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Technical Specification

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

Reference

RTS/INT-00048

Keywords

BICC, ISUP, SIP, testing, TSS&TP

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Contents

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Foreword

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

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

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

Part 2: "Test Suite Structure and Test Purposes (TSS&TP)";

Part 3: "Abstract Test Suite (ATS) and partial Protocol Implementation eXtra Information for Testing (PIXIT)".

1 Scope

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

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

2 References

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

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

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

2.1 Normative references

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

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

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

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

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

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

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

[7] ISO/IEC 9646-3 (1998): "Information technology - Open Systems Interconnection - Conformance testing methodology and framework - Part 3: The Tree and Tabular Combined Notation (TTCN)".

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

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

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

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

[12] Void.

2.2 Informative references

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

[i.1] Void.

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

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

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

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

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

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

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

[i.9] Void.

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

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

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

[i.13] ITU-T Recommendation T.38: "Procedures for real-time Group 3 facsimile communication over IP networks".

[i.14] ITU-T Recommendation Q.1902.4: "Bearer independent call control protocol (Capability Set 2): Basic call procedures".

[i.15] IETF RFC 4040: "RTP Payload Format for a 64 kbit/s Transparent Call".

[i.16] Void.

3 Definitions and abbreviations

3.1 Definitions

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

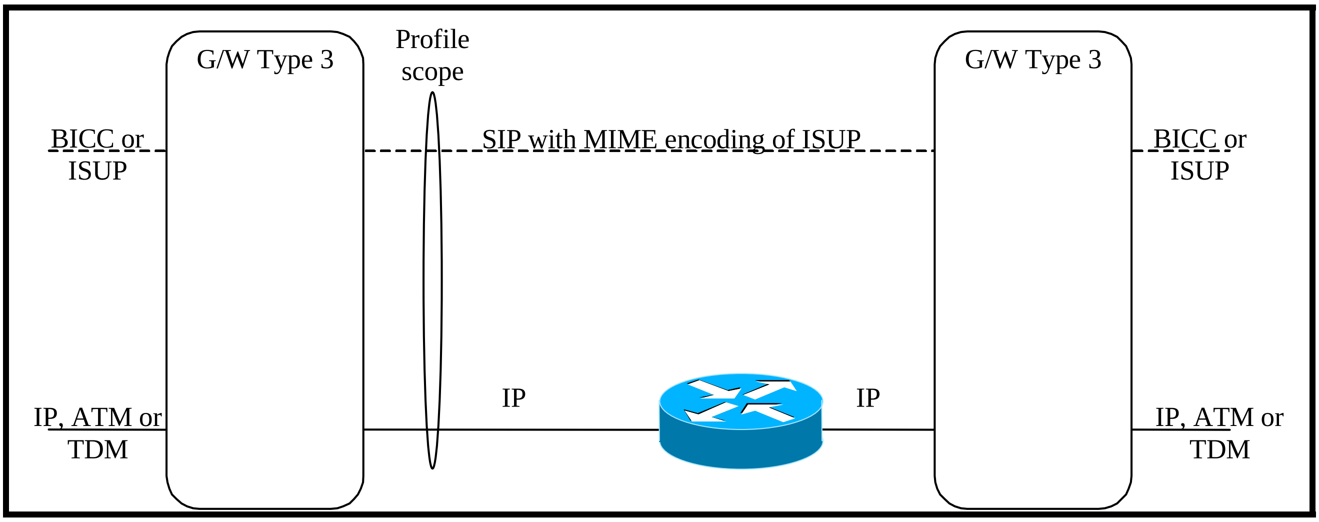

Figure 1: Profile scope for SIP with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateways

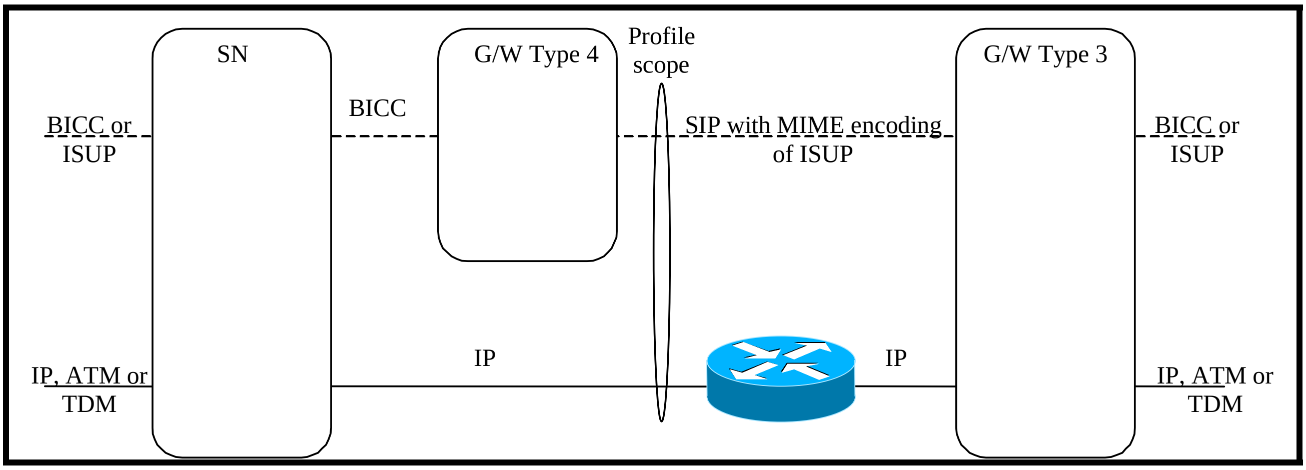

Figure 2: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 3 and 4 gateways

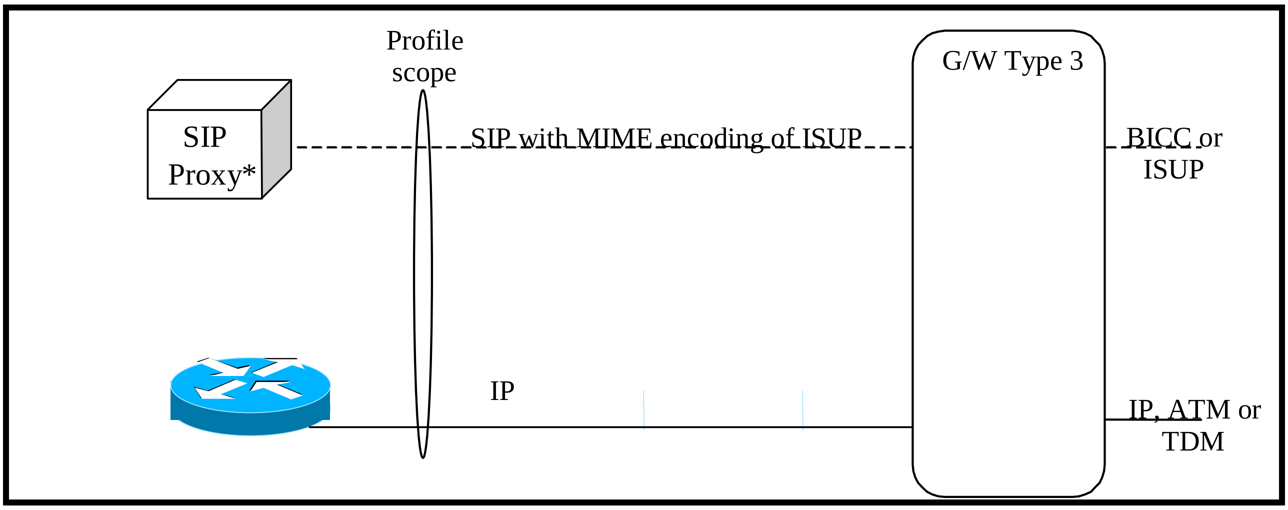

Figure 3: Profile scope for SIP with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateways

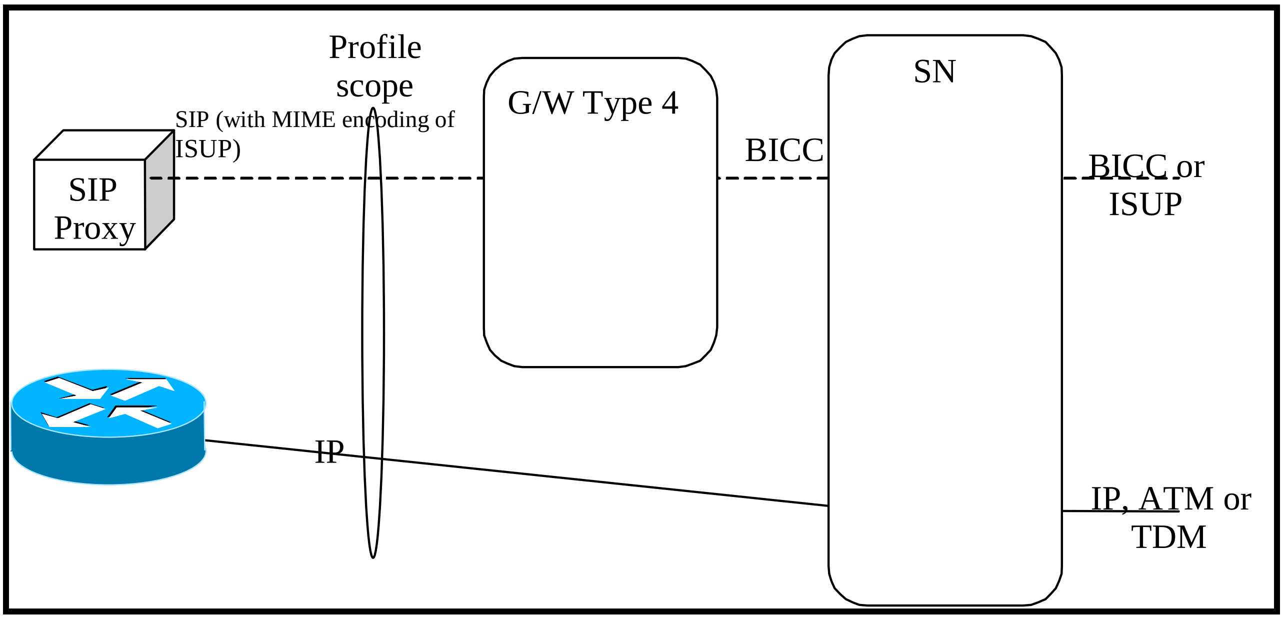

Figure 4: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 4 gateway

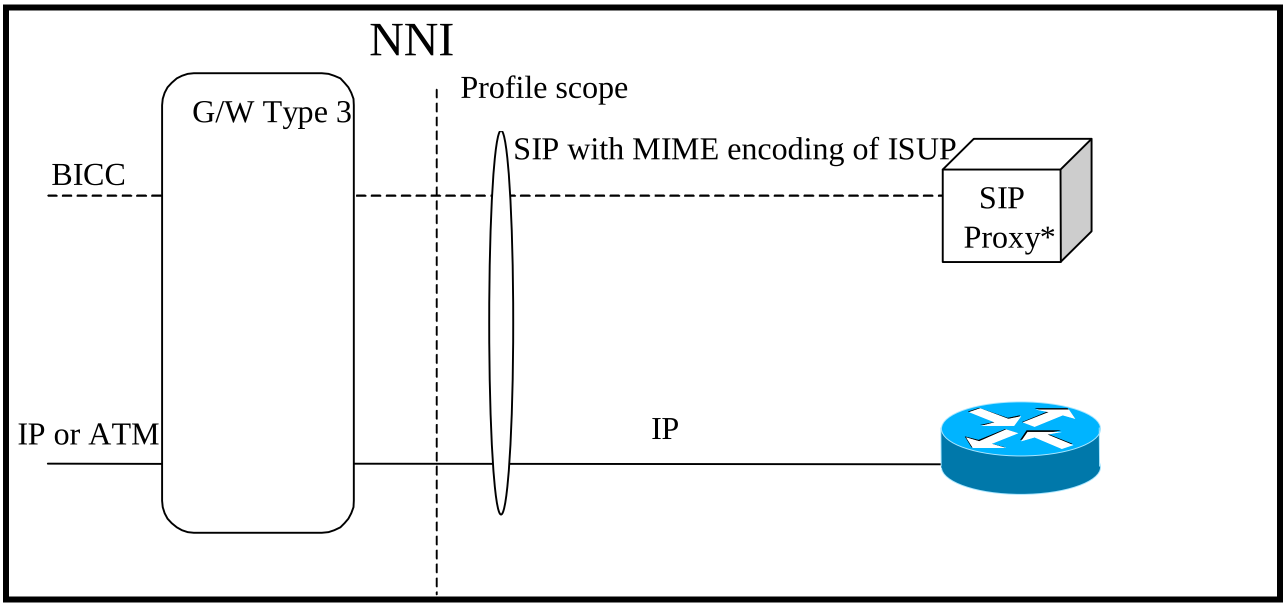

Figure 5: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateway

3.2 Abbreviations

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

3PTY	Three-Party
ACM	Address Complete Message
ANM	ANswer Message
ASP	Abstract Service Primitive
ATC	Abstract Test Case
ATM	Abstract Test Method
ATP	Access Transport Parameter
ATS	Abstract Test Suite
AVP	Attribute-Value Pairs
BC	Bearer Capability
BCI	Backward Call Indicators
BICC	Bearer Independent Call Control protocol
BLA	BLocking Acknowledgement message
BLO	BLOcking message
CC	Country Code
CCBS	Completion of Communication to Busy Subscriber
CD	Call Deflection
CDIV	Call DIVersion
CFB	Call Forwarding Busy
CFN	ConFusioN message
CFNR	Communications Forwarding No Reply
CFU	Call Forwarding Unconditional
CGB	Circuit Group Blocking
CGBA	Circuit Group Blocking Acknowledgement message
CGU	Circuit Group Unblocking message
CGUA	Circuit Group Unblocking Acknowledgement message
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
COL	COnnected Line
COLP	COnnected Line identification Presentation
COLR	COnnected Line identification Restriction
CON	CONnect message
CONF	CONFerence calling

COT	COnTinuity message
CPG	Call Progress Message
CPS	Calling Party's Category
CTNb	ConnecTed Number
CUG	Closed User Group
CW	Call Waiting
DSS1	Digital Subscriber System no. 1
ECT	Explicit Call Transfer
FAA	FAcility Accepted message
FAC	FACility message
FAR	FAcility Request message
FCI	Forward Call Indicators
FRJ	Facility ReJect message
GRA	circuit Group Reset Acknowledgement message
GRS	Group ReSet
HLC	High Layer Compatibility
HOLD	Call HOLD
IAM	Initial Address Message
IDR	IDentification Request message
I-IWU	Incoming InterWorking Unit
I-MGCF	Incoming Media Gateway Control Function
IRS	Identification ResponSe message
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
ITU	International Telecommunication Union
IUT	Implementation Under Test
LOP	LOop Prevention message
MCID	Malicious Call IDentification
MGCF	Media Gateway Control Function
MHS	Message Handling System
MIME	Multi-purpose Internet Mail Extension
MOT	Means Of Testing
NCI	Nature of Connection Indicators
NDC	National Destination Code
OBCI	Optional Backward Call Indicators
O-IWU	Outgoing InterWorking Unit
OLE	Originating Local Exchange
O-MGCF	Outgoing Media Gateway Control Function
OSI	Open Systems Interconnection
PCMA	Pulse Code Modulation A-law
PCMU	Pulse Code Modulation µ-law
PCO	Point of Control and Observation
PICS	Protocol Implementation Conformance Statement
PIXIT	Protocol Implementation eXtra Information for Testing
PT	Pay load Type
PTC	Parallel Test Component
REL	RELease message
RES	RESUME
RLC	ReLease Complete message
RSC	ReSet Circuit
RTP	Real Time Protocol
SAM	Subsequent Address Message
SDP	Session Description Protocol
SGM	SeGmentation Message
SIP	Session Initiation Protocol
SIP-I	Session Initiation Protocol with encapsulated ISUP
SN	Subscriber Number
SS	Supplementary Services
SUB	SUBaddressing
SUS	SUSPEND
SUT	System Under Test
TMR	Transmission Medium Requirement

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

4 Test Suite Structure (TSS)

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

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

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

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

4.3 Supplementary services supported by encapsulation

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

5 Test Purposes (TP)

5.1 Introduction

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

5.1.1 Test Purpose (TP) naming convention

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

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

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

5.1.2 Source of test purpose definition

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

5.1.3 Test purpose structure

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

5.2 Test purposes for the basic cal

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

5.2.1.1 Sending of the Initial Address Message (IAM)

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

TP101002	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 6.1.2 (i,2ai)**	
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection criteria	PICS 4/4 AND PICS 4/5				
ISUP selection criteria	PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, **with a SDP offer** 100rel extensions and preconditions extensions in the SIP Supported header: • the SUT shall delete µ-law (PCMU), if present, from the media description that it will send back in the SDP answer; • the IAM shall be sent out immediately on the BICC side with the coding of the Nature of Connection Indicators parameter: **"COT to be expected".**				
SIP parameter values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 8				
ISUP parameter values	IAM Continuity Indicator: **COT to be expected**, USI: A-law or absent COT Continuity Indicator: **continuity**				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
			Preconditions met		
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101003	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clause 6.1.2 (i,2ai)**		
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection criteria	PICS 4/4 AND PICS 4/5				
ISUP selection criteria	PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, **with a SDP offer** 100rel extensions and preconditions extensions in the SIP 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

TP101004	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clause 6.1.2 (i,2aii)**		
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)/				
SIP selection criteria	PICS 4/4 AND PICS 4/5				
ISUP selection criteria	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, **with a SDP offer** 100rel extensions and preconditions extensions in the SIP Supported header: • the SUT shall delete µ-law (PCMU), if present, from the media description that it will send back in the SDP answer; • the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "**continuity check required on this circuit**" or set to "**continuity check performed on previous circuit**".				
SIP parameter values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 8				
ISUP parameter values	IAM Continuity Indicator: **continuity check required on this circuit or continuity check performed on previous circuit**, USI: A-law or absent COT Continuity Indicator: **continuity check successful**				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
			Preconditions met		
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
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Q.1912.5 [1], 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" or set to "continuity check performed on previous circuit".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">SIP INVITE: Audio RTP/AVP 0 8
200 OK: Audio RTP/AVP 8</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM Continuity Indicator: continuity check required on this circuit or continuity check performed on previous circuit, USI: A-law or absent
COT Continuity Indicator: continuity check successful</td></tr>
<tr><td rowspan="14">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>COT</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Preconditions met</td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP101006	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clause 6.1.2 (i,2b)**		
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection criteria	PICS 4/4 AND PICS 4/5				
ISUP selection criteria	NOT PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, **with a SDP offer** 100rel extensions and preconditions extensions in the SIP Supported header: • the SUT shall delete µ-law (PCMU), if present, from the media description that it will send back in the SDP answer; • the IAM shall be deferred until all preconditions have been met.				
SIP parameter values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 8				
ISUP parameter values	IAM USI: A-law or absent				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	IAM
	200 OK UPDATE	←			
			Preconditions met		
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101007	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clause 6.1.2 (i,2b)**		
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection 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

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

<table>
<tr><th>TP101009</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 6.1.2 (i,2ai)</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4 AND PICS 1/6 AND PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer 100rel extensions and preconditions extensions in the SIP 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>TP101010</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 6.1.2 (i,2ai)</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4 AND PICS 1/6 AND PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer 100rel extensions and preconditions extensions in the SIP Require header:
• the SUT shall delete A-law (PCMA) if both A-law (PCMA) and µ-law (PCMU) were present in the offer of the media description, that it will send it back in the SDP answer;
• the IAM shall be sent out immediately on the BICC side with the coding of the Nature of Connection Indicators parameter: "COT to be expected".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">SIP INVITE: Audio RTP/AVP 0 8
200 OK: Audio RTP/AVP 0</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM USI: µ-law; Nature of Connection Indicators parameter: "COT to be expected" COT; Continuity Indicator: continuity</td></tr>
<tr><td rowspan="14">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>COT</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Preconditions met</td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP101011	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 6.1.2 (i,2aii)**
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection criteria	PICS 4/4 AND PICS 4/5				
ISUP selection criteria	PICS 1/5 AND PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, **with a SDP offer** 100rel extensions and preconditions extensions in the SIP Supported header: • the SUT shall delete A-law (PCMA) if both A-law (PCMA) and µ-law (PCMU) were present in the offer of the media description, that it will send it back in the SDP answer; • the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "**continuity check required on this circuit**" or is set to "**continuity check performed on previous circuit**".				
SIP parameter values	SIP INVITE: Audio RTP/AVP 0 8 200 OK: Audio RTP/AVP 0				
ISUP parameter values	IAM: USI: µ-law; Continuity check indicator "**continuity check required on this circuit**" or **continuity** check performed on previous circuit COT Continuity Indicator: **continuity check successful**				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
			Preconditions met		
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

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

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

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

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

TP101015A	**SIP reference: RFC 3261 [4]**				**ISUP reference: ES 283 027 [11], clause 7.2.3.1.2.5 TS 129 164 [10], clause 6.2.4.1.3.2**
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address message (IAM)				
SIP selection criteria	Based on table 1A				
ISUP selection criteria					
Test purpose	*Mapping of SDP into the TMR* Ensure that the SUT in the Idle state on receipt of an INVITE message containing the media description defined in table 1 with the "a =" "b =" and "m=" lines set to a_b_m_LINE_VALUE: and the media description does not match the TMR and USI value sends an IAM message, with the Transmission Medium Requirement (TMR) parameter set to **TMR_VALUE** derived from the media description. The Information transfer capability in the USI is set in parallel to the TMR value.				
SIP Parameter values	INVITE; a_b_m_LINE_VALUE				
ISUP Parameter values	IAM; TMR: ISUP_TMR				
Comments	**SIP**		SUT		**ISUP**
	INVITE (IAM)	→		→	IAM
	180 Ringing (ACM)	←		←	ACM
	Ringing tone				
	200 OK INVITE (ANM)	←		←	ANM
	ACK	→			
	Conversation				
	BYE (REL)	→		→	REL
	200 OK BYE (RLC)	←		←	RLC

Table 1A

Values for test purposes TP101015A						
	a_b_m_LINE_VALUE					
m= line				**b= line**	**a= line**	**TMR_VALUE**
test purposes	<media>	<transport>	<fmt-list>	<modifier>:<bandwidth-value>	rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>	TMR codes
VA_01	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMU/8000	*"3,1 KHz audio"*
VA_02	audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A	*"3,1 KHz audio"*
VA_03	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMA/8000	*"3,1 KHz audio"*
VA_04	audio	RTP/AVP	Dynamic PT	AS: 64 kbit/s	rtpmap:<dynamic-PT> CLEARMODE/8000 (see note 2)	*"64 kbit/s unrestricted"*
VA_05	image	udptl	t38	N/A or up to 64 kbit/s	Based on ITU-T T.38 [i.13]	*"3,1 KHz audio"*
VA_06	image	tcptl	t38	N/A or up to 64 kbit/s	Based on ITU-T T.38 [i.13]	*"3,1 KHz audio"*
VA_07	audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A	*"3,1 KHz audio"*
NOTE 1: In this table the codec G.711 is used only as an example. Other codecs are possible. NOTE 2: CLEARMODE is specified in RFC 4040 [i.15]. NOTE 3: If the b=line indicates a bandwidth greater than 64 kbit/s then the call may use compression techniques or reject the call with a 415 response indicating that only one media stream of 64 kbit/s is supported. NOTE 4: <bandwidth value> for <modifier> of AS is in units of kbit/s.						

P101016	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 6.1.3.5	
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection criteria	NOT PICS 4/4 and NOT PICS 4/5				
ISUP selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of an 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

TP101017	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 6.1.3.5	
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection 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 TP1010017	
VA_01	HLC_VALUE = Telephony USI= speech
VA_02	HLC_VALUE = Facsimile Group 2/3 USI= 3,1 kHz audio
VA_03	HLC_VALUE == Facsimile Group 4 Class I USI= Unrestricted digital information
VA_04	HLC_VALUE == Teletex service, basic and mixed mode of operation and facsimile service Group 4, Classes II and III USI= Unrestricted digital information
VA_05	HLC_VALUE == Teletex service, basic and processable mode of operation USI= Unrestricted digital information
VA_06	HLC_VALUE = Teletex service, basic mode of operation USI= Unrestricted digital information
VA_07	HLC_VALUE = Syntax based Videotex USI= Unrestricted digital information
VA_08	HLC_VALUE = International Videotex interworking via gateways or interworking units USI= Unrestricted digital information
VA_09	HLC_VALUE = Telex service USI= Unrestricted digital information
VA_10	HLC_VALUE = Message Handling Systems (MHS) USI= Unrestricted digital information
VA_11	HLC_VALUE = OSI application USI= Unrestricted digital information
VA_12	HLC_VALUE = Audio visual USI= Unrestricted digital information

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

TP101019	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 6.1.3.1
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection criteria	PICS 1/9 AND NOT PICS 4/4 and NOT PICS 4/5				
ISUP selection criteria	NOT PICS 1/7				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a Called party number contained in the user info component of the Request-URI. Send an IAM Message with the called party number coded as follows: • Nature of address indicator: Analyse the information contained in received URI with user=phone, and if it is in the format: **+CC NDC SN** where CC is the country code of the network in which the next hop terminates, then set Nature of Address indicator to "**National (significant) number**", remove "+CC" and use the remaining digits to fill the Address signals. • Internal Network Number Indicator: routing to internal network number not allowed. • **Numbering plan Indicator 001 ISDN (Telephony) numbering plan.** • Address Signals: **NDC SN.**				
SIP parameter values					
ISUP parameter values	IAM: Called party number				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

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

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

TP101022	**SIP reference: RFC 3261 [4]**		**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**		
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9				
ISUP selection criteria	PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for µ-Law and a-Law 100rel extensions and preconditions extensions in the SIP Supported header, **then independent from the received order of preference:** • the IAM shall be sent out immediately on the BICC side with the coding of the Nature of Connection Indicators parameter: "**COT to be expected**"; • the G.711 a-law codec shall be returned in the SDP answer as preferred codec.				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 0 8 Answer: m=audio 4712 RTP/AVP 8 0				
ISUP parameter values	IAM Continuity Indicator: **COT to be expected**, USI: A-law or absent COT Continuity Indicator: **continuity**				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP101023</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 1/9</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4 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 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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">Offer: m=audio 4711 RTP/AVP 0 8
Answer: m=audio 4712 RTP/AVP 8 0</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM Continuity Indicator: COT to be expected, USI: A-law or absent
COT Continuity Indicator: continuity</td></tr>
<tr><td rowspan="13">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>COT</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

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

TP101025	**SIP reference: RFC 3261 [4]**				**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9				
ISUP selection criteria	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer 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**" or "**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 required on this circuit or continuity check performed on previous circuit**, USI: A-law or absent COT Continuity Indicator: **continuity check successful**				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101026	**SIP reference: RFC 3261 [4]**				**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9				
ISUP selection criteria	PICS 1/5 AND NOT PICS 1/6 AND NOT PICS 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for µ-Law and a-Law 100rel extensions and preconditions extensions in the SIP 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

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

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

TP101029	SIP reference: RFC 3261 [4]		ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2		
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9				
ISUP selection criteria	PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no µ-Law 100rel extensions and preconditions extensions in the SIP Supported header, **then independent the normal offer answer procedures apply:** • the IAM shall be sent out immediately on the BICC side with the coding of the Nature of Connection Indicators parameter: "**COT to be expected**"; • the G.711 a-law codec shall be returned in the SDP answer.				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 8 Answer: m=audio 4711 RTP/AVP 8				
ISUP parameter values	IAM Continuity Indicator: **COT to be expected**, USI: A-law or absent COT Continuity Indicator: continuity				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
			Preconditions met		
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101030	SIP reference: RFC 3261 [4]		ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2		
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9				
ISUP selection criteria	PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no µ-Law 100rel extensions and preconditions extensions in the SIP Require header, **then independent the normal offer answer procedures apply:** • the IAM shall be sent out immediately on the BICC side with the coding of the Nature of Connection Indicators parameter: "**COT to be expected**"; • the G.711 a-law codec shall be returned in the SDP answer.				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 8 Answer: m=audio 4711 RTP/AVP 8				
ISUP parameter values	IAM Continuity Indicator: **COT to be expected**, USI: A-law or absent COT Continuity Indicator: **continuity**				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
			Preconditions met		
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

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

TP101032	**SIP reference: RFC 3261 [4]**		**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**		
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9				
ISUP selection criteria	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no µ-Law 100rel extensions and preconditions extensions in the SIP Require header, **then independent the normal offer answer procedures apply:** • the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "**continuity check required on this circuit**" or "**continuity check performed on previous circuit**"; • the G.711 a-law codec shall be returned in the SDP answer.				
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 or continuity check performed on previous circuit**, USI: A-law or absent COT Continuity Indicator: **continuity check successful**				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
			Preconditions met		
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP101033	**SIP reference: RFC 3261 [4]**		**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**		
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9				
ISUP selection criteria	NOT PICS 1/6 AND NOT PICS 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no µ-Law 100rel extensions and preconditions extensions in the SIP 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

TP101034	SIP reference: RFC 3261 [4]				ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9				
ISUP selection criteria	NOT PICS 1/6 AND NOT PICS 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no µ-Law 100rel extensions and preconditions extensions in the SIP Require header, **then independent the normal offer answer procedures apply:** • the IAM shall be deferred until all preconditions have been met; • the G.711 a-law codec shall be returned in the SDP answer.				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 8 Answer: m=audio 4711 RTP/AVP 8				
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	IAM
	200 OK UPDATE	←			
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

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

<table>
<tr><th>TP101036</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 1/9</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4 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 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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">Offer: m=audio 4711 RTP/AVP 0
m=audio 4712 RTP/AVP 8
Answer: m=audio 0 RTP/AVP 0</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM Continuity Indicator: COT to be expected, USI: A-law or absent
COT Continuity Indicator: continuity</td></tr>
<tr><td rowspan="14">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>COT</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Preconditions met</td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP101037</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 1/9</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4 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 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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">Offer: m=audio 4711 RTP/AVP 0
m=audio 4712 RTP/AVP 8
Answer: m=audio 0 RTP/AVP 0</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM Continuity Indicator: COT to be expected, USI: A-law or absent
COT Continuity Indicator: continuity</td></tr>
<tr><td rowspan="14">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>COT</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Preconditions met</td><td></td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

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

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

TP101040	**SIP reference: RFC 3261 [4]**			**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**	
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9				
ISUP selection criteria	NOT PICS 1/6 AND NOT PICS 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer m line **without a-law codec** 100rel extensions and preconditions extensions in the SIP Supported header: • the IAM shall be deferred until all preconditions have been met; • **the u-law codec shall be rejected.**				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 0 m=audio 4712 RTP/AVP 8 Answer: m=audio 0 RTP/AVP 0				
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	IAM
	200 OK UPDATE	←			
		Preconditions met			
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

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

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

TP101042	SIP reference: RFC 3261 [4]	ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)	
SIP selection criteria	NOT PICS 4/4 AND NOT PICS 4/5 AND PICS 1/9 AND PICS 4/19	
ISUP selection criteria	NOT PICS 1/6	
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer **with more than one media streams and based on operator policy then:** • **the call is refused with a 415 Unsupported media type response.**	
SIP parameter values	Offer: m=audio 4711 RTP/AVP 8 m= audio 4712 RTP/AVP 8	
ISUP parameter values		

Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	415 Unsupported media type	←			
	ACK	→			

TP101043	**SIP reference: RFC 3261 [4]**				**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND PICS 4/19				
ISUP selection criteria	NOT PICS 1/6				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer **with more than one media streams** 100rel extensions and preconditions extensions in the SIP Supported header **and based on operator policy then:** • **the call is refused with a 415 Unsupported media type response.**				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 8 m= audio 4712 RTP/AVP 8				
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	415 Unsupported media type	←			
	ACK	→			

TP101044	**SIP reference: RFC 3261 [4]**				**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND PICS 4/19				
ISUP selection criteria	NOT PICS 1/6				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer **with more than one media streams** 100rel extensions and preconditions extensions in the SIP Require header **and based on operator policy then:** • **the call is refused with a 415 Unsupported media type response.**				
SIP parameter values	Offer: m=audio 4711 RTP/AVP 8 m= audio 4712 RTP/AVP 8				
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	415 Unsupported media type	←			
	ACK	→			

<table>
<tr><td>TP101045</td><td colspan="5">SIP reference: RFC 3261 [4]</td><td>ISUP reference:
EN 383 001 [2], clause 6.1.3.5.2.2</td></tr>
<tr><td>TSS reference</td><td colspan="6">SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6">NOT PICS 4/4 AND NOT PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams and based on operator policy then:
• if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected;
• if the SDP offer contains several audio type media streams, the IWU shall only consider one, and reject the other streams.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">Offer: m=audio 4711 RTP/AVP 8
m= audio 4712 RTP/AVP 8
m= video 4713 RTP/AVP 31

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

<table>
<tr><th>TP101046</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="6">SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6">NOT PICS 4/4 AND NOT PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6">PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="6">Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams 100rel extensions and preconditions extensions in the SIP Supported header and based on operator policy then:
• the IAM shall be sent out immediately on the BICC side with the coding of the Nature of Connection Indicators parameter: "COT to be expected";
• if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected;
• if the SDP offer contains several audio type media streams, the IWU shall only consider one, and reject the other streams.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">Offer: m=audio 4711 RTP/AVP 8
m= audio 4712 RTP/AVP 8
m= video 4713 RTP/AVP 31

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

TP101047	**SIP reference: RFC 3261 [4]**	**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)	
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19	
ISUP selection criteria	PICS 1/4 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 **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.	
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: **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

TP101048	**SIP reference: RFC 3261 [4]**	**ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2**
TSS reference	SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)	
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19	
ISUP selection criteria	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1	
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer **with more than one media streams**100rel extensions and preconditions extensions in the SIP Supported header **and based on operator policy then:** • the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "**continuity check required on this circuit**" or "**continuity check performed on previous circuit**"; • if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected; • if the SDP offer contains several audio type media streams, the IWU shall only consider one, and reject the other streams.	
SIP parameter values	Offer: m=audio 4711 RTP/AVP 8 m= audio 4712 RTP/AVP 8 m= video 4713 RTP/AVP 31 Answer: m=audio 4711 RTP/AVP 8 m=audio 0 RTP/AVP 8 m=video 0 RTP/AVP 31	
ISUP parameter values	IAM Continuity Indicator: **continuity check required on this circuit or continuity check performed on previous circuit**, USI: A-law or absent COT Continuity Indicator: **continuity check successful**	
Comments	(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><th>TP101049</th><th>SIP reference: RFC 3261 [4]</th><th>ISUP reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="2">SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)</td></tr>
<tr><td>SIP selection criteria</td><td colspan="2">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="2">PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="2">Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams 100rel extensions and preconditions extensions in the SIP Require header and based on operator policy then:
• the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "continuity check required on this circuit" or "continuity check performed on previous circuit";
• if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected;
• if the SDP offer contains several audio type media streams, the IWU shall only consider one, and reject the other streams.</td></tr>
<tr><td>SIP parameter values</td><td colspan="2">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="2">IAM Continuity Indicator: continuity check required on this circuit or continuity check performed on previous circuit, USI: A-law or absent
COT Continuity Indicator: continuity check successful</td></tr>
<tr><td>Comments</td><td colspan="2">
<table>
<tr><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>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>
</td></tr>
</table>

<table>
<tr><th>TP101050</th><th>SIP reference: RFC 3261 [4]</th><th>ISUP reference:
EN 383 001 [2], clause 6.1.3.5.2.2</th></tr>
<tr><td>TSS reference</td><td colspan="2">SIP-ISUP/Basic call/Sending of the Initial Address Message (IAM)</td></tr>
<tr><td>SIP selection criteria</td><td colspan="2">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="2">NOT PICS 1/6 AND NOT PICS 4/1</td></tr>
<tr><td>Test purpose</td><td colspan="2">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="2">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="2"></td></tr>
<tr><td>Comments</td><td colspan="2">
<table>
<tr><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td></td><td></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>
</td></tr>
</table>

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

5.2.1.2 Sending of the Subsequent Address Message (SAM)

TP102001	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 6.2 a)**
TSS reference	SIP-ISUP/Basic call/Sending of the Subsequent Address Message (SAM)			
SIP selection criteria	PICS 3/4			
ISUP selection criteria	PICS 3/8			
Test purpose	Ensure that the SUT receives an INVITE with the same Call-ID and From tag as a previous INVITE which was associated with a BICC/ISUP call/bearer control instance currently existing on the BICC/ISUP side whereby the number of digits in the Request-URI is **greater** than the number of digits already accumulated for the call, sends a SAM and pass it to outgoing BICC/ISUP procedures. The SAM shall contain in its Subsequent Number parameter only the additional digits received in this Request-URI compared with the digits already accumulated for the call.			
SIP parameter values				
ISUP parameter values	SAM; **subsequent number** (PIXIT)			
Comments	SIP-I		SUT	ISUP
	INVITE	→	→	IAM
	INVITE	→	→	SAM
	INVITE	→	→	SAM
	180 Ringing	←	←	ACM
	200 OK INVITE	←	←	ANM
	ACK	→		
		Conversation		
	BYE(REL)	→	→	REL
	200 OK BYE(RLC)	←	←	RLC

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

5.2.1.3 Sending of COT

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

TP103002	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 6.3**	
TSS reference	SIP-ISUP/Basic call/COT				
SIP selection criteria	PICS 4/4 AND PICS 4/5				
ISUP selection criteria	PICS 1/5 AND PICS 4/1				
Test purpose	Ensure that the when the SUT determines that all the preconditions on the incoming SIP side have been met and any continuity procedures on the outgoing ISUP side have been successfully completed: • the I-IWU shall send the COT message where the Continuity Indicator in the COT message shall be set to "**Continuity check successful**".				
SIP parameter values					
ISUP parameter values	COT continuity indicator: Continuity check successful				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	183 Session Progress	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→		→	COT
	200 OK UPDATE	←			
			Preconditions met		
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

5.2.1.4 Receipt of the Address Complete Message (ACM)

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

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

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

5.2.1.5 Receipt of the Call progress message (CPG)

TP105001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 6.6**
TSS reference	SIP-ISUP/Basic call/Receipt of the Call progress message (CPG).				
SIP selection criteria					
ISUP selection criteria					
Test purpose	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="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 6.6</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/Receipt of the Call progress message (CPG).</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the event information parameter event indicator is set to "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="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 6.6</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/Receipt of the Call progress message (CPG).</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the event information parameter event indicator is set to "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 (Inband 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 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], 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, (see ITU-T Recommendation Q.1902.4 [i.14]) is successfully completed; and
• the I-IWU determines (using the procedures defined in RFC 3312 [5]) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable).
In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "notification not required", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312 [5]) that sufficient preconditions have been met for the session to proceed.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">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 [4]				ISUP reference: Q.1912.5 [1], clauses 6.4, 6.7
TSS reference	SIP-ISUP/Basic call/Receipt of the CONNECT message (CON).				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**SDP offer was received** in the initial INVITE. Ensure that the SUT, on receipt of an CON message: • sends a 200 OK INVITE; • the received CON is encapsulated in the 200 OK INVITE. The bearer path shall be connected in both directions when both of the following conditions are satisfied: • the BICC outgoing bearer set-up procedure, (see ITU-T Recommendation Q.1902.4 [i.14]) is successfully completed; and • the I-IWU determines (using the procedures defined in RFC 3312 [5]) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable). In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "*notification not required*", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312 [5]) that sufficient preconditions have been met for the session to proceed.				
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 [4]				ISUP reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-ISUP/Basic call/Receipt of the Release message (REL)				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, 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 to 47) (47 is class default)
VA_21	500 Server Internal Error	Cause Value No. 50 ("requested facility not subscribed")
VA_22	500 Server Internal Error (SIP-I only)	Cause Value No. 55 ("incoming calls barred within CUG")
VA_23	500 Server Internal Error	Cause Value No. 57 ("bearer capability not authorized")
VA_24	500 Server Internal Error	Cause Value No. 58 ("bearer capability not presently")
VA_25	500 Server Internal Error	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_26	500 Server Internal Error	Cause Value in the Class 100 (service or option not implemented Cause Value No. 65 79) (79 is class default)
VA_27	500 Server Internal Error	Cause Value No. 87 ("user not member of CUG")
VA_28	500 Server Internal Error	Cause Value No. 88 ("incompatible destination")
VA_29	500 Server Internal Error	Cause Value No. 90 ("Non-existent CUG")
VA_30	404 Not Found	Cause Value No. 91 ("invalid transit network selection")
VA_31	500 Server Internal Error	Cause Value No. 95 ("invalid message") (Class default)
VA_32	500 Server Internal Error	Cause Value No. 97 ("Message type non-existent or not implemented")
VA_33	500 Server Internal Error	Cause Value No. 99 ("information element/parameter non-existent or not implemented")
VA_34	480 Temporarily unavailable	Cause Value No. 102 ("recovery on timer expiry")
VA_35	500 Server Internal Error	Cause Value No. 103 ("Parameter non-existent or not implemented, pass on")
VA_36	500 Server Internal Error	Cause Value No. 110 ("Message with unrecognized Parameter, discarded")
VA_37	500 Server Internal Error	Cause Value No. 111 ("protocol error, unspecified") (Class default)
VA_38	480 Temporarily unavailable	Cause Value No. 127 ("interworking unspecified") (Class default)

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

Table 2

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

TP108003	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause 6.11.2
TSS reference	SIP-ISUP/Basic call/Receipt of the Release message (REL)	
SIP selection criteria		
ISUP selection criteria		
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message where the CPS indicator is set to "**subscriber free**", having sent a 180 Ringing message on receipt of an ISUP REL: • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA with the encapsulated REL message.	
SIP parameter values	SIP Statue-Code: SIP_FAILURE_VA (PIXIT)	
ISUP parameter values	REL; **cause value**: CV_ISUP (PIXIT)	
Comments		

SIP-I		SUT		ISUP
INVITE(IAM)	→		→	IAM
180 Ringing(ACM)	←		←	ACM
SIP_FAILURE_VA(REL)	←		←	REL
ACK	→		→	RLC

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

SIP-I		SUT		ISUP
INVITE(IAM)	→		→	IAM
183 Session Progress(ACM)	←		←	ACM(no indication)
180 Ringing(CPG)	←		←	CPG(ALERTING)
SIP_FAILURE_VA(REL)	←		←	REL
ACK	→		→	RLC

Table 3

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

<table>
<tr><th>TP108005</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clause 6.11.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/Receipt of the Release message (REL)</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, 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="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clause 6.11.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">SIP-ISUP/Basic call/Receipt of the Release message (REL)</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, 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 [4]**				**ISUP reference:** **Q.1912.5 [1], clause 6.11.2**
TSS reference	SIP-ISUP/Basic call/Receipt of the Release message (REL)				
SIP selection criteria	PICS 4/21				
ISUP selection criteria					
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, on receipt of an ISUP REL with cause value 23 the SUT shall: • the SUT immediately requests the redirection to the new destination according the ISUP/BICC procedures.				
SIP parameter values					
ISUP parameter values	REL; **cause value**: 23				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM(Destination 1)
				←	REL(new Destination)
				→	RLC
				→	IAM(Destination 2)
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	←		←	REL
	200 OK BYE(RLC)	→		→	RLC

5.2.1.9 Autonomous release at I-IWU

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

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

TP109003	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 6.11.3
TSS reference	SIP-ISUP/Basic call/Autonomous release at I-IWU				
SIP selection criteria					
ISUP selection criteria	PICS 3/4				
Test purpose	Ensure that the SUT on receipt of insufficient digits received in an INVITE messages: • sends a 484 Address Incomplete message.				
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	484 Address incomplete	←			
	ACK	→			

TP109004	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 6.11.3
TSS reference	SIP-ISUP/Basic call/Autonomous release at I-IWU				
SIP selection criteria	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 [4]				ISUP reference: Q.1912.5 [1], clause 6.11.3
TSS reference	SIP-ISUP/Basic call/Autonomous release at I-IWU				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT in congestion on receipt of INVITE message: • sends a 480 Temporarily Unavailable message.				
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→			
	480 Temporarily unavailable	←			
	ACK	→			

TP109006	SIP reference: RFC 3261 [4]		ISUP reference: Q.1912.5 [1], clause 6.11.3		
TSS reference	SIP-ISUP/Basic call/Autonomous 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(any unknown parameter)
	500 Server internal error(REL)	←		→	REL
	ACK	→		←	RLC

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

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

5.2.1.10 Receipt of the Release message BYE/CANCEL

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

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

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

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

TP111001	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses 6.11.4 and 5
TSS reference	SIP-ISUP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a RSC message sends: • a BYE message if the SUT has already received an ACK for the 200 OK INVITE message which had it sent.			
SIP parameter values				
ISUP parameter values				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
		Conversation			
	BYE(REL)	←		←	RSC
	200 OK BYE(RLC)	→		→	RLC

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

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

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

TP111005	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clauses 6.11.4 and 5**		
TSS reference:	SIP-ISUP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria:					
ISUP selection criteria:					
Test purpose:	Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a GRS message sends 200 OK INVITE if the SUT has not yet received an ACK for the 200 OK INVITE: • the SUT shall wait until it receives the ACK for the 200 OK INVITE before sending the BYE.				
SIP parameter values:					
ISUP parameter values:					
Comments:	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
				←	GRS
	ACK	→		→	GRA
	BYE	←			
	200 OK BYE	→			

TP111006	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 6.11.4
TSS reference	SIP-ISUP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT, when at least one backward ISUP message relating to the call has already been received on receipt of a CGB message, with the Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented", sends 200 OK INVITE if the SUT has not yet received an ACK for the 200 OK INVITE: • the SUT shall wait until it receives the ACK for the 200 OK INVITE before sending the BYE.				
SIP parameter values					
ISUP parameter values	Circuit Group Supervision Message Type Indicator "hardware failure oriented"				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
				←	CGB(hardware failure)
	ACK	→		→	CGBA
	BYE	←			
	200 OK BYE	→			

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

TP111008	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses 6.11.4 and 5
TSS reference	SIP-ISUP/Basic call/Receipt of 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

TP111009	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 6.11.4
TSS reference	SIP-ISUP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that the SUT, when at least one backward ISUP message relating to the call has already been received on receipt of a CGB message, with the Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented", sends: • a 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 [4]			ISUP reference: Q.1912.5 [1], clauses 6.11.4 and 5	
TSS reference	SIP-ISUP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving more than one INVITE sending an IAM message for each call association on receipt of a GRS message were the Range and Status Parameter value is bigger than *"1":* • the SUT shall send a BYE requests for each call association.				
SIP parameter values					
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM) 1	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
	INVITE(IAM) 2	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
	BYE 1	←		←	GRS
	200 OK BYE	→		→	GRA
	BYE 2	←			
	200 OK BYE	→			

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

5.2.1.12 Receipt of the Suspend message (SUS) network initiated

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

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

5.2.1.13 Receipt of the RESume message (RES) network initiated

TP113001	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause 6.10
TSS reference	SIP-ISUP/Basic call/	
SIP selection criteria		
ISUP selection criteria		
Test purpose	Ensure that the SUT, on receipt of a RESUME message containing the suspend/resume indicator set to "network initiated": • the RES is transferred in an INFO message.	
SIP parameter values		
ISUP parameter values	RES; **Suspend indicator**: network initiated	
Comments		

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

5.2.1.14 Receipt of Confusion message

TP114001	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause A.1.1.3
TSS reference	ISUP-SIP/ISUP Messages for special consideration/Confusion message	
SIP selection criteria		
ISUP selection criteria		
Test purpose	Ensure that the SUT after receiving the INVITE with encapsulated IAM that contains an unknown parameter, sending an IAM message as received encapsulated in the INVITE request. Ensure that when the succeeding node discards an unknown parameter and send back a Confusion message if indicated in the parameter compatibility information and the sending of a Confusion message is requested, the CFN message is transported through the SIP network encapsulated in the 183 Session Progress.	
SIP parameter values	180 Ringing containing an ACM with an unknown parameter	
ISUP parameter values	INFO with encapsulated CFN	
Comments		

SIP-I				**ISUP**
INVITE	→		→	IAM
183 Session Progress(CFN)	←		←	CFN
180 Ringing(ACM)	←		←	ACM
200 OK INVITE(ANM)	←		←	ANM
ACK	→			
		Communication		
BYE(REL)	→		→	REL
200 OK BYE(RLC)	←		←	RLC

5.2.1.15 Segmentation

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

5.2.2 Interworking from ISUP to SIP-I

5.2.2.1 Sending of the INVITE message

TP301001	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1 1 a)
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete **called party number** and the **sending complete** indication: • sends the INVITE message with the encapsulated IAM in the MIME body.				
SIP parameter values					
ISUP parameter values	IAM; **Called party number**: with sending complete indication				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP301002	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1 1 b)
TSS reference	**ISUP-SIP/Basic call/Sending of the INVITE message**				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan: • sends the INVITE message.				
SIP parameter values					
ISUP parameter values	**IAM; Called party number 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 [4]				ISUP reference: Q.1912.5 [1], 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 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.1 1 d)**
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the **complete called party number** where the end of address signalling is determined by the expiration timer T_{OIW1} after the receipt of the latest address message: • sends the INVITE message.			
SIP parameter values				
ISUP parameter values				

Comments	ISUP/BICC		SUT		SIP-I
	IAM	→			
			T_{OIW1} expiry		
				→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

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

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

TP301006	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1 A)
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria	NOT PICS 4/4 AND NOT PICS 4/5 AND NOT PICS 4/15				
ISUP selection criteria	PICS 1/5				
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message 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 [4]				ISUP reference: Q.1912.5 [1], clause 7.1 A)
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria	NOT PICS 4/4 AND NOT PICS 4/5 AND NOT PICS 4/15				
ISUP selection criteria	PICS 1/5				
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message 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)

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<tr><th>TP301008</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1 A)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/4 AND NOT PICS 4/5 AND NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/5</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in Idle state, on receipt of an IAM message 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>
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<tr><th>TP301009</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1 A)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/4 AND NOT PICS 4/5 AND NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/5</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in Idle state, on receipt of an IAM message 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>

TP301010	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1 B)
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 4/15				
ISUP selection criteria	PICS 1/5 AND PICS 4/2				
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message where the Continuity Check indicator in the Nature of Connection Indicators parameter in the IAM is set to indicate **"continuity check not required"**: • sends an INVITE message without precondition using the SDP offer in the INVITE.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

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

TP301012	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.1 B)**
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message			
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 4/15			
ISUP selection criteria	PICS 1/5 AND PICS 4/2			
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message where the Continuity Check indicator in the Nature of Connection Indicators parameter in the IAM is set to indicate **"continuity check 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 [4]**	**ISUP reference: Q.1912.5 [1], clause 7.1 B)**
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 4/15	
ISUP selection criteria	PICS 1/5 AND PICS 4/2	
Test purpose	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

TP301014	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.1 B)**	
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 4/15				
ISUP selection criteria	PICS 1/5 AND PICS 4/2				
Test purpose	The SUT in Idle state, receives an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to **"continuity check required on this circuit"** and sends an INVITE message with precondition using the SDP offer in the INVITE. The ISUP Timer **T8 expires**. The call has been cleared **before** an early dialogue has been established. Ensure that the SUT: • sends CANCEL on the SIP side.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	100 Trying
			T8 expires		
	REL(#47)	←		→	CANCEL
	RLC	→		←	200 OK CANCEL
				←	487 Request Terminated
				→	ACK

TP301015	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.1 C)**	
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria	NOT PICS 4/15				
ISUP selection criteria	PICS 1/4				
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "**COT to be expected**": • The sending of the INVITE is delayed until all the following conditions are satisfied: o Continuity message, with the Continuity Indicators parameter set to "**continuity**" shall be received; o Bearer Set-up indication - for the forward bearer set-up case where the incoming Connect Type is "notification not required" was received.				
SIP parameter values					
ISUP parameter values					
Comments	BICC		SUT		SIP-I
	IAM	→			
	COT(successful)	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP301016</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1 C)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected":
• The sending of the INVITE is delayed until all the following conditions are satisfied:
o Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received;
o 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>TP301017</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1 C)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected":
• The sending of the INVITE delays until all the following conditions are satisfied:
o Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received;
o Bearer Set-up Connect indication - for the backward bearer set-up case was received.</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>TP301018</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], 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:
o Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received;
o 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>TP301019</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1 C)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected":
• 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="6">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>
</table>

<table>
<tr><th>TP301020</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1 D)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4 AND PICS 4/2</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected" sends an INVITE message with precondition using the SDP offer in the INVITE. The precondition signalling is concluded upon sending the (within an SDP offer-answer exchange) confirmation of a precondition being met. The SDP offer or answer carrying the confirmation of a precondition being met is sent when all of the following conditions are satisfied when:
• Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received;
• Bearer Set-up indication - for the forward bearer set-up case where the incoming Connect Type is "notification not required" was received.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="14">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>183 Session Progress</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK PRACK</td></tr>
<tr><td>COT(successful)</td><td>→</td><td></td><td>→</td><td>UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK UPDATE</td></tr>
<tr><td></td><td colspan="3">Preconditions met</td><td></td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP301021</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.1 D) 2.2</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 4/4 AND PICS 4/5 AND PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4 AND PICS 4/2</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected" sends an INVITE message with precondition using the SDP offer in the INVITE. The precondition signalling is concluded upon sending the (within an SDP offer-answer exchange) confirmation of a precondition being met. The SDP offer or answer carrying the confirmation of a precondition being met is sent when all of the following conditions are satisfied when:
• Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received;
• APM with Action indicator set to "Connected" - for the forward bearer set-up cases (with, or without bearer control tunnelling) where the incoming Connect Type is "notification required", and for the fast set-up (backward) case.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="14">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>183 Session Progress</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK PRACK</td></tr>
<tr><td>COT(successful)</td><td>→</td><td></td><td>→</td><td>UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK UPDATE</td></tr>
<tr><td></td><td colspan="3">Preconditions met</td><td></td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

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

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

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

TP301024	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause 7.1 D)
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS 4/15	
ISUP selection criteria	PICS 1/4 AND PICS 4/2	
Test purpose	The SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "**COT to be expected**", sends an INVITE message with precondition using the SDP offer in the INVITE: • ensure that the SUT sends CANCEL if the ISUP timer **T8 expires** if the call has been cleared **before** an early dialogue has been established.	
SIP parameter values		
ISUP parameter values		
Comments		

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

<table>
<tr><th>TP301025</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], 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 1/4 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 containing Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "COT to be expected". Ensure that the SUT:
sends CANCEL if on the SIP side the internal resource reservation was unsuccessful and if the call has been cleared before an early dialogue with the message has been established;
a REL with Cause Value 47 (resource unavailable, unspecified) shall be sent on the ISUP side of the O-IWU.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="8">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>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>TP301026</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], 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>TP301027</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], 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 5: Void

Table 6

Values for test purposes TP301026						
	ISUP	**SDP - a_b_m_LINE_VALUE**				
	TMR parameter	m= line			b= line	a= line
	TMR codes	<media>	<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 TP301027									
VA	**ISUP**				**SDP - a_b_m_LINE_VALUE**				
		USI parameter		HLC IE in ATP	m= line			b= line	a= line
	TMR	Information Transfer Capability	User Information Layer 1 Protocol Indicator	High Layer Characteristics Identification	<media>	<transport>	<fmt-list>	<modifier>:<bandwidth-value>	rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>
VA_01	"speech"	"Speech"	"G.711 µ-law"	Ignore	audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)
VA_02	"speech"	"Speech"	"G.711 A-law"	Ignore	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000
VA_03	"3,1 KHz audio"	USI Absent		Ignore	audio	RTP/AVP	0 and/or 8	AS:64	rtpmap:0 PCMU/8000 and/or rtpmap:8 PCMA/8000
VA_04	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 µ-law"		audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)
VA_05	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 A-law"		audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000
VA_06	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 A-law"	"Facsimile Group 2/3"	image	tcptl	t38	AS:64	Based on T.38.
VA_07	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 A-law"	"Facsimile Group 2/3"	image	udptl	t38	AS:64	Based on T.38.
VA_08	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 µ-law"	"Facsimile Group 2/3"	image	udptl	t38	AS:64	Based on T.38.
VA_09	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 µ-law"	"Facsimile Group 2/3"	image	tcptl	t38	AS:64	Based on T.38.
VA_10	"64 kbit/s unrestricted"	"Unrestricted digital inf. W/tone/ann."	N/A	Ignore	audio	RTP/AVP	9	AS:64	Rtpmap:9 G722/8000
VA_11	"64 kbit/s unrestricted"	"Unrestricted digital information"	N/A	Ignore	Audio	RTP/AVP	Dynamic PT	AS:64	rtpmap:<dynamic-PT> CLEARMODE/8000

TP301027A	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1.2 TS 129 164 [10], clause 6.2.4.2
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT is changing the received User information layer 1 protocol contained in the USI from A-law to µ-law if transcoding is performed from A-law to µ-law in the outgoing IWU				
SIP parameter values	INVITE: IAM USI User information layer 1 protocol=µ-Law				
ISUP parameter values	IAM: USI User information layer 1 protocol=A-Law				
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)

TP301027B	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1.2 TS 129 164 [10], clause 6.2.4.2
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT is changing the received User information layer 1 protocol contained in the USI from µ -law to A -law if transcoding is performed from µ -law to A -law in the outgoing IWU				
SIP parameter values	INVITE: IAM USI User information layer 1 protocol= A-Law				
ISUP parameter values	IAM: USI User information layer 1 protocol= µ-Law				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP301028	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1.2
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter of the IAM: • to the addr-spec component of the **To header field** in the INVITE message.				
SIP parameter values	INVITE: To: …				
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

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

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

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

TP301032	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.1.4**
TSS reference	ISUP-SIP/Basic call/Sending of the Initial Address Message (IAM)				
SIP selection criteria					
ISUP selection criteria	PICS 4/3				
Test purpose	The O-IWU acting as an independent exchange shall perform the normal BICC/ISUP Hop Counter procedure as it constructs the outgoing encapsulated IAM.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP301033	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.1.2**
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria	PICS 1/9				
ISUP selection criteria	PICS 1/8				
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, **Nature of address = "International number"** of the IAM to the addr-spec component of the **To header field** in the INVITE message. The format of the To header field is "+CC+NDC+SN": • the forward address information is derived from the user info component of the INVITE Request-URI.				
SIP parameter values	INVITE: To: …				
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP301034	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1.2
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria	PICS 1/9				
ISUP selection criteria	NOT PICS 1/8				
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, **Nature of address = "National (significant) number"** of the IAM: • to the addr-spec component of the **To header field** in the INVITE message; • the format of the To header field is "+CC+NDC+SN"; • the forward address information is derived from the user info component of the INVITE Request-URI.				
SIP parameter values	INVITE: To: URI, tag, parameters				
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP301035	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.1.2
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria	PICS 1/9				
ISUP selection criteria	PICS 1/8				
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, **Nature of address = "International number"** of the IAM and the and the followed SAM: • to the addr-spec component of the **To header field**; • the format of the To header field is "+CC+NDC+SN"; • the forward address information is derived from the user info component of the INVITE Request-URI.				
SIP parameter values	INVITE: To: URI, tag, parameters				
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→			
	SAM	→			
	SAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP301036	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.1.2	
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message				
SIP selection criteria	PICS 1/9				
ISUP selection criteria	NOT PICS 1/8				
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, **Nature of address = "National (significant) number"** of the IAM and the followed SAM: • to the addr-spec component of the **To header field**; • the format of the To header field is "+CC+NDC+SN"; • the forward address information is derived from the user info component of the INVITE Request-URI.				
SIP parameter values	INVITE: To:				
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→			
	SAM	→			
	SAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

5.2.2.2 Receipt of the SAM message after INVITE has been send

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

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

<table>
<tr><th>TP302003</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Receipt of SAM after invite has been sent</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 3/2 AND NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/5 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="13">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></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>INVITE1(IAM)</td></tr>
<tr><td>SAM</td><td>→</td><td></td><td>→</td><td>INVITE2(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>484 Address Incomplete (1)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing (2) (ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE (2) (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>TP302004</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Receipt of SAM after invite has been sent</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 3/2 AND NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/5 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 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.</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="13">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></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 1 (IAM)</td></tr>
<tr><td>SAM</td><td>→</td><td></td><td>→</td><td>INVITE 2 (IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>484 Address Incomplete (1)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing (2) (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>

TP302005	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.2.1
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent				
SIP selection criteria	PICS 3/2 AND NOT PICS 4/15				
ISUP selection criteria	PICS 1/5 AND PICS 4/2				
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "**continuity check required on this circuit**" sending of INVITE is delayed. INVITE message shall not be sent after the Continuity message was received with the Continuity Indicators parameter set to "**continuity check failed**". On receipt of a SAM from the ISUP the SUT shall: 1) Stop timer TOIW3 (if it is running); 2) TOIW2 shall be restarted.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→			
	SAM	→			
	COT	→			

TP302006	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.2.1
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent				
SIP selection criteria	PICS 3/2 AND NOT PICS 4/15				
ISUP selection criteria	PICS 1/5 AND PICS 4/2				
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "**continuity check required on this circuit**" sending of INVITE is delayed. INVITE shall not be sent after the ISUP timer T8 expires. On receipt of a SAM from the ISUP the SUT shall: 1) Stop timer TOIW3 (if it is running); 2) TOIW2 shall be restarted.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→			
	SAM	→			
		T8 expires			
	REL	←			
	RLC	→			

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

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

TP302008	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause 7.2.1
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria	PICS 3/2 AND PICS 4/5 AND PICS 4/15	
ISUP selection criteria	PICS 1/5 AND PICS 4/2	
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set 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.	
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 an 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	→		→	INVITE2 (IAM and digits from SAM X + SAM Y)
			←	484 Address Incomplete (1)
			→	ACK
ACM	←		←	180 Ringing2 (ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP302009</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Receipt of SAM after invite has been sent</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 3/2 AND NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4 AND NOT PICS 4/2</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected".
The sending of the INVITE is delayed until all the following conditions are satisfied:
• Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received;
• Bearer Set-up indication - for the forward bearer set-up case where the incoming Connect Type is "notification not required" was received.
On receipt of a SAM from the BICC the SUT shall:
1) Stop timer TOIW3 (if it is running);
2) TOIW2 shall be restarted and the SUT shall invoke the following procedures:
a) the Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call;
b) a new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent;
c) the new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question;
d) all other contents of the new INVITE are interworked from the parameters of the original IAM.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="11">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>SAM x</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>COT</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>SAM y</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP302010	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause 7.2.1
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria	PICS 3/2 AND NOT PICS 4/15	
ISUP selection criteria	PICS 1/4 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 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.	
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>TP302011</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Receipt of SAM after invite has been sent</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 3/2 AND NOT PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/4 AND PICS 4/2</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected".
The sending of the INVITE delays until all the following conditions are satisfied:
• Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received;
• Bearer Set-up Connect indication - for the backward bearer set-up case was received.
On receipt of a SAM from the BICC the SUT shall:
1) Stop timer TOIW3 (if it is running);
2) TOIW2 shall be restarted and the SUT shall invoke the following procedures:
a) the Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call;
b) a new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent;
c) the new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question;
d) all other contents of the new INVITE are interworked from the parameters of the original IAM.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5"></td></tr>
<tr><td rowspan="11">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>SAM x</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>COT</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>SAM y</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP302012	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clause 7.2.1**
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria	PICS 3/2 AND NOT PICS 4/15	
ISUP selection criteria	PICS 1/4 AND PICS 4/2	
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "**COT to be expected**". The sending of the INVITE delays until all the following conditions are satisfied: • Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received; • BNC set-up success indication for cases using bearer control tunnelling was received On receipt of a SAM from the BICC the SUT shall: 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)

TP302013	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause 7.2.1
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria	PICS 3/2 AND PICS 4/5 AND PICS 4/15	
ISUP selection criteria	PICS 1/4 AND PICS 4/2	
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "**COT to be expected**". Sends the INVITE message. The events: • Continuity message, with the Continuity Indicators parameter set to "**continuity**" was received; • Bearer Set-up indication - for the forward bearer set-up case where the incoming Connect Type is "**notification not required**" was received; are indicating the successful completion of bearer set-up. On receipt of a SAM from the BICC the SUT shall: 1) Stop timer TOIW3 (if it is running); 2) TOIW2 shall be restarted and the SUT shall invoke the following procedures: a) the Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call; b) a new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent; c) the new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question; d) all other contents of the new INVITE are interworked from the parameters of the original IAM.	
SIP parameter values	INVITE2: Request URI contains digits from the IAM and digits from SAM x and SAM y. The IAM is also contained	
ISUP parameter values		
Comments	The O-IWU should initiate the precondition signalling procedure using the SDP Offer in the INVITE. The precondition signalling is concluded upon sending (within an SDP offer-answer exchange) the confirmation of a precondition being met. The SDP Offer or Answer carrying the confirmation of a precondition being met is sent when the conditions to send a INVITE message are satisfied.	

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

<table>
<tr><th>TP302014</th><th>SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clause 7.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="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 PICS 4/5 AND 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".
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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="2">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="2"></td></tr>
<tr><td>Comments</td><td colspan="2">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.
<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>INVITE1(IAM)</td></tr>
<tr><td>SAM</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 with digits from SAM X + SAM Y)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>484 Address Incomplete (1)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing2(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table></td></tr>
</table>

<table>
<tr><th>TP302015</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clause 7.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP/Basic call/Receipt of SAM after invite has been sent</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6">PICS 3/2 AND PICS 4/5 AND PICS 4/15</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6">PICS 1/4 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".
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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">INVITE2: Request URI contains digits from the IAM and digits from SAM x and SAM y. The IAM is also contained</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6"></td></tr>
<tr><td rowspan="17">Comments</td><td colspan="6">The O-IWU should initiate the precondition signalling procedure using the SDP Offer in the INVITE. The precondition signalling is concluded upon sending (within an SDP offer-answer exchange) the confirmation of a precondition being met. The SDP Offer or Answer carrying the confirmation of a precondition being met is sent when the conditions to send a INVITE message are satisfied.</td></tr>
<tr><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td colspan="2">SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE1(IAM)</td></tr>
<tr><td>SAM</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">183 Session Progress without encapsulated ACM</td></tr>
<tr><td>COT</td><td>→</td><td></td><td>→</td><td colspan="2">UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK UPDATE</td></tr>
<tr><td>SAM</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE2 (IAM with digits from SAM X + SAM Y)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">484 Address Incomplete (1)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing2(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="2"></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td colspan="2">BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK BYE(RLC)</td></tr>
</table>

<table>
<tr><th>TP302016</th><th>SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clause 7.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="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 PICS 4/5 AND 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 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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="2">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="2"></td></tr>
<tr><td>Comments</td><td colspan="2">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>
</table>

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

TP302017	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.2.1
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent				
SIP selection criteria	PICS 3/2				
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 [4]				ISUP reference: Q.1912.5 [1], clause 7.2.1
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent				
SIP selection criteria	PICS 3/1				
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 [4]**	**ISUP reference: Q.1912.5 [1], clause 7.1, Q.764 [i.12], clause 2.1.4.8**
TSS reference	ISUP-SIP/Basic call/Sending of the ACM message	
SIP selection 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)

<table>
<tr><th>TP303002</th><th colspan="4">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clause 7.1,
Q.764 [i.12], clause 2.1.4.8</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Sending of the ACM message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 1/3</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 4/9</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan:
• sends the INVITE message to called user;
• sends the ACM message with;
• the CPS indicator set to "no indication (00)";
• the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)";
• the interworking indicator set to "INT_IND_VAL";
• the ISUP indicator set to "ISUP_IND_ID";
• the ISDN access indicator set to "ISDN_ACC_IND_VAL".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM; Called party number: complete number
ACM, Backward call indicator is set to the value in the encapsulated ACMCPS indicator no indication (00)
Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)
interworking indicator: INT_IND_VAL (PIXIT)
ISUP indicator: ISUP_IND_ID (PIXIT)
ISDN access indicator ISDN_ACC_IND_VAL (PIXIT)</td></tr>
<tr><td rowspan="9">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM(no indication)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CPG(Alerting)</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

TP303003	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause 7.1, Q.764 [i.12], clause 2.1.4.8
TSS reference	ISUP-SIP/Basic call/Sending of the ACM 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** where the end of address signalling is determined by analysis of the called party number to indicate that a **sufficient number of digits has been received to route the call to the called party:** • sends the INVITE message to called user; • sends the ACM message with the **CPS indicator** set to "no indication (00)", the **Called party's category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the **interworking indicator** set to "INT_IND_VAL", the **ISUP indicator** set to "ISUP_IND_ID", the **ISDN access indicator** set to "ISDN_ACC_IND_VAL".	
SIP parameter values		
ISUP parameter values	IAM; **Called party number**: complete number ACM, **CPS indicator** no indication (00) **Called party's category indicator:** no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator:** INT_IND_VAL (PIXIT) **ISUP indicator:** ISUP_IND_ID (PIXIT) **ISDN access indicator** ISDN_ACC_IND_VAL (PIXIT)	

Comments	ISUP/BICC		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)

TP303004	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clauses 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)

TP303005	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clauses 7.1, 7.3.1**
TSS reference	ISUP-SIP/Basic call/Sending of the ACM message	
SIP selection criteria	PICS 1/3	
ISUP selection criteria	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): • 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".	
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	→			
	SAM	→			
	SAM	→		→	INVITE(IAM)
			T_{OIW2} expiry		
	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>TP303006</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses 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 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses 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 ACM 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><th>TP303011</th><th>SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clauses 7.1, 7.3.1, 7.4</th></tr>
<tr><td>TSS reference</td><td colspan="2">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="2">PICS 1/3</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="2">PICS 4/2 AND NOT PICS 4/9</td></tr>
<tr><td>Test purpose</td><td colspan="2">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="2"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="2">IAM; Called party number: complete number
ACM,
CPS indicator no indication (00)
Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)interworking indicator: INT_IND_VAL (PIXIT)
ISUP indicator: ISUP_IND_ID (PIXIT)
ISDN access indicator ISDN_ACC_IND_VAL (PIXIT)</td></tr>
<tr><td>Comments</td><td colspan="2">
<table>
<tr><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>183 Session Progress without encapsulated ACM</td></tr>
<tr><td>COT</td><td>→</td><td></td><td>→</td><td>UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK UPDATE</td></tr>
<tr><td></td><td colspan="3">T_{OIW1} expiry</td><td></td></tr>
<tr><td>ACM(no indication)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CPG(Alerting, BCi)</td><td>←</td><td></td><td>←</td><td>180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>
</td></tr>
</table>

<table>
<tr><th>TP303012</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clauses 7.1, 7.3.1, 7.4</th></tr>
<tr><td>TSS reference</td><td colspan="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 PICS 4/5 AND PICS 4/4 AND PICS 4/15</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 if overlap addressing is to be used toward the SIP network, on receipt of an IAM message containing the minimum number of digits required for routing the call has been received (start timer TOIW2 and invoke the appropriate outgoing SIP signalling procedure) and the continuity check is performed (ISUP) or COT is expected (BICC). After the expiry of T_{oiw2}:
• sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "INT_IND_VAL", the ISUP indicator set to "ISUP_IND_ID", the ISDN access indicator set to "ISDN_ACC_IND_VAL".</td></tr>
<tr><td>SIP parameter values</td><td colspan="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="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_{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, 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>

<table>
<tr><th>TP303013</th><th colspan="4">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clauses 7.1, 7.3.1, 7.4</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Sending of the 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 4/2 AND NOT PICS 4/9</td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number, the continuity check is performed (ISUP) or COT is expected (BICC) indication receipt of a 180 Ringing message:
• Sends the ACM message with:
o the CPS indicator set to the value in the encapsulated ACM;
o the Called party's category indicator set to the value in the encapsulated ACM;
o the interworking indicator set to the value in the encapsulated ACM;
o the ISUP indicator set to the value in the encapsulated ACM;
o the ISDN access indicator set to the value in the encapsulated ACM.</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="12">Comments</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td></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>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>

TP303014	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clauses 7.1, 7.3.1, 7.4
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria	PICS 1/3 AND PICS 3/2 AND NOT PICS 4/15	
ISUP selection criteria	PICS 3/8 AND 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 T_{OIW2} and invoke the appropriate outgoing SIP signalling procedure) and the continuity check is performed (ISUP) or COT is expected (BICC). After the expiry of T_{oiw2}: • sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "INT_IND_VAL", the ISUP indicator set to "ISUP_IND_ID", the ISDN access indicator set to "ISDN_ACC_IND_VAL".	
SIP parameter values		
ISUP parameter 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	→			
COT	→		→	INVITE(IAM)
		T_{OIW2} expiry		
ACM(no indication)	←			
CPG(Alerting)	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
		Conversation		
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

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

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

5.2.2.4 Sending of the CPG message

TP304001	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clauses 7.1, 7.3.1**
TSS reference	ISUP-SIP/Basic call/Sending of the CPG message	
SIP selection criteria	PICS 3/1	
ISUP selection criteria	PICS 3/8	
Test purpose	Ensure that the SUT, having sent a ACM message with called party status "no indication" on receipt of a 180 Ringing with a encapsulated ISUP message: • sends the CPG message with the **event indicator** set to "Alerting".	
SIP parameter values		
ISUP parameter values	ACM: BCi called party status indicator = no indication CPG: Event Indicator = ALERTING, BCi as received from the encapsulated ACM	

Comments	ISUP/BICC		SUT		SIP-I
	IAM	→			
	SAM	→			
	SAM	→		→	INVITE(IAM)
			T_{OIW2} expiry		
	ACM(no indication)	←			
	CPG(Alerting BCi)	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

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

5.2.2.5 Sending of the ANM message

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

5.2.2.6 Sending of the CON message

TP306001	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 7.5, 7.5.1
TSS reference	ISUP-SIP/Basic call/Sending of the Connect Message (CON)				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT, having not sent the ACM message, on receipt of a 200 OK INVITE for this call, it shall stop timer TOIW2 (if running): • send CON as determined by BICC/ISUP procedures. Stop any existing awaiting answer indication (e.g. ringing tone) BCI encoded as received in the encapsulated CON.				
SIP parameter values	200 OK INVITE;				
ISUP parameter values	CON; **interworking indicator**: INT_IND_VAL (PIXIT) **ISUP indicator**: ISUP_IND_ID (PIXIT) **ISDN access indicator** ISDN_ACC_IND_VAL (PIXIT) **CPS indicator**: no indication				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	CON	←		←	200 OK INVITE(CON)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

5.2.2.7 Receipt of the Release message (REL)

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

TP307002	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause 7.7.1 2)**	
TSS reference	ISUP-SIP/Basic call/Receipt of the Release message (REL)				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message **before** any response message has been received which establishes a confirmed dialogue: • the SUT shall hold the REL message until a SIP response has been received; • the SUT shall send a BYE request.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	REL	→			
	RLC	←			
				←	200 OK INVITE(CON)
				→	ACK
				→	BYE(REL)
				←	200 OK BYE(RLC)

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

<table>
<tr><th>TP307004</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.7.1 2) 3)</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/Basic call/Receipt of the Release message (REL)/</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message before 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</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/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>100 Trying</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>
</table>

<table>
<tr><th>TP307005</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.7.1 4)</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 after a 200 OK response message has been received:
• the SUT shall send a BYE request. The received REL is encapsulated in the BYE.</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/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></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>

<table>
<tr><th>TP307006</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause 7.7.1 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 after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established and before dialog has been confirmed:
• the SUT shall send a CANCEL request which is answered by 200 OK CANCEL and INVITE request will be terminated by 487.</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>ACM</td><td>←</td><td></td><td>←</td><td>SIP_MESSAGE_VA</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 colspan="3"></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>
</table>

Table 8

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

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

TP308001	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.7.2
TSS reference	ISUP-SIP/Basic call/Sending of the 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		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)

TP308002	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.7.6
TSS reference	ISUP-SIP/Basic call/Sending of the Release message (REL)				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM sends out an INVITE message. On receipt of a Failure message (4xx, 5xx, 6xx) defined as SIP_Failure_VA: • sends a REL message constructed from the encapsulated REL.				
SIP parameter values					
ISUP parameter values	REL; **cause value**: CV_ISUP				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	100 Trying
	REL	←		←	SIP_Failure_VA(REL)
	RLC	→		→	ACK

Table 9

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

TP308003	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause 7.7.6**
TSS reference	ISUP-SIP/Basic call/Sending of the 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
	RLC	←		←	200 OK CANCEL
				←	487 Request Terminated
				→	ACK

TP308004	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.7.6
TSS reference	ISUP-SIP/Basic call/Sending of the Release message (REL)/				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM sends out an INVITE message, a SIP message defined as **SIP MESSAGE_VA** has been received, on receipt of a Failure message (4xx, 5xx, 6xx) defined as **SIP_Failure_VA**: • sends a REL message constructed from the encapsulated REL.				
SIP parameter values					
ISUP parameter values	REL; **cause value**: CV_ISUP				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	SIP MESSAGE_VA(ACM)
	REL	←		←	SIP_Failure_VA(REL)
	RLC	→		→	ACK

Table 10

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

Table 11

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

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

Table 12

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

TP308006	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 7.7.6
TSS reference	ISUP-SIP/Basic call/Sending of the Release message (REL)/				
SIP selection criteria	NOT PICS 4/21				
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM sends out an INVITE message. On receipt of a response message (3xx) defined as **SIP_Response_VA,** the SUT: • sends a REL message with the **Cause value** CV_ ISUP.				
SIP parameter values					
ISUP parameter values	REL; **cause value**: CV_ISUP				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	100 Trying
	REL	←		←	SIP_Response_VA
	RLC	→		→	ACK

Table 13

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

Mapping of Cause Indicators parameter into SIP Reason header fields.

Table 14

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

5.2.2.9 Autonomous release at O-IWU

5.2.2.9.1 Receipt of Reset Circuit message (RSC)

TP309001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 7.7.1, 1), 7.7.4, 7.7.5**
TSS reference	ISUP-SIP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM but before an INVITE has been sent on receipt of a RSC message: • no action is required on the SIP side other than to terminate local procedures if any are in progress.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→			
	RSC	→			
	RLC	←			

TP309002	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.5
TSS reference	ISUP-SIP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message on receipt RSC message **before** a **SIP MESSAGE_VA** response message has been received: • the SUT shall hold the RSC message until a SIP response has been received; • the SUT shall send a CANCEL request. The RSC is not encapsulated.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	RSC	→			
	RLC	←			
				←	SIP_MESSAGE_VA
				→	CANCEL
				←	200 OK CANCEL
				←	487 Request terminated
				→	ACK

Table 15

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

TP309003	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.5
TSS reference	ISUP-SIP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt RSC message **before** a 200 OK response message has been received: • on subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent The RSC is not encapsulated.				
SIP parameter values	BYE: A REL is encapsulated with cause 31				
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	RSC	→			
	RLC	←			
				←	200 OK INVITE(CON)
				→	ACK
				→	BYE(REL#31)
				←	200 OK BYE(RLC)

TP309005	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.5**
TSS reference	ISUP-SIP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message with the complete called party number, sending a BYE message on receipt RSC message **after** a 200 OK response message has been received: • the SUT shall send a BYE request The RSC is not encapsulated.			
SIP parameter values	BYE: A REL is encapsulated with cause 31			
ISUP parameter values				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
	RSC	→		→	BYE(REL#31)
	RLC	←		←	200 OK BYE(RLC)

TP309006	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.5**
TSS reference	ISUP-SIP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt RSC message **after** an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established. The SUT shall send a CANCEL request The RSC is not encapsulated.			
SIP parameter values				
ISUP parameter values				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	SIP_MESSAGE_VA(ACM)
	RSC	→			
	RLC	←			
				→	CANCEL
				←	200 OK CANCEL
				←	487 Request terminated
				→	ACK

Table 16

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

5.2.2.9.2 Receipt of Circuit group reset message (GRS)

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

TP309008	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.5**
TSS reference	ISUP-SIP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt GRS message **before SIP MESSAGE_VA** response message has been received: • the SUT shall hold the GRS message until a SIP response has been received; • the SUT shall send a CANCEL request The GRS is not encapsulated.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	GRS	→			
	GRA	←			
				←	SIP_MESSAGE_VA
	CASE A				
				→	CANCEL
				←	200 OK CANCEL
				←	487 Request terminated
				→	ACK
	CASE B				
				→	BYE(REL#31)
				←	200 OK BYE(RLC)
				←	487 Request terminated
				→	ACK

Table 17

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

TP309009	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 7.7.1 3), 7.7.4, 7.7.5**
TSS reference	ISUP-SIP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt GRS message **before** a 200 OK response message has been received: • the SUT shall hold the GRS message until a response has been received. A CANCEL is sent The GRS is not encapsulated; • on subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		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 [4]				ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.5
TSS reference	ISUP-SIP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message with the complete called party number, sending a INVITE message on receipt GRS message **after** a 200 OK response message has been received: • the SUT shall send a BYE request The GRS is not encapsulated.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
	GRS	→		→	BYE(REL#31)
	GRA	←		←	200 OK BYE(RLC)

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

Table 18

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

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

5.2.2.9.3 Receipt of Circuit group blocking message (CGB)

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

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

ISUP/BICC		SUT		SIP-I
IAM	→		→	INVITE(IAM)
CGB	→			
CGBA	←			
			←	SIP_MESSAGE_VA
			→	CANCEL
			←	200 OK CANCEL
			←	487 Request terminated
			→	ACK

Table 19

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

TP309016	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clauses 7.7.1 3), 7.7.4**	
TSS reference	ISUP-SIP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" **before** a 200 OK response message has been received: • on subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent The CGB is not encapsulated.				
SIP parameter values					
ISUP parameter values	CGB(hardware failure oriented)				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
				←	100 Trying
	CGB	→			
	CGBA	←		→	CANCEL
				←	200 OK INVITE(CON)
				→	ACK
				←	200 OK CANCEL
				→	BYE(REL#31)
				←	200 OK BYE(RLC)

TP309017	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4**	
TSS reference	ISUP-SIP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" **after** a 200 OK response message has been received: • the SUT shall send a BYE request The CGB is not encapsulated.				
SIP parameter values					
ISUP parameter values	CGB(hardware failure oriented)				
Comments	ISUP/BICC		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
	CGB	→		→	BYE(REL#31)
	CGBA	←		←	200 OK BYE(RLC)

TP309018	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4
TSS reference	ISUP-SIP/Basic call/Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" **after** an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established: • the SUT shall send a CANCEL 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	←		←	SIP_MESSAGE_VA(ACM)
	CGB	→			
	CGBA	←			
				→	CANCEL
				←	200 OK CANCEL
				←	487 Request terminated
				→	ACK

Table 20

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

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

5.2.2.10 Receipt of Confusion message

TP310001	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause A.1.1.3
TSS reference	ISUP-SIP/ISUP Messages for special consideration/Confusion message	
SIP selection criteria		
ISUP selection criteria		
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number and contains an unknown parameter, sending a INVITE message with the complete called party number and encapsulated IAM as received. Ensure that when the succeeding node discards an unknown parameter and send back a Confusion message if indicated in the parameter compatibility information and the sending of a Confusion message is requested, the CFN message encapsulated in a 183 Session Progress is sent. Ensure ISUP message is transported through the SIP network encapsulated in the 183 Session Progress.	
SIP parameter values	183 Session Progress with encapsulated CFN	
ISUP parameter values	CFN	
Comments		

ISUP				SIP-I
IAM	→		→	INVITE(IAM with unknown parameter)
CFN	←		←	183 Session Progress(CFN)
ACM	←		←	180 Ringing(ACM)
ANM	←		←	200 OK INVITE(ANM)
			→	ACK
	Communication			
REL	→		→	BYE(REL)
RLC	←		←	200 OK BYE(RLC)

5.2.2.11 Receipt of "Suspend" or "Resume" message

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

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

5.2.2.12 Receipt of a Blocking message

TP312001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause A.1.1.3.1**
TSS reference	ISUP-SIP/ISUP Messages for special consideration/Receipt of a Blocking message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the blocking/unblocking procedure can be correctly initiated. Ensure the BLO messages are not encapsulated within SIP messages				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	BLO	→			
	BLA	←			
	UBL	→			
	UBA	←			

TP312002	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause A.1.1.3.1**
TSS reference	ISUP-SIP/ISUP Messages for special consideration/Receipt of a Blocking message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the blocking from both ends; removal of blocking from one end can be correctly initiated. Ensure the BLO messages are not encapsulated within SIP messages.				
SIP parameter values					
ISUP parameter values					
Comments	ISUP/BICC		SUT		SIP-I
	BLO	→			
	BLA	←			
	BLO	←			
	BLA	→			
	UBL	→			
	UBA	←			

<table>
<tr><th>TP312003</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause A.1.1.3.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/ISUP Messages for special consideration/Receipt of a Blocking message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">CGB and CGU sent
Ensure that the SUT is able to respond on a Circuit group blocking message with a CGBA and on a Circuit group unblocking message (both maintenance oriented) with a CGUA.
Ensure the CGB/CGU messages are not encapsulated within SIP messages.</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</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>CGB</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>CGBA</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>CGU</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>CGUA</td><td>←</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><th>TP312004</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause A.1.1.3.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/ISUP Messages for special consideration/Receipt of a Blocking message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that the SUT on receipt of a CGB, which is received encapsulated within SIP messages, discards the ISUP information.</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="2">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>INFO(CGB)</td></tr>
</table>

<table>
<tr><th>TP312005</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause A.1.1.3.1
Q.784 [i.11], clause 1.3.2.4</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP/ISUP Messages for special consideration/Receipt of a Blocking message</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that a received IAM will unblock a remotely blocked circuit.</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>BLO</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>BLA</td><td>←</td><td></td><td></td><td></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>REL</td><td>→</td><td></td><td>→</td><td>BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE(RLC)</td></tr>
</table>

5.2.2.13 Receipt of a user part test message

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

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

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

5.2.2.14 Segmentation

TP314001	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause A.1.1.3.1
TSS reference	ISUP-SIP/ISUP Messages for special consideration/Receipt of a user part test message				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that a call can be successfully completed if segmentation applies in forward direction.				
SIP parameter values	INVITE - encapsulated IAM: Optional 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 [4]				ISUP reference: Q.1912.5 [1], clauses 7.1.3, B.1
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Calling Party number network provided, transferred in O-MGCF* Ensure that the SUT can successfully transmit a call having a **calling party number** with the screening indicator set to "network provided" and the presentation restricted indicator set to "presentation allowed".				
SIP parameter values					
ISUP parameter values	IAM; **Calling party number parameter** Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '11'B presentation restricted indicator = presentation allowed, '00'B				
Comments	**ISUP**		**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)

TP401002	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 7.1.3, B.1
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Calling Party number network provided, Calling Subaddress transferred in O-MGCF* Ensure that the SUT can successfully transmit a call having a **calling party number** with the screening indicator set to "network provided" and an **access transport** parameter containing the **calling sub-address.**				
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)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP401003	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 7.1.3, B.1
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Calling Party Number user provided transferred in O-MGCF* Ensure that the SUT can successfully transmit a call having the **calling party number** with the screening indicator set to "user provided, verified and passed" and the presentation restricted indicator set to "presentation allowed".				
SIP parameter values					
ISUP parameter values	IAM; **Calling party number parameter** Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '01'B presentation restricted indicator = presentation allowed, '00'B				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP401004</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clauses 7.1.3, B.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Calling Party Number user provided and calling subaddress transferred in O-MGCF

Ensure that the SUT can successfully transmit a call having a calling party number with the screening indicator set to "user provided, verified and passed" and an access transport parameter containing the calling sub-address.</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="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>

TP401005	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 7.1.3, B.1**
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Calling Party Number network provided and additional calling party number user provided not verified transferred in O-MGCF.* Ensure that the SUT can successfully transmit a call having a default **calling party number** with the screening indicator set to "network provided" and a **generic number** containing the additional calling party number with the screening indicator set to "user provided, not verified" and the presentation restricted indicator set to "presentation allowed".				
SIP parameter values					
ISUP parameter values	IAM; **Calling party number 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	**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)

TP401006	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clauses 7.1.3, B.1
TSS reference	ISUP-SIP-ISUP/SS/CLIP	
SIP selection criteria		
ISUP selection criteria		
Test purpose	*Calling Party Number network provided, additional calling party number user provided not verified and calling subaddress transferred in O-MGCF.* Ensure that the SUT can successfully transmit a call having a default calling party number with the screening indicator set to "network provided", a generic number containing the additional calling party number with the screening indicator set to "user provided, not verified" and an access transport parameter containing the calling sub-address.	
SIP parameter values		
ISUP parameter values	IAM; **Calling party number parameter** Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '11'B **Generic number parameter** Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Access transport parameter including the subaddress information	
Comments	(see message flow below)	

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)

TP401007	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clauses 6.1.3.6, B.1
TSS reference	ISUP-SIP-ISUP/SS/CLIP	
SIP selection criteria		
ISUP selection criteria	PICS 6/8	
Test purpose	*Calling party number discarded to due bilateral agreement in the I-MGCF.* Ensure that the calling party number is discarded in case of bilateral agreements, if the address presentation restricted indicator is set to "presentation allowed" (see note).	
SIP parameter values		
ISUP parameter values	IAM; **No calling party number parameter**	
Comments	(see message 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

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.

<table>
<tr><th>TP401008</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses 6.1.3.6, B.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 6/7</td></tr>
<tr><td>Test purpose</td><td colspan="5">Additional Calling party number is discarded to due bilateral agreements in the I-MGCF

Ensure that the additional calling party number in the generic number is discarded in case of bilateral agreements, if the address presentation restricted indicator is set to "presentation allowed".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM;
No calling party number parameter</td></tr>
<tr><td rowspan="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>
<tr><td colspan="6">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.</td></tr>
</table>

<table>
<tr><th>TP401009</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses 6.1.3.6, B.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 6/6</td></tr>
<tr><td>Test purpose</td><td colspan="5">Calling party number is omitted if the presentation restriction indicator is set to address not available in the I-MGCF

Ensure that the calling party number is omitted, if the address presentation restricted indicator is set to "address not available".</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>

TP401010	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 6.1.3.6, B.1
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Calling party number is sent as received* Ensure that the calling party number in the sent IAM is generated from the calling party number in the encapsulated IAM.				
SIP parameter values					
ISUP parameter values					
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP401011	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 6.1.3.6, B.1
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Additional calling party number is sent as received* Ensure that the additional calling party number in the sent IAM is generated from the additional calling party number in the encapsulated IAM.				
SIP parameter values					
ISUP parameter values					
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP401012	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 6.1.3.6, B.1
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Additional calling party number is omitted in the I-MGCF* Ensure that if the **calling party number** is not sent, then an additional calling party number in a **generic number** will be omitted.				
SIP parameter values	INVITE: No calling party number included in the encapsulated IAM, additional calling party number included.				
ISUP parameter values	IAM; **No calling party number parameter** **No generic number parameter**				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP401013	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 6.1.3.6, B.1, Q.731 [i.2], clause 3.5
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	PICS 1/7				
Test purpose	*Convert the Calling party number into the international format in the I-MGCF* Ensure that the SUT can convert the **calling party number** into an international number, setting the nature of address indicator to "international number" and can pass on the address presentation restricted indicator and the screening indicator transparently.				
SIP parameter values					
ISUP parameter values	IAM; **Calling party number parameter** Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000100'B Screening indicator = '11'B Presentation restricted indicator =presentation allowed, '00'B				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP401014	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 6.1.3.6, B.1, 3.5/Q.731 [i.2]
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	PICS 1/7				
Test purpose	*Converting the additional calling party number to international format in the I-MGCF* Ensure that the SUT can convert the additional calling party number in the **generic number** into an international number, if the numbering plan indicator is "ISDN Telephony", setting the nature of address indicator to "international number" and can pass on the address presentation restricted indicator and the screening indicator transparently.				
SIP parameter values					
ISUP parameter values	**IAM** **Calling party number parameter** Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000100'B Screening indicator = '11'B Presentation restricted indicator =presentation allowed, '00'B **Generic number parameter** Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000100'B Screening indicator = '00'B Presentation restricted indicator =presentation allowed, '00'B				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP401015	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 6.1.3.6, B.1, 3.5/Q.731 [i.2]
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	PICS 1/7 AND NOT PICS 1/9				
Test purpose	*Discarding an incomplete calling party number in the I-MGCF* Ensure that the calling party number is discarded, if it is received with the calling party number incomplete indicator set to "incomplete" (see note).				
SIP parameter values					
ISUP parameter values	IAM: **No calling party number parameter**				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			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 [4]**				**ISUP reference:** **Q.1912.5 [1], clauses 6.1.3.6, B.1, 3.5/Q.731 [i.2]**
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	PICS 1/8				
Test purpose	*Converting the calling party number to national format, if necessary in the O-MGCF* Ensure that the country code in the address signals of the **calling party number** is removed if it is the network's own country code. The nature of address indicator shall be set to "national (significant) number". The address presentation restricted indicator shall be transferred transparently.				
SIP parameter 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)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP401017	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 6.1.3.6, B.1, 3.5/Q.731 [i.2]
TSS reference:	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	PICS 1/8				
Test purpose	*Converting the additional calling party number to national format, if necessary in the O-MGCF* Ensure that the country code in the address signals of the **generic number** coded as an "additional calling party number", if the numbering plan indicator is "ISDN Telephony" is removed if it is the network's own country code. The nature of address indicator shall be set to "national (significant) number". The address presentation restricted indicator shall be transferred transparently.				
SIP parameter 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 = '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**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP401018	SIP reference: RFC 3261 [4]		ISUP reference: Q.1912.5 [1], clauses 6.1.3.6, B.1, 3.5/Q.731 [i.2]	
TSS reference	ISUP-SIP-ISUP/SS/CLIP			
SIP selection criteria				
ISUP selection criteria	PICS 1/7			
Test purpose	*Adding a prefix to an international calling party number in the I-MGCF* Ensure that a prefix is added to the **calling party number** and the nature of address indicator is set to "unknown" (see note).			
SIP parameter values				
ISUP parameter values				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC
NOTE: The coding "unknown" is a national option (@).					

TP401019	SIP reference: RFC 3261 [4]		ISUP reference: Q.1912.5 [1], clauses 6.1.3.6, B.1, 3.5/Q.731 [i.2]		
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Handling of address presentation restricted indicator set to "address not available" in the I-MGCF* Ensure that the screening indicator shall be set to "network provided" if the address presentation restricted indicator in **calling party number** is set to "address not available" (see note).				
SIP parameter values					
ISUP parameter values	IAM; **Calling party number parameter** Address signals = PIXIT1 Numbering plan indicator = ' *'B Nature of address indicator = '*'B Screening indicator = '11'B Presentation restricted indicator =address not available, '10'B				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC
NOTE: The coding "address not available" is a national option (@).					

TP401020	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 7.1.3, B.1
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*O-MGCF: Calling party number and Additional calling party number not received* 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.invalid				
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)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP401021	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 7.1.3, B.1
TSS reference:	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*O-MGCF: Setting of From header* Ensure that when the SUT has received an IAM message whereby **Calling Party Number** parameter is **not applicable** and the **Generic Number is applicable** whereby the address presentation restriction parameter is set to "presentation allowed" and the Nature of Address Indicator is set to NoAS_VALUE. Sends an INVITE message without the "P-Asserted-Identity header field", a "From header field" where the user portion of the addr-spec is set to value of the additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN and no "Privacy Header field".				
SIP parameter values	INVITE: no P-Asserted-Identity, no Privacy header, From header contains the value of the additional calling party number				
ISUP parameter values	IAM; no Calling party number present, Additional calling party number present				
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)

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

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

Values for test purpose TP401021, TP401022, TP401023		
NoAS_VALUE	**ISUP parameter values**	**SIP parameter values:**
VA_01	**IAM** NoAS_VALUE: ***"national (significant) number" (NDC+SN)***	**INVITE** FHf_Addr_SPEC_ID: CC (of the country where the IWU is located) is added to the Generic Number Address Signals and then mapped to user portion of URI scheme
VA_02	**IAM** NoAS_VALUE: ***"international number" ("+"CC+NDC+SN)***	**INVITE** FHf_Addr_SPEC_ID: the complete GenericNumber Address Signals is mapped to the user portion of URI scheme used

TP401024	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clauses 6.1.3.6, B.1**
TSS reference	ISUP-SIP-ISUP/SS/CLIP			
SIP selection criteria				
ISUP selection criteria	PICS 1/7			
Test purpose	*Calling party derived from the P-Asserted-Identity international number* Ensure when no calling party number is included in the encapsulated IAM or the calling party number in the in the encapsulated IAM is not identical to the P-Asserted-Identity, no Privacy value "id" received. Send an IAM the calling party number is derived from SIP P-Asserted-Identity. The Address Presentation Restricted Indicator is set to Presentation allowed.			
SIP parameter values	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value "id" is not present			
ISUP parameter values	**IAM** message with the **Calling party number parameter** coded Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: "international number"			

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

<table>
<tr><th>TP401025</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses 6.1.3.6, B.1</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">NOT PICS 1/7</td></tr>
<tr><td>Test purpose</td><td colspan="5">Calling party derived from the P-Asserted-Identity national (significant) number

Ensure when no calling party number is included in the encapsulated IAM or the calling party number in the in the encapsulated IAM is not identical to the P-Asserted-Identity, no Privacy value "id" received.
Send an IAM the calling party number is derived from SIP P-Asserted-Identity. The Address Presentation Restricted Indicator is set to Presentation allowed.</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 not present</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">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"</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>TP401026</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses 6.1.3.6, B.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/7</td></tr>
<tr><td>Test purpose</td><td colspan="5">Additional calling party number derived from the From header international number

Ensure when no additional calling party number is included in the encapsulated IAM or the additional calling party number in the in the encapsulated IAM is not identical to the From header field, no Privacy value "id" received.
Send an IAM the additional calling party number is derived from From header field. The Address Presentation Restricted Indicator is set to Presentation allowed.</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 not 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 allowed
NoAS: "international number"</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>

TP401027	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clauses 6.1.3.6, B.1**		
TSS reference:	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	NOT PICS 1/7				
Test purpose	*Additional calling party number derived from the From header national (significant) number* Ensure when no additional calling party number is included in the encapsulated IAM or the additional calling party number in the in the encapsulated IAM is not identical to the From header field, no Privacy value "id" received. Send an IAM the additional calling party number is derived from From header field. The Address Presentation Restricted Indicator is set to Presentation allowed.				
SIP parameter values	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value "id" is not present				
ISUP parameter values	**IAM** message with the **Additional Calling party number parameter** coded Address signals = number derived from SIP From header Screening indicator = User provided, not verified Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: "national (significant) number"				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

5.3.2 Calling Line Identification Restriction (CLIR)

TP402001	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clauses 7.1.3, B.1, Q.731 [i.2], clause 4.5.2.1.1**		
TSS reference:	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Calling party number network provided presentation restricted is passed.* Ensure that the SUT can successfully transmit a call having a **calling party number** with the screening indicator set to "network provided" and the address presentation restricted indicator set to "presentation restricted".				
SIP parameter values					
ISUP parameter values	IAM; **Calling party number parameter** Screening indicator = '11'B Address presentation restricted parameter = '01'B **Generic number parameter** not present **Access transport parameter is not** including the subaddress information				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP402002	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses 7.1.3, B.1, Q.731 [i.2], clause 4.5.2.1.1	
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Restricted calling party 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)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP402003	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses 7.1.3, B.1, Q.731 [i.2], clause 4.5.2.1.1	
TSS reference:	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Restricted calling party 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)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP402004	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clauses 7.1.3, B.1, Q.731 [i.2], clause 4.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/CLIR	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Restricted calling party 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)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP402005	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 7.1.3, B.1, Q.731 [i.2], clause 4.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Restricted calling party 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)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP402006	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 7.1.3, B.1, Q.731 [i.2], clause 4.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Restricted calling party 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)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP402007	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 6.1.3.6, B.1, Q.731 [i.2], clause 4.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection 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
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP402008</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses 6.1.3.6, B.1,
Q.731 [i.2], clause 4.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="7">ISUP-SIP-ISUP/SS/CLIR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="7"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="7">PICS 6/4 AND PICS 6/5</td></tr>
<tr><td>Test purpose</td><td colspan="7">Discarding the additional calling party number if the presentation is restricted
Ensure that the additional calling party number in the generic number is discarded in case of bilateral agreements, if the address presentation restricted indicator is set to "presentation restricted".</td></tr>
<tr><td>SIP parameter values</td><td colspan="7"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="7">IAM;
No Calling party number parameter
No Generic number parameter</td></tr>
<tr><td rowspan="8">Comments</td><td>SIP-I</td><td></td><td colspan="2">SUT</td><td></td><td colspan="2">ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td colspan="2"></td><td>→</td><td colspan="2">IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td colspan="2"></td><td>←</td><td colspan="2">ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td colspan="2"></td><td>←</td><td colspan="2">ANM</td></tr>
<tr><td>ACK</td><td>→</td><td colspan="2"></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td colspan="4">Conversation</td><td colspan="2"></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td colspan="2"></td><td>→</td><td colspan="2">REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td colspan="2"></td><td>←</td><td colspan="2">RLC</td></tr>
</table>

<table>
<tr><th>TP402009</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses 6.1.3.6, B.1,
Q.731 [i.2], clause 4.5.2.1.1</th></tr>
<tr><td>TSS reference:</td><td colspan="7">ISUP-SIP-ISUP/SS/CLIR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="7"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="7"></td></tr>
<tr><td>Test purpose</td><td colspan="7">I-MGCF: Calling party number received in the INVITE is sent in the IAM

Ensure that the calling party number contained in the encapsulated IAM is unchanged sent in the ISUP IAM.</td></tr>
<tr><td>SIP parameter values</td><td colspan="7"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="7"></td></tr>
<tr><td rowspan="8">Comments</td><td>SIP-I</td><td></td><td colspan="2">SUT</td><td></td><td colspan="2">ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td colspan="2"></td><td>→</td><td colspan="2">IAM</td></tr>
<tr><td>180 Ringing(ACM)</td><td>←</td><td colspan="2"></td><td>←</td><td colspan="2">ACM</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td colspan="2"></td><td>←</td><td colspan="2">ANM</td></tr>
<tr><td>ACK</td><td>→</td><td colspan="2"></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td colspan="4">Conversation</td><td colspan="2"></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td colspan="2"></td><td>→</td><td colspan="2">REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td colspan="2"></td><td>←</td><td colspan="2">RLC</td></tr>
</table>

TP402010	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 6.1.3.6, B.1, Q.731 [i.2], clause 4.5.2.1.1**
TSS reference:	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*I-MGCF: Additional calling party number received in the INVITE is sent in the IAM* Ensure that the additional calling party number contained in the encapsulated IAM is unchanged sent in the ISUP IAM.				
SIP parameter values					
ISUP parameter values					
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP402011	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 7.1.3, B.1**
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that when the SUT has received an IAM message, the **Calling Party Number is applicable** whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to **presentation restricted** and the **Generic Number is not applicable** Sends an INVITE message with: • the "P-Asserted-Identity header field" where the user portion of the addr-spec is set to value of the additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN; • a "From header field" set to "anonymous@anonymous.invalid". • and with "Privacy Header field" set to "id".				
SIP parameter values	INVITE: P-Asserted-Identity, From Header: anonymous@anonymous.invalid, 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)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><td>TP402012</td><td colspan="2">SIP reference: RFC 3261 [4]</td><td colspan="3">ISUP reference:
Q.1912.5 [1], clauses 7.1.3, B.1</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CLIR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation restricted and the Generic Number is applicable.
Sends an INVITE message with:
• the "P-Asserted-Identity header field", where the user portion of the addr-spec is set to value of the calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN;
• "From header field" where the user portion of the addr-spec is set to value of the additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN;
• and with "Privacy Header field" is set to "id".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: P-Asserted-Identity, From header field, Privacy "id"</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: Calling party number. additional calling party number</td></tr>
<tr><td rowspan="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><td colspan="3">Values for test purpose TP401012</td></tr>
<tr><td>NoAS_VALUE</td><td>ISUP parameter values</td><td>SIP parameter values:</td></tr>
<tr><td>VA_01</td><td>IAM
NoAS_VALUE: "national (significant) number" (NDC+SN)</td><td>INVITE
FHf_Addr_SPEC_ID: CC (of the country where the IWU is located) is added to the Generic Number Address Signals and then mapped to user portion of URI scheme</td></tr>
<tr><td>VA_02</td><td>IAM
NoAS_VALUE: "international number" ("+"CC+NDC+SN)</td><td>INVITE
FHf_Addr_SPEC_ID: the complete GenericNumber Address Signals is mapped to the user portion of URI scheme used.</td></tr>
</table>

TP402013	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses 6.1.3.6, B.1
TSS reference	ISUP-SIP-ISUP/SS/CLIR			
SIP selection criteria				
ISUP selection criteria	PICS 1/7			
Test purpose	Ensure when no 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
				→	ACK
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP402014	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses 6.1.3.6, B.1
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
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP402015	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 6.1.3.6, B.1
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria	PICS 1/7				
Test purpose	Ensure when no additional calling party number is included in the encapsulated IAM or the additional calling party number in the encapsulated IAM is not identical to the From header field, Privacy value "id" received. Send an IAM the additional calling party number is derived from From header field. The Address Presentation Restricted Indicator is set to Presentation restricted.				
SIP parameter values	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value "id" is present				
ISUP parameter values	**IAM** message with the **Additional Calling party number parameter** coded Address signals = number derived from SIP From header Screening indicator = User provided, not verified Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: "international number"				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP402016	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 6.1.3.6, B.1
TSS reference	ISUP-SIP-ISUP/SS/CLIR				
SIP selection criteria					
ISUP selection criteria	NOT PICS 1/7				
Test purpose	Ensure when no additional calling party number is included in the encapsulated IAM or the additional calling party number in the encapsulated IAM is not identical to the From header field, Privacy value "id" received. Send an IAM the additional calling party number is derived from From header field. The Address Presentation Restricted Indicator is set to Presentation restricted.				
SIP parameter values	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value "id" is present				
ISUP parameter values	**IAM** message with the **Additional Calling party number parameter** coded Address signals = number derived from SIP From header Screening indicator = User provided, not verified Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: "national (significant) number"				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

5.3.3 Connected line identification presentation (COLP)

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

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

<table>
<tr><td>TP403003</td><td colspan="5">SIP reference: RFC 3261 [4]</td><td>ISUP reference:
Q.1912.5 [1], clause B.2,
Q.731 [i.2], clause 5.5.2.1.1</td></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">Connected number (user provided, not verified) without connected sub-address
Ensure that the SUT passes transparently a default connected number with the screening indicator set to "network provided", a generic number containing the additional connected number with the screening indicator set to "user provided, not verified" without an access transport parameter containing the connected sub-address.</td></tr>
<tr><td>SIP parameter values</td><td colspan="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 = '00'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '11'B
Address signals = PIXIT
Additional connected number present
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '00'B
Address signals = PIXIT

b)
CON;
Connected number parameter
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '11'B
Address signals = PIXIT
Additional connected number present
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '00'B
Address signals = PIXIT</td></tr>
<tr><td rowspan="12">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>ACK</td><td>→</td><td></td><td></td><td colspan="2"></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>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>

TP403004	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clause B.2, Q.731 [i.2], clause 5.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/COLP	
SIP selection criteria		
ISUP selection criteria	PICS 1/7	
Test purpose	**Converting the connected number to national format, if necessary** Ensure that the country code in the address signals of the **connected number** is 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.	
SIP parameter values	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	
ISUP parameter values	**IAM;** **optional forward call indicators** Connected line identity request indicator: requested a) **ANM;** **Connected number parameter** Address presentation restricted parameter = '00'B Nature of address indicator = '0000100'B Numbering plan indicator = '001'B Screening indicator = 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	
Comments		

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

TP403005	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause B.2, Q.731 [i.2], clause 5.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/COLP	
SIP selection criteria		
ISUP selection criteria	PICS 1/7	
Test purpose	**Converting the additional connected number to national format, if necessary** Ensure that the country code in the address signals of the **generic number** coded as an "additional connected number", if the numbering plan indicator is "ISDN Telephony" is removed if it is the network's 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.	
SIP parameter values	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	
ISUP parameter values	**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	

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

<table>
<tr><th>TP403006</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause B.2,
Q.731 [i.2], clause 5.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/COLP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/8 AND PICS 7/5</td></tr>
<tr><td>Test purpose</td><td colspan="5">Adding a prefix to an international connected number
Ensure that a prefix is added to the connected number and the nature of address indicator is set to "unknown" (see note).</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">200 OK INVITE with encapsulated ANM or CON
Connected number parameter
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '11'B
Address signals = PIXIT</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">ANM/CON:
Connected number parameter
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000010'B
Numbering plan indicator = '001'B
Screening indicator = '11'B
Address signals = Prefix+PIXIT</td></tr>
<tr><td rowspan="12">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></td><td></td><td></td><td>→</td><td>ACK</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></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 colspan="6">NOTE: The coding "unknown" is a national option (@).</td></tr>
</table>

<table>
<tr><th>TP403007</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clause B.2,
Q.731 [i.2], clause 5.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/COLP</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/8 AND PICS 7/3</td></tr>
<tr><td>Test purpose</td><td colspan="5">Discarding the connected number in case of bilateral agreements
Ensure that the connected number is discarded in case of bilateral agreements, if the address presentation restricted indicator is set to "presentation allowed" (see note).</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">200 OK INVITE with encapsulated ANM or CON
Connected number parameter
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '11'B
Address signals = PIXIT</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM
optional forward call indicators
Connected line identity request indicator: requested
a)
ANM
No Connected number parameter
b)
CON;
No Connected number parameter</td></tr>
<tr><td rowspan="12">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></td><td></td><td></td><td>→</td><td>ACK</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></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 colspan="6">NOTE: This bilateral agreement prohibits the transferral of the connected number in any case. The test with the address presentation restricted indicator set to "presentation restricted" is a COLR test.</td></tr>
</table>

<table>
<tr><td>TP403008</td><td colspan="5">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1], clause B.2,
Q.731 [i.2], 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 AND PICS 7/4</td></tr>
<tr><td>Test purpose</td><td colspan="7">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="7">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="7">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="12">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td colspan="3">SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td colspan="3">INVITE(IAM)</td></tr>
<tr><td colspan="7">CASE A</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td colspan="3">180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td colspan="3">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="3">ACK</td></tr>
<tr><td colspan="7">CASE B</td></tr>
<tr><td>CON</td><td>←</td><td></td><td>←</td><td colspan="3">200 OK INVITE(CON)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="3">ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="3"></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td colspan="3">BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td colspan="3">200 OK BYE(RLC)</td></tr>
<tr><td colspan="8">NOTE: This bilateral agreement prohibits the transferral of the additional connected number in the generic number in any case.</td></tr>
</table>

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

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

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

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

TP403012	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.2, Q.731 [i.2], clause 5.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/COLP				
SIP selection criteria					
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)
				→	ACK
	CASE B				
	CON	←		←	200 OK INVITE(CON)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><td>TP403013</td><td colspan="5">SIP reference: RFC 3261 [4]</td><td colspan="2">ISUP reference:
Q.1912.5 [1], clause B.2,
Q.731 [i.2], 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"></td></tr>
<tr><td>Test purpose</td><td colspan="7">O-MGCF: connected number and additional connected number transferred transparently

Ensure that an ANM or CON encapsulated in a 200 OK INVITE is sent on the ISUP side without changing. The connected number is unchanged. The ATP contained the connected sub address is included.</td></tr>
<tr><td>SIP parameter values</td><td colspan="7">200 OK INVITE: encapsulated ANM or CON included</td></tr>
<tr><td>ISUP parameter values</td><td colspan="7">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="13">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td colspan="3">SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td colspan="3">INVITE(IAM)</td></tr>
<tr><td colspan="7">CASE A</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td colspan="3">180 Ringing(ACM)</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td colspan="3">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="3">ACK</td></tr>
<tr><td colspan="7">CASE B</td></tr>
<tr><td>CON</td><td>←</td><td></td><td>←</td><td colspan="3">200 OK INVITE(CON)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="3">ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td colspan="3"></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td colspan="3">BYE(REL)</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td colspan="3">200 OK BYE(RLC)</td></tr>
</table>

5.3.4 Connected Line Identification Restriction (COLR)

<table>
<tr><th>TP404001</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause B.2,
Q.731 [i.2], clause 6.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/COLR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM;
optional forward call indicators
Connected line identity request indicator: requested
a)
ANM;
Connected number parameter
Address presentation restricted parameter = '01' B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '01'B
Address signals = PIXIT
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</td></tr>
<tr><td rowspan="12">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>ACK</td><td>→</td><td></td><td></td><td></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>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>TP404002</th><th>SIP reference: RFC 3261 [4]</th><th colspan="4">ISUP reference:
Q.1912.5 [1], clause B.2,
Q.731 [i.2], clause 6.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/COLR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">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="5"></td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM;
optional forward call indicators
Connected line identity request indicator: requested
a)
ANM;
Connected number parameter
Address presentation restricted parameter = '01' B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '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="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 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>ACK</td><td>→</td><td></td><td></td><td></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>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>

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

TP404004	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause B.2, Q.731 [i.2], clause 6.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/COLR	
SIP selection 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
	ACK	→			
	CASE B				
	200 OK INVITE(CON)	←		←	CON
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP404005</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause B.2,
Q.731 [i.2], clause 6.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/COLR</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5">PICS 7/2</td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Discarding the additional connected number in the generic number if the presentation is restricted
Ensure that the additional connected number in the generic number is discarded in case of bilateral agreements, if the address presentation restricted indicator is set to "presentation restricted".</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">200 INVITE: encapsulated ANM or CON
No Additional Connected number parameter included</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 parameter
Address presentation restricted parameter = '01'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '11'B
Address signals = PIXIT

b)
CON;
Connected number parameter present
Additional Connected number parameter
Address presentation restricted parameter = '01'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '11'B
Address signals = PIXIT</td></tr>
<tr><td rowspan="12">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>ACK</td><td>→</td><td></td><td></td><td></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>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>

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

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

5.3.5 Terminal Portability (TP)

TP405001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause B.13, Q.733 [i.6], clause 4.5.2.1**
TSS reference:	ISUP-SIP-ISUP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Terminal portability, requested by the calling party** Ensure that SUT informs the called party that a suspend and a resume have been requested by the calling party upon receipt of user initiated **SUS** and **RES** messages.				
SIP parameter values	INFO: Content-Type: application/ISUP ; SUS and RES encapsulated in the MIME body				
ISUP parameter values					
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
		Conversation			
	SUS	→		→	INFO(SUS)
				←	200 OK INFO
	RES	→		→	INFO(RES)
				←	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP405002	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause B.13, Q.733 [i.6], clause 4.5.2.1**
TSS reference:	ISUP-SIP-ISUP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Terminal portability, requested by the called party** Ensure that SUT informs the calling party that a suspend and a resume have been requested by the called party upon receipt of user initiated **SUS** and **RES** messages.				
SIP parameter values	INFO: Content-Type: application/ISUP ; SUS and RES encapsulated in the MIME body				
ISUP parameter values					
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
		Conversation			
	SUS	←		←	INFO(SUS)
				→	200 OK INFO
	RES	←		←	INFO(RES)
				→	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP405003	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.13, Q.733 [i.6], clause 4.5.2.1
TSS reference	ISUP-SIP-ISUP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Terminal portability, requested by local served user, no Resume after Suspend** Ensure that the call is released with cause #102 (recovery on timer expiry) by the SUT if timer T2 expires because the local served user does not resume the call.				
SIP parameter values	INFO: Content-Type: application/ISUP ; SUS encapsulated in the MIME body BYE: Content-Type: application/ISUP ; REL encapsulated in the MIME body				
ISUP parameter values					
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	SUS	→		→	INFO(SUS)
				←	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP405004	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.13, Q.733 [i.6], clause 4.5.2.1
TSS reference	ISUP-SIP-ISUP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Terminal portability, release suspended call** Ensure that a suspended call can be released, if the remote user releases the call.				
SIP parameter values	INFO: Content-Type: application/ISUP ; SUS encapsulated in the MIME body BYE: Content-Type: application/ISUP ; REL encapsulated in the MIME body				
ISUP parameter values					
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	SUS	→		→	INFO(SUS)
				←	200 OK INFO
	REL	←		←	BYE(REL)
	RLC	→		→	200 OK BYE(RLC)

5.3.6 SUB-addressing (SUB)

TP406001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause B.5, Q.731 [i.2], clause 8.5.2.1.1/**
TSS reference:	ISUP-SIP-ISUP/SS/SUB				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Sending the called sub-address in the access transport parameter* Ensure that the SUT can include the called sub-address in the **access transport** parameter in the encapsulated IAM.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP, 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)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP406002	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause B.5, Q.731 [i.2], clause 8.5.2.1.1/**
TSS reference	ISUP-SIP-ISUP/SS/SUB				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Receiving the called sub-address in the access transport parameter* Ensure that the SUT can include the called sub-address in the **access transport** parameter in the ISUP IAM.				
SIP parameter values					
ISUP parameter values					
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP406003	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.5, Q.731 [i.2], clause 8.5.2.1.1/
TSS reference	ISUP-SIP-ISUP/SS/SUB				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Sending the calling sub-address in the access transport parameter* Ensure that the SUT can include the calling sub-address in the **access transport** parameter in the encapsulated IAM.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP multipart/mixed, Content-Type: application/ISUP, 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)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP406004	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.5, Q.731 [i.2], clause 8.5.2.1.1/
TSS reference	ISUP-SIP-ISUP/SS/SUB				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Receiving the calling sub-address in the access transport parameter* Ensure that the SUT can include the calling sub-address in the **access transport** parameter in the ISUP IAM.				
SIP parameter values					
ISUP parameter values					
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

5.3.7 Malicious Call Identification (MCID)

TP407001	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause B.4, Q.731.7 [i.3], clause 7.5.2.1.1**
TSS reference:	ISUP-SIP-ISUP/SS/MCID				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Successful MCID request O-MGCF Ensure that the SUT can successfully pass on a 183 Session Progress containing an encapsulated **IDR** having the **MCID request indicator** set to "MCID request" and pass on an **IRS** with **MCID response indicator** set to "MCID included" and the **calling party number** included. ISUP to SIP-I interworking.				
SIP parameter values	183 Session Progress: Content-Type: application/ISUP; IDR encapsulated in the MIME body INFO: Content-Type: application/ISUP; IRS encapsulated in the MIME body				
ISUP parameter values					
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	IDR	←		←	183 Session Progress(IDR)
	IRS	→		→	INFO(IRS)
				←	200 OK INFO
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP407002	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause B.4, Q.731.7 [i.3], clause 7.5.2.1.1**
TSS reference	ISUP-SIP-ISUP/SS/MCID				
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
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP407003	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.4, Q.731.7 [i.3], clause 7.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/MCID				
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
	ACK	→			
	CASE B				
	183 Session Progress(ACM)	←		←	ACM(early)
	183 Session Progress(IDR)	←		←	IDR
	INFO(IRS)	→		→	IRS
	200 OK INFO	←			
	180 Ringing(CPG)	←		←	CPG(alerting)
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC
NOTE: This situation may occur e.g. if the call has been forwarded before reaching the destination.					

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

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

TP407006	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause B.4, Q.731.7 [i.3], clause 7.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/MCID			
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**

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

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

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

TP407008	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.4, Q.731.7 [i.3], clause 7.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/MCID				
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
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

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

TP407010	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.4, Q.731.7 [i.3], clause 7.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/MCID				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**MCID timer (T39) expiry O-MGCF** Ensure that call setup is continued (user is alerted) if no **IRS** is received within timer T39 expiry, after having sent the **IDR** with **MCID request indicator** set to "MCID requested". 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
				T39 expiry	
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

5.3.8 Call hold (HOLD)

TP408001	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.733 [i.6], clauses 2.5.2.1.1.1; 2.5.2.1.1.2
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Call hold after answer, requested by the originating user* Ensure that the notifications that a call is placed on hold and retrieved are sent with **CPG** messages having the **event indicator** set to "progress". 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 [4]		ISUP reference: Q.1912.5 [1], Q.733 [i.6], clauses 2.5.2.1.1.1; 2.5.2.1.1.2		
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Call hold after answer, requested by the originating user* Ensure that the notifications that a call is placed on hold and retrieved are sent with **CPG** messages having the **event indicator** set to "progress". I-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: application/ISUP; CPG encapsulated in the MIME body				
ISUP parameter values					
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	INVITE(CPG, sendonly)	→		→	CPG(progress, hold)
	200 OK INVITE	←			
	ACK	→			
	INVITE(CPG, sendrecv)	→		→	CPG(progress, retrieve)
	200 OK INVITE	←			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

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

TP408004	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], Q.733 [i.6], clauses 2.5.2.1.1.1; 2.5.2.1.1.2**
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Call hold after answer, requested by the terminating user* Ensure that the notifications that a call is placed on hold and retrieved are sent with **CPG** messages having the **event indicator** set to "progress". I-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: application/ISUP; CPG encapsulated in the MIME body				
ISUP parameter values					
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
			Conversation		
	INVITE(CPG, sendonly)	←		←	CPG(progress, hold)
	200 OK INVITE	→			
	ACK	←			
	INVITE(CPG, sendrecv)	←		←	CPG(progress, retrieve)
	200 OK INVITE	→			
	ACK	←			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

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

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

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

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

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

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

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

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

5.3.9 Call Waiting (CW)

TP409001	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.9, Q.733 [i.6], clause 1.5.2.1.1
TSS reference:	ISUP-SIP-ISUP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Call waiting indication in 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 bodyMIME 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)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP409002	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause B.9, Q.733 [i.6], clause 1.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/CW			
SIP selection criteria				
ISUP selection criteria				
Test purpose	*Call waiting indication in 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
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP409003	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause B.9, Q.733 [i.6], clause 1.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/CW			
SIP selection criteria				
ISUP selection criteria				
Test purpose	*Call waiting indication in 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)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP409004</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clause B.9,
Q.733 [i.6], clause 1.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP-ISUP/SS/CW</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 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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">180 Ringing: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">CPG: Generic notification indicator "Call is a waiting call"</td></tr>
<tr><td rowspan="9">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td colspan="2">ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td colspan="2">IAM</td></tr>
<tr><td>183 Session Progress ACM)</td><td>←</td><td></td><td>←</td><td colspan="2">ACM</td></tr>
<tr><td>180 Ringing(CPG)</td><td>←</td><td></td><td>←</td><td colspan="2">CPG(waiting)</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>TP409005</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clause B.9,
Q.733 [i.6], clause 1.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP-ISUP/SS/CW</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">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">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</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">ACM or CPG: Generic notification indicator "Call is a waiting call"
REL: Cause #21 (call rejected)</td></tr>
<tr><td rowspan="5">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(waiting)</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing(ACM)</td></tr>
<tr><td>REL(#21)</td><td>←</td><td></td><td>←</td><td colspan="2">480 Temporarily Unavailable(REL)</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td>→</td><td colspan="2">ACK</td></tr>
</table>

TP409006	**SIP reference: RFC 3261 [4]**			**ISUP reference:** **Q.1912.5 [1], clause B.9,** **Q.733 [i.6], clause 1.5.2.1.1**	
TSS reference	ISUP-SIP-ISUP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**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 Message 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

TP409008	**SIP reference: RFC 3261 [4]**			**ISUP reference:** **Q.1912.5 [1], clause B.9,** **Q.733 [i.6], clause 1.5.2.1.1**	
TSS reference	ISUP-SIP-ISUP/SS/CW				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call waiting 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				
				→	REL(#19)
				←	RLC
	480 Temporarily Unavailable	←			
	ACK	→			

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

<table>
<tr><th>TP410001</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses B.6, B.7,
Q.732 [i.4], clause 2.5</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/Call Diversion</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">"Call is diverting" indication received in 180 Ringing

Verify that a call can be successfully established, if diversion occurs. The ACM contains the generic notification indicator set to "call is diverting", the call diversion information and the redirection number.
The Redirection reason is set to CV_redirection_reason.
CPG (alerting) is coded as if it has been mapped from the 180 Ringing (CPG).
O-MCGF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">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</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">ACM: BCI Called party status indicator "No indication"
Generic notification
Call diversion information
Redirection number
CPG: Event indicator=alerting</td></tr>
<tr><td rowspan="9">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td>SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE(IAM)</td></tr>
<tr><td>ACM(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><th>TP410002</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses B.6, B.7,
Q.732 [i.4], clause 2.5</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/Call Diversion</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">"Call is diverting" indication received in CPG

Verify that a call can be successfully established, if diversion occurs. The ACM contains the generic notification indicator set to "call is diverting", the call diversion information and the redirection number.
The Redirection reason is set to CV_redirection_reason.
180 Ringing (CPG (alerting)) is coded as if it has been mapped from the CPG.
I-MCGF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">183 Session Progress: Content-Type: application/ISUP; ACM encapsulated in the Message body
180 Ringing: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">ACM: BCI Called party status indicator "No indication"
Generic notification
Call diversion information
Redirection number
CPG: Event indicator=alerting</td></tr>
<tr><td rowspan="9">Comments</td><td>SIP-I</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE(IAM)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>183 Session Progress(ACM)</td><td>←</td><td></td><td>←</td><td>ACM(no indication)</td></tr>
<tr><td>180 Ringing(CPG)</td><td>←</td><td></td><td>←</td><td>CPG(alerting)</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

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

TP410003	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses B.6, B.7, Q.732 [i.4], clause 2.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**"Call diversion may occur" received in** 180 Ringing**(ACM)** Verify that a call can be successfully established, if diversion may occur. The encapsulated ACM indicates that "call diversion may occur" in the optional backward call indicators. The following CPG contains the generic notification indicator set to "call is diverting", the call diversion information and the redirection number, if diversion occurs. The CPG (progress) contains **CV_redirection_reason** in call diversion information and also Redirection number. The CPG (alerting) is coded as if it has been mapped from ACM, with RnNbRes parameter (optional). O-MCGF interworking.				
SIP parameter 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)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP410004</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clauses B.6, B.7,
Q.732 [i.4], clause 2.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP-ISUP/SS/Call Diversion</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">"Call diversion may occur" received in ACM
Verify that a call can be successfully established, if diversion may occur. The ACM indicates that "call diversion may occur" in the optional backward call indicators. The following CPG contains the generic notification indicator set to "call is diverting", the call diversion information and the redirection number, if diversion occurs.
The CPG (progress) contains CV_redirection_reason in call diversion information and also Redirection number. The CPG (alerting) is coded as if it has been mapped from ACM, with RnNbRes parameter (optional).
I-MCGF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body
183 Session Progress: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">ACM: BCI Called party status indicator "subscriber free", Optional backward call indicator: "Call diversion may occur"
CPG: Event information=progress, Call diversion information; Generic notification; Redirection number
CPG: Event information=alerting</td></tr>
<tr><td rowspan="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>180 Ringing(ACM)</td><td>←</td><td colspan="2"></td><td>←</td><td>ACM(free)</td></tr>
<tr><td>183 Session Progress(CPG)</td><td>←</td><td colspan="2"></td><td>←</td><td>CPG</td></tr>
<tr><td>183 Session Progress(CPG)</td><td>←</td><td colspan="2"></td><td>←</td><td>CPG(alerting)</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td colspan="2"></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td colspan="2"></td><td></td><td></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>

CV_redirection_reason TP410003, TP410004	
VA_1	No reply
VA_2	Deflection during alerting

<table>
<tr><th>TP410005</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses B.6, B.7,
Q.732 [i.4], clause 2.4.2</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">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="5">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="5">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="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(no indication)</td><td>←</td><td></td><td>←</td><td>183 Session Progress(ACM)</td></tr>
<tr><td>CPG1</td><td>←</td><td></td><td>←</td><td>183 Session Progress(CPG)</td></tr>
<tr><td>CPG2(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><th>TP410006</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clauses B.6, B.7,
Q.732 [i.4], 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.
I-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
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</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>183 Session Progress(ACM)</td><td>←</td><td></td><td>←</td><td colspan="2">ACM(no indication)</td></tr>
<tr><td>183 Session Progress(CPG)</td><td>←</td><td></td><td>←</td><td colspan="2">CPG1</td></tr>
<tr><td>180 Ringing(CPG)</td><td>←</td><td></td><td>←</td><td colspan="2">CPG2(alerting)</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>

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 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses B.6, B.7,
Q.732 [i.4], clause 2.5.2.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/Call Diversion</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Notification procedures for a diverting call - after the diverting exchange

Verify that the IUT can successfully pass on in both directions (on the leg after the diversion) all the diversion information from the diverting exchange.

It has to be checked that the following signalling information is passed on in the forward direction:
redirecting number (see note);
original called number (see note);
redirection information.
It has to be checked that the following signalling information is passed on in the backward direction:
redirection number restriction parameter (in ACM/CPG/ANM/CON).
O-MCGF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; IAM encapsulated in the MIME body
200 OK INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; ANM encapsulated in the MIME body</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="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 colspan="5">CASE A</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>183 Session Progress(ACM,no indication)</td></tr>
<tr><td>CPG</td><td>←</td><td></td><td>←</td><td>180 Ringing(CPG,alerting)</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 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></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 colspan="5"></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 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses B.6, B.7,
Q.732 [i.4], clause 2.5.2.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/Call Diversion</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Notification procedures for a diverting call - after the diverting exchange

Verify that the IUT can successfully pass on in both directions (on the leg after the diversion) all the diversion information from the diverting exchange
It has to be checked that the following signalling information is passed on in the forward direction:
redirecting number (see note);
original called number (see note);
redirection information.
It has to be checked that the following signalling information is passed on in the backward direction:
redirection number restriction parameter (in ACM/CPG/ANM/CON).
I-MCGF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; IAM encapsulated in the MIME body
200 OK INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; ANM encapsulated in the MIME body</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="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 colspan="5">CASE A</td></tr>
<tr><td>183 Session Progress(ACM)</td><td>←</td><td></td><td>←</td><td>ACM(no indication)</td></tr>
<tr><td>180 Ringing(CPG)</td><td>←</td><td></td><td>←</td><td>CPG(alerting)</td></tr>
<tr><td>200 OK INVITE(ANM)</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td 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>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>
<tr><td colspan="5"></td></tr>
<tr><td colspan="6">NOTE: Altered in Gateways.</td></tr>
</table>

TP410009	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses B.6, B.7, Q.731 [i.2], clause 3.5.2.4.1
TSS reference	ISUP-SIP-ISUP/SS/Call 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: multipart/mixed, 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
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP410010	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses B.6, B.7, Q.731 [i.2], clause 4.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/Call 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: multipart/mixed, 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
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP410011</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clauses B.6, B.7,
Q.731 [i.2], clause 4.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/Call Diversion</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/7</td></tr>
<tr><td>Test purpose</td><td colspan="5">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.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; IAM containing an Original called number called number encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: No original called number present</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>TP410012</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clauses B.6, B.7,
Q.731 [i.2], clause 4.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/Call Diversion</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/8</td></tr>
<tr><td>Test purpose</td><td colspan="5">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).</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; IAM containing an Original called number called number encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: Original called number "National number"</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>TP410013</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses B.6, B.7,
Q.731 [i.2], clause 4.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/Call Diversion</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 10/2 AND PICS 1/7</td></tr>
<tr><td>Test purpose</td><td colspan="5">Redirecting number in the outgoing international gateway
Verify that the outgoing international gateway checks and manipulates the redirecting number according to the procedures as defined for CLIP:
Discarding the redirecting number if case of bilateral agreements.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; IAM containing a Redirecting number encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: No Redirecting number present</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>TP410014</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses B.6, B.7,
Q.731 [i.2], clause 4.5.2.1.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/Call Diversion</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/7</td></tr>
<tr><td>Test purpose</td><td colspan="5">Redirecting number in the outgoing international gateway
Verify that the outgoing international gateway checks and manipulates the redirecting number according to the procedures as defined for CLIP:
Discarding the redirecting number, if the address is marked not available.
The PTC will send an IAM with an "address not available" RgNb.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; IAM containing a Redirecting number encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: No Redirecting number present</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>TP410015</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses B.6, B.7,
Q.732 [i.4], clause 2.5.2.3,
Q.731 [i.2], clause 3.5.2.3</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISUP-SIP-ISUP/SS/Call Diversion</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/7</td></tr>
<tr><td>Test purpose</td><td colspan="5">Redirecting number in the outgoing international gateway
Verify that the outgoing international gateway checks and manipulates the redirecting number according to the procedures as defined for CLIP:
Converting the redirecting number to international format with transparent transferral of address presentation restriction indicator.
The PTC will send an IAM with a national significant RgNb.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; IAM containing a Redirecting number "National number" encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: Redirecting number "International number"</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>TP410016</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clauses B.6, B.7,
Q.732 [i.4], clause 2.5.2.3,
Q.731 [i.2], clause 3.5.2.3</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/Call Diversion</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5">PICS 1/8</td></tr>
<tr><td>Test purpose</td><td colspan="5">Redirecting number in the incoming international gateway
Verify that the incoming international gateway checks and manipulates the redirecting number according to the procedures as defined for CLIP:
Converting the redirecting number to national format, if necessary (own country code).
The PTC will send an IAM with RgNb.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; ; IAM containing a Redirecting number "International number" encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: Redirecting number "national number"</td></tr>
<tr><td rowspan="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>

TP410017	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses B.6, B.7, Q.732 [i.4], clause 2.5.2.3, Q.731 [i.2], clause 3.5.2.3
TSS reference	ISUP-SIP-ISUP/SS/Call 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: multipart/mixed, 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
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP410018	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses B.6, B.7, Q.732 [i.4], clause 2.5.2.4, Q.731 [i.2], clause 3.5.2.4
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion				
SIP selection criteria					
ISUP selection criteria	PICS 10/5 AND PICS 1/8				
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: multipart/mixed, 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)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP410019	**SIP reference: RFC 3261 [4]**			**ISUP reference:** **Q.1912.5 [1], clauses B.6, B.7,** **Q.732 [i.4], clause 2.5.2.3,** **Q.731 [i.2], clause 3.5.2.3**	
TSS reference	ISUP-SIP-ISUP/SS/Call 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)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP410020	**SIP reference: RFC 3261 [4]**			**ISUP reference:** **Q.1912.5 [1], clauses B.6, B.7,** **Q.732 [i.4], clause 2.5.2.3,** **Q.731 [i.2], clause 3.5.2.3**	
TSS reference	ISUP-SIP-ISUP/SS/Call 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)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP410021	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses B.6, B.7, Q.731 [i.2], clause 5.5.2.3.1
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion				
SIP selection criteria					
ISUP selection criteria	PICS 1/8 AND PICS 10/6				
Test purpose	**Redirection number in the outgoing international gateway** Verify that the outgoing international gateway checks and manipulates the **redirection 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)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

5.3.11 CONF

TP411001	**SIP reference: RFC 3261 [4]**		**ISUP reference: Q.1912.5 [1], clause B.14, Q.734 [i.7], clause 1.6.15**		
TSS reference	ISUP-SIP-ISUP/SS/CONF				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Generic notification transfer "conference established" and "other party added" To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message: 1. assist a call set up from ISUP to SIP-I; 2. check that the notification "conference established" is received in the CPG from conferee at SIP-I; 3. check the notification "other party added" in the CPG. O-MGCF interworking.				
SIP parameter values	INFO/INVITE: Content-Type: multipart/mixed, 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)
				→	ACK
			Conversation		
	CASE A				
	CPG(conference established)	→		→	INVITE(CPG,sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	CASE B				
	CPG(conference established)	→		→	INFO(CPG)
				←	200 OK (INFO)
	CASE A				
	CPG(other party added)	→		→	INVITE(CPG,sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	CASE B				
	CPG(other party added)	→		→	INFO(CPG)
				←	200 OK (INFO)
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP411002	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause B.14, Q.734 [i.7], clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/CONF			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Generic notification transfer "conference established" and "other party added" To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message: 1. Assist a call set up from SIP-I to ISUP; 2. Check that the notification "conference established" is received in the CPG from conferee at the ISUP; 3. Check the notification "other party added" in the CPG. I-MGCF interworking.			
SIP parameter values	INFO/INVITE: Content-Type: multipart/mixed, 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	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	CASE A				
	INFO(CPG)	→		→	CPG(conference established)
	200 OK INFO	←			
	CASE B				
	INVITE (CPG)	→		→	CPG(conference established)
	200 OK INVITE	←			
	ACK	→			
	CASE A				
	INFO(CPG)	→		→	CPG(other party added)
	200 OK INFO	←			
	CASE B				
	INVITE (CPG)	→		→	CPG(other party added)
	200 OK INFO	←			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP411003	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.14, Q.734 [i.7], clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/CONF				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Generic notification transfer "conference established" and "isolated" To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message: 1. Assist a call set up from ISUP to SIP-I; 2. Check that the notification "conference established" is received in the CPG from conferee at the SIP-I; 3. Check the notification "isolated" in the CPG. O-MGCF interworking.				
SIP parameter values	INFO/INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; CPG encapsulated in the MIME body				
ISUP parameter values	CPG: Generic notification: conference established CPG: Generic notification: isolated				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	CASE A				
	CPG(conference established)	→		→	INFO(CPG)
				←	200 OK INFO
	CPG(isolated)	→		→	INFO(CPG)
				←	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)
	CASE B				
	CPG(conference established)	→		→	INVITE(CPG,sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	CPG(isolated)	→		→	INVITE(CPG,sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP411004	SIP reference: RFC 3261 [4]	ISUP reference: Q.1912.5 [1], clause B.14, Q.734 [i.7], clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/CONF	
SIP selection criteria		
ISUP selection criteria		
Test purpose	Generic notification transfer "conference established" and "isolated" To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message: 1. assist a call set up from SIP-I to ISUP; 2. check that the notification "conference established" is received in the CPG from conferee at SIP-I; 3. check the notification "isolated" in the CPG. I-MGCF interworking.	
SIP parameter values	INFO/INVITE: Content-Type: multipart/mixed, 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
	ACK	→			
			Conversation		
	CASE A				
	INFO(CPG)	→		→	CPG(conference established)
	200 OK INFO	←			
	INFO(CPG)	→		→	CPG(isolated)
	200 OK INFO	←			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC
	CASE B				
	INVITE (CPG)	→		→	CPG(conference established)
	200 OK INVITE	←			
	ACK	→			
	INVITE (CPG)	→		→	CPG(isolated)
	200 OK INVITE	←			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP411005	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause B.14, Q.734 [i.7], clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/CONF			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Generic notification transfer "conference established", "isolated" and "reattached" To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message: 1. Assist a call set up from ISUP to SIP-I; 2. Check that the notification "conference established" is received in the CPG from conferee at SIP-I; 3. Check the notification "reattached" in the CPG. O-MGCF interworking.			
SIP parameter values	INFO/INVITE: Content-Type: multipart/mixed, 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	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	CASE A				
	CPG(conference established)	→		→	INFO(CPG)
				←	200 OK INFO
	CPG(isolated)	→		→	INFO(CPG)
				←	200 OK INFO
	CPG(reattached)	→		→	INFO(CPG)
				←	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)
	CASE B				
	CPG(conference established)	→		→	INVITE(CPG,sendrecv)
				←	200 OK INVITE(sendrecv
				→	ACK
	CPG(isolated)	→		→	INVITE(CPG,sendonly)
				←	200 OK INVITE(recvonly)
				→	ACK
	CPG(reattached)	→		→	INFO(CPG,sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP411006	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause B.14, Q.734 [i.7], clause 1.6.15	
TSS reference	ISUP-SIP-ISUP/SS/CONF				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Generic notification transfer "conference established", "isolated" and "reattached" To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message: 1. Assist a call set up from SIP-I to ISUP; 2. Check that the notification "conference established" is received in the CPG from conferee at SIP-I; 3. Check the notification "reattached" in the CPG. I-MGCF interworking.				
SIP parameter values	INFO/INVITE: Content-Type: multipart/mixed, 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
	ACK	→			
			Conversation		
	CASE A				
	INFO(CPG)	→		→	CPG(conference established)
	200 OK INFO	←			
	INFO(CPG)	→		→	CPG(isolated)
	200 OK INFO	←			
	INFO(CPG)	→		→	CPG(reattached)
	200 OK INFO	←			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC
	CASE B				
	INVITE(CPG,sendrecv)	→		→	CPG(conference established)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
	INVITE(CPG,sendonly)	→		→	CPG(isolated)
	200 OK INVITE(recvonly)	←			
	ACK	→			
	INVITE(CPG,sendrecv)	→		→	CPG(reattached)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP411007	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause B.14, Q.734 [i.7], clause 1.6.15**	
TSS reference	ISUP-SIP-ISUP/SS/CONF				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Generic notification transfer "conference established", "other party added" and "other party disconnected" To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message: 1. assist a call set up from ISUP to SIP-I; 2. check the notification "other party disconnected" in the CPG. O-MGCF interworking.				
SIP parameter values	INFO/INVITE: Content-Type: multipart/mixed, 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	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	CASE A				
	CPG(conference established)	→		→	INFO(CPG)
				←	200 OK INFO
	CPG(other party added)	→		→	INFO(CPG)
				←	200 OK INFO
	CPG(other party disconnected)	→		→	INFO(CPG)
				←	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)
	CASE B				
	CPG(conference established)	→		→	INVITE(CPG,sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	CPG(other party added)	→		→	INVITE(CPG, sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	CPG(other party disconnected)	→		→	INVITE(CPG, sendrecv)
				←	200 OK INVITE (sendrecv)
				→	ACK
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP411008	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.14, Q.734 [i.7], clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/CONF				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Generic notification transfer "conference established", "other party added" and "other party disconnected" To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message: 1. Assist a call set up from SIP-I to ISUP; 2. Check the notification "other party disconnected" in the CPG. I-MGCF interworking.				
SIP parameter values	INFO/INVITE: Content-Type: multipart/mixed, 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
	ACK	→			
			Conversation		
	CASE A				
	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
	CASE B				
	INVITE(CPG,sendrecv)	→		→	CPG(conference established)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
	INVITE(CPG,sendrecv)	→		→	CPG(other party added)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
	INVITE(CPG,sendrecv)	→		→	CPG(other party disconnected)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

<table>
<tr><th>TP411009</th><th colspan="4">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clause B.14,
Q.734 [i.7], clause 1.6.15</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CONF</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Generic notification transfer "conference established", and disconnect the conference

To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message:
1. Assist a call set up from ISUP to SIP-I;
2. Check that the notification "conference established" is received in the CPG from conferee at ISUP;
3. Release the conference.
O-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INFO/INVITE: Content-Type: multipart/mixed, 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</td></tr>
<tr><td rowspan="20">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>CASE A</td><td></td><td></td><td></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>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>CASE B</td><td></td><td></td><td></td><td></td></tr>
<tr><td>CPG(conference established)</td><td>→</td><td></td><td>→</td><td>INVITE(CPGsendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(sendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>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>

TP411010	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause B.14, Q.734 [i.7], clause 1.6.15**	
TSS reference	ISUP-SIP-ISUP/SS/CONF				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Generic notification transfer "conference established", and disconnect the conference To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message: 1. Assist a call set up from SIP-I to ISUP; 2. Check that the notification "conference established" is received in the INFO(CPG) from conferee at SIP-I; 3. Release the conference. I-MGCF interworking.				
SIP parameter values	INFO/INVITE: Content-Type: multipart/mixed, 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
	ACK	→			
			Conversation		
	CASE A				
	INFO(CPG)	→		→	CPG(conference established)
	200 OK INFO	←			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC
	CASE B				
	INVITE(CPG,sendrecv)	→		→	CPG(conference established)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

5.3.12 ECT

TP412001	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8, Q.732.7 [i.5], clause 7.5.2.1.1.1 a)**
TSS reference	ISUP-SIP-ISUP/SS/ECT	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Capability of sending a call transfer number for the active user** Verify that the IUT is able to send the Generic notification parameter "Call transfer active", the service activation parameter "call transfer" and the call transfer number, received in the ISUP FAC, in an INFO request for the active user. O-MGCF interworking.	
SIP parameter values	INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body	
ISUP parameter values	FAC: Generic notification=call transfer active, Service activation=call transfer, Call transfer number (PIXIT)	
Comments		

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

TP412002	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8, Q.734 [i.7], clause 1.6.15**
TSS reference	ISUP-SIP-ISUP/SS/ECT	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Capability of sending the call transfer number for the active user** Verify that the IUT is able to send the Generic notification parameter "Call transfer active", the service activation parameter "call transfer" and the call transfer number, received in the INFO request containing the encapsulated FAC, in a ISUP FAC for the active user. I-MGCF interworking.	
SIP parameter values	INFO: Content-Type: application/ISUP; FAC encapsulated in the Message body	
ISUP parameter values	FAC: Generic notification=call transfer active, Service activation=call transfer, Call transfer number (PIXIT)	
Comments		

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

TP412005	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8, Q.732.7 [i.5], clause 7.5.2.1.1.1 a)
TSS reference	ISUP-SIP-ISUP/SS/ECT			
SIP selection criteria				
ISUP selection criteria				
Test purpose	**Capability of sending the call transfer number for the held user** Verify that the IUT is able to transfer the call transfer number received in an ISUP FAC and sent in INFO request containing the FAC for the held user. O-MGCF interworking.			
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body new session with new INVITE to CTNb			
ISUP parameter values	CPG: Event indicator=progress, Generic notification=hold FAC: Generic notification=call transfer active, Service activation=call transfer, Call transfer number(PIXIT)			
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	CPG(hold)	→		→	INVITE(CPG, sendonly)
				←	200 OK INVITE(recvonly)
				→	ACK
	FAC(call transfer active, CTNb)	→		→	INFO(FAC)
				←	200 OK INFO
				→	INVITE(sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP412006</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8,
Q.734 [i.7], clause 1.6.15</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/CONF</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Capability of sending the call transfer number for the active user
Verify that the IUT is able to transfer the call transfer number received in an ISUP FAC and sent in INFO request containing the FAC for the held user.
I-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; CPG encapsulated in the MIME body
INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">CPG: Event indicator=progress, Generic notification=hold
FAC: Generic notification=call transfer active, Service activation=call transfer, Call transfer number(PIXIT)</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>ACK</td><td>→</td><td></td><td></td><td></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>

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TP412009	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8, Q.732.7 [i.5], clause 7.5.2.1.1.1 a)**	
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Loop prevention procedure – initiation** Verify that the SUT is able to transfer the loop request received in an ISUP LOP in an INFO request containing the LOP message. SUT is able to transfer the loop response received in an ISUP LOP in an SIP INFO request containing the ISUP LOP message. O-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body 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, Service activation=call transfer, Call transfer number(PIXIT)				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	CPG(hold)	→		→	INVITE(CPG, sendonly)
				←	200 OK INVITE(recvonly)
				→	ACK
	LOP(request)	→		→	INFO(LOP)
				←	200 OK INFO
	LOP(response)	←		←	INFO(LOP)
				→	200 OK INFO
	FAC(call transfer active, CTNb)	→		→	INFO(FAC)
				←	200 OK INFO
				→	INVITE(sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	REL	→		→	BYE(REL)
	RLC(RLC)	←		←	200 OK BYE

TP412010	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8, Q.734 [i.7], clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Loop prevention procedure – initiation** Verify that the SUT is able to transfer the loop request received in an INFO request containing the ISUP LOP message. Verify that the SUT is able to transfer the loop response received in an ISUP LOP message in the SIP INFO request containing the ISUP LOP message. I-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body 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, Service activation=call transfer, Call transfer number(PIXIT)				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	INVITE(CPG, sendonly)	→		→	CPG(hold)
	200 OK INVITE(recvonly)	←			
	ACK	→			
	INFO(LOP)	→		→	LOP(request)
	200 OK INFO	←			
	INFO(LOP)	←		←	LOP(response)
	200 OK INFO	→			
	INFO(FAC)	→		→	FAC(call transfer active, CTNb)
	200 OK INFO	←			
	INVITE(sendrecv)	→			
	200 OK INVITE(sendrecv)	←			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP412011	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8, Q.732.7 [i.5], clause 7.5.2.1.1.1 a)
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Loop prevention procedure - unsuccessful on timer expiry** To verify that the SUT is able to transfer the loop request received in an ISUP LOP in an INFO request containing the LOP message. Verify that the connection is unsuccessful if the loop detection procedure is unsuccessful. The connection is released from the remote end. O-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; LOP encapsulated in the MIME body				
ISUP parameter values	CPG: Event indicator=progress, 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)
				→	ACK
			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)

<table>
<tr><th>TP412012</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8,
Q.734 [i.7], clause 1.6.15</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP-ISUP/SS/CONF</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">Loop prevention procedure - unsuccessful on timer expiry
To verify that SUT is able to transfer the loop request received in an INFO request containing the LOP message in an ISUP LOP message. Verify that the connection is unsuccessful if the loop detection procedure is unsuccessful. The connection is released from the remote end.
I-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; CPG encapsulated in the MIME body
INFO: Content-Type: application/ISUP; LOP encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">CPG: Event indicator=progress, Generic notification=hold
LOP: request: Call transfer reference</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>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>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(LOP)</td><td>→</td><td></td><td>→</td><td colspan="2">LOP(request)</td></tr>
<tr><td>200 OK INFO</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td colspan="2">REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td colspan="2">RLC</td></tr>
</table>

TP412013	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8, Q.732.7 [i.5], clause 7.5.2.1.1.1 a)
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Loop prevention procedure - successful on timer expiry** Verify that the SUT is able to transfer the loop request received in an ISUP LOP in an INFO request containing the LOP message. Verify that the connection is successful if the loop detection procedure is unsuccessful. O-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body 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, Service activation=call transfer, Call transfer number(PIXIT)				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			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 [4]				ISUP reference: Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8, Q.734 [i.7], clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Loop prevention procedure - successful on timer expiry** Verify that the SUT is able to transfer the loop request received in an INFO request containing the LOP message in an ISUP LOP message. Verify that the connection is successful if the loop detection procedure is unsuccessful. I-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body 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, Service activation=call transfer, Call transfer number(PIXIT)				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			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 [4]				ISUP reference: Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8, Q.732.7 [i.5], clause 7.5.2.1.1.1 a)
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Facility message with generic notification sent to the remote user** Verify that the SUT is able to transfer the generic notification "call transfer, active" or "call transfer, alerting" and the **service activation** parameter set to "call transfer" received in an ISUP FAC in a SIP INFO request containing the ISUP FAC. O-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body				
ISUP parameter values	CPG: Event indicator=progress, Generic notification=hold FAC: Generic notification=call transfer active, Service activation=call transfer, Call transfer number(PIXIT)				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			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 [4]				ISUP reference: Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8, Q.734 [i.7], clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Facility message with generic notification sent to the remote user** Verify that the SUT is able to transfer the generic notification **generic notification** set to "call transfer, active" or "call transfer, alerting" and the **service activation** parameter set to "call transfer" received in a SIP-I INFO request containing the ISUP FAC message in an ISUP FAC message. I-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body				
ISUP parameter values	CPG: Event indicator=progress, Generic notification=hold FAC: Generic notification=call transfer active, Service activation=call transfer, Call transfer number(PIXIT)				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			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

<table>
<tr><th>TP412017</th><th colspan="5">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8,
Q.732.7 [i.5], clause 7.5.2.1.1.1 a)</th></tr>
<tr><td>TSS reference</td><td colspan="6">ISUP-SIP-ISUP/SS/ECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="6"></td></tr>
<tr><td>Test purpose</td><td colspan="6">Call progress message with generic notification sent to the remote user
Verify that the transfer the CPG with the generic notification set to "call transfer, active" and the service activation parameter set to "call transfer" in a SIP-I INFO request containing the ISUP CPG message.
O-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="6">INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="6">CPG: Generic notification=call transfer active, Service activation=call transfer, Call transfer number (PIXIT)</td></tr>
<tr><td rowspan="12">Comments</td><td>ISUP</td><td></td><td>SUT</td><td></td><td colspan="2">SIP-I</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE(IAM)</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing(ACM)</td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>CPG(call transfer active, CTNb)</td><td>→</td><td></td><td>→</td><td colspan="2">INFO(CPG)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK INFO</td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE(ANM)</td></tr>
<tr><td></td><td></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>TP412018</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8,
Q.734 [i.7], clause 1.6.15</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/ECT</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Call progress message with generic notification sent to the remote user
Verify that the SUT is able to transfer the ISUP CPG with the generic notification set to "call transfer, active" and the service activation parameter set to "call transfer" contained in SIP-I INFO request in an ISUP CPG. The held user is retrieved by receiving a re-INVITE sendrecv.
I-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; CPG encapsulated in the MIME body
INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">CPG: Event indicator=progress, Generic notification=hold
CPG: Generic notification=call transfer active, Service activation=call transfer, Call transfer number(PIXIT)</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>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>INVITE(CPG, sendonly)</td><td>→</td><td></td><td>→</td><td>CPG(hold)</td></tr>
<tr><td>200 OK INVITE(recvonly)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>INFO(CPG)</td><td>→</td><td></td><td>→</td><td>CPG(call transfer active, CTNb)</td></tr>
<tr><td>200 OK INFO</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>INVITE(sendrecv)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK INVITE(sendrecv)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE(REL)</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE(RLC)</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP412019	SIP reference: RFC 3261 [4]					ISUP reference: Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8, Q.734 [i.7], clause 1.6.15
TSS reference	ISUP-SIP-ISUP/SS/ECT					
SIP selection criteria						
ISUP selection criteria						
Test purpose	**Call transfer number - removal of number** Verify that the exchange removes the **call transfer number** in the SIP-I INFO request containing a **FAC** or **CPG** before sending it to the next exchange, if its indicator is set to "presentation restricted" and there is no bilateral agreement to transfer the number.					
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC(CTNb=restricted) encapsulated in the MIME body					
ISUP parameter values	CPG: Event indicator=progress, Generic notification=hold FAC: Generic notification=call transfer active, Service activation=call transfer, no Call transfer number(PIXIT)					
Comments	**SIP-I**		**SUT**		**ISUP**	
	INVITE(IAM)	→		→	IAM	
	180 Ringing(ACM)	←		←	ACM	
	200 OK INVITE(ANM)	←		←	ANM	
	ACK	→				
			Conversation			
	INVITE(CPG, sendonly)	→		→	CPG(hold)	
	200 OK INVITE(recvonly)	←				
	ACK	→				
	INFO(FAC)	→		→	FAC(call transfer active)	
	200 OK INFO	←				
	INVITE(sendrecv)	→				
	200 OK INVITE(sendrecv)	←				
	ACK	→				
	BYE(REL)	→		→	REL	
	200 OK BYE(RLC)	←		←	RLC	

TP412020	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8, Q.734 [i.7], clause 1.6.15**
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call transfer number - conversion to international number** Verify that the IUT converts the **call transfer number** contained in the SIP-I INFO request into international format. The nature of address indicator shall be set to "international number".				
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC(CTNb=national) encapsulated in the Message body				
ISUP parameter values	CPG: Event indicator=progress, Generic notification=hold FAC: Generic notification=call transfer active, Service activation=call transfer, Call transfer number=international(PIXIT)				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	INVITE(CPG, sendonly)	→		→	CPG(hold)
	200 OK INVITE(recvonly)	←			
	ACK	→			
	INFO(FAC)	→		→	FAC(call transfer active, CTNb)
	200 OK INFO	←			
	INVITE(sendrecv)	→			
	200 OK INVITE(sendrecv)	←			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP412021	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8, Q.732.7 [i.5], clause 7.5.2.1.1.1 a)
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Call transfer number - removal of own country code** Verify that the IUT removes the country code in the address signals of the **call transfer number** if it is the network's own country code contained in the ISUP FAC message. The nature of address indicator shall be set to "national (significant) number"				
SIP parameter values	INVITE: Content-Type: multipart/mixed, 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, Service activation=call transfer, Call transfer number=international(PIXIT)				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			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 [4]				ISUP reference: Q.1912.5 [1], clauses 5.4.3, 5.4.3.2, B.8, Q.732.7 [i.5], clause 7.5.2.1.1.1 a)
TSS reference	ISUP-SIP-ISUP/SS/ECT				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**ECT - Interaction with SUB** Verify that if the IUT is able to transfer the sub-address in the **access transport** parameter in the ISUP **FAC** message contained in the SIP-I INFO request in ISUP FAC message and vice versa received in an ISUP FAC message in a SIP-I INFO request containing the ISUP FAC message. These are the calling sub-address for incoming calls and the connected sub-address for outgoing calls. O-MGCF interworking.				
SIP parameter values	INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body				
ISUP parameter values	FAC: Generic notification=call transfer active, Service activation=call transfer, Call transfer number(PIXIT) FAC: ATP contained the connected sub address				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			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)

3PTY

<table>
<tr><th>TP413001</th><th colspan="3">SIP reference: RFC 3261 [4]</th><th colspan="2">ISUP reference:
Q.1912.5 [1], clause B.15,
Q.734.2 [i.8], clause 2.4; 2.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/3PTY</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Served user initiates 3PTY
Verify that the served user with two active calls is located, can successfully join this call (remote held user) to a three-way conversation, and notify the implied remote party accordingly.
The IUT should transfer an ISUP CPG message with the generic notification indicator set to "conference established" in a SIP-I INFO request containing the ISUP CPG message. The event indicator in the CPG should be set to "progress":
1. setup a call to user B;
2. put this call on hold;
3. join this call to a conference.
O-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: multipart/mixed, 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</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></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Conversation</td><td></td></tr>
<tr><td>CPG(hold)</td><td>→</td><td></td><td>→</td><td>INVITE(CPG, sendonly)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(recvonly)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>CPG(conference established)</td><td>→</td><td></td><td>→</td><td>INVITE(CPG, sendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(sendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td 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>TP413002</th><th colspan="2">SIP reference: RFC 3261 [4]</th><th colspan="3">ISUP reference:
Q.1912.5 [1], clause B.15,
Q.734.2 [i.8], clause 2.4; 2.2.1</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/3PTY</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Served user initiates 3PTY
Verify that the served user with two active calls is located, can successfully join this call (remote held user) to a three-way conversation, and notify the implied remote party accordingly.
The IUT should send a CPG message with the generic notification indicator set to "conference established" to both implied parties. The event indicator in the CPG should be set to "progress":
1. setup a call to user B;
2. put this call on hold;
3. join this call to a conference.
I-MGCF interworking</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: multipart/mixed, 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</td></tr>
<tr><td rowspan="16">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>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 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>

TP413003	**SIP reference: RFC 3261 [4]**	**ISUP reference: Q.1912.5 [1], clause B.15, Q.734.2 [i.8], clause 2.4; 2.2.1**
TSS reference	ISUP-SIP-ISUP/SS/3PTY	
SIP selection criteria		
ISUP selection criteria		
Test purpose	**Served user initiates 3PTY** Verify that the served user with two active calls is located, can successfully join this call (remote active user) to a three-way conversation, and notify the implied remote party accordingly. The IUT should send a **CPG** message with the **generic notification indicator** set to "conference established" to both implied parties. The **event indicator** in the **CPG** should be set to "progress": setup a call to user B; establish a conference. O-MGCF interworking.	
SIP parameter values	INFO/INVITE: Content-Type: multipart/mixed, 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)
				→	ACK
			Conversation		
	CASE A				
	CPG(conference established)	→		→	INFO(CPG)
				←	200 OK INFO
	CASE B				
	CPG(conference established)	→		→	INVITE(CPG, sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP413004	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.15, Q.734.2 [i.8], clause 2.4; 2.2.1
TSS reference	ISUP-SIP-ISUP/SS/3PTY				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Served user initiates 3PTY** Verify that the served user with two active calls is located, can successfully join this call (remote 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/INVITE: Content-Type: multipart/mixed, 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
	ACK	→			
			Conversation		
	INFO(CPG)	→		→	CPG(conference established)
	200 OK INFO	←			
	CASE B				
	INVITE(CPG,sendrecv)	→		→	CPG(conference established)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP413005	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause B.15, Q.734.2 [i.8], clause 2.5.2.1.1.3 a	
TSS reference	ISUP-SIP-ISUP/SS/3PTY				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Served user creates a private communication with a remote user** Verify that a 3PTY call can successfully create private communication with the active-held user. The appropriate notification received in a ISUP CPG and is sent in INVITE/INFO (**CPG)** messages to the SIP-I. O-MGCF interworking.				
SIP parameter values	INFO/INVITE: Content-Type: multipart/mixed, 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)
				→	ACK
			Conversation		
	CASE A				
	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)

TP413005	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause B.15, Q.734.2 [i.8], clause 2.5.2.1.1.3 a**
	CASE B				
	CPG 1(hold)	→		→	INVITE(CPG, sendonly)
				←	200 OK INVITE(recvonly)
				→	ACK
	CPG 2(conference established)	→		→	INVITE(CPG, sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	CPG 3(conference disconnected)	→		→	INVITE(CPG,sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	CPG 4(hold)	→		→	INVITE(CPG, sendonly)
				←	200 OK INVITE(recvonly)
				→	ACK
	CPG 5(retrieve)	→		→	INVITE(CPG, sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
		Conversation			
	CPG 6(conference established)	→		→	INVITE(CPG,sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP413006	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.15, Q.734.2 [i.8], clause 2.5.2.1.1.3 a
TSS reference	ISUP-SIP-ISUP/SS/3PTY				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Served user creates a private communication with a remote user** Verify that a 3PTY call can successfully create private communication with the active-held user. The appropriate notification received in a INVITE/INFO (CPG) and is sent in **CPG** messages to the ISUP. I-MGCF interworking.				
SIP parameter values	INFO/INVITE: Content-Type: multipart/mixed, 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
	ACK	→			
			Conversation		
	CASE A				
	INVITE(CPG, sendonly)	→		→	CPG(hold)
	200 OK INVITE(recvonly)	←			
	ACK	→			
	INVITE(CPG, sendrecv)	→		→	CPG(conference established)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
	INFO(CPG)	→		→	CPG(conference disconnected)
	200 OK INFO	←			
	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

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

TP413007	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.15, Q.734.2 [i.8], clause 2.5.2.1.1.3 a
TSS reference	ISUP-SIP-ISUP/SS/3PTY				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Served user creates a private 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/INVITE: Content-Type: multipart/mixed, 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)
				→	ACK
			Conversation		
	CASE A				
	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)
	CASE B				
	CPG(conference established)	→		→	INVITE(CPG,sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	CPG(conference disconnected)	→		→	INVITE(CPG,sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
			Conversation		
	CPG(conference established)	→		→	INVITE(CPG,sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP413008	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause B.15, Q.734.2 [i.8], clause 2.5.2.1.1.3 a
TSS reference	ISUP-SIP-ISUP/SS/3PTY			
SIP selection criteria				
ISUP selection criteria				
Test purpose	**Served user creates a private 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/INVITE: Content-Type: multipart/mixed, 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
	ACK	→			
			Conversation		
	CASE A				
	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
	CASE B				
			Conversation		
	INVITE(CPG,sendrecv)	→		→	CPG(conference established)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
	INVITE(CPG,sendrecv)	→		→	CPG(conference disconnected)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
			Conversation		
	INVITE(CPG,sendrecv)	→		→	CPG(conference established)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP413009	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause B.15, Q.734.2 [i.8], clause 2.5.2.1.1.3 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/INVITE: Content-Type: multipart/mixed, 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)
				→	ACK
			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)
	CASE B				
	CPG(conference established)	→		→	INVITE(CPG,sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	CPG(conference disconnected)	→		→	INVITE(CPG,sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP413010	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.15, Q.734.2 [i.8], clause 2.5.2.1.1.3 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/INVITE: Content-Type: multipart/mixed, 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
	ACK	→			
			Conversation		
	CASE A				
	INVITE(CPG,sendrecv)	→		→	CPG(conference established)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
	INVITE(CPG,sendrecv)	→		→	CPG(conference disconnected)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC
	CASE B				
	INVITE(CPG,sendrecv)	→		→	CPG(conference established)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
	INVITE(CPG,sendrecv)	→		→	CPG(conference disconnected)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP413011	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.15, Q.734.2 [i.8], clause 2.5.2.1.1.3 b
TSS reference	ISUP-SIP-ISUP/SS/3PTY				
SIP selection criteria					
ISUP selection criteria					
Test purpose	**Served user disconnects 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/INVITE: Content-Type: multipart/mixed, 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)
				→	ACK
			Conversation		
	CASE A				
	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)
	CASE B				
	CPG(hold)	→		→	INVITE(CPG, sendonly)
				←	200 OK INVITE(recvonly)
				→	ACK
	CPG(conference established)	→		→	INVITE(CPG, sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	CPG(conference disconnected)	→		→	INVITE(CPG,sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	CPG(hold)	→		→	INVITE(CPG, sendonly)
				←	200 OK INVITE(recvonly)
				→	ACK
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP413012	**SIP reference: RFC 3261 [4]**				**ISUP reference:** **Q.1912.5 [1], clause B.15,** **Q.734.2 [i.8], clause 2.5.2.1.1.3 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/INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; CPG encapsulated in the MIME body				
ISUP parameter values	CPG: Event indicator=progress, Generic notification=hold CPG: Event indicator=progress, Generic notification=conference established CPG: Event indicator=progress, Generic notification=conference disconnected				
Comments	**SIP-I**		**SUT**		**ISUP**
	INVITE(IAM)	→		→	IAM
	180 Ringing(ACM)	←		←	ACM
	200 OK INVITE(ANM)	←		←	ANM
	ACK	→			
			Conversation		
	CASE A				
	INVITE(CPG, sendonly)	→		→	CPG(hold)
	200 OK INVITE(recvonly)	←			
	ACK	→			
	INVITE(CPG, sendrecv)	→		→	CPG(conference established)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
	INFO(CPG)	→		→	CPG(conference disconnected)
	200 OK INFO	←			
	INVITE(CPG, sendonly)	→		→	CPG(hold)
	200 OK INVITE(recvonly)	←			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC
	CASE B				
	INVITE(CPG, sendonly)	→		→	CPG(hold)
	200 OK INVITE(recvonly)	←			
	ACK	→			
	INVITE(CPG, sendrecv)	→		→	CPG(conference established)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
	INVITE(CPG,sendrecv)	→		→	CPG(conference disconnected)
	200 OK INVITE(sendrecv)	←			
	ACK	→			
	INVITE(CPG, sendonly)	→		→	CPG(hold)
	200 OK INVITE(recvonly)	←			
	ACK	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

5.3.14 User-to-user service

5.3.14.1 User-to-user service 1

TP414001	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.21, Q.737 [i.10], clauses 1.1.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Service 1 implicit request: User-to-user information in the IAM* Ensure that the SUT can successfully transfer the User-to-user service 1 implicit request in the encapsulated IAM. O-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, 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)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP414002	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.21, Q.737 [i.10], clauses 1.1.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Service 1 implicit request: User-to-user information in the INVITE* Ensure that the SUT can successfully transfer the User-to-user service 1 implicit request in the encapsulated IAM. I-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, 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
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP414003	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause B.21, Q.737 [i.10], clauses 1.1.5.2.3 and 4	
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Service 1 explicit request: User-to-user information in the INVITE* 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: multipart/mixed, 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 not essential				
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)

TP414004	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause B.21, Q.737 [i.10], clauses 1.1.5.2.3 and 4	
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Service 1 explicit request: User-to-user indicator in the INVITE* Ensure that the SUT can successfully transfer the User-to-user service 1 explicit request essential received in the IAM. O-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; IAM containing the user-to-user indicator parameter encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM containing the user-to-user indicator parameter encapsulated in the MIME body				
ISUP parameter values	IAM: User-to-user information parameter, User-to-user indicator = service 1 explicit request essential 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)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP414005	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.21, Q.737 [i.10], clauses 1.1.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Service 1 explicit request: User-to-user indicator in the INVITE* Ensure that the SUT can successfully transfer the User-to-user service 1 explicit request essential received in the encapsulated IAM. I-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; IAM containing the user-to-user indicator parameter encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM containing the user-to-user indicator parameter encapsulated in the MIME body				
ISUP parameter values	IAM: User-to-user information parameter, User-to-user indicator = service 1 explicit request essential 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
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP414006	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause B.21, Q.737 [i.10], clauses 1.1.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Service 1 implicit response: User-to-user information in the ACM* Ensure that the SUT can successfully transfer the User-to-user service 1 implicit response in the encapsulated ACM. O-MGCF interworking.				
SIP parameter values	Service 1 implicit response: User-to-user information in the 180 Ringing INVITE: Content-Type: multipart/mixed, 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)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP414007	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause B.21, Q.737 [i.10], clauses 1.1.5.2.3 and 4**
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Service 1 implicit response: User-to-user information in the ACM Ensure that the SUT can successfully transfer the User-to-user service 1 implicit response in the encapsulated ACM. I-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, 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
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP414008	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clause B.21, Q.737 [i.10], clauses 1.1.5.2.3 and 4**
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	*Service 1 explicit response service 1 not provided in the ACM* Ensure that the SUT can successfully transfer the User-to-user service 1 explicit response not provided in the encapsulated ACM. O-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, 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 not essential ACM: User-to-user indicator set to service 1 not provided response				
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)

TP414009	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause B.21, Q.737 [i.10], clauses 1.1.5.2.3 and 4	
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Service 1 explicit response service 1 not provided in the ACM Ensure that the SUT can successfully transfer the User-to-user service 1 explicit response not provided in the ACM. I-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, 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 not essential ACM: User-to-user indicator set to service 1 not provided response				
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

TP414010	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause B.21, Q.737 [i.10], clauses 1.1.5.2.3 and 4	
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can 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: multipart/mixed, 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)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP414011	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause B.21, Q.737 [i.10], clauses 1.1.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS1			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that the SUT can 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: multipart/mixed, 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
	ACK	→			
			Conversation		
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

5.3.14.2 User-to-user service 2

TP414101	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 5.4.3, B.21, Q.737 [i.10], clauses 1.2.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS2				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Service 2 request not essential transferred in the INVITE Ensure that the SUT can successfully transfer the User-to-user service 2 explicit request and User-to user information in the encapsulated IAM. An additional User-to-user information is sent in a USR message encapsulated in an INFO request. O-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; IAM containing the user-to-user indicator and User-to-user information encapsulated in the MIME body INFO: Content-Type: application/ISUP; USR containing the User-to-user information parameter encapsulated in the MIME body				
ISUP parameter 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)
				→	ACK
			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 [4]**			**ISUP reference: Q.1912.5 [1], clauses 5.4.3, B.21, Q.737 [i.10], clauses 1.2.5.2.3 and 4**	
TSS reference	ISUP-SIP-ISUP/SS/UUS2				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Service 2 request not essential transferred in the IAM Ensure that the SUT can successfully transfer the User-to-user service 2 explicit request in the encapsulated IAM. An additional User-to-user information is sent in a USR message encapsulated in an INFO request. O-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; IAM containing the user-to-user indicator encapsulated in the MIME body INFO: Content-Type: application/ISUP; USR containing the User-to-user information parameter encapsulated in the MIME body				
ISUP parameter 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
	ACK	→			
			Conversation		
	INFO(USR)	→		→	USR
	200 OK INFO	←			
	INFO(USR)	←		←	USR
	200 OK INFO	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP414103	**SIP reference: RFC 3261 [4]**			**ISUP reference: Q.1912.5 [1], clause B.21, Q.737 [i.10], clauses 1.2.5.2.3 and 4**	
TSS reference	ISUP-SIP-ISUP/SS/UUS2				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Service 2 response not provided transferred in the ACM Ensure that the SUT can successfully transfer the User-to-user service 2 explicit response not provided in the encapsulated ACM. I-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; IAM containing the user-to-user information parameter encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM containing the user-to-user indicator parameter encapsulated in the MIME body				
ISUP parameter values	IAM: User-to-user information parameter, User-to-user indicator set to service 2 request ACM: User-to-user indicator set to service 2 not provided response				
Comments	**ISUP**		**SUT**		**SIP-I**
	IAM	→		→	INVITE(IAM)
	ACM	←		←	180 Ringing(ACM)
	ANM	←		←	200 OK INVITE(ANM)
				→	ACK
			Conversation		
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

<table>
<tr><th>TP414104</th><th colspan="4">SIP reference: RFC 3261 [4]</th><th>ISUP reference:
Q.1912.5 [1], clause B.21,
Q.737 [i.10], clauses 1.2.5.2.3 and 4</th></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/UUS2</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Service 2 response not provided transferred in the ACM

Ensure that the SUT can successfully transfer the User-to-user service 2 explicit response not provided in the encapsulated ACM. I-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; IAM containing the user-to-user indicator parameter encapsulated in the MIME body
180 Ringing: Content-Type: application/ISUP; ACM containing the user-to-user indicator parameter encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">IAM: User-to-user information parameter, User-to-user indicator
ACM: User-to-user indicator set to service 2 not provided response</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>

5.3.14.3 User-to-user service 3

TP414201	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 5.4.3, B.21, Q.737 [i.10], clauses 1.2.5.2.3 and 4**
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that 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: multipart/mixed, Content-Type: application/ISUP ; IAM containing the user-to-user indicator encapsulated in the MIME body INFO: Content-Type: application/ISUP; USR containing the User-to-user information parameter encapsulated in the MIME body				
ISUP parameter 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)
				→	ACK
			Conversation		
	USR	→		→	INFO(USR)
				←	200 OK INFO
	USR	←		←	INFO(USR)
				→	200 OK INFO
	USR	←		←	INFO(USR)
				→	200 OK INFO
	USR	→		→	INFO(USR)
				←	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP414202	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 5.4.3, B.21, Q.737 [i.10], clauses 1.2.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can successfully transfer the User-to-user service 3 explicit request in the encapsulated IAM. Additional User-to-user information is sent in several USR message encapsulated in an INFO request. I-MGCF interworking.				
SIP parameter values	INVITE: Content-Type: multipart/mixed, Content-Type: application/ISUP ; IAM containing the user-to-user indicator encapsulated in the MIME body INFO: Content-Type: application/ISUP; USR containing the User-to-user information parameter encapsulated in the MIME body				
ISUP parameter 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
	ACK	→			
			Conversation		
	INFO(USR)	→		→	USR
	200 OK INFO	←			
	INFO(USR)	←		←	USR
	200 OK INFO	→			
	INFO(USR)	←		←	USR
	200 OK INFO	→			
	INFO(USR)	→		→	USR
	200 OK INFO	←			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP414203	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 5.4.3, B.21, Q.737 [i.10], clauses 1.2.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that a User-to-user request service 3 encapsulated in an INFO request during the confirmed state can successful proceeded. The User-to-user information is passed on in several encapsulated USR messages. O-MGCF interworking.				
SIP parameter 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)
				→	ACK
			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)

<table>
<tr><td>TP414204</td><td colspan="2">SIP reference: RFC 3261 [4]</td><td colspan="3">ISUP reference:
Q.1912.5 [1], clauses 5.4.3, B.21,
Q.737 [i.10], clauses 1.2.5.2.3 and 4</td></tr>
<tr><td>TSS reference</td><td colspan="5">ISUP-SIP-ISUP/SS/UUS3</td></tr>
<tr><td>SIP selection criteria</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria</td><td colspan="5"></td></tr>
<tr><td>Test purpose</td><td colspan="5">Ensure that a User-to-user request service 3 encapsulated in an INFO request during the confirmed state can successful proceeded. The User-to-user information is passed on in several encapsulated USR messages.
I-MGCF interworking.</td></tr>
<tr><td>SIP parameter values</td><td colspan="5">INFO: Content-Type: application/ISUP; FAR containing the user-to-user indicator encapsulated in the MIME body
INFO: Content-Type: application/ISUP; FAA containing the user-to-user indicator encapsulated in the MIME body
INFO: Content-Type: application/ISUP; USR containing the User-to-user information parameter encapsulated in the MIME body</td></tr>
<tr><td>ISUP parameter values</td><td colspan="5">FAR: User-to-user indicator service 3 request not essential
FAA: User-to-user indicator service 3 response provided
USR: User-to-user information</td></tr>
<tr><td rowspan="26">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>INFO(FAR)</td><td>→</td><td></td><td>→</td><td>FAR</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(FAA)</td><td>←</td><td></td><td>←</td><td>FAA</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>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>

TP414205	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause B.21, Q.737 [i.10], clauses 1.2.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS3			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that the SUT can successfully transfer the User-to-user service 3 explicit response in the encapsulated ANM. O-MGCF interworking.			
SIP parameter values	INVITE: Content-Type: multipart/mixed, 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)
				→	ACK
		Conversation			
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP414206	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause B.21, Q.737 [i.10], clauses 1.2.5.2.3 and 4
TSS reference	ISUP-SIP-ISUP/SS/UUS3			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Ensure that the SUT can successfully transfer the User-to-user service 3 explicit response in the encapsulated ANM. O-MGCF interworking.			
SIP parameter values	INVITE: Content-Type: multipart/mixed, 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
	ACK	→			
		Conversation			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

TP414207	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 5.4.3, B.21, Q.737 [i.10], clauses 1.2.5.2.3 and 4**
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that a User-to-user request service 3 encapsulated in an INFO request during the confirmed state can successful proceeded. The user to user 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)
				→	ACK
			Conversation		
	FAR	→		→	INFO(FAR)
				←	200 OK INFO
	FRJ	←		←	INFO(FRJ)
				→	200 OK INFO
	REL	→		→	BYE(REL)
	RLC	←		←	200 OK BYE(RLC)

TP414208	**SIP reference: RFC 3261 [4]**				**ISUP reference: Q.1912.5 [1], clauses 5.4.3, B.21, Q.737 [i.10], clauses 1.2.5.2.3 and 4**
TSS reference	ISUP-SIP-ISUP/SS/UUS3				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that a User-to-user request service 3 encapsulated in an INFO request during the confirmed state can successful proceeded. The user to user 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
	ACK	→			
			Conversation		
	INFO(FAR)	→		→	FAR
	200 OK INFO	←			
	INFO(FRJ)	←		←	FRJ
	200 OK INFO	→			
	BYE(REL)	→		→	REL
	200 OK BYE(RLC)	←		←	RLC

Annex A (informative): Bibliography

ITU-T Recommendation Q.761: "Signalling System No. 7 - ISDN User Part functional description".

ITU-T Recommendation Q.762: "Signalling System No. 7 - ISDN User Part general functions of messages and signals".

ITU-T Recommendation Q.763: "Signalling System No. 7 - ISDN User Part formats and codes".

ITU-T Recommendation Q.1902.1: "Bearer Independent Call Control protocol (Capability Set 2): Functional description".

ITU-T Recommendation Q.1902.2: "Bearer Independent Call Control protocol (Capability Set 2) and Signalling System No.7 ISDN User Part: General functions of messages and parameters".

ITU-T Recommendation Q.1902.3: "Bearer Independent Call Control protocol (Capability Set 2) and Signalling System No.7 ISDN User Part: Formats and codes".

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

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

IETF RFC 3311 (2002): "The Session Initiation Protocol (SIP) UPDATE Method".

ITU-T Recommendation Q.730: "ISDN user part supplementary services".

ITU-T Recommendation Q.735: "Closed user group (CUG)".

ETSI EN 300 403-1: "Integrated Services Digital Network (ISDN); Digital Subscriber Signalling System No. one (DSS1) protocol; Signalling network layer for circuit-mode basic call control; Part 1: Protocol specification [ITU-T Recommendation Q.931 (1993), modified]".

Annex B (informative): Change history

Date	WG Doc.	CR	Rev	CAT	Title/Comment	Current Version	New Version
10-06-09	21PTD096r1	001		F	Update of test description and message flows	1.1.1	1.2.1
					Publication	1.2.1	1.2.1

History

Document history		
V3.1.1	June 2010	Publication
V3.2.1	March 2011	Publication