUT can include the called sub-address in the **access transport** parameter in the encapsulated IAM				
SIP parameter values	INVITE: Content-Type: 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				ISUP reference: Q.1912.5 Q.731 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 called sub-address in the **access transport** parameter in the ISUP IAM				
SIP parameter values					
ISUP parameter values					
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	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**				**ISUP reference: Q.1912.5 Q.731.7 clause 7.5.2.1.1**
TSS reference:	ISUP-SIP-ISUP/SS/MCID				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Successful MCID request O-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. 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**				**ISUP reference: Q.1912.5 Q.731.7 clause 7.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/MCID				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Successful MCID request 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				ISUP reference: Q.1912.5 Q.731.7 clause 7.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/MCID				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Successful MCID request - 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				ISUP reference: Q.1912.5 Q.731.7 clause 7.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/MCID				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**MCID request - MCID not supported by the OLE** O-MGCF Ensure that the SUT rejects a MCID request by sending a **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				ISUP reference: Q.1912.5 Q.731.7 clause 7.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/MCID				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**MCID request - MCID not supported by the OLE** I-MGCF Ensure that the SUT rejects a MCID request by sending a **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				ISUP reference: Q.1912.5 Q.731.7 clause 7.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/MCID				
SIP selection criteria					
ISUP selection criteria	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				ISUP reference: Q.1912.5 Q.731.7 clause 7.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/MCID				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Successful MCID request with calling sub-address O-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. 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				ISUP reference: Q.1912.5 Q.731.7 clause 7.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/MCID				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Successful MCID request 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><td>TP407009</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISUP reference:
Q.1912.5
Q.731.7 clause 7.5.2.1.1</td></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>

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<tr><td>TP407010</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISUP reference:
Q.1912.5
Q.731.7 clause 7.5.2.1.1</td></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)

TP408001	**SIP reference: RFC 3261**			**ISUP reference:** **Q.1912.5** **Q.733 clause 2.5.2.1.1.1; 2.5.2.1.1.2**	
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call hold after answer, requested by the local user** Ensure that the notifications that a call is placed on hold and retrieved are sent with **CPG** messages having the **event indicator** set to "progress". 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)

TP408002	**SIP reference: RFC 3261**			**ISUP reference:** **Q.1912.5** **Q.733 clause 2.5.2.1.1.1; 2.5.2.1.1.2**	
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call hold after answer, requested by the local user** Ensure that the notifications that a call is placed on hold and retrieved are sent with **CPG** messages having the **event indicator** set to "progress". I-MGCF interworking				
SIP parameter values	INVITE: Content-Type: 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**				**ISUP reference: Q.1912.5 Q.733 clause 2.5.2.1.1.1; 2.5.2.1.1.2**
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call hold after answer, requested by the remote user** Ensure that the notifications that a call is placed on hold and retrieved are sent with **CPG** messages having the **event indicator** set to "progress". 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**				**ISUP reference: Q.1912.5 Q.733 clause 2.5.2.1.1.1; 2.5.2.1.1.2**
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call hold after answer, requested by the remote user** Ensure that the notifications that a call is placed on hold and retrieved are sent with **CPG** messages having the **event indicator** set to "progress". I-MGCF interworking				
SIP parameter values	INVITE: Content-Type: 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		ISUP reference: Q.1912.5 Q.733 clause 2.2.1; 2.5.2.1.1.1; 2.5.2.1.1.2		
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria	PICS 8/1				
Test purpose	**Call hold after alerting, requested by the local user** Ensure that when a outgoing call is placed on hold and retrieved after alerting the notifications are sent with **CPG** messages. 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		ISUP reference: Q.1912.5 Q.733 clause 2.2.1; 2.5.2.1.1.1; 2.5.2.1.1.2		
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria	PICS 8/1				
Test purpose	**Call hold after alerting, requested by the local use***r* Ensure that when a outgoing call is placed on hold and retrieved after alerting the notifications are sent with **CPG** messages. I-MGCF interworking				
SIP parameter 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**				**ISUP reference: Q.1912.5 Q.764 clause 2.3**
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call hold after answer, release of the call by the local 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**				**ISUP reference: Q.1912.5 Q.764 clause 2.3**
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call hold after answer, release of the call by the local 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				ISUP reference: Q.1912.5 Q.764 clause 2.3
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call hold after answer, release of the call by the non-served user** Ensure that a call in the held state can be released by the user who did not activate the Call hold service. O-MGCF interworking				
SIP parameter values	INVITE: Content-Type: application/ISUP; CPG encapsulated in the MIME body				
ISUP parameter values					
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	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				ISUP reference: Q.1912.5 Q.764 clause 2.3
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call hold after answer, release of the call by the non-served user** Ensure that a call in the held state can be released by the user who did not activate the Call hold service. 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

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Q.1912.5
Q.764 clause 2.3</td></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 alerting, release of the call by the local served user
Ensure that a held call can be released by the user who activated the Call hold service without retrieving the call. O-MGCF interworking</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="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>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td></td><td colspan="3">Ringing</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>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>

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Q.1912.5
Q.764 clause 2.3</td></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 alerting, release of the call by the local served user
Ensure that a held call can be released by the user who activated the Call hold service without retrieving the call. I-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="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>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td></td><td colspan="3">Ringing</td><td></td></tr>
<tr><td>INVITE(CPG, sendonly)</td><td>→</td><td></td><td>→</td><td>CPG(progress, hold)</td></tr>
<tr><td>200 OK INVITE</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>

5.3.9 Call Waiting (CW)

TP409001	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.733 clause 1.5.2.1.1
TSS reference:	ISUP-SIP-ISUP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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				ISUP reference: Q.1912.5 Q.733 clause 1.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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**				**ISUP reference: Q.1912.5 Q.733 clause 1.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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**				**ISUP reference: Q.1912.5 Q.733 clause 1.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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				ISUP reference: Q.1912.5 Q.733 clause 1.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**User rejects the waiting call** Ensure that the SUT pass on a **REL** with cause #21 (call rejected) if a busy user rejects the waiting call. 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				ISUP reference: Q.1912.5 Q.733 clause 1.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**User rejects the waiting call** Ensure that the SUT pass on a **REL** with cause #21 (call rejected) if a busy user rejects the waiting call. I-MGCF interworking				
SIP parameter values	180 Ringing: Content-Type: application/ISUP ; ACM or CPG encapsulated in the MIME body 480 Temporarily unavailable: Content-Type: application/ISUP ; REL encapsulated in the MIME body				
ISUP parameter values	ACM or CPG: Generic notification indicator "Call is a waiting call" REL: Cause #21 (call rejected)				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM(waiting)
	480 Temporarily Unavailable(REL)	←		←	REL(#21)
	ACK	→		→	RLC

TP409007	SIP reference: RFC 3261		ISUP reference: Q.1912.5 Q.733 clause 1.5.2.1.1	
TSS reference	ISUP-SIP-ISUP/SS/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		ISUP reference: Q.1912.5 Q.733 clause 1.5.2.1.1		
TSS reference	ISUP-SIP-ISUP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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**	**ISUP reference: Q.1912.5, Q.732 clause 2.5**
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**"Call is diverting" indication received in ACM** Verify that a call can be successfully established, if diversion occurs. The **ACM** contains the **generic notification indicator** set to "call is diverting", the **call diversion information** and the **redirection number** The Redirection reason is set to **CV_redirection_reason**. CPG (alerting) is coded as if it has been mapped from ACM. 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**	**ISUP reference: Q.1912.5, Q.732 clause 2.5**
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**"Call is diverting" indication received in ACM** Verify that a call can be successfully established, if diversion occurs. The **ACM** contains the **generic notification indicator** set to "call is diverting", the **call diversion information** and the **redirection number** The Redirection reason is set to **CV_redirection_reason**. CPG (alerting) is coded as if it has been mapped from ACM. I-MCGF interworking	
SIP parameter 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

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

TP410004	SIP reference: RFC 3261				ISUP reference: Q.1912.5, Q.732 clause 2.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**"Call diversion may occur" received in ACM** Verify that a call can be successfully established, if diversion may occur. The ACM indicates that "call diversion may occur" in the optional backward call indicators. The following CPG contains the generic notification indicator set to "call is diverting", the call diversion information and the redirection number, if diversion occurs. The CPG (progress) contains **CV_redirection_reason** in call diversion information and also Redirection number. The CPG (alerting) is coded as if it has been mapped from ACM, with RnNbRes parameter (optional). I-MCGF interworking				
SIP parameter values	180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 183 Session Progress: Content-Type: application/ISUP; CPG encapsulated in the MIME body				
ISUP parameter values	ACM: BCI Called party status indicator "subscriber free", Optional backward call indicator: "Call diversion may occur" CPG: Event information=progress, Call diversion information; Generic notification; Redirection number CPG: Event information=alerting				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM(free)
	183 Session Progress(CPG)	←		←	CPG
	183 Session Progress(CPG)	←		←	CPG(alerting)
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

CV_redirection_reason TP410003, TP410004	
VA_1	No reply
VA_2	Deflection during alerting

<table>
<tr><th>TP410005</th><th colspan="5">SIP reference: RFC 3261</th><th>ISUP reference:
Q.1912.5, Q.732 clause 2.4.2</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">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</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">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</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">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</td></tr>
<tr><td rowspan="9">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(no indication)</td><td>←</td><td></td><td>←</td><td colspan="2">183 Session Progress(ACM)</td></tr>
<tr><td>CPG1</td><td>←</td><td></td><td>←</td><td colspan="2">183 Session Progress(CPG)</td></tr>
<tr><td>CPG2(alerting)</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing(CPG)</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>

TP410006	SIP reference: RFC 3261	ISUP reference: Q.1912.5, Q.732 clause 2.4.2
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Multiple diversions -Verify that a call can be successfully established, if multiple diversion occur** Several messages each 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="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5, Q.732 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>

<table>
<tr><th>TP410008</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5, Q.732 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)
I-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>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>
<tr><td colspan="6">NOTE: Altered in Gateways.</td></tr>
</table>

TP410009	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 Q.731 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**				**ISUP reference: Q.1912.5 Q.731 clause 4.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion				
SIP selection criteria					
ISUP selection criteria	PICS 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**				**ISUP reference: Q.1912.5 Q.731 clause 4.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion				
SIP selection criteria					
ISUP selection criteria	PICS 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**				**ISUP reference: Q.1912.5 Q.731 clause 4.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion				
SIP selection criteria					
ISUP selection criteria	PICS 1/8				
Test purpose	**Original called number in the incoming international gateway** Verify that the incoming international gateway checks and manipulates the **original called number** according to the procedures as defined for CLIP.Applicable tests: Converting the **original called number** to national format, if necessary (own country code)				
SIP parameter values	INVITE: Content-Type: 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**				**ISUP reference: Q.1912.5 Q.731 clause 4.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion				
SIP selection criteria					
ISUP selection criteria	PICS 10/2 AND PICS 1/7				
Test purpose	**Redirecting number in the outgoing international gateway** Verify that the outgoing international gateway checks and manipulates the **redirecting 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**				**ISUP reference: Q.1912.5 Q.731 clause 4.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion				
SIP selection criteria					
ISUP selection criteria	PICS 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				ISUP reference: Q.1912.5, Q.732 clause 2.5.2.3Q.731clause 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

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

TP410017	SIP reference: RFC 3261				ISUP reference: Q.1912.5, Q.732 clause 2.5.2.3, Q.731clause 3.5.2.3
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion				
SIP selection criteria					
ISUP selection criteria	PICS 1/8 AND 10/4				
Test purpose	**Redirecting number in the incoming international gateway** Verify that the incoming international gateway checks and manipulates the **redirecting numbe**r according to the procedures as defined for CLIP Adding a prefix to an international **redirecting number** The PTC will send an IAM with RgNb				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM containing a Redirecting number encapsulated in the MIME body				
ISUP parameter values	IAM: Redirecting 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

TP410018	SIP reference: RFC 3261				ISUP reference: Q.1912.5, Q.732 clause 2.5.2.4, Q.731clause 3.5.2.4
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion				
SIP selection criteria					
ISUP selection criteria	PICS 10/5 AND PICS 1/8				
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 Discarding the **redirection number** in case of bilateral agreements removes the **redirection number restriction parameter**				
SIP parameter values	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				
ISUP parameter values	ACM: Called party status=no indication Generic notification Call diversion information Redirection reason unconditional No Redirection number ANM: No Redirection number restriction parameter				
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)

TP410019	SIP reference: RFC 3261				ISUP reference: Q.1912.5, Q.732 clause 2.5.2.3, Q.731clause 3.5.2.3
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion				
SIP selection criteria					
ISUP selection criteria	PICS 1/7				
Test purpose	**Redirection number in the outgoing international gateway** Verify that the outgoing international gateway checks and manipulates the **redirection numbe**r according to the procedures defined for COLP Converting the **redirection number** to national format, if necessary (own country code) 1. The PTC will provide the necessary stimulus 2. ACM with CDInf, GenNot = "call is diverting" and an international RnNb with own CC				
SIP parameter 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 "National 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)

TP410020	SIP reference: RFC 3261				ISUP reference: Q.1912.5, Q.732 clause 2.5.2.3, Q.731clause 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	ISUP reference: Q.1912.5, Q.731 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 number** according to the procedures defined for COLP Adding a prefix to an international **redirection number** The PTC will provide the necessary stimulus.ACM with CDInf, GenNot = "call is diverting" and an international RnNb with foreign country code	
SIP parameter 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	ISUP reference: Q.1912.5 Q.734 clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/CONF	
SIP selection criteria		
ISUP selection criteria		
Test purpose	To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message 1. Assist a call set up from SP A to SPD. 2. Check that the notification "conference established" is received in the CPG from conferee at SPC 3. Check the notification "other party added" in the CPG. O-MGCF interworking	
SIP parameter values	INFO: Content-Type: application/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</th><th colspan="2">ISUP reference:
Q.1912.5
Q.734 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">To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message
1. Assist a call set up from SP A to SPD
2. Check that the notification "conference established" is received in the CPG from conferee at SPC
3. Check the notification "other party added" in the CPG.
I-MGCF interworking</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><th>TP411003</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5
Q.734 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">To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message
1. Assist a call set up from SPA to SPD
2. Check that the notification "conference established" is received in the CPG from conferee at SPC
3. Check the notification "isolated" in the CPG
O-MGCF interworking</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>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(isolated)</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>

TP411004	**SIP reference: RFC 3261**				**ISUP reference:** **Q.1912.5** **Q.734 clause 1.6.15**
TSS reference	ISUP-SIP-ISUP/SS/CONF				
SIP selection criteria					
ISUP selection criteria					
Test purpose	To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message 1. Assist a call set up from SPA to SPD 2. Check that the notification "conference established" is received in the CPG from conferee at SPC 3. Check the notification "isolated" 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				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	INFO(CPG)	→		→	CPG(conference established)
	200 OK INFO	←			
	INFO(CPG)	→		→	CPG(isolated)
	200 OK INFO	←			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP411005</th><th colspan="5">SIP reference: RFC 3261</th><th>ISUP reference:
Q.1912.5
Q.734 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">To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message
1. Assist a call set up from SPA to SPD
2. Check that the notification "conference established" is received in the CPG from conferee at SPC
3. Check the notification "reattached" in the CPG
O-MGCF interworking</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**			**ISUP reference: Q.1912.5 Q.734 clause 1.6.15**	
TSS reference	ISUP-SIP-ISUP/SS/CONF				
SIP selection criteria					
ISUP selection criteria					
Test purpose	To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message 1. Assist a call set up from SPA to SPD 2. Check that the notification "conference established" is received in the CPG from conferee at SPC 3. Check the notification "reattached" in the CPG 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(other party added)
	200 OK INFO	←			
	INFO(CPG)	→		→	CPG(isolated)
	200 OK INFO	←			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP411007</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5
Q.734 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">To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message
1. Assist a call set up from SPA to SPD
2. Check the notification "other party disconnected" in the CPG
O-MGCF interworking</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>

TP411008	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.734 clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/CONF				
SIP selection criteria					
ISUP selection criteria					
Test purpose	To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message 1. Assist a call set up from SPA to SPD 2. Check the notification "other party disconnected" in the CPG 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: other party added CPG: Generic notification: other party 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(other party added)
	200 OK INFO	←			
	INFO(CPG)	→		→	CPG(other party disconnected)
	200 OK INFO	←			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP411009	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.734 clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/CONF				
SIP selection criteria					
ISUP selection criteria					
Test purpose	To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message 1. Assist a call set up from SPA to SPD 2. Check that the notification "conference established" is received in the CPG from conferee at SPC 3. 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			ISUP reference: Q.1912.5 Q.734 clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/CONF			
SIP selection criteria				
ISUP selection criteria				
Test purpose	To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message 1. Assist a call set up from SPA to SPD 2. Check that the notification "conference established" is received in the CPG from conferee at SPC 3. 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			ISUP reference: Q.1912.5 Q.732.7 clause 7.5.2.1.1.1 a)
TSS reference	ISUP-SIP-ISUP/SS/ECT			
SIP selection criteria				
ISUP selection criteria				
Test purpose	**Capability of sending a additional calling party number in the call transfer number** Verify that the IUT is able to store the additional calling party number in the **generic number** when the **calling party number** and the **generic number** have been received from the remote user. This information is sent by the IUT to the other remote user in the **call transfer number** in either the **FAC** or **CPG** when the call transfer is activated. O-MGCF interworking			
SIP parameter values	INFO: Content-Type: application/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**				**ISUP reference: Q.1912.5 Q.734 clause 1.6.15**
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Capability of sending the additional calling party number in the call transfer number** Verify that the IUT is able to store the additional calling party number in the **generic number** when the **calling party number** and the **generic number** have been received from the remote user. This information is sent by the IUT to the other remote user in the **call transfer number** in either the **FAC** or **CPG** when the call transfer is activated. I-MGCF interworking				
SIP parameter values	INFO: Content-Type: 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**				**ISUP reference:** **Q.1912.5** **Q.732.7 clause 7.5.2.1.1.1 a)**
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Capability of storing and sending the additional connected number in the call transfer number** 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. 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><td>TP412006</td><td colspan="4">SIP reference: RFC 3261</td><td>ISUP reference:
Q.1912.5
Q.734 clause 1.6.15</td></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">Capability of storing and sending the additional connected number in the call transfer number
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.
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</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">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>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, 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><td>TP412009</td><td colspan="5">SIP reference: RFC 3261</td><td>ISUP reference:
Q.1912.5
Q.732.7 clause 7.5.2.1.1.1 a)</td></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">Loop prevention procedure - initiation
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.
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
INFO: Content-Type: application/ISUP; LOP encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">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)</td></tr>
<tr><td rowspan="24">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(hold)</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(CPG, sendonly)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK INVITE(recvonly)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>LOP(request)</td><td>→</td><td></td><td>→</td><td colspan="2">INFO(LOP)</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>LOP(response)</td><td>←</td><td></td><td>←</td><td colspan="2">INFO(LOP)</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>FAC(call transfer active, CTNb)</td><td>→</td><td></td><td>→</td><td colspan="2">INFO(FAC)</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></td><td></td><td></td><td>→</td><td colspan="2">INVITE(sendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK INVITE(sendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">ACK</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(RLC)</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK BYE</td></tr>
</table>

TP412010	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.734 clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Loop prevention procedure - initiation** 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. 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				ISUP reference: Q.1912.5 Q.732.7 clause 7.5.2.1.1.1 a)
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Loop prevention procedure - unsuccessful on timer expiry** To verify that the local exchange controlling the ECT rejects the call transfer if no **LOP** is received within $\mathbf{T_{ECT}}$ expiry. O-MGCF interworking				
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; LOP encapsulated in the MIME body				
ISUP parameter values	CPG: Event indicator=progress, 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				ISUP reference: Q.1912.5 Q.734 clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/CONF				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Loop prevention procedure - unsuccessful on timer expiry** To verify that the local exchange controlling the ECT rejects the call transfer if no **LOP** is received within $\mathbf{T_{ECT}}$ expiry. 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				ISUP reference: Q.1912.5 Q.732.7 clause 7.5.2.1.1.1 a)
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Loop prevention procedure - successful on timer expiry** Verify that the local exchange controlling the ECT completes the call transfer if no **LOP** is received within $\mathbf{T_{ECT}}$ expiry. 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)

TP412014	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.734 clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Loop prevention procedure - successful on timer expiry** Verify that the local exchange controlling the ECT completes the call transfer if no **LOP** is received within $\mathbf{T_{ECT}}$ expiry. 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 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(FAC)	→		→	FAC(call transfer active, CTNb)
	200 OK INFO	←			
	INVITE(sendrecv)	→			
	200 OK INVITE(sendrecv)	←			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP412015	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.732.7 clause 7.5.2.1.1.1 a)
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Facility message with generic notification sent to the remote user** Verify that the local exchange controlling the ECT can successfully initiate a call transfer by sending **FAC** with the **generic notification** set to "call transfer, active" or "call transfer, alerting" and the **service activation** parameter set to "call transfer". 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)

TP412016	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.734 clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Facility message with generic notification sent to the remote user** Verify that the local exchange controlling the ECT can successfully initiate a call transfer by sending **FAC** with the **generic notification** set to "call transfer, active" or "call transfer, alerting" and the **service activation** parameter set to "call transfer". 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	**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

TP412017	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.732.7 clause 7.5.2.1.1.1 a)
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call progress message with generic notification sent to the remote user** Verify that the local exchange (controlling the ECT) can successfully initiate a call transfer by sending **CPG** with the **generic notification** set to "call transfer, active" and the **service activation** parameter set to "call transfer". O-MGCF interworking				
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body				
ISUP parameter values	CPG: Generic notification=call transfer active, Call transfer number (PIXIT)				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	CPG(call transfer active, CTNb)	→		→	INFO(CPG)
				←	200 OK INFO
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

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

<table>
<tr><th>TP412019</th><th colspan="5">SIP reference: RFC 3261</th><th>ISUP reference:
Q.1912.5
Q.734 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">Call transfer number - removal of number
Verify that the exchange removes the call transfer number in the FAC or CPG before sending it to the next exchange, if its indicator is set to "presentation restricted" and there is no bilateral agreement to transfer the number</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(CTNb=restricted) 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, 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 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)</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>

TP412020	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 Q.734 clause 1.6.15**
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call transfer number - conversion to international number** Verify that the IUT converts the **call transfer number** to international format. The nature of address indicator shall be set to "international number".				
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC(CTNb=national) encapsulated in the MIME body				
ISUP parameter values	CPG: Event indicator=progress, Generic notification=hold FAC: Generic notification=call transfer active, Call transfer number=international(PIXIT)				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	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				ISUP reference: Q.1912.5 Q.732.7 clause 7.5.2.1.1.1 a)
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call transfer number - removal of own country code** Verify that the IUT removes the country code in the address signals of the **call transfer 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	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)

TP412022	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 Q.732.7 clause 7.5.2.1.1.1 a)**
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**ECT - Interaction with SUB** Verify that if the IUT is able to receive and re-send the sub-address in the **access transport** parameter in the **FAC** message in either direction after activating the call transfer service. These are the calling sub-address for incoming calls and the connected sub-address for outgoing calls. O-MGCF interworking				
SIP parameter values	INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body				
ISUP parameter values	FAC: Generic notification=call transfer active, Call transfer number(PIXIT) FAC: ATP contained the connected sub address				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	FAC(call transfer active, CTNb)	→		→	INFO(FAC)
				←	200 OK INFO
	FAC(ATP=SUB)	←		←	INFO(FAC)
				→	200 OK INFO
	FAC(ATP=SUB)	→		→	INFO(FAC)
				←	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP412023	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.734 clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**ECT - Interaction with SUB** Verify that if the IUT is able to receive and re-send the sub-address in the **access transport** parameter in the **FAC** message in either direction after activating the call transfer service. These are the calling sub-address for incoming calls and the connected sub-address for outgoing calls. O-MGCF interworking				
SIP parameter values	INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body				
ISUP parameter values	CPG: Event indicator=progress, Generic notification=hold FAC: Generic notification=call transfer active, Call transfer number(PIXIT) FAC: ATP contained the connected sub address				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	INVITE(sendonly,CPG hold)	←		←	CPG(hold)
	200 OK INVITE(recvonly)	→			
	ACK	←			
	INFO(FAC)	→		→	FAC(call transfer active, CTNb)
	200 OK INFO	←			
	INFO(FAC)	←		←	FAC(ATP=SUB)
	200 OK INFO	→			
	INFO(FAC)	→		→	FAC(ATP=SUB)
	200 OK INFO	←			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

5.3.13 3PTY

TP413001	**SIP reference: RFC 3261**	**ISUP reference: Q.1912.5 Q.734.2 clause 2.4; 2.2.1**
TSS reference	ISUP-SIP-ISUP/SS/3PTY	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**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 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)

TP413002	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 Q.734.2 clause 2.4; 2.2.1**
TSS reference	ISUP-SIP-ISUP/SS/3PTY				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Served user initiates 3PTY** Verify that the served user with two active calls is located, can successfully join this call (remote held user) to a three-way conversation, and notify the implied remote party accordingly The IUT should send a **CPG** message with the **generic notification indicator** set to "conference established" to both implied parties. The **event indicator** in the **CPG** should be set to "progress" 1. Setup a call to user B 2. Put this call on hold 3. Join this call to a conference I-MGCF interworking				
SIP parameter 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	**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	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP413003	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.734.2 clause 2.4; 2.2.1
TSS reference	ISUP-SIP-ISUP/SS/3PTY				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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				ISUP reference: Q.1912.5 Q.734.2 clause 2.4; 2.2.1
TSS reference	ISUP-SIP-ISUP/SS/3PTY				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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

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

TP413006	**SIP reference: RFC 3261**	**ISUP reference:** **Q.1912.5** **Q.734.2 clause 2.5.2.1.1.3 a**
TSS reference	ISUP-SIP-ISUP/SS/3PTY	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Served user creates a private communication with a remote user** Verify that the IUT (controlling the conference) on a 3PTY call can successfully create private communication with the active-held user. The appropriate notification is sent in **CPG** messages to the user 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=hold CPG: Event indicator=progress, Generic notification=retrieve 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	→			
	INVITE(CPG, sendrecv)	→		→	CPG(retrieve)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
			Conversation		
	INFO(CPG)	→		→	CPG(conference established)
	200 OK INFO	←			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP413007	**SIP reference: RFC 3261**		**ISUP reference:** **Q.1912.5** **Q.734.2 clause 2.5.2.1.1.3 a**		
TSS reference	ISUP-SIP-ISUP/SS/3PTY				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Served user creates a private communication with a remote user** Verify that the IUT (controlling the conference) on a 3PTY call can successfully create private communication with the active-idle user. The appropriate notification is sent in **CPG** messages to the user O-MGCF interworking				
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body				
ISUP parameter values	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(conference established)	→		→	INFO(CPG)
				←	200 OK INFO
	CPG(conference disconnected)	→		→	INFO(CPG)
				←	200 OK INFO
			Conversation		
	CPG(conference established)	→		→	INFO(CPG)
				←	200 OK INFO
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP413008	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 Q.734.2 clause 2.5.2.1.1.3 a**
TSS reference	ISUP-SIP-ISUP/SS/3PTY				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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

TP413009	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.734.2 clause 2.5.2.1.1.3 b
TSS reference	ISUP-SIP-ISUP/SS/3PTY				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Served user disconnects one remote user and retains the other** Verify that the IUT (controlling the conference) on a 3PTY call can successfully disconnect the active-held user and retain and notify the other user appropriately using **CPG** messages The IUT should send to the appropriate remote users **CPG** messages with a **generic notification indicator**. The **event indicator** in the **CPG** should be set to "progress". O-MGCF interworking				
SIP parameter 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	**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(conference disconnected)	→		→	INFO(CPG)
				←	200 OK INFO
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP413010	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.734.2 clause 2.5.2.1.1.3 b
TSS reference	ISUP-SIP-ISUP/SS/3PTY				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Served user disconnects one remote user and retains the other** Verify that the IUT (controlling the conference) on a 3PTY call can successfully disconnect the active-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**				**ISUP reference: Q.1912.5 Q.734.2 clause 2.5.2.1.1.3 b**
TSS reference	ISUP-SIP-ISUP/SS/3PTY				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Served user disconnects one remote user and retains the other** Verify that the IUT (controlling the conference) on a 3PTY call can successfully disconnect the active-idle user and retain and notify the other user appropriately using **CPG** messages The IUT should send to the appropriate remote users **CPG** messages with a **generic notification indicator**. The **event indicator** in the **CPG** should be set to "progress" O-MGCF interworking				
SIP parameter 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)

<table>
<tr><td>TP413012</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISUP reference:
Q.1912.5
Q.734.2 clause 2.5.2.1.1.3 b</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 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</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=conference established
CPG: Event indicator=progress, Generic notification=conference disconnected</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>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 colspan="3"></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 User-to-user service

5.3.14.1 User-to-user service 1

TP414001	**SIP reference: RFC 3261**			**ISUP reference: Q.1912.5 A.1.1 1.1.5.2.3 and 4/Q.737**	
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can 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**			**ISUP reference: Q.1912.5 A.1.1 1.1.5.2.3 and 4./Q.737**	
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can 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			ISUP reference: Q.1912.5 A.1.1 1.1.5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS1			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that the SUT can 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			ISUP reference: Q.1912.5 A.1.1 1.1.5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS1			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that the SUT can successfully transfer the User-to-user service 1 explicit request essential 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			
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

TP414005	SIP reference: RFC 3261				ISUP reference: Q.1912.5 A.1.1 1.1.5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can successfully transfer the User-to-user service 1 implicit response 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 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)

TP414006	SIP reference: RFC 3261				ISUP reference: Q.1912.5 A.1.1 1.1.5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can 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

TP414007	SIP reference: RFC 3261				ISUP reference: Q.1912.5 A.1.1 1.1.5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can successfully transfer the User-to-user service 1 explicit response 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 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)

TP414008	SIP reference: RFC 3261				ISUP reference: Q.1912.5 A.1.1 1.1.5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can successfully transfer the User-to-user service 1 explicit 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 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 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

TP414009	SIP reference: RFC 3261			ISUP reference: Q.1912.5 A.1.1 1.1.5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS1			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that the SUT can 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)

TP414010	SIP reference: RFC 3261				ISUP reference: Q.1912.5 A.1.1 1.1.5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can successfully transfer the User-to-user service 1 discarded by the network 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 ACM: User-to-user indicator set to discarded by the network 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

5.3.14.2 User-to-user service 2

TP414101	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 A.1.1 1.2.5.2.3 and 4./Q.737**
TSS reference	ISUP-SIP-ISUP/SS/UUS2				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can successfully transfer the User-to-user service 2 explicit request and User-to user information in the encapsulated IAM. An additional User-to-user information is sent in a USR message encapsulated in an INFO request. O-MGCF interworking				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM containing the user-to-user indicator and User-to-user information encapsulated in the MIME body INFO: Content-Type: application/ISUP; USR containing the User-to-user information parameter encapsulated in the MIME body				
ISUP parameter 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
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP414102	SIP reference: RFC 3261			ISUP reference: Q.1912.5 A.1.1 1.2.5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS2			
SIP selection criteria				
ISUP selection criteria				
Test purpose	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			ISUP reference: Q.1912.5 A.1.1 1.2.5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS2			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that the SUT can successfully transfer the User-to-user service 2 explicit 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 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 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				ISUP reference: Q.1912.5 A.1.1 1.2.5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS2				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can successfully transfer the User-to-user service 2 explicit request 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				
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				ISUP reference: Q.1912.5 A.1.1 1.3.5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can 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)

<table>
<tr><th>TP414202</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5 A.1.1
1.3.5.2.3 and 4./Q.737</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 request in the encapsulated IAM. Additional User-to-user information is sent in several USR message encapsulated in an INFO request.
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 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">IAM: User-to-user information parameter, User-to-user indicator
USR: User-to-user information</td></tr>
<tr><td rowspan="19">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(USR)</td><td>→</td><td></td><td>→</td><td>USR</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(USR)</td><td>←</td><td></td><td>←</td><td>USR</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(USR)</td><td>←</td><td></td><td>←</td><td>USR</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(USR)</td><td>→</td><td></td><td>→</td><td>USR</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>

TP414203	SIP reference: RFC 3261				ISUP reference: Q.1912.5 A.1.1 1.3.5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that an User-to-user request service 3 encapsulated in an INFO request during the confirmed state can successful proceeded. The User-to-user information is passed on in several encapsulated USR messages. O-MGCF interworking				
SIP parameter 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	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
			Conversation		
	FAR	→		→	INFO(FAR)
				←	200 OK INFO
	FAA	←		←	INFO(FAA)
				→	200 OK INFO
	USR	→		→	INFO(USR)
				←	200 OK INFO
	USR	←		←	INFO(USR)
				→	200 OK INFO
	USR	←		←	INFO(USR)
				→	200 OK INFO
	USR	→		→	INFO(USR)
				←	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP414204	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 A.1.1 1.3.5.2.3 and 4./Q.737**
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that an User-to-user request service 3 encapsulated in an INFO request during the confirmed state can successful proceeded. The User-to-user information is passed on in several encapsulated USR messages 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

TP414205	SIP reference: RFC 3261				ISUP reference: Q.1912.5 A.1.1 1.3.5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can successfully transfer the User-to-user service 3 explicit response in the encapsulated ANM. O-MGCF interworking				
SIP parameter values	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				
ISUP parameter values	IAM: User-to-user indicator set to service 3 request ANM: User-to-user indicator set to service 3 provided 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)

TP414206	SIP reference: RFC 3261				ISUP reference: Q.1912.5 A.1.1 1.3.5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can successfully transfer the User-to-user service 3 explicit response in the encapsulated ANM O-MGCF interworking				
SIP parameter values	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				
ISUP parameter values	IAM: User-to-user indicator set to service 3 request ANM: User-to-user indicator set to service 3 provided 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

TP414207	SIP reference: RFC 3261				ISUP reference: Q.1912.5 A.1.1 1.3.5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that an 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				ISUP reference: Q.1912.5 A.1.1 1.3.5.2.3 and 4./Q.737
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that an 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</th><th colspan="2">ISDN reference:
Q.1912.5 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**	**ISDN reference: Q.1912.5 clause 6.1.2 (i,2b**
TSS reference:	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message	
SIP selection 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) 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) 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	(see flow below)	

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**				**ISDN reference: Q.1912.5 clause 6.1.2 (i,1)**
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection 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**				**ISDN reference: Q.1912.5 clause 6.1.2 (i,2b)**
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection 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				ISDN reference: Q.1912.5 clause 6.1.2 (i,1)
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection 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				ISDN reference: Q.1912.5 clause 6.1.2 (i,2b)
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection 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				ISDN reference: Q.1912.5 clause 6.1.2 (i,1)
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection 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				ISDN reference: Q.1912.5 clause 6.1.2 (i,2b)
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection 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

TP501009	**SIP reference: RFC 3261**				**ISDN reference: EN 383 001 clause 6.1.3.5.2.2**
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				
ISDN selection criteria					
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for µ-Law and A-Law, **then independent from the received order of preference** Sends a SETUP message 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				
ISDN parameter values					
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	180 Ringing(ACM)	←		←	ALERTING
	200 OK INVITE(ANM)	←		←	CONNECT
	ACK	→			
			Conversation		
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

TP501010	**SIP reference: RFC 3261**				**ISDN reference: EN 383 001 clause 6.1.3.5.2.2**
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				
ISDN selection criteria					
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for µ-Law and A-Law 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				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 0 8 Answer: m=audio 4712 RTP/AVP 8 0				
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

TP501011	SIP reference: RFC 3261				ISDN reference: EN 383 001 clause 6.1.3.5.2.2
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				
ISDN selection criteria					
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for µ-Law and A-Law 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				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 0 8 Answer: m=audio 4712 RTP/AVP 8 0				
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

TP501012	SIP reference: RFC 3261				ISDN reference: Q.1912.5 clause 6.1.2 (i,1)
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection criteria	NOT PICS 4/4 AND NOT 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 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)	→		→	SETUP
	180 Ringing(ACM)	←		←	ALERTING
	200 OK INVITE(ANM)	←		←	CONNECT
	ACK	→			
			Conversation		
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

TP501013	SIP reference: RFC 3261				ISDN reference: Q.1912.5 clause 6.1.2 (i,1)
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection 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				ISDN reference: Q.1912.5 clause 6.1.2 (i,1)
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection 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				ISDN reference: Q.1912.5 clause 6.1.2 (i,1)
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection 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**				**ISDN reference: Q.1912.5 clause 6.1.2 (i,1)**
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection 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**				**ISDN reference: Q.1912.5 clause 6.1.2 (i,1)**
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection 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><td>TP501018</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISDN reference:
Q.1912.5 clause 6.1.3</td></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				ISDN reference: Q.1912.5 clause 6.1.3.5
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message containing an encapsulated IAM, sends an SETUP message with the **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				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				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				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.1.1.1.1.2 Sending of the INFO

TP502001	SIP reference: RFC 3261				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				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				ISDN reference: Q.1912.5 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				ISDN reference: Q.1912.5 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				ISDN reference: Q.1912.5 clause 6.5 2)
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in call state 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				ISDN reference: Q.1912.5 clause 6.5 2)
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection criteria	PICS 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				ISDN reference: Q.1912.5 clause 6.5 2)
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in call state 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

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

TP503007	SIP reference: RFC 3261				ISDN reference: Q.1912.5 clause 6.5 2)
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection criteria	PICS 1/6				
Test purpose	Ensure that the SUT in call state N9, on receipt of the ALERTING message containing a **progress indicator** set to PI_VALUE, the 180 Ringing SIP response is sent				
SIP parameter 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
	180 Ringing(ACM)	←		←	ALERTING(PI)
	BYE(REL)	→		→	DISCONNECT
	200 OK BYE(RLC)	←		←	RELEASE
				→	RELEASE COMPLETE

TP503008	SIP reference: RFC 3261				ISDN reference: Q.1912.5 clause 6.5 2)
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection criteria	PICS 1/6				
Test purpose	Ensure that the SUT in 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				ISDN reference: Q.1912.5 clause 6.5 2)
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection criteria	PICS 1/6				
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message sends out a SETUP message in state N6, where the end of address signalling is determined by analysis of the called party number to indicate that **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**				**ISDN reference: Q.1912.5 clause 6.5 2)**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in call state N25, on receipt of a PROGRESS message containing no **progress indicator**, • a 183 Session Progress with an encapsulated ACM is sent to the previous entity				
SIP parameter values	183 Session Progress encapsulated ACM: BCi Called party status = no indication				
ISDN parameter values	CALL PROCEEDING				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
	183 Session Progress(ACM)	←		←	CALL PROCEEDING
				←	PROGRESS
	CANCEL(REL)	→		→	RELEASE
	200 OK CANCEL(RLC)	←		←	RELEASE COMPLETE

TP503011	**SIP reference: RFC 3261**				**ISDN reference: Q.1912.5 clause 6.6**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection 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**				**ISDN reference: Q.1912.5 clause 6.6**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection 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**				**ISDN reference: Q.1912.5 clause 6.6**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection 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 GPG: 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				ISDN reference: Q.1912.5 clause 6.6
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection 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 GPG: 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				ISDN reference: Q.1912.5 clause 6.6
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection 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				ISDN reference: Q.1912.5 clause 6.6
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message				
SIP selection criteria					
ISDN selection 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				ISDN reference: Q.1912.5 clause 6.7
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of CONNECT_message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message sends out a SETUP message, receives an ALERTING message, on receipt of a CONNECT message • sends a 200 OK INVITE to the previous entity The bearer path shall be connected in both directions when the following condition is satisfied: • The BICC outgoing bearer set-up procedure, (Q.1902.4) is successfully completed In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "*notification not required*", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312) that sufficient preconditions have been met for the session to proceed				
SIP parameter values					
ISDN parameter values					
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	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				ISDN reference: Q.1912.5 clause 6.7
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of CONNECT_message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message sends out a SETUP message, receives 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				ISDN reference: Q.1912.5 clause 6.7
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of CONNECT_message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	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				ISDN reference: Q.1912.5 clause 6.4, 6.7
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of CONNECT_message				
SIP selection criteria					
ISDN selection criteria					
Test purpose	**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				ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, 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	→			

TP505002	SIP reference: RFC 3261				ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, on receipt of a RELEASE with the **Cause value** CV_ISDN, location LOC_ISDN • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN REL_COMP is returned to the ISDN side • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA				
SIP parameter values	SIP Statue-Code: SIP_FAILURE_VA (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

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

<table>
<tr><th>TP505004</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISDN reference:
Q.1912.5 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, 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><th>TP505005</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISDN reference:
Q.1912.5 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, 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>

<table>
<tr><td>TP505006</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISDN reference:
Q.1912.5 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 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="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="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>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>TP505007</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISDN reference:
Q.1912.5 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, 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</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="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>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>INFO</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>

TP505008	SIP reference: RFC 3261				ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection 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 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	SIP Statue-Code: SIP_FAILURE_VA (PIXIT)				
ISDN parameter values	RELEASE; **cause value**: CV_ISDN (PIXIT)				
Comments	**SIP-I**		**SUT**		**ISDN**
	INVITE(IAM)	→		→	SETUP
				←	SETUP ACK
	INVITE(IAM)	→		→	INFO
	SIP_FAILURE_VA(REL)	←		←	RELEASE
	ACK	→		→	RELEASE COMPLETE

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

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

TP505010	SIP reference: RFC 3261				ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, and receives a CALL PROCEEDING message, on receipt of a RELEASE 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 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
	SIP_FAILURE_VA(REL)	←		←	RELEASE COMPLETE
	ACK	→			

TP505011	SIP reference: RFC 3261				ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, and receives a CALL PROCEEDING message, on receipt of a RELEASE message with the **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 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
	SIP_FAILURE_VA(REL)	←		←	RELEASE
	ACK	→		→	RELEASE COMPLETE

TP505012	**SIP reference: RFC 3261**				**ISDN reference: Q.1912.5 clause 6.11.2**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection 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**				**ISDN reference: Q.1912.5 clause 6.11.2**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, 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</th><th colspan="2">ISDN reference:
Q.1912.5 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</th><th colspan="2">ISDN reference:
Q.1912.5 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. 38-47) (47 is class default)
VA_9	500 Server Internal Error	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_10	500 Server Internal Error	Cause Value No. 88 ("incompatible destination")
VA_11	500 Server Internal Error	Cause Value No. 111 ("protocol error, unspecified") (Class default)

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

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

TP505018	SIP reference: RFC 3261				ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a SETUP ACKNOWLEDGE message followed by a ALERTING message, having sent a 180 Ringing message on receipt of an ISDN DISCONNECT message • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE is returned to the ISDN side • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA				
SIP 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
	180 Ringing(ACM)	←		←	ALERTING
	SIP_FAILURE_VA(REL)	←		←	DISCONNECT
	ACK	→		→	RELEASE
				←	RELEASE COMPLETE

TP505019	SIP reference: RFC 3261				ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message followed by a ALERTING message, having sent a 180 Ringing message on receipt of an ISDN RELEASE 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				ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message followed by a ALERTING message, having sent a 180 Ringing message on receipt of an ISDN RELEASE message • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA				
SIP parameter values	183 Session Progress 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

<table>
<tr><th>TP505021</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISDN reference:
Q.1912.5 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 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</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)</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>180 Ringing(CPG)</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>

<table>
<tr><th>TP505022</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISDN reference:
Q.1912.5 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 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</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">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>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>180 Ringing(CPG)</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>

TP505023	SIP reference: RFC 3261	ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE	
SIP selection criteria	NOT PICS 4/10	
ISDN selection criteria		
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message, 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	ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE	
SIP selection criteria	NOT PICS 4/10	
ISDN selection criteria		
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message, 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

<table>
<tr><td>TP505025</td><td colspan="2">SIP reference: RFC 3261</td><td colspan="2">ISDN reference:
Q.1912.5 clause 6.11.2</td></tr>
<tr><td>TSS reference</td><td colspan="4">SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="4">NOT PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="4"></td></tr>
<tr><td>Test purpose</td><td colspan="4">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</td></tr>
<tr><td>SIP parameter values</td><td colspan="4">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="4">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>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>180 Ringing(CPG)</td><td>←</td><td></td><td>←</td><td>ALERTING</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 COMPLETE</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><td>TP505026</td><td colspan="2">SIP reference: RFC 3261</td><td colspan="2">ISDN reference:
Q.1912.5 clause 6.11.2</td></tr>
<tr><td>TSS reference</td><td colspan="4">SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE</td></tr>
<tr><td>SIP selection criteria</td><td colspan="4">NOT PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="4"></td></tr>
<tr><td>Test purpose</td><td colspan="4">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</td></tr>
<tr><td>SIP parameter values</td><td colspan="4">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="4">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>180 Ringing(CPG)</td><td>←</td><td></td><td>←</td><td>ALERTING</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>TP505027</th><th colspan="4">SIP reference: RFC 3261</th><th>ISDN reference:
Q.1912.5 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 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</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="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>180 Ringing(CPG)</td><td>←</td><td></td><td>←</td><td>ALERTING</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 22

<table>
<tr><th colspan="3">Values for test purposes TP108016 and TP108027</th></tr>
<tr><th colspan="2">←SIP Message
SIP_FAILURE_VA</th><th>← REL
Cause Indicators parameter
CV_ISDN,</th></tr>
<tr><td>VA_1</td><td>480 Temporarily unavailable</td><td>Cause Value No. 21 ("all rejected")</td></tr>
<tr><td>VA_2</td><td>480 Temporarily unavailable</td><td>Cause Value No. 31 ("normal unspecified")
(Class default)</td></tr>
<tr><td>VA_4</td><td>500 Server Internal Error</td><td>Cause Value No. 38 ("Network out of order")</td></tr>
<tr><td>VA_4</td><td>500 Server Internal Error</td><td>Cause Value No. 41 ("Temporary failure ")</td></tr>
<tr><td>VA_5</td><td>500 Server Internal Error</td><td>Cause Value No. 111 ("protocol error, unspecified")
(Class default)</td></tr>
</table>

TP505028	SIP reference: RFC 3261	ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE	
SIP selection criteria	NOT PICS 4/10	
ISDN selection criteria		
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a CALL PROCEEDING message, 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	ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE	
SIP selection criteria	NOT PICS 4/10	
ISDN selection criteria		
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a CALL PROCEEDING message, 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

<table>
<tr><th>TP505030</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISDN reference:
Q.1912.5 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 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</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">183 Session Progress encapsulated ACM: BCi Called party status = no indication</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>TP505031</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISDN reference:
Q.1912.5 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, 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</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></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>

TP505032	SIP reference: RFC 3261				ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection 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				ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out 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				ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, via a broadcast data link, after time-out of **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				ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends 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				ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, 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 (“Misdialed trunk prefix”)
VA_6	500 Server Internal Error Cause Value No. 8	Cause Value No. 8 (“Preemption”)
VA_7	500 Server Internal Error Cause Value No. 9	Cause Value No. 9 (“Preemption-circuit reserved for reuse”)
VA_8	486 Busy Here Cause Value No. 17	Cause Value No. 17 (“user busy”)
VA_9	480 Temporarily unavailable Cause Value No. 18	Cause Value No. 18 (“no user responding”)
VA_10	480 Temporarily unavailable Cause Value No. 19	Cause Value No. 19 (“no answer from the user”)
VA_11	480 Temporarily unavailable Cause Value No. 20	Cause Value No. 20 (“subscriber absent”)
VA_12	480 Temporarily unavailable Cause Value No. 21	Cause Value No. 21 (“all rejected”)
VA_13	410 Gone Cause Value No. 22	Cause Value No. 22 (“number changed”)
VA_14	480 Temporarily unavailable Cause Value No. 25	Cause Value No. 25 (“Exchange routing error”)
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”)
VA_18	480 Temporarily unavailable Cause Value No. 31	Cause Value No. 31 (“normal unspecified”) (Class default)

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

TP505037	SIP reference: RFC 3261				ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, and receives a CALL PROCEEDING message, on receipt of a 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				ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, and receives a CALL PROCEEDING message, on receipt of a 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				ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, and receives a CALL PROCEEDING message, on receipt of a DISCONNECT message with the **Cause value** CV_ISDN, **location** LOC_ISDN • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE message is returned to the ISDN side • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA • The ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field				
SIP parameter values	183 Session Progress 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				ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message 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	ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE	
SIP selection criteria	PICS 4/10	
ISDN selection criteria		
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message 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	ISDN reference: Q.1912.5 clause 6.11.2
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE	
SIP selection criteria	PICS 4/10	
ISDN selection criteria		
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message 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. 38-47) (47 is class default)
VA_9	500 Server Internal Error Cause Value No. 63	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_10	500 Server Internal Error Cause Value No. 88	Cause Value No. 88 ("incompatible destination")
VA_11	500 Server Internal Error Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)

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

TP505044	**SIP reference: RFC 3261**				**ISDN reference: Q.1912.5 clause 6.11.2**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends a SETUP message, receives a SETUP ACKNOWLEDGE message followed by a ALERTING message, having sent a 180 Ringing message on receipt of an ISDN RELEASE message • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RLC is returned to the ISDN side • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA • The ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field				
SIP parameter values	SIP Statue-Code: SIP_FAILURE_VA (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**				**ISDN reference: Q.1912.5 clause 6.11.2**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends a SETUP message, receives a SETUP ACKNOWLEDGE message followed by a ALERTING message, having sent a 180 Ringing message on receipt of an ISDN 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. 38-47) (47 is class default)
VA_9	500 Server Internal Error Cause Value No. 63	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_10	500 Server Internal Error Cause Value No. 88	Cause Value No. 88 ("incompatible destination")
VA_11	500 Server Internal Error Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)

TP505046	**SIP reference: RFC 3261**				**ISDN reference: Q.1912.5 clause 6.11.2**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message, receives a CONNECT message, a 200 OK message is sent, on receipt of a RELEASE COMPLETE message with the **Cause value** CV_ISDN, **location** LOC_ISDN • the SUT immediately requests the disconnection of the internal bearer path. • the SUT shall send a BYE message • The ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field				
SIP parameter 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
	200 OK INVITE(ANM)	←		←	CONNECT
			Communication		
	BYE(REL)	←		←	RELEASE COMPLETE
	200 OK BYE(RLC)	→			

TP505047	**SIP reference: RFC 3261**				**ISDN reference: Q.1912.5 clause 6.11.2**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message, receives a CONNECT message, a 200 OK message is sent, on receipt of a RELEASE message with the Cause value CV_ISDN, location LOC_ISDN • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RLC is returned to the ISDN side • the SUT shall send a BYE message • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field				
SIP parameter values	183 Session Progress 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
	200 OK INVITE(ANM)	←		←	CONNECT
			Communication		
	BYE(REL)	←		←	RELEASE
	200 OK BYE(RLC)	→		→	RELEASE COMPLETE

TP505048	**SIP reference: RFC 3261**				**ISDN reference: Q.1912.5 clause 6.11.2**
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message, receives a CONNECT message, a 200 OK message is sent, on receipt of a DISCONNECT message with the **Cause value** CV_ISDN, **location** LOC_ISDN • the SUT immediately requests the disconnection of the internal bearer path When the bearer channel is available for re-selection, an ISDN RELEASE is returned to the ISDN side • the SUT shall send a BYE message • The ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field				
SIP parameter values	183 Session Progress 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
	200 OK INVITE(ANM)	←		←	CONNECT
			Communication		
	BYE(REL)	←		←	DISCONNECT
	200 OK BYE(RLC)	→		→	RELEASE
				←	RELEASE COMPLETE

<table>
<tr><th>TP505049</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISDN reference:
Q.1912.5 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</th><th colspan="2">ISDN reference:
Q.1912.5 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>

<table>
<tr><th>TP505051</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISDN reference:
Q.1912.5 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 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">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="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>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 27

<table>
<tr><th colspan="3">Values for test purposes TP108047 and TP108052</th></tr>
<tr><th colspan="2">←SIP Message
SIP_FAILURE_VA
CV_SIP</th><th>← REL
Cause Indicators parameter
CV_ISDN,</th></tr>
<tr><td>VA_1</td><td>BYE
Cause Value No. 16</td><td>Cause Value No. 16</td></tr>
<tr><td>VA_2</td><td>BYE
Cause Value No. 31</td><td>Cause Value No. 31 ("normal unspecified")
(Class default)</td></tr>
<tr><td>VA_3</td><td>BYE
Cause Value No. 38</td><td>Cause Value No. 38 ("Network out of order")</td></tr>
<tr><td>VA_4</td><td>BYE
Cause Value No. 41</td><td>Cause Value No. 41 ("Temporary failure ")</td></tr>
<tr><td>VA_5</td><td>BYE
Cause Value No. 111</td><td>Cause Value No. 111 ("protocol error, unspecified")
(Class default)</td></tr>
</table>

A.1.1.1.1.6 Receipt of BYE / CANCEL messages

TP506001	SIP reference: RFC 3261				ISDN reference: Q.1912.5 clause 6.11.1
TSS reference	SIP-I-ISDN/Basic_call/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 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				ISDN reference: Q.1912.5 clause 6.11.1
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out 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				ISDN reference: Q.1912.5 clause 6.11.1
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, 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				ISDN reference: Q.1912.5 clause 6.11.1
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a 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				ISDN reference: Q.1912.5 clause 6.11.1
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a 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				ISDN reference: Q.1912.5 clause 6.11.1
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message, 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	→			

<table>
<tr><td>TP506007</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISDN reference:
Q.1912.5 clause 6.11.1</td></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL</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 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</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</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">DISC: Cause value and location mapped from the encapsulated REL in the received CANCEL</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>183 Session Progress(CPG)</td><td>←</td><td></td><td>←</td><td>PROGRESS(PI)</td></tr>
<tr><td>CANCEL(REL)</td><td>→</td><td></td><td>→</td><td>DISCONNECT</td></tr>
<tr><td>200 OK CANCEL</td><td>←</td><td></td><td>←</td><td>RELEASE</td></tr>
<tr><td>487 Request Terminated</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>
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<tr><td>TP506008</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISDN reference:
Q.1912.5 clause 6.11.1</td></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL</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 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</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">DISC: Cause value and location mapped from the encapsulated REL in the received CANCEL</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></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>CANCEL(REL)</td><td>→</td><td></td><td>→</td><td>DISCONNECT</td></tr>
<tr><td>200 OK CANCEL</td><td>←</td><td></td><td>←</td><td>RELEASE</td></tr>
<tr><td>487 Request Terminated</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>

TP506009	SIP reference: RFC 3261				ISDN reference: Q.1912.5 clause 6.11.1
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_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				ISDN reference: Q.1912.5 clause 6.11.1
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message, 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				ISDN reference: Q.1912.5 clause 6.11.1
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out 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				ISDN reference: Q.1912.5 clause 6.11.1
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message, 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				ISDN reference: Q.1912.5 clause 6.11.1
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message followed by a PROGRESS message, on receipt of SIP **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				ISDN reference: Q.1912.5 clause 6.11.1
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a 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				ISDN reference: Q.1912.5 clause 6.11.1
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message, 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				ISDN reference: Q.1912.5 clause 6.11.1
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message, 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**	**ISDN/ISDN reference: Q.1912.5 clause 7.1 1 a)**
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria		
ISDN selection criteria		
Test purpose	Ensure that the SUT in Idle state, on receipt of a SETUP message containing the complete **called party number** and the **sending complete** indication • Sends the INVITE message	
SIP parameter values		
ISDN parameter values	SETUP; **Called party 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**	**ISDN/ISDN reference: Q.1912.5 clause 7.1 1 b)**
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria		
ISDN selection criteria		
Test purpose	Ensure that the SUT in Idle state, on receipt of a SETUP message containing the **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	→			

<table>
<tr><td>TP601003</td><td colspan="4">SIP reference: RFC 3261</td><td>ISDN/ISDN reference:
Q.1912.5 clause 7.1 1 c)</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Sending of the INVITE 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 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</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">SETUP; Called party number: sufficient number of digits to route to the called party</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>
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<tr><td>TP601004</td><td colspan="4">SIP reference: RFC 3261</td><td>ISDN/ISDN reference:
Q.1912.5 clause 7.1 1 d)</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Sending of the INVITE 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 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</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-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>
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<table>
<tr><th>TP601005</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISDN/ISDN reference:
Q.1912.5 clause 7.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Sending of the INVITE 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 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="5"></td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">INVITE: a_b_m_LINE_VALUE, IAM encapsulated in a MIME-body</td></tr>
<tr><td rowspan="9">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 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

<table>
<tr><td>TP601006</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISDN/ISDN reference:
Q.1912.5 clause 7.1.2</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Sending of the INVITE 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 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</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: To: sip: ….; user=phone</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-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>
<tr><td>TP601007</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISDN/ISDN reference:
Q.1912.5 clause 7.1.2</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Sending of the INVITE 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 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</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: To:</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></td></tr>
<tr><td>INFO</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>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>

TP601008	**SIP reference: RFC 3261**	**ISDN/ISDN reference: Q.1912.5 clause 7.1.2**
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria		
ISDN selection criteria		
Test purpose	Ensure that the SUT is mapping 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	(see message flow below)	

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**	**ISDN/ISDN reference: Q.1912.5 clause 7.1.2**
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria		
ISDN selection criteria		
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, TON **= "International number"** of the SETUP • to the addr-spec component of the **To header field** in the INVITE message • The format of the To header field is "+CC+NDC+SN" • the forward address information is derived from the user info component of the INVITE Request-URI	
SIP parameter values	INVITE: To: …	
ISDN parameter values		
Comments	(see message flow below)	

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>TP601010</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISDN/ISDN reference: Q.1912.5 clause 7.1.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Sending of the INVITE 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 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</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: To: …</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-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>
<tr><th>TP601011</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISDN/ISDN reference: Q.1912.5 clause 7.1.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Sending of the INVITE 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 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</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: To: …</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-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>
<tr><td>TP601012</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISDN/ISDN reference: Q.1912.5 clause 7.1.2</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Sending of the INVITE 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 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</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: To: …</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></td></tr>
<tr><td>INFO</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>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><td>TP601013</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISDN/ISDN reference: Q.1912.5 clause 7.1.2</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISDN-SIP /Basic call/Sending of the INVITE 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 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</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: To: …</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></td></tr>
<tr><td>INFO</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>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>

TP601014	SIP reference: RFC 3261	ISDN/ISDN reference: Q.1912.5 clause 7.1.2
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria		
ISDN selection criteria		
Test purpose	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	ISDN/ISDN reference: Q.1912.5 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	→			

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

TP602003	SIP reference: RFC 3261				ISDN/ISDN reference: Q.1912.5 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				ISDN/ISDN reference: Q.1912.5 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	→			

<table>
<tr><th>TP602005</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISDN/ISDN reference:
Q.1912.5 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 1/9 AND 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 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</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="22">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>404/484</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>INFO</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>404/484</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>INFO</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>404/484</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>INFO</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</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>

TP602006	**SIP reference: RFC 3261**				**ISDN/ISDN reference: Q.1912.5 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**				**ISDN/ISDN reference: Q.1912.5 clause 7.1 1) a), 7.3.1**
TSS reference	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING				
SIP selection criteria	PICS 3/1				
ISDN selection criteria					
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the complete **called party number** and the **sending complete** indication • Sends the INVITE message to called user • Sends the CALL PROCEDING message • Sends the PROGRESS message, with the with progress description set to PI_VAL				
SIP parameter values	183 Session Progress with encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with Progress indicator				
ISDN parameter values	CALL PROCEEDING PROGRESS(PI value=PI_VAL)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	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</th><th colspan="2">ISDN/ISDN reference:
Q.1912.5 clause 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></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>

<table>
<tr><th>TP603003</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISDN/ISDN reference:
Q.1912.5 clause 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="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></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			ISDN/ISDN reference: Q.1912.5 clause 7.1 1) a), 7.3.1	
TSS reference	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING				
SIP selection criteria	PICS 3/1				
ISDN selection criteria					
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the complete **called party number** and the **sending complete** indication, on receipt of a 180 Ringing message • Sends the ALERTING message with the with the with progress description value PI_VAL				
SIP parameter values	180 Ringing encapsulated 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			ISDN/ISDN reference: Q.1912.5 clause 7.1 1 a)	
TSS reference	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING				
SIP selection criteria	PICS 3/2				
ISDN selection criteria					
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the complete **called party number** where the end of address signalling is determined by the expiration timer T_{OIW2} after the receipt of the latest address message on receipt of a 183 Session Progress 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	→			

<table>
<tr><th>TP603006</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISDN/ISDN reference: Q.1912.5 clause 7.1 1 a)</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 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</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">183 Session Progress encapsulated ACM: BCi Called party status = no indication, without ATP</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></td></tr>
<tr><td>INFO</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></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(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>TP603007</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISDN/ISDN reference: Q.1912.5 clause 7.1 1 a)</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 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</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">183 Session Progress without encapsulated ISUP message</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></td></tr>
<tr><td>INFO</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></td><td></td><td></td><td>←</td><td>183 Session Progress</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>

A.1.1.1.2.4 Sending of the CONNECT message

TP604001	SIP reference: RFC 3261			ISDN/ISDN reference: Q.1912.5 clause 7.5
TSS reference	ISDN-SIP /Basic call/Sending_of_CONNECT			
SIP selection criteria				
ISDN selection criteria				
Test purpose	Ensure that the SUT having sent the ALERTING message, on receipt of a 200 OK INVITE for this call, it shall stop timer TOIW2 (if running) • Send CONNECT as determined by ISDN procedures • Stop any existing awaiting answer indication (e.g. ringing tone)			
SIP parameter values	200 OK INVITE: encapsulated ANM in the MIME body			
ISDN parameter values	CONNECT			
Comments	**ISDN**		**SUT**			**SIP-I**

	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	ISDN/ISDN reference: Q.1912.5 clause 7.5
TSS reference	ISDN-SIP /Basic call/Sending_of_CONNECT	
SIP selection criteria		
ISDN selection criteria		
Test purpose	Ensure that the SUT does not having sent the ALERTING message, on receipt of a 200 OK INVITE for this call, it shall stop timer TOIW2 (if running) • Send CON as determined by ISDN procedures • Stop any existing awaiting answer indication (e.g. ringing tone)	
SIP parameter values	200 OK INVITE: encapsulated CON in the MIME body	
ISDN parameter values	CONNECT	
Comments		

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

TP605002	**SIP reference: RFC 3261**				**ISDN/ISDN reference: Q.1912.5 clause 7.7.1, 1)**
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP but before an INVITE has been sent. On receipt of a RELEASE message • no action is required on the SIP side other than to terminate local procedures if any are in progress				
SIP parameter values					
ISDN parameter values	RELEASE; **cause value**: (PIXIT), **location** (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→			
	RELEASE	→			
	RELEASE COMPLETE	←			

TP605003	**SIP reference: RFC 3261**				**ISDN/ISDN reference: Q.1912.5 clause 7.7.1, 1)**
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP but before an INVITE has been sent. On receipt of a 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	→			

TP605004	SIP reference: RFC 3261				ISDN/ISDN reference: Q.1912.5 clause 7.7.1 2)
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE ISDN message **before** a 200 OK (any) response message has been received which establishes a confirmed dialogue • The SUT shall hold the release procedure until a SIP 200 OK INVITE response has been received • The SUT shall send a BYE request				
SIP parameter values					
ISDN parameter values	RELEASE COMPLETE; **cause value**: (PIXIT), **location** (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	RELEASE COMPLETE	→			
				←	200 OK INVITE(ANM)
				→	ACK
				→	BYE(REL)
				←	200 OK BYE(RLC)

TP605005	SIP reference: RFC 3261				ISDN/ISDN reference: Q.1912.5 clause 7.7.1 2)
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection 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** a 200 OK (any) response message has been received which establishes a confirmed dialogue • The SUT shall hold the release procedure until a SIP 200 OK INVITE response has been received • The SUT shall send a BYE request				
SIP parameter values					
ISDN parameter values	RELEASE; **cause value**: (PIXIT), **location** (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
	RELEASE	→			
	RELEASE COMPLETE	←			
				←	200 OK INVITE(ANM)
				→	ACK
				→	BYE(REL)
				←	200 OK BYE(RLC)

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

TP605007	**SIP reference: RFC 3261**				**ISDN/ISDN reference: Q.1912.5 clause 7.7.1 2) 3)**
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE ISDN message **before** a 200 OK SIP response message has been received • The SUT shall hold the release procedure. A CANCEL is sent when any SIP response was been received • On subsequently receiving 200 OK INVITE messages , the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent				
SIP parameter values					
ISDN parameter values	RELEASE COMPLETE; **cause value**: (PIXIT), **location** (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
				←	100 Trying
	RELEASE COMPLETE	→			
				→	CANCEL(REL)
				←	200 OK INVITE(ANM)
				→	ACK
				←	200 OK CANCEL
				→	BYE(REL)
				←	200 OK BYE(RLC)

TP605008	SIP reference: RFC 3261	ISDN/ISDN reference: Q.1912.5 clause 7.7.1 2) 3)
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT	
SIP selection 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	ISDN/ISDN reference: Q.1912.5 clause 7.7.1 2) 3)
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT	
SIP selection criteria	NOT PICS 4/10	
ISDN selection criteria		
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a 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)

TP605010	**SIP reference: RFC 3261**			**ISDN/ISDN reference: Q.1912.5 clause 7.7.1 2) 3)**	
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection criteria	NOT PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE message ISDN message **before** an early dialogue with the message defined as SIP_MESSAGE_VA has been established • The SUT shall hold the release procedure until a **SIP_MESSAGE_VA** response has been received • The SUT shall send a CANCEL or BYE request.				
SIP parameter values					
ISDN parameter values	RELEASE COMPLETE; **cause value**: (PIXIT), **location** (PIXIT)				
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
				←	100 Trying
	RELEASE COMPLETE	→			
				←	SIP_MESSAGE_VA
				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><th>TP605011</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISDN/ISDN reference:
Q.1912.5 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 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; cause value: (PIXIT), location (PIXIT)</td></tr>
<tr><td rowspan="16">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></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>

TP605012	SIP reference: RFC 3261				ISDN/ISDN reference: Q.1912.5 clause 7.7.1 2) 3)
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection 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	**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				ISDN/ISDN reference: Q.1912.5 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	**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				ISDN/ISDN reference: Q.1912.5 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 reques				
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				ISDN/ISDN reference: Q.1912.5 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				ISDN/ISDN reference: Q.1912.5 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				ISDN/ISDN reference: Q.1912.5 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**			**ISDN/ISDN reference: Q.1912.5 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**			**ISDN/ISDN reference: Q.1912.5 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**				**ISDN/ISDN reference: Q.1912.5 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**				**ISDN/ISDN reference: Q.1912.5 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	→			

TP605022	**SIP reference: RFC 3261**			**ISDN/ISDN reference: Q.1912.5 clause 7.7.1 2) 3**	
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE message with **Cause value** CV_ISDN, **location** LOC_ISDN **before** a 200 OK response message has been received • The SUT shall hold the REL message a **CANCEL** is sent when any SIP response was been received • On subsequently receiving 200 OK INVITE messages , the SUT shall send an ACK for the 200 OK INVITE and subsequently send a **BYE** request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP				
SIP parameter 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)
				←	100 Trying
	RELEASE COMPLETE	→			
				→	CANCEL(REL)
				←	200 OK INVITE(ANM)
				→	ACK
				←	200 OK CANCEL
				→	BYE(REL)
				←	200 OK BYE(RLC)

TP605023	**SIP reference: RFC 3261**			**ISDN/ISDN reference: Q.1912.5 clause 7.7.1 2) 3**	
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection 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 message **before** a 200 OK response message has been received • The SUT shall hold the release procedure. A **CANCEL** is sent when any SIP response was been received • On subsequently receiving 200 OK INVITE messages , the SUT shall send an ACK for the 200 OK INVITE and subsequently send a **BYE** request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP				
SIP parameter 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	←		→	CANCEL(REL)
				←	200 OK INVITE(ANM)
				→	ACK
				←	200 OK CANCEL
				→	BYE(REL)
	→			←	200 OK BYE(RLC)

<table>
<tr><td>TP605024</td><td colspan="4">SIP reference: RFC 3261</td><td>ISDN/ISDN reference: Q.1912.5 clause 7.7.1 2) 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 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</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN DISCONNECT</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">DISCONNECT; 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>DISCONNECT</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>RELEASE</td><td>←</td><td></td><td>→</td><td>CANCEL(REL)</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>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>TP605025</th><th colspan="2">SIP reference: RFC 3261</th><th colspan="3">ISDN/ISDN reference:
Q.1912.5 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>

<table>
<tr><th>TP605026</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISDN/ISDN reference:
Q.1912.5 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 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</td></tr>
<tr><td>ISDN parameter values</td><td colspan="5">RELEASE; cause value: (PIXIT), location (PIXIT)</td></tr>
<tr><td rowspan="17">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></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>

<table>
<tr><td>TP605027</td><td colspan="2">SIP reference: RFC 3261</td><td colspan="2">ISDN/ISDN reference: Q.1912.5 clause 7.7.1 3)</td></tr>
<tr><td>TSS reference</td><td colspan="4">ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="4">PICS 4/10</td></tr>
<tr><td>ISDN selection criteria</td><td colspan="4"></td></tr>
<tr><td>Test purpose</td><td colspan="4">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</td></tr>
<tr><td>SIP parameter values</td><td colspan="4">BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN DISCONNECT</td></tr>
<tr><td>ISDN parameter values</td><td colspan="4">DISCONNECT; cause value: (PIXIT), location (PIXIT)</td></tr>
<tr><td rowspan="17">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td></tr>
</table>

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

<table>
<tr><td>TP605028</td><td>SIP reference: RFC 3261</td><td>ISDN/ISDN reference: Q.1912.5 clause 7.7.1 3)</td></tr>
<tr><td>TSS reference</td><td colspan="2">ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="2">PICS 4/10</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 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</td></tr>
<tr><td>SIP parameter values</td><td colspan="2">BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN RELEASE COMPLETE</td></tr>
<tr><td>ISDN parameter values</td><td colspan="2">RELEASE COMPLETE; cause value: (PIXIT), location (PIXIT)</td></tr>
<tr><td>Comments</td><td colspan="2"></td></tr>
</table>

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				ISDN/ISDN reference: Q.1912.5 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				ISDN/ISDN reference: Q.1912.5 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				ISDN/ISDN reference: Q.1912.5 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

TP605032	**SIP reference: RFC 3261**				**ISDN/ISDN reference: Q.1912.5 clause 7.7.1 3)**
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection 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** 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				
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

TP605032	SIP reference: RFC 3261				ISDN/ISDN reference: Q.1912.5 clause 7.7.1 3)
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection 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**" (Note 1)	Protocol-cause	"**cause= XX**" (Note 1)
-	-	Reason-text	Should be filled with the definition text as stated in Q.850 (Note 2)
NOTE 1: "XX" is the Cause Value as defined in Q.850. NOTE 2: Due to the fact that the Cause Indications parameter does not include the definition text as defined in Table 1/Q.850 this is based on provisioning in the O-IWU.			

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

TP606001	**SIP reference: RFC 3261**				**ISDN/ISDN reference: Q.1912.5 clause 7.7.2**
TSS reference	ISDN-SIP /Basic call/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	←			

TP606002	SIP reference: RFC 3261				ISDN/ISDN reference: Q.1912.5 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 a INVITE message and on receipt of a BYE message where a Reason header field with Q.850 Cause Value is included • sends a DISCONNECT message constructed from the encapsulated REL in the received BYE MIME body.				
SIP parameter values	BYE: ISUP REL encapsulated in the MIME body, Reason header value = (PIXIT)				
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 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**" (Note 1)	Cause Value	constructed from the encapsulated REL
-	-	Location	constructed from the encapsulated REL
NOTE: "XX" is the Cause Value as defined in Q.850.			

TP606003	**SIP reference: RFC 3261**				**ISDN/ISDN reference: Q.1912.5 clause 7.7.6**
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. 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**				**ISDN/ISDN reference: Q.1912.5 clause 7.7.6**
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. 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**				**ISDN/ISDN reference: Q.1912.5 clause 7.7.6**
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP 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				ISDN/ISDN reference: Q.1912.5 clause 7.7.6
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria	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				ISDN/ISDN reference: Q.1912.5 clause 7.7.6
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria	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**			**ISDN/ISDN reference: Q.1912.5 clause 7.7.6**
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE			
SIP selection criteria	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**			**ISDN/ISDN reference: Q.1912.5 clause 7.7.6**	
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message, on receipt of a Failure message **487 Request terminated** where a Reason header field with Q.850 Cause Value is **not** included • no action is taken on the ISDN if a CANCEL request was previously sent before answer to an INVITE				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**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**			**ISDN/ISDN reference: Q.1912.5 clause 7.7.6**	
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message, on receipt of a Failure message **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				ISDN/ISDN reference: Q.1912.5 clause 7.7.6
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria	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				ISDN/ISDN reference: Q.1912.5 clause 7.7.6
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message a 180 ringing message has been received on receipt of a Failure message (4xx, 5xx, 6xx) defined as **SIP_Failure_VA** where a Reason header field with Q.850 Cause Value is **not** included • sends a DISCONNECT message with the **Cause value** CV_ ISDN				
SIP parameter 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**				**ISDN/ISDN reference: Q.1912.5 clause 7.7.6**
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria	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**			**ISDN/ISDN reference: Q.1912.5 clause 7.7.6**	
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria	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**			**ISDN/ISDN reference: Q.1912.5 clause 7.7.6**	
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria	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

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

TP606017	**SIP reference: RFC 3261**				**ISDN/ISDN reference: Q.1912.5 clause 7.7.6**
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria	NOT PICS 4/17				
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT in state N3, before having received an backward message, on receipt of a response message (3xx) defined as **SIP_Response_VA,** the SUT • sends a DISC message with the **Cause value** 127 Interworking				
SIP parameter values					
ISDN parameter values	REL; **cause value**: CV_ISDN				
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	←			

TP606018	SIP reference: RFC 3261	ISDN/ISDN reference: Q.1912.5 clause 7.7.6
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE	
SIP selection criteria	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	ISDN/ISDN reference: Q.1912.5 clause 7.7.6
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE	
SIP selection criteria	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**				**ISDN/ISDN reference: Q.1912.5 clause 7.7.6**
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE				
SIP selection criteria	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><td>TP606021</td><td colspan="3">SIP reference: RFC 3261</td><td colspan="2">ISDN/ISDN reference: Q.1912.5 clause 7.7.6</td></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				ISDN/ISDN reference: Q.1912.5 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 TOIW3. The RELEASE or DISCONNECT message shall only be sent if TOIW3 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				ISDN/ISDN reference: Q.1912.5 clause 7.7.6.1
TSS reference	ISDN-SIP /Basic call/Autonomous_release				
SIP selection criteria	NOT PICS 3/4				
ISDN selection criteria					
Test purpose	Ensure that the SUT a On receipt of a 484 Address Incomplete response for the current INVITE (i.e. there are no other pending INVITE transactions for this call), if the O-IWU is not configured to propagate overlap signalling into the SIP network then the timer shall not be started and the • DISCONNECT message shall be sent immediately to the ISDN network				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	INVITE(IAM)
				←	484 Address incomplete
	CASE A			→	ACK
	RELEASE	←			
	RELEASE COMPLETE	→			
	CASE B				
	DISCONNECT	←			
	RELEASE	→			
	RELEASE COMPLETE	←			

TP607003	**SIP reference: RFC 3261**				**ISDN/ISDN reference: Q.1912.5 clause 7.7.3**
TSS reference	ISDN-SIP /Basic call/Autonomous_release				
SIP selection 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**				**ISUP reference: Q.1912.5 Q.731 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**				**ISUP reference: Q.1912.5 Q.731 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**				**ISUP reference: Q.1912.5 Q.731 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				ISUP reference: Q.1912.5 Q.731 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				ISUP reference: Q.1912.5 Q.731 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			ISUP reference: Q.1912.5 Q.731 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			ISUP reference: Q.1912.5 Q.731 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**			**ISUP reference:** **Q.1912.5** **Q.731 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**			**ISUP reference:** **Q.1912.5** **Q.731 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	→			

TP702004	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.731 clause 4.5.2.1.1
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Restricted calling party number (user provided, verified and passed) with calling sub-address** Verify that the IUT can successfully originate a call having a **calling party number** with the screening indicator set to "user provided, verified and passed", the address presentation restricted indicator set to "presentation restricted" and an **access transport** parameter containing the calling sub-address				
SIP parameter 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	→			

TP702005	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.731 clause 4.5.2.1.1
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Restricted calling party number (user provided, not verified)** Verify that the IUT can successfully originate a call having a default **calling party number** with the screening indicator set to "network provided" and a **generic number** containing the additional calling party number with the screening indicator set to "user provided, not verified", both having the address presentation restricted indicator set to "presentation restricted"				
SIP parameter 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	→			

TP702006	SIP reference: RFC 3261				ISUP reference: Q.1912.5 Q.731 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				ISUP reference: Q.1912.5 Q.731 clause 4.2.1
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria	DLE				
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**				**ISUP reference: Q.1912.5 Q.731 clause 4.2.1**
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria	DLE				
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**				**ISUP reference: Q.1912.5 Q.731 clause 4.2.1**
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria	DLE				
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				ISUP reference: Q.1912.5 Q.731 clause 4.2.1
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria	DLE				
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				ISUP reference: Q.1912.5 Q.731 clause 5.5.2.5.1 i)
TSS reference	ISDN-(ISUP)-SIP/SS/COLP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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**			**ISUP reference:** **Q.1912.5** **Q.731 clause 5.5.2.5.1 i)**	
TSS reference	ISDN-(ISUP)-SIP/SS/COLP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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**			**ISUP reference:** **Q.1912.5** **Q.731 clause 5.5.2.5.1 i)**	
TSS reference	ISDN-(ISUP)-SIP/SS/COLP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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				ISUP reference: Q.1912.5 Q.731 clause 5.5.2.5.1 i)
TSS reference	ISDN-(ISUP)-SIP/SS/COLP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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				ISUP reference: Q.1912.5 Q.731 clause 5.5.2.5.1 i)
TSS reference	ISDN-(ISUP)-SIP/SS/COLP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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				ISUP reference: Q.1912.5 Q.731 clause 5.5.2.5.1 i)
TSS reference	ISDN-(ISUP)-SIP/SS/COLP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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				ISUP reference: Q.1912.5 Q.731 clause 5.5.2.5.1 i)
TSS reference	ISDN-(ISUP)-SIP/SS/COLP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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**				**ISUP reference: Q.1912.5 Q.731 clause 5.5.2.5.1 i)**
TSS reference	ISDN-(ISUP)-SIP/SS/COLP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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**				**ISUP reference: Q.1912.5 Q.731 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**	**ISUP reference: Q.1912.5 Q.731 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**	**ISUP reference: Q.1912.5 Q.731 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**				**ISUP reference: Q.1912.5 Q.731 clause 6.5.2.1.**
TSS reference	ISDN-(ISUP)-SIP/SS/COLR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Restricted connected number (user provided, verified and passed) with connected sub-address** Verify that the IUT can provide a **connected number** with the screening indicator set to "user provided, verified and passed" and with the address presentation restricted indicator set to "presentation restricted", if the user provided COL is valid. Additionally, an **access transport** parameter containing the connected sub-address shall also be provided				
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body				
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP-I**
	SETUP	→		→	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**				**ISUP reference: Q.1912.5 Q.731 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	→			

TP704006	SIP reference: RFC 3261	ISUP reference: Q.1912.5 Q.731 clause 6.5.2.1.2
TSS reference	ISDN-(ISUP)-SIP/SS/COLR	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Restricted connected number (user provided, not verified)** Verify that the IUT can provide a default **connected number** with the screening indicator set to "network provided" and a **generic number** containing the additional connected number with the screening indicator set to "user provided, not verified" - both having the address presentation restricted indicator set to "presentation restricted"	
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body	
ISDN parameter values		

Comments	ISDN		SUT		SIP-I
	SETUP	→		→	INVITE(IAM)
	CASE A				
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
				→	ACK
	CASE B				
	CONN	←		←	200 OK INVITE(CON)
				→	ACK
			Communication		
	DISC	→		→	BYE(REL)
	REL	←		←	200 OK BYE(RLC)
	REL_COM	→			

TP704007	SIP reference: RFC 3261	ISUP reference: Q.1912.5 Q.731 clause 6.5.2.1.2
TSS reference	ISDN-(ISUP)-SIP/SS/COLR	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Restricted connected number (user provided, not verified) with connected sub-address** Verify that the IUT can provide a default **calling party number** with the screening indicator set to "network provided", a **generic number** containing the additional connected number with the screening indicator set to "user provided, not verified" - both having the address presentation restricted indicator set to "presentation restricted" and additionally an **access transport** parameter containing the connected sub-address	
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body	
ISDN parameter values		

Comments	**ISDN**		**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.1.1.2.5 Terminal Portability (TP)

TP705001	SIP reference: RFC 3261				ISUP reference: Q.1912.5 4.5.2.1.1 a)/Q.733
TSS reference	ISDN-(ISUP)-SIP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Terminal portability, requested by the calling party** To verify that the calling party can suspend and resume an outgoing call and that user initiated SUS and RES messages are sent to the succeeding exchange				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP	→		→	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				ISUP reference: Q.1912.5 4.5.2.1.1 b)/Q.733
TSS reference	ISDN-(ISUP)-SIP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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	→			

TP705003	SIP reference: RFC 3261				ISUP reference: Q.1912.5 4.5.2.1.2/Q.733
TSS reference	ISDN-(ISUP)-SIP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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				
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
			T2 expiry		
				→	BYE(REL)
				←	200 OK BYE

TP705004	SIP reference: RFC 3261				ISUP reference: Q.1912.5 4.5.2.1.1/Q.733
TSS reference	ISDN-(ISUP)-SIP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Terminal portability, 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				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP	→		→	INVITE(IAM)
	ALERTING	←		←	180 Ringing(ACM)
	CONN	←		←	200 OK INVITE(ANM)
			Communication		
	SUSPEND	→		→	INFO(SUS)
				←	200 OK INFO
				←	BYE(REL)
				→	200 OK BYE

TP705005	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 4.5.2.5.1 a)/Q.733**
TSS reference	ISDN-(ISUP)-SIP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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**				**ISUP reference: Q.1912.5 4.5.2.5.1 b)/Q.733**
TSS reference	ISDN-(ISUP)-SIP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Terminal portability, requested by the called party** To verify that the called party can suspend and resume an incoming call and that user initiated **SUS** and **RES** messages are sent to the preceding exchange				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP	←		←	INVITE(IAM)
	ALERTING	→		→	180 Ringing(ACM)
	CONN	→		→	200 OK INVITE(ANM)
			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**					**ISUP reference: Q.1912.5 1.1.5.2.1.1.1; 1.1.5.2.1.1.3; 1.1.5.2.2-4.1/Q.737**
TSS reference	ISDN-(ISUP)-SIP/SS/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**				**ISUP reference: Q.1912.5 .1.5.2.5.2.3; 1.1.5.2.2-4.2/Q.737**
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS1 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.					

TP706003	SIP reference: RFC 3261	ISUP reference: Q.1912.5 1.1.5.2.5.2.3; 1.1.5.2.3-5.2/Q.737
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**UUS1 implicit - discarded but no indication received** To verify that the IUT can successfully initiate/transit a call with an UUS1 implicit request, and complete the call if no indication is provided in the backward direction (see note).	
SIP parameter values		
ISDN parameter values		
Comments		

ISDN		SUT		SIP
SETUP(UUInf)	→		→	INVITE(IAM UUInf)
ALERTING	←		←	180 Ringing(ACM)
CONN	←		←	200 OK INVITE(ANM)
			→	ACK
		Communication		
DISC	←		←	BYE(REL)
REL	→		→	200 OK BYE
REL_COM	←			

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.

TP706004	SIP reference: RFC 3261	ISUP reference: Q.1912.5 1.1.5.2.1.1.1; 1.1.5.2.1.1.3; 1.1.5.2.3-5.1/Q.737
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**UUS1 implicit - acceptance** To verify that the IUT can successfully transit/accept a call with an UUS1 implicit request, and transfer/include the **user-to-user information** parameter in the **ACM**, **CPG, ANM or CON** as implicit acceptance (no **user-to-user indicators**)	
SIP parameter values		
ISDN parameter values		
Comments		

ISDN		SUT		SIP
SETUP(UUInf)	→		→	INVITE(IAM UUInf)
CASE A				
ALERTING(UUInf)	←		←	180 Ringing(ACM UUInf)
CASE B				
ALERTING	←		←	180 Ringing(ACM)
NOTIFY(UUInf)			←	183 Session Progress(CPG UUInf)
CASE C				
CONN(UUInf)	←		←	200 OK INVITE(CON UUInf)
			→	ACK
CASE D				
ALERTING	←		←	180 Ringing(ACM)
CONN(UUInf)	←		←	200 OK INVITE(ANM UUInf)
			→	ACK
		Communication		
DISC	←		←	BYE(REL)
REL	→		→	200 OK BYE
REL_COM	←			

TP706005	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 1.1.5.2.5.2.3; 1.1.5.2.3-5.2/Q.737**
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS1 implicit - discard with indication generated** To verify that the IUT can successfully transit/accept a call with an UUS1 implicit request and set the **user-to-user indicators** to "user-to-user information discarded by the network" in the first backward message, if the network is unable to support it (see note).				
SIP parameter values					
ISDN parameter values					
Comments	**SIP**		**SUT**		**ISDN**
	INVITE(IAM UUInf)	→		→	SETUP
	180 Ringing(ACM UUInd)	←		←	ALERTING
	200 OK INVITE(ANM)	←		←	CONN
	ACK	→			
			Communication		
	BYE(REL)	←		←	DISC
	200 OK BYE	→		→	REL
				←	REL_COM
NOTE: The user-to-user information is discarded because the network does not support it.					

TP706006	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 1.1.5.2.1.1.2; 1.1.5.2.2-4.1/Q.737**
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS1 explicit non-essential - request** To verify that the IUT can successfully initiate/transit a call with an UUS1 explicit non-essential request, by including/transferring the **user-to-user information** parameter and the **user-to-user indicators** in the **IAM** set to "request, not essential"				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP(FAC uus1reqness)				INVITE(IAM UUInd, UUInf)
	ALERTING(FAC uus1rr)				180 Ringing(ACM)
	CONN				200 OK INVITE(ANM)
					ACK
			Communication		
	BYE(REL)	←		←	DISC
	200 OK BYE	→		→	REL
				←	REL_COM

<table>
<tr><td>TP706007</td><td colspan="5">SIP reference: RFC 3261</td><td>ISUP reference:
Q.1912.5
1.1.5.2.5.2.3;
1.1.5.2.2-4.2/Q.737</td></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 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).</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="18">Comments</td><td>ISDN</td><td></td><td>SUT</td><td></td><td colspan="2">SIP</td></tr>
<tr><td>SETUP(FAC uus1reqness)</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(IAM UUInd, UUInf)</td></tr>
<tr><td>CASE A</td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>ALERTING(FAC uus1rr)</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing(ACM UUInd s1 prov)</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE</td></tr>
<tr><td>CASE B</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(ACM)</td></tr>
<tr><td>ALERTING(FAC uus1rr)</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing(CPG UUInd s1 prov)</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE</td></tr>
<tr><td>CASE C</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(FAC uus1rr)</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE(ANM UUInd s1 prov)</td></tr>
<tr><td>CASE D</td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>CONN(FAC uus1rr)</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE(ANM UUInd s1 prov)</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">DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td colspan="2">REL</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">REL_COM</td></tr>
<tr><td colspan="7">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>

<table>
<tr><th>TP706008</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5
1.1.5.2.5.2.3;
1.1.5.2.2-4.2/Q.737</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				ISUP reference: Q.1912.5 1.1.5.2.1.1.2; 1.1.5.2.3-5.1/Q.737
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS1 explicit non-essential - acceptance** To verify that the IUT can successfully transit/accept a call with an UUS1 explicit non-essential request, by transferring/including the **user-to-user indicators** parameter in the **ACM, CPG, ANM** or **CON** set to "service provided"				
SIP parameter values					
ISDN parameter values					
Comments	**SIP**		**SUT**		**ISDN**
	INVITE(IAM UUInd, UUInf)	→		→	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				ISUP reference: Q.1912.5 1.1.5.2.5.2.2; 1.1.5.2.2-5.2/Q.737
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS1 explicit 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**			**ISUP reference: Q.1912.5 1.1.5.2.1.1.2; 1.1.5.2.2-5.1/Q.737**	
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS1 explicit essential - 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**			**ISUP reference: Q.1912.5 1.1.5.2.5.2.2; 1.1.5.2.2-5.2/Q.737**	
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS1 explicit essential - 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**					**ISUP reference: Q.1912.5 1.1.5.2.1.1.2; 1.1.5.2.2-5.1/Q.737**
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1					
SIP selection criteria						
ISUP selection criteria						
Test purpose	**UUS1 explicit essential - acceptance** To verify that the IUT can successfully complete a call with an UUS1 explicit essential request having the **user-to-user indicators** parameter in the **ACM**, **CPG**, **ANM** or **CON** set to "service provided"					
SIP parameter values						
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**					**ISUP reference: Q.1912.5 1.1.5.2.5.2.2; 1.1.5.2.2-5.2/Q.737**
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1					
SIP selection criteria						
ISUP selection criteria						
Test purpose	**UUS1 explicit essential - rejection** To verify that the service can be rejected with a **REL** having the **Cause value** 29 "facility rejected" or 69 "requested facility not implemented", either with diagnostics (specifying the name of the user-to-user indicator parameter) (see note).					
SIP parameter values						
ISDN parameter values						
Comments	**ISDN**		**SUT**		**SIP**	
	SETUP(FAC uus1reqess)	→		→	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**		**ISUP reference: Q.1912.5 1.2.5.2.1.1.2; 1.2.5.2.2-5.1/Q.737**		
TSS reference	ISDN-(ISUP)-SIP/SS/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**		**ISUP reference: Q.1912.5 1.2.5.2.5.2.2; 1.2.5.2.2-5.2/Q.737**		
TSS reference	ISDN-(ISUP)-SIP/SS/UUS2				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS2 explicit non-essential - explicit rejection (service not provided)** To verify that the UUS2 service can be rejected and the **user-to-user indicators** in the **ACM** or **CPG** are set to "service 2 not provided" (see note).				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP(FAC 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**				**ISUP reference: Q.1912.5 1.2.5.2.5.2.3; 1.2.5.2.2-5.2/Q.737**
TSS reference	ISDN-(ISUP)-SIP/SS/UUS2				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS2 explicit non-essential - implicit rejection (no indication)** To verify that the IUT can successfully complete a call with an UUS2 explicit non-essential request, if no indication is provided in the backward direction (see note).				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP(FAC 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**				**ISUP reference: Q.1912.5 1.2.5.2.1.1.2; 1.2.5.2.2-5.1/Q.737**
TSS reference	ISDN-(ISUP)-SIP/SS/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				ISUP reference: Q.1912.5 1.2.5.2.1.1.2; 1.2.5.2.2-5.1/Q.737
TSS reference	ISDN-(ISUP)-SIP/SS/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				ISUP reference: Q.1912.5 1.2.5.2.5.2.1; 1.2.5.2.2-5.2/Q.737
TSS reference	ISDN-(ISUP)-SIP/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**				**ISUP reference: Q.1912.5 1.2 5.2.5.2.1; 1.2.5.2.2-5.2/Q.737**
TSS reference	ISDN-(ISUP)-SIP/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)

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

TP706202	SIP reference: RFC 3261				ISUP reference: Q.1912.5 1.3.5.2.1.1.2; 1.3.5.2.2-5.1/Q.737
TSS reference	ISDN-(ISUP)-SIP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**UUS3 explicit essential - request and acceptance** To verify that the IUT can successfully originate/transit a call with an UUS3 explicit essential request, having in the **IAM** the **user-to-user indicators** set to "request, essential" and the ISDN user part preference indicator in the **forward call indicators** set to "ISUP required all the way" To verify that the IUT can successfully complete a call with an UUS3 explicit essential request having in the **ANM** or **CON** the Service 3 field of the **user-to-user indicators** parameter set to "service provided"				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP(UU3 req ess)	→		→	INVITE(IAM UU3 ess)
	ALERTING	←		←	180 Ringing(ACM)
	CONN(UU3 ret res)	←		←	200 OK INVITE(ANM UU3 prov)
			Communication		
	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	→			

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

TP706204	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 1.3.5.2.1.1.2; 1.3.5.2.2-5.1/Q.737**
TSS reference	ISDN-(ISUP)-SIP/SS/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	→			

TP706205	SIP reference: RFC 3261				ISUP reference: Q.1912.5 1.3.5.2.5.2.2/Q.737
TSS reference	ISDN-(ISUP)-SIP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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"				
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 err)	←		←	INFO(FRJ resp UU3 not prov)
				→	200 OK INFO
			Communication		
	DISC	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE
	REL_COM	→			

TP706206	SIP reference: RFC 3261				ISUP reference: Q.1912.5 1.3.5.2.5.2.2/ITU-T Q.737
TSS reference	ISDN-(ISUP)-SIP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria	PICS 11/3				
Test purpose	**UUS3 explicit non-essential - implicit rejection during call (no indication - discard FAA or FRJ)** To verify that the IUT can successfully complete a call with an UUS3 request in the **FAR**, if the **FAA** or **FRJ** are discarded				
SIP parameter values					
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP	→		→	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
			Communication		
	DISC	→		→	BYE(REL)
	RELEASE	←		←	200 OK BYE
	REL_COM	→			

A.1.1.2.7 Closed User Group (CUG)

TP707001	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 1.5.2.1.1 i) a)/Q.735**
TSS reference	ISDN-(ISUP)-SIP/SS/CUG				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**CUG without outgoing 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**				**ISUP reference: Q.1912.5 1.5.2.5.1; Table 1-2/Q.735**
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				ISUP reference: Q.1912.5 1.5.2.5.1; Table 1-2/Q.735
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				ISUP reference: Q.1912.5 1.5.2.5.1; Table 1-2/Q.735
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**				**ISUP reference: Q.1912.5 1.5.2.5.1; Table 1-2/Q.735**
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**				**ISUP reference: Q.1912.5 1.5.2.5.1; Table 1-2/Q.735**
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**			**ISUP reference: Q.1912.5 Q.73**
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**			**ISUP reference: Q.1912.5 1.5.2.5.1; Table 1-2/Q.735**
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**			**ISUP reference: Q.1912.5 1.5.2.5.1; Table 1-2 /Q.735**
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				ISUP reference: Q.1912.5 1.5.2.5.1; Table 1-2/Q.735
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				ISUP reference: Q.1912.5 1.5.2.5.1; Table 1-2/Q.735
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**			**ISUP reference: Q.1912.5 1.5.2.5.1; Table 1-2/Q.735**	
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**			**ISUP reference: Q.1912.5 1.5.2.5.1; Table 1-2/Q.735**	
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**			**ISUP reference: Q.1912.5 1.5.2.5.1; Table 1-2/Q.735**	
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				ISUP reference: Q.1912.5 8.5.2.1.1/Q.731
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				ISUP reference: Q.1912.5 8.5.2.5.1/Q.731
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)

TP709001	**SIP reference: RFC 3261**			**ISUP reference: Q.1912.5 7.5.2.1.1/Q.731.7**	
TSS reference	ISDN-(ISUP)-SIP/SS/MCID				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Successful MCID request** To verify that the IUT can successfully reply to an **IDR** having the **MCID request indicator** set to "MCID request" by sending an **IRS** with **MCID response indicator** set to "MCID included" and the **calling party number** included				
SIP parameter values	INFO: The encapsulated IDR contains the MCID Request indicator "requested" INFO: The encapsulated IRS contains the MCID response indicator "included" and the Calling party number of the originating User				
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP	→		→	INVITE(IAM)
				←	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	→			

TP709002	**SIP reference: RFC 3261**			**ISUP reference: Q.1912.5 7.5.2.1.1/Q.731.7**	
TSS reference	ISDN-(ISUP)-SIP/SS/MCID				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Successful MCID request** To verify that the IUT can successfully reply to an **IDR** having the **MCID request indicator** set to "MCID request" by sending an **IRS** with **MCID response indicator** set to "MCID included" and the **calling party number** included				
SIP parameter values	INFO: The encapsulated IDR contains the MCID Request indicator "requested" INFO: The encapsulated IRS contains the MCID response indicator "included" and the Calling party number of the originating User				
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP	←		←	INVITE(IAM)
				→	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	←			

<table>
<tr><th>TP709003</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5
7.5.2.1.1/Q.731.7</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>

TP709004	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 7.5.2.1.1/Q.731.7**
TSS reference	ISDN-(ISUP)-SIP/SS/MCID				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Successful MCID request - after ACM** To verify that the IUT will accept and reply correctly to an MCID request after ACM has been received. The IUT should reply to an **IDR** having the **MCID request indicator** set to "MCID request" by sending an **IRS** with **MCID response indicator** set to "MCID included" and the **calling party number** included (see note).				
SIP parameter values	INFO: The encapsulated IDR contains the MCID Request indicator "requested" INFO: The encapsulated IRS contains the MCID response indicator "included" and the Calling party number of the originating User				
ISDN parameter values					
Comments	**ISDN**		**SUT**		**SIP**
	SETUP	←		←	INVITE(IAM)
	ALERTING	→		→	180 Ringing(ACM)
				→	INFO(IDR requested)
				←	200 OK INFO
				←	INFO(IRS included)
				→	200 OK INFO
	CONN	→		→	200 OK INVITE(ANM)
				←	ACK
			Communication		
	DISC	←		←	BYE(REL)
	REL	→		→	200 OK BYE
	REL_COM	←			
NOTE: This situation may occur e.g. if the call has been forwarded before reaching the destination.					

<table>
<tr><th>TP709005</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5
7.5.2.1.1/Q.731.7</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 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</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", the Calling party number and the calling sub-address 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>TP70906</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5
7.5.2.1.2/Q.731.7</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">PICS 9/1</td></tr>
<tr><td>Test purpose</td><td colspan="5">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"</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 "not included"</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 not 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>

TP70907	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 7.5.2.1.2/Q.731.7**
TSS reference	ISDN-(ISUP)-SIP/SS/MCID				
SIP selection criteria					
ISUP selection criteria	PICS 9/1				
Test purpose	**MCID request - 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**				**ISUP reference: Q.1912.5 7.5.2.5.2/Q.731.7**
TSS reference	ISDN-(ISUP)-SIP/SS/MCID				
SIP selection criteria					
ISUP selection criteria	PICS 5/9				
Test purpose	**MCID timer (T39) expiry** To verify that call setup is 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**				**ISUP reference: Q.1912.5 Q.734 clause 1.5.2.1.1.2**	
TSS reference	ISDN-(ISUP)-SIP/SS/CONF					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Verify that the 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**	**ISUP reference: Q.1912.5 Q.734 clause 1.5.2.1.1.2**
TSS reference	ISDN-(ISUP)-SIP/SS/CONF	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Verify that the 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**	**ISUP reference: Q.1912.5 Q.734 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	ISUP reference: Q.1912.5 Q.734 clause 1.5.2.1.1.5
TSS reference:	ISDN-(ISUP)-SIP/SS/CONF	
SIP selection criteria		
ISUP selection criteria		
Test purpose	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/Q.734.

TP710005	**SIP reference: RFC 3261**	**ISUP reference: Q.1912.5 Q.734 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**	**ISUP reference: Q.1912.5 Q.734 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	ISUP reference: Q.1912.5 Q.734 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**	**ISUP reference: Q.1912.5 Q.734 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	ISUP reference: Q.1912.5 Q.734 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	ISUP reference: Q.1912.5 7.5.2.1.1.1 a)/Q.732.7
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

TP711002	**SIP reference: RFC 3261**						**ISUP reference: Q.1912.5 7.5.2.1.1.1 a)/Q.732.7**
TSS reference	ISDN-(ISUP)-SIP/SS/ECT						
SIP selection criteria							
ISUP selection criteria							
Test purpose	**Capability of storing and sending the calling party number in the call transfer number** To verify that the IUT is able to store the **calling party number** when only this CLI has been received from the remote user. This information is sent by the IUT to the other remote user in the **call transfer number** in either the **FAC** or **CPG** when the call transfer is activated						
SIP parameter values	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)						
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

TP711003	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 7.5.2.1.1.1 b)/Q.732.7**		
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						ISUP reference: Q.1912.5 7.5.2.1.1.1 b)/Q.732.
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	ISUP reference: Q.1912.5 7.5.2.1.1.2.1/Q.732.7

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						ISUP reference: Q.1912.5 7.5.2.1.1.2.2 a)/Q.732.7	
TSS reference	ISDN-(ISUP)-SIP/SS/ECT							
SIP selection criteria								
ISUP selection criteria								
Test purpose	**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**				**ISUP reference: Q.1912.5 7.5.2.1.1.2.2 a)/Q.732.7**		
TSS reference	ISDN-(ISUP)-SIP/SS/ECT						
SIP selection criteria							
ISUP selection criteria							
Test purpose	**Call progress message with generic notification sent to the remote user** To verify that the local exchange (controlling the ECT) can successfully initiate a call transfer by sending **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	ISUP reference: Q.1912.5 7.5.2.1.1.2.2 b)/Q.732.7
TSS reference	ISDN-(ISUP)-SIP/SS/ECT	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**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**					**ISUP reference: Q.1912.5 7.5.2.1.1.2.2 b)/Q.732.7**
TSS reference	ISDN-(ISUP)-SIP/SS/ECT					
SIP selection criteria						
ISUP selection criteria						
Test purpose	**Capability of sending the additional connected number in the call transfer number parameter when the ECT is invoked while one call is alerting** To verify that, in case the ECT is invoked while one call is alerting, the **FAC** sent to the 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**				**ISUP reference: Q.1912.5 7.5.2.1.1.2.2 b)/Q.732.7**		
TSS reference	ISDN-(ISUP)-SIP/SS/ECT						
SIP selection criteria							
ISUP selection criteria							
Test purpose	**Capability of sending the 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						ISUP reference: Q.1912.5 7.3; 7.5.2.3.1/Q.732.7
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					ISUP reference: Q.1912.5 7.3; 7.5.2.4.1/Q.732.7	
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	ISUP reference: Q.1912.5 2.5.2.1.1/Q.732
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV	
SIP selection criteria		
ISUP selection criteria		
Test purpose	"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	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**				**ISUP reference:** **Q.1912.5** **2.5.2.1.1/Q.732**
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV				
SIP selection criteria					
ISUP selection criteria					
Test purpose	"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**				**ISUP reference:** **Q.1912.5**
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	→			

TP712004	**SIP reference: RFC 3261**	**ISUP reference: Q.1912.5 2.4.2; Table 2-1/Q.732**
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Redirection number - presentation restricted - according to the notification subscription option** To verify that the originating exchange does not make the redirection number available to the calling access signalling system, if the notification subscription option of the call diversion information is coded "001 presentation not allowed", "011 presentation allowed without redirection number" or "000 unknown"	
SIP parameter values	183 Session Progress encapsulated ACM call diversion information notification subscription option "presentation allowed without redirection number"	
ISDN parameter values	NOTIFY: notification indicator "call is diverted" ALERTING: no redirection number CONNECT: no 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	→			

TP712005	**SIP reference: RFC 3261**	**ISUP reference: Q.1912.5 2.4.2; Table 2-1/Q.732**
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Redirection number - presentation restricted - according to redirection number restriction parameter** To verify that the originating exchange does not make the redirection number available to the calling access signalling system, if the redirection number restriction parameter indicates "01 Presentation restricted" The notification subscription option of the call diversion information is coded "010 Presentation allowed with redirection number"	
SIP parameter values	183 Session Progress encapsulated ACM call diversion information notification subscription option "presentation allowed with redirection number" 200 OK INVITE: encapsulated ANM redirection number restriction "presentation restricted	
ISDN parameter values	CONNECT: no redirection number	
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	→			

TP712006	**SIP reference: RFC 3261**				**ISUP reference: Q.1912.5 2.4.2; Table 2-1/Q.732**
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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"				
SIP parameter values	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				
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><td>TP712007</td><td colspan="5">SIP reference: RFC 3261</td><td>ISUP reference:
Q.1912.5
2.4.2/Q.732</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 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><th>TP712008</th><th colspan="3">SIP reference: RFC 3261</th><th colspan="2">ISUP reference:
Q.1912.5
2.4.2/Q.732</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">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="5">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="5">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>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></td><td></td><td></td><td>←</td><td>183 Session Progress(CPG)</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>
<tr><td colspan="6">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**					**ISUP reference: Q.1912.5 2.5.2.5.1.1/Q.732**	
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV						
SIP selection criteria							
ISUP selection criteria							
Test purpose	**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**			**ISUP reference: Q.1912.5 2.5.2.5.1.1/Q.732**	
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Setting of redirection number 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**			**ISUP reference:** **Q.1912.5** **2.5.2.5.1.1/Q.732**	
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Setting 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	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**					**ISUP reference:** **Q.1912.5** **2.5.2.5.1.2 b) 2)/Q.732**	
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	**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**					**ISUP reference: Q.1912.5**	
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**	**ISUP reference: Q.1912.5 2.5.2.5.1.2 b) 5)/Q.732**
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**	**ISUP reference: Q.1912.5 2.5.2.5.1.2 c) ii); iii)/Q.732**
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Call diversion may occur in 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**				**ISUP reference: Q.1912.5 2.5.2.1.1.1; 2.5.2.1.1.2/Q.733**
TSS reference	ISDN-(ISUP)-SIP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call hold after answer, 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**				**ISUP reference: Q.1912.5 2.5.2.1.1.1; 2.5.2.1.1.2/Q.733**
TSS reference	ISDN-(ISUP)-SIP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call hold after answer, 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**				**ISUP reference: Q.1912.5 2.2.1; 2.5.2.1.1.1; 2.5.2.1.1.2/Q.733**
TSS reference	ISDN-(ISUP)-SIP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria	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**				**ISUP reference: Q.1912.5 2.2.1; 2.5.2.5.1/Q.733**
TSS reference	ISDN-(ISUP)-SIP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria	PICS 8/1				
Test purpose	**Call hold after alerting, 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				ISUP reference: Q.1912.5 2.3/Q.764
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				ISUP reference: Q.1912.5 2.3/Q.764
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				ISUP reference: Q.1912.5 2.3/Q.764
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				ISUP reference: Q.1912.5 2.3/Q.764
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**				**ISUP reference: Q.1912.5 1.5.2.1.1/Q.733**
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**				**ISUP reference: Q.1912.5 1.5.2.1.1/Q.733**
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**				**ISUP reference: Q.1912.5 1.5.2.5.1/Q.733**
TSS reference	ISDN-(ISUP)-SIP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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**				**ISUP reference: Q.1912.5 1.5.2.5.2/Q.733**
TSS reference	ISDN-(ISUP)-SIP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call waiting rejected** To verify that the IUT 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**					**ISUP reference: Q.1912.5 2.4; 2.2.1/Q.734.2**	
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	ISUP reference: Q.1912.5 2.5.2.1.1.3 a)/Q.734.2
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**	**ISUP reference: Q.1912.5 2.5.2.1.1.3 b)/Q.734.2**
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					ISUP reference: Q.1912.5 2.5.2.1.1.3/Q.734.2	
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**				**ISUP reference: Q.1912.5 2.2.1/Q.734.2**		
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	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**				**ISUP reference: Q.1912.5 2.4; 2.2.1/Q.734.2**
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	←			

History

Document history		
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