

**Telecommunications and Internet converged Services and
Protocols for Advanced Networking (TISPAN);
SIP-ISUP Interworking between the
IP Multimedia (IM) Core Network (CN) subsystem
and Circuit Switched (CS) networks;
Part 2: Test Suite Structure and Test Purposes (TSS&TP)**

iteh STANDARD PREVIEW
(standards.iteh.ai)
Full standard:
<https://standards.iteh.ai/catalog/standards/sist/8473ce95-12ff-44c5-aa62-5f5f186784d/etsi-ts-186-009-2-v2.1.1-2009-03>



Reference

DTS/TISPAN-06025-2-NGN-R2

Keywords

BICC, CTS, interworking, SIP, testing, TSS&TP

ETSI

650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
Association à but non lucratif enregistrée à la
Sous-Préfecture de Grasse (06) N° 7803/88

Important notice

Individual copies of the present document can be downloaded from:
<http://www.etsi.org>

The present document may be available in more than one electronic version or in print. In any case of existing or perceived difference in contents between such versions, the reference version is the Portable Document Format (PDF). In case of dispute, the reference shall be the printing on ETSI printers of the PDF version kept on a specific network drive within ETSI Secretariat.

Users of the present document should be aware that the document may be subject to revision or change of status.

Information on the current status of this and other ETSI documents is available at

<http://portal.etsi.org/tb/status/status.asp>

If you find errors in the present document, please send your comment to one of the following services:

http://portal.etsi.org/chaircor/ETSI_support.asp

Copyright Notification

No part may be reproduced except as authorized by written permission.
The copyright and the foregoing restriction extend to reproduction in all media.

© European Telecommunications Standards Institute 2009.
All rights reserved.

DECT™, PLUGTESTS™, UMTS™, TIPHON™, the TIPHON logo and the ETSI logo are Trade Marks of ETSI registered for the benefit of its Members.

3GPP™ is a Trade Mark of ETSI registered for the benefit of its Members and of the 3GPP Organizational Partners.

LTE™ is a Trade Mark of ETSI currently being registered
for the benefit of its Members and of the 3GPP Organizational Partners.

GSM® and the GSM logo are Trade Marks registered and owned by the GSM Association.

Contents

Intellectual Property Rights	5
Foreword.....	5
1 Scope	6
2 References	6
2.1 Normative references	6
2.2 Informative references.....	8
3 Definitions and abbreviations.....	8
3.1 Definitions.....	8
3.2 Abbreviations	9
4 Implementation under test and test methods	9
4.1 Identification of the system and implementation under test	9
5 Test Suite Structure (TSS).....	10
5.1 Interworking from SIP to ISUP (outgoing call)	10
5.2 Interworking from ISUP to SIP (incoming call).....	10
5.3 Supplementary Services - Interworking from SIP to ISUP (outgoing call).....	11
5.4 Supplementary Services - Interworking from ISUP to SIP (incoming call).....	11
6 Test purposes (TP)	11
6.1 Introduction	11
6.1.1 Test purpose (TP) naming convention	11
6.1.2 Source of test purpose definition.....	12
6.1.3 Test purpose structure	12
6.2 Test purposes for the basic call	12
6.2.1 Interworking from SIP to ISUP (Outgoing Call)	12
6.2.1.1 Sending of the Initial Address Message (IAM)	12
6.2.1.2 Overlap procedure at the I-MGCF.....	36
6.2.1.3 Sending of COT	37
6.2.1.4 Receipt of the Address Complete Message (ACM)	38
6.2.1.5 Receipt of the Call progress message (CPG)	53
6.2.1.6 Receipt of the Answer message (ANM).....	64
6.2.1.7 Receipt of the Connect message (CON)	70
6.2.1.8 Receipt of the REL message	77
6.2.1.9 Autonomous release at I-MGCF	83
6.2.1.10 Receipt of the Release message BYE / CANCEL.....	86
6.2.1.11 Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	88
6.2.1.12 Receipt of the Suspend message (SUS) network initiated	93
6.2.1.13 Receipt of the Resume message (RES) network initiated	93
6.2.2 Interworking from ISUP to SIP	94
6.2.2.1 Sending of the INVITE message.....	94
6.2.2.2 Receipt of the SAM message after INVITE has been send.....	117
6.2.2.3 Sending of the ACM message.....	123
6.2.2.4 Sending of the CPG message	152
6.2.2.5 Sending of the ANM message.....	167
6.2.2.6 Sending of the CON message.....	173
6.2.2.7 Receipt of the Release message (REL)	179
6.2.2.8 Sending of a REL message / receipt of a backward BYE	183
6.2.2.9 Autonomous release at O-MGCF.....	188
6.2.2.10 Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	191
6.2.2.10.1 Receipt of Reset Circuit message (RSC)	191
6.2.2.10.2 Receipt of Circuit group reset message (GRS)	194
6.2.2.10.3 Receipt of Circuit group blocking message (CGB)	198
6.3.1 Interworking from SIP to ISUP (Incoming Call)	203

6.3.1.1	Calling Line Identification (CLI)	203
6.3.1.2	Call Hold (HOLD)	227
6.3.1.3	Terminal portability (TP)	231
6.3.1.4	Conference calling (CONF)	232
6.3.1.5	Three Party service (3PTY).....	245
6.3.1.6	Connected line identification (COL).....	248
6.3.1.7	Malicious call identification MCID	262
6.3.1.8	Sub-addressing (SUB).....	263
6.3.1.9	Call diversion (CDIV).....	264
6.3.1.10	Call waiting (CW).....	290
6.3.1.11	User to user signalling (UUS)	291
6.3.1.12	Explicit call transfer (ECT).....	293
6.3.1.13	Completion of Call to Busy Subscriber (CCBS).....	294
6.3.1.14	Completion of Calls on No reply (CCNR)	295
6.3.1.15	Anonymous Call Rejection (ACR)	295
6.3.1.16	Closed user group (CUG).....	296
6.3.2	Interworking from ISUP to SIP (Outgoing Call)	301
6.3.2.1	Calling Line Identification (CLI)	301
6.3.2.2	Call Hold (HOLD)	309
6.3.2.3	Terminal portability (TP)	313
6.3.2.4	Conference calling (CONF)	314
6.3.2.5	Three Party service (3PTY).....	326
6.3.2.6	Connected line identification (COL).....	328
6.3.2.7	Sub-addressing (SUB).....	338
6.3.2.8	Closed user group (CUG).....	341
6.3.2.9	Call diversion (CDIV).....	343
6.3.2.10	User to user signalling (UUS)	366
6.3.2.11	Explicit call transfer (ECT).....	376
6.3.2.12	Anonymous Call Rejection (ACR)	380
6.3.2.13	Call waiting (CW)	381
6.3.2.14	Malicious call identification (MCID).....	381
Annex A (informative):	Bibliography.....	385
History	387

<https://standards.etsi.org/standards/sis/873ce95-12ff-44c5-aa62-5f518678a209/v2.1.1/>

Intellectual Property Rights

IPRs essential or potentially essential to the present document may have been declared to ETSI. The information pertaining to these essential IPRs, if any, is publicly available for **ETSI members and non-members**, and can be found in ETSI SR 000 314: *"Intellectual Property Rights (IPRs); Essential, or potentially Essential, IPRs notified to ETSI in respect of ETSI standards"*, which is available from the ETSI Secretariat. Latest updates are available on the ETSI Web server (<http://webapp.etsi.org/IPR/home.asp>).

Pursuant to the ETSI IPR Policy, no investigation, including IPR searches, has been carried out by ETSI. No guarantee can be given as to the existence of other IPRs not referenced in ETSI SR 000 314 (or the updates on the ETSI Web server) which are, or may be, or may become, essential to the present document.

Foreword

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

The present document is part 2 of a multi-part deliverable covering the Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); SIP-ISUP Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks;

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

Part 2: "**T**est **S**uite **S**tructure and **T**est **P**urposes (TSS&TP)";

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

iTeh STANDaRD PREv1.1
Full standard:
<https://standards.iteh.ai/catalog/standards/sist/8473c&t/12ff-44c5-aa62-5f5f186784d/etsi-ts-186-009-2-v2.1.1-2009-03>

1 Scope

The present document specifies the Test Suite Structure and Test Purposes (TSS&TP) for SIP-ISUP Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks ES 283 027 [1]. The references [1] and [16] are identical.

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

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication and/or edition number or version number) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version 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 indispensable for the application of the present document. For dated references, only the edition cited applies. For non-specific references, the latest edition of the referenced document (including any amendments) applies.

- [1] ETSI ES 283 027 (V2.5.1): "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]".
- [2] ETSI TS 129 163: "Digital cellular telecommunications system (Phase 2+) Universal Mobile Telecommunications System (UMTS) Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (3GPP TS 29.163 version 7.9.0 Release 7)".
- [3] ITU-T Recommendations Q.761 to Q.764 (2000): "Signalling System No.7 ISDN User Part (ISUP)".
- [4] Void.
- [5] ITU-T Recommendation Q.850 (1998): "Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part".
- [6] IETF RFC 3261 (2002): "SIP: Session Initiation Protocol".
- [7] IETF RFC 3312 (2002): "Integration of Resource Management and Session Initiation Protocol (SIP)".
- [8] ISO/IEC 9646-1 (1994): "Conformance testing methodology and framework - Part 1: General Concepts".

- [9] ISO/IEC 9646-3 (1992): "Conformance testing methodology and framework - Part 3: The Tree and Tabular Combined Notation".
- [10] ISO/IEC 9646-7 (1994): "Conformance testing methodology and framework - Part 7: Implementation Conformance Statement".
- [11] ITU-T Recommendation E.164: "The international public telecommunication numbering plan".
- [12] Void.
- [13] ITU-T Recommendation Q.939: "Typical DSS 1 service indicator codings for ISDN telecommunications services".
- [14] ETSI TS 183 008: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); PSTN/ISDN simulation services Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR); Protocol specification".
- [15] ETSI TS 129 163: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (3GPP TS 29.163 version 8.5.0 Release 8)".
- [16] ETSI TS 129 527 (V8.2.0): "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); 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] (3GPP TS 29.527 version 8.2.0 Release 8)".
- [17] IETF RFC 3556: "Session Description Protocol (SDP) Bandwidth Modifiers for RTP Control Protocol (RTCP) Bandwidth".
- [18] IETF RFC 3264: "An Offer/Answer Model with Session Description Protocol (SDP)".
- [19] IETF RFC 4040: "RTP Payload Format for a 64 kbit/s Transparent Call".
- [20] ITU-T Recommendation F.182: "Operational provisions for the international public facsimile service between subscribers with Group 3 facsimile terminals (Telefax 3)".
- [21] ITU-T Recommendation F.184: "Operational provisions for the international public facsimile service between subscriber stations with group 4 facsimile terminals (telefax 4)".
- [22] ITU-T Recommendation F.230: "Service requirements unique to the mixed mode (MM) used within the teletex service".
- [23] ITU-T Recommendation F.220: "Service requirements unique to the processable mode number eleven (PM11) used within the teletex service".
- [24] ITU-T Recommendation F.200: "Teletex service".
- [25] ITU-T Recommendation F.300: "Videotex service".
- [26] ITU-T Recommendation F.60: "Operational provisions for the international telex service".
- [27] ITU-T Recommendation F.721: "Videotelephony teleservice for ISDN".
- [28] ETSI ETS 300 356-1: "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 1: Basic services [ITU-T Recommendations Q.761 to Q.764 (1999) modified]".
- [29] ITU-T Recommendation X.213: "Information technology - Open Systems Interconnection - Network service definition".
- [30] ISO/IEC 8348: "Information technology - Open Systems Interconnection - Network service definition".

- [31] ITU-T Recommendation T.38: "Procedures for real-time Group 3 facsimile communication over IP networks".
- [32] ITU-T Recommendation Q.1912.5: "Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part".
- [33] ITU-T Recommendation Q.737.1: "Stage 3 description for additional information transfer supplementary services using Signalling System No. 7 : User-to-user signalling (UUS)".
- [34] ITU-T Recommendation Q.734.1: "Stage 3 description for multiparty supplementary services using Signalling System No. 7: Conference calling".
- [35] ITU-T Recommendation Q.734.2: "Stage 3 description for multiparty supplementary services using Signalling System No. 7: Three-party service".

2.2 Informative references

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

Not applicable.

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in SIP / ISUP interworking reference specification, in ISDN layer 3 reference specification, in ISO/IEC 9646-1 [8], in ISO/IEC 9646-3 [9], in ISO/IEC 9646-7 [10] and the following apply:

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 [11]

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

3.2 Abbreviations

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

ATC	Abstract Test Case
ATM	Abstract Test Method
ATP	Access Transport Parameter
ATS	Abstract Test Suite
BCI	Backward Call Indicators
CPS	Calling Party's Category
DSS1	Digital Subscriber System No. 1
FCI	Forward Call Indicators
HLC	High Layer Compatibility
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
IUT	Implementation Under Test
MOT	Means Of Testing
NCI	Nature of Connection Indicators
OBCI	Optional Backward Call Indicators
OFCI	Optional Forward Call Indicator
PICS	Protocol Implementation Conformance Statement
PIXIT	Protocol Implementation eXtra Information for Testing
SUT	System Under Test
TMR	Transmission Medium Requirement
TP	Test Purpose
TSS	Test Suite Structure
TTCN	Tree and Tabular Combined Notation

NOTE: The ISUP message acronyms can be found in table 2/ ITU-T Recommendation Q.762 [3].

4 Implementation under test and test methods

4.1 Identification of the system and implementation under test

FFS

5 Test Suite Structure (TSS)

The Test Suite Structure is in close alignment with ES 283 027 [1].

5.1 Interworking from SIP to ISUP (outgoing call)

SIP -ISUP Basic call		
	Sending of the Initial address message (IAM)	101xxx
	Sending of the Subsequent address message (SAM)	102xxx
	Sending of COT	103xxx
	Receipt of the Address complete message (ACM)	104xxx
	Receipt of the Call progress message (CPG)	105xxx
	Receipt of the answer message (ANM)	106xxx
	Receipt of the Connect message (CON)	107xxx
	Receipt of the Release message (REL)	108xxx
	Autonomous release at I-MGCF	109xxx
	Receipt of the BYE, CANCEL message / sending of a REL message	110xxx
	Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	111xxx
	Receipt of the SUSPEND Message (SUS)	111xxx
	Receipt of the RESUME Message (RES)	112xxx

**Figure 1: Basic call -
Test suite structure for interworking between SIP to ISUP (outgoing call)**

5.2 Interworking from ISUP to SIP (incoming call)

ISUP-SIP Basic call		
	Sending of the INVITE message	301xxx
	Receipt of the Subsequent address message (SAM)	302xxx
	Sending of the Address complete message (ACM)	303xxx
	Sending of the Call progress message (CPG)	304xxx
	Sending of the answer message (ANM)	305xxx
	Sending of the Connect message (CON)	306xxx
	Receipt of the Release message (REL)	307xxx
	Sending of the Release Message (REL)	308xxx
	Autonomous release	309xxx
	Receipt of Reset circuit message (RSC)	310xxx
	Receipt of Circuit group reset message (GRS)	311xxx
	Receipt of Circuit group blocking message (CGB) with the indication hardware failure oriented	312xxx

**Figure 2: Basic call -
Test suite structure for interworking between ISUP to SIP (incoming call)**

5.3 Supplementary Services - Interworking from SIP to ISUP (outgoing call)

SIP-ISUP Supplementary Services	
Calling Line Identification (CLI)	501xxx
Call Hold (HOLD)	502xxx
Terminal Portability (TP)	503xxx
Conference Calling (CONF)	504xxx
Three-Party (3PTY)	505xxx
Connected Line Identification (COL)	506xxx
Malicious call identification (MCID)	507xxx
Subaddressing (SUB)	508xxx
Call Diversion (CDIV)	509xxx
Call Waiting (CW)	510xxx
User to User Signalling (UUS)	511xxx
Explicit Call transfer (ECT)	512xxx
Completion of Call to Busy Subscriber (CCBS)	513xxx
Completion of Calls on No reply (CCNR)	514xxx
Anonymous Call Rejection (ACR)	515xxx
Closed user group (CUG)	516xxx

Figure 3: Supplementary Services - Test suite structure for interworking between SIP to ISUP (outgoing call)

5.4 Supplementary Services - Interworking from ISUP to SIP (incoming call)

ISUP-SIP	
Calling Line Identification (CLI)	601xxx
Call Hold (HOLD)	602xxx
Terminal Portability (TP)	603xxx
Conference Calling (CONF)	604xxx
Three-Party (3PTY)	605xxx
Connected Line Identification (COL)	606xxx
Subaddressing (SUB)	607xxx
Closed User Group (CUG)	608xxx
Call Diversion (CDIV)	609xxx
User to User Signalling (UUS)	610xxx
Explicit Call transfer (ECT)	611xxx
Anonymous Call Rejection (ACR)	612xxx
Call waiting (CW)	613xxx
Malicious call identification (MCID)	614xxx

Figure 4: Supplementary Services - Test suite structure for interworking between ISUP to SIP (outgoing call)

6 Test purposes (TP)

6.1 Introduction

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

6.1.1 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.

6.1.2 Source of test purpose definition

The test purposes have been developed based on ES 283 027 [1] as an endorsement of TS 129 163 [15].

6.1.3 Test purpose structure

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

6.2 Test purposes for the basic call

6.2.1 Interworking from SIP to ISUP (Outgoing Call)

6.2.1.1 Sending of the Initial Address Message (IAM)

TP101001	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.1
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	<p><i>Normal call setup without precondition rerequirement</i></p> <p>Ensure that if the SIP precondition extension is not included in the Supported or Require header, the I-MGCF shall send an IAM immediately after the reception of the INVITE. The I-MGCF shall set the continuity indicators to "Continuity check not required".</p>	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	<p>SIP</p> <p>INVITE → 180 Ringing ←</p> <p>200 OK INVITE ← ACK →</p> <p>BYE → 200 OK BYE ←</p>	<p>SUT</p> <p>Ringing tone</p> <p>Conversation</p> <p>ISUP</p> <p>→ IAM ← ACM</p> <p>← ANM</p> <p>→ REL ← RLC</p>

TP101002	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.1
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:		
Test purpose:	<p><i>Call setup with precontion tag in the Supported header and preconditions are fullfield successful</i></p> <p>Ensure if a Continuity Check procedure is supported in the ISUP network and SIP precondition extension are included in the SIP Supported header and the preconditions are indicated as fullfield in the SDP, the I-MGCF shall send the IAM immediately after the reception of the INVITE. The preconditions met is sent in the 200 OK INVITE.</p>	
SIP Parameter values:	<p>INVITE: Supported: 100rel, precondition SDP a=curr:qos local sendrecv a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos none remote sendrecv</p> <p>200 OK INVITE SDP a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv</p>	
ISUP Parameter values:	IAM: Continuity indicator: Continuity check not required	
Comments:	<p>SIP</p> <p>INVITE → SUT 180 Ringing ←</p> <p>200 OK INVITE ← ACK → Ringing tone</p> <p>BYE → 200 OK BYE ←</p> <p>ISUP</p> <p>→ IAM ← ACM</p> <p>← ANM</p> <p>→ REL ← RLC</p>	

*ITEH STANDARD PRELIMINARY
Full standard:
https://standards.iteh.ai/catalog/standards/sis/8473e95-
12ff-44c5-aa62-5b5f18674d/etsi-ts-186-009-2-v2.1.1
200903*