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Technical Specification

Telecommunications and Internet Converged Services and Protocols for Advanced Networking (TISPAN); Network Integration Testing between SIP and ISDN/PSTN network signalling protocols Part 1: Test Suite Structure and Test Purposes (TSS&TP) for SIP-ISDN

Reference
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Keywords
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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

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Foreword

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

The present document is part 1 of a multi-part standard covering Network Integration Testing between SIP and ISDN/PSTN network signalling protocols:

Part 1: "Test Suite Structure and Test Purposes (TSS&TP) for SIP-ISDN";

Part 2: "Abstract Test Suite (ATS) and partial Protocol Implementation eXtra Information for Testing (PIXIT) proforma specification";

Part 3: "Test Suite Structure and Test Purposes (TSS&TP) for SIP-SIP".

1 Scope

The present document specifies the Test Suite Structure and Test Purposes (TSS&TP) for Network Integration Testing (NIT) to verify the overall compatibility of SIP, ISDN and non-ISDN (PSTN) over the national or international ISDN networks. The TSS&TP specification covers the procedures described in ITU-T Recommendation Q.1912.5 [51] or EN 383 001 [49] or ES 283 027 [48] and EN 300 899-1 [23]. For SIP and SDP specific terminology, reference shall be made to ES 283 003 [42] respectively RFC 3261 [28].

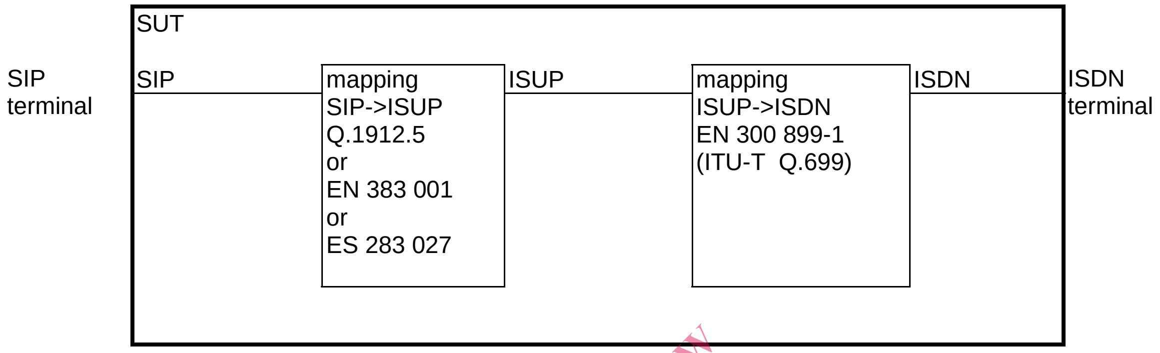

Figure 1: SIP-ISDN inter-working testing architecture

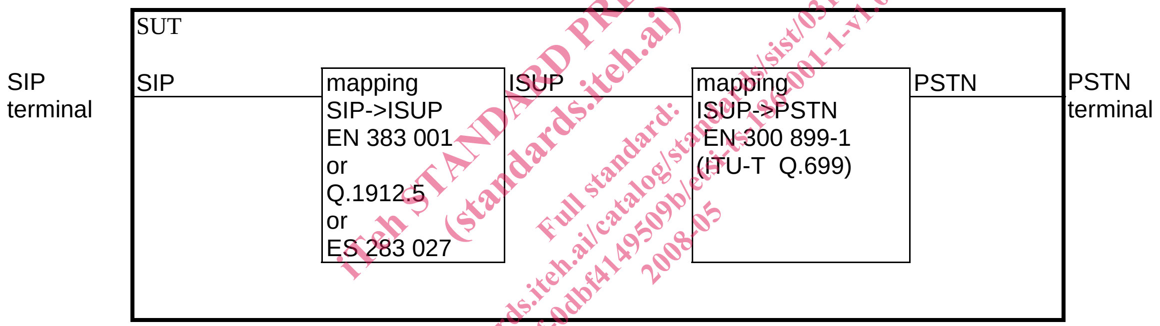

Figure 2: SIP-PSTN inter-working testing architecture

2 References

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  - for informative references.

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2.1 Normative references

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

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

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

[3] ITU-T Recommendation Q.731.7 (06/97): "Stage 3 description for number identification supplementary services using Signalling System No. 7: Malicious call identification (MCID)".

[4] ITU-T Recommendation Q.732.2 (12/99): "Stage 3 description for call offering supplementary services using Signalling System No. 7: Call diversion services: Call Forwarding Busy (CFB)".

[5] ITU-T Recommendation Q.732.3 (12/99): "Stage 3 description for call offering supplementary services using Signalling System No. 7: Call Forwarding No Reply (CFNR)".

[6] ITU-T Recommendation Q.732.4 (12/99): "Stage 3 description for call offering supplementary services using Signalling System No. 7: Call Forwarding Unconditional (CFU)".

[7] ITU-T Recommendation Q.732.5 (12/99): "Stage 3 description for call offering supplementary services using Signalling System No. 7: Call Deflection (CALL DEFLECTION)".

[8] ITU-T Recommendation Q.732.7 (07/96): "Stage 3 description for call offering supplementary services using Signalling System No. 7: Explicit Call Transfer".

[9] ITU-T Recommendation Q.733.1 (02/92): "Stage 3 description for call completion supplementary services using Signalling System No. 7: Call waiting (CW)".

[10] ITU-T Recommendation Q.733.2 (03/93): "Stage 3 description for call completion supplementary services using Signalling System No. 7: Call hold (HOLD)".

[11] ITU-T Recommendation Q.733.3 (06/97): "Stage 3 description for call completion supplementary services using Signalling System No. 7: Completion of calls to busy subscriber (CCBS)".

[12] ITU-T Recommendation Q.733.4 (03/93): "Stage 3 description for call completion supplementary services using Signalling System No. 7: Terminal portability (TP)".

[13] ITU-T Recommendation Q.733.5 (12/99): "Stage 3 description for call completion supplementary services using Signalling System No. 7: Completion of calls on no reply".

[14] ITU-T Recommendation Q.734.1 (03/93): "Stage 3 description for multiparty supplementary services using Signalling System No. 7: Conference calling".

[15] ITU-T Recommendation Q.734.2 (07/96): "Stage 3 description for multiparty supplementary services using Signalling System No. 7: Three-party service".

[16] ITU-T Recommendation Q.735.1 (03/93): "Stage 3 description for community of interest supplementary services using Signalling System No. 7: Closed user group (CUG)".

[17] ITU-T Recommendation Q.735.3 (03/93): "Stage 3 description for community of interest supplementary services using Signalling System No. 7: Multi-level precedence and preemption".

[18] ITU-T Recommendation Q.735.6 (07/96): "Stage 3 description for community of interest supplementary services using Signalling System No. 7: Global Virtual Network Service (GVNS)".

[19] ITU-T Recommendation Q.736.1 (10/95): "Stage 3 description for charging supplementary services using Signalling System No. 7: International Telecommunication Charge Card (ITCC)".

[20] ITU-T Recommendation Q.736.3 (10/95): "Stage 3 description for charging supplementary services using Signalling System No. 7: Reverse charging (REV)".

[21] ITU-T Recommendation Q.737.1 (06/97): "Stage 3 description for additional information transfer supplementary services using Signalling System No. 7: User-to-user signalling (UUS)".

[22] ITU-T Recommendation Q.850 (05/98): "Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part".

[23] ETSI EN 300 899-1: "Integrated Services Digital Network (ISDN); Signalling System No.7; Interworking between ISDN User Part (ISUP) version 2 and Digital Subscriber Signalling System No. one (DSS1); Part 1: Protocol specification [ITU-T Recommendation Q.699, modified]".

[24] IETF RFC 2046 (1996): "Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types".

[25] IETF RFC 2327 (1998): "SDP: Session Description Protocol".

[26] IETF RFC 2806 (2000): "URLs for Telephone Calls".

[27] IETF RFC 3204 (2001): "MIME media types for ISDNand QSIG Objects".

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

[29] IETF RFC 3262 (2002): "Reliability of Provisional Responses in the Session Initiation Protocol (SIP)".

[30] IETF RFC 3264 (2002): "An Offer/Answer Model with the Session Description Protocol (SDP)".

[31] IETF RFC 3311 (2002): "The Session Initiation Protocol UPDATE Method".

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

[33] IETF RFC 3323 (2002): "A Privacy Mechanism for the Session Initiation Protocol (SIP)".

[34] IETF RFC 3325 (2002): "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks".

[35] IETF RFC 3326 (2002): "The Reason Header Field for the Session Initiation Protocol".

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

[37] ISO/IEC 9646-2 (1994): "Conformance testing methodology and framework - Part 2: Abstract Test Suite Specification".

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

[39] ISO/IEC 9646-3/DAM 1 (1992): "Conformance testing methodology and framework - Part 3: The Tree and Tabular Combined Notation; Amendment 1: TTCN extensions".

[40] ISO/IEC 9646-5 (1994): "Conformance testing methodology and framework - Part 5: Requirements on test laboratories and clients for the conformance assessment process".

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

[42] ETSI ES 283 003: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IP Multimedia Call Control Protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) Stage 3 [3GPP TS 24.229 [Release 7], modified]".

[43] ETSI TS 183 007: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); PSTN/ISDN simulation services; Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR); Protocol specification".

[44] 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".

[45] ETSI TS 183 004: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); PSTN/ISDN simulation services: Communication Diversion (CDIV); Protocol specification".

[46] ETSI TS 183 005: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); PSTN/ISDN simulation services: Conference (CONF); Protocol specification".

[47] ETSI TS 183 010: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); NGN Signalling Control Protocol; Communication HOLD (HOLD) PSTN/ISDN simulation services; Protocol specification".

[48] 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]".

[49] 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].

[50] IETF RFC 4967 (2007): "Dial String Parameter for the Session Initiation Protocol Uniform Recource Identifier".

[51] ITU-T Recommendation Q.1912.5 (2004): "Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control Protocol or ISDN User Part".

[52] ITU-T Recommendation Q.699 (09/97): "Interworking between ISDN access and non-ISDN access over ISDN User Part of Signalling System No. 7".

[53] ITU-T Recommendation Q.931 (05/98): "ISDN user-network interface layer 3 specification for basic call control".

[54] ETSI TS 134 229-1: "Universal Mobile Telecommunications System (UMTS); Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Part 1: Protocol conformance specification (3GPP TS 34.229-1 version 6.3.0 Release 6)".

[55] ETSI TS 124 229: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS);Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3 (3GPP TS 24.229 version 7.9.0 Release 7)".

[56] IETF RFC 2833: " RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals ".

2.2 Informative references

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

Not applicable.

3 Definitions and abbreviations

3.1 Definitions

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

For BICC or ISUP specific terminology, reference shall be made to ITU-T Recommendation Q.1902.2 [2]. For SIP and SDP specific terminology, reference shall be made to RFC 3261 [28] and RFC 2327 [25] respectively. Definitions for additional terminology used in this interworking Recommendation are as follows:

Adjacent SIP Node (ASN): SIP node (e.g. SIP Proxy or Back-to-Back User Agent or the SIP side of an IWU) that has established a direct trust relation (association) with Incoming or Outgoing IWU entities

NOTE: The SIP Proxy and Back-to-Back User Agent are defined in accordance with RFC 3261 [28].

Basic Call Control (BCC): signalling protocol associated with the DSS1 - ISDN Basic Call control procedures of ITU-T recommendation Q.931 [53] (EN 300 403-1)

incoming or outgoing: direction of a call (not signalling information) with respect to a reference point

Incoming Interworking Unit (I-IWU): physical entity, (which can be combined with a BICC ISN or ISUPexchange) that terminates incoming calls using SIP and originates outgoing calls using the BICC or ISUP protocols

incoming SIP or BICC/ISUP (network): network, from which the incoming calls are received, that uses the SIP or BICC/ISUP protocol (without the term "network", it simply refers to the protocol)

inopportune: specifies a test purpose covering a signalling procedure where an inopportune message (type of message not expected in the IUT current state) is sent to the IUT

Outgoing Interworking Unit (O-IWU): physical entity, (which can be combined with a BICC ISN or ISUP exchange) that terminates incoming calls using BICC or ISUP protocols and originates outgoing calls using the SIP

outgoing SIP or BICC/ISUP (network): network, to which the outgoing calls are sent, that uses the SIP or BICC/ISDNprotocol

NOTE: Without the term "network", it simply refers to the protocol.

SIP precondition: indicates the support of the SIP "precondition procedure" as defined in RFC 3312 [32]

syntactically invalid: specifies a test purpose covering a signalling procedure where a valid (expected in the current status of the IUT) but not correctly encoded (unknown or incorrect parameter values) message is sent to the IUT, wSIch shall react correctly and eventually reject the message

test purpose: non-formal test description, mainly using text

NOTE: TSIs test description can be used as the basis for a formal test specification (e.g. Abstract Test Suite in TTCN). See ISO 9646.

valid: specifies a test purpose covering a signalling procedure where all the messages sent to or received from the IUT are valid (expected in the current status of the IUT) and correctly encoded

3.1.1 Conventions for representation of SIP/SDP information

1) All letters of SIP method names are capitalized.

EXAMPLE 1: INVITE, INFO.

2) SIP header fields are identified by the unabbreviated header field name as defined in the relevant RFC, including capitalization and enclosed hyphens but excluding the following colon.

EXAMPLE 2: To, From, Call-ID.

3) Where it is necessary to refer with finer granularity to components of a SIP message, the component concerned is identified by the ABNF rule name used to designate it in the defining RFC (generally 25/RFC 3261 [28]), in plain text without surrounding angle brackets.

EXAMPLE 3: Request-URI, the user info portion of a sip: URI.

4) URI types are represented by the lower-case type identifier followed by a colon and the abbreviation "URI"

EXAMPLE 4: sip: URI, tel: URI.

5) SIP provisional responses and final responses other than 2XX are represented by the status code followed by the normal reason phrase for that status code, with initial letters capitalized.

EXAMPLE 5: 100 Trying, 484 Address Incomplete.

6) Because of potential ambiguity within a call flow about which request a 200 OK final response answers, 200 OK is always followed by the method name of the request.

EXAMPLE 6: 200 OK INVITE, 200 OK PRACK.

7) A particular line of an SDP session description is identified by the two initial characters of the line -- that is, the line type character followed by "="

EXAMPLE 7: m=line, a=line.

8) Where it is necessary to refer with finer granularity to components of a session description, the component concerned is identified by its rule name in the ABNF description of the SDP line concerned, delimited with angle brackets.

EXAMPLE 8: the <media> and <fmt> components of the m= line.

3.2 Abbreviations

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

GW	GateWay
I	Inopportune
IUT	Implementation Under Test
PER	Packed Encoding Rules
PICS	Protocol Implementation Conformance Statement
PIXIT	Protocol Implementation eXtra Information for Testing
S	Syntactically invalid
TP	Test Purpose
TSS	Test Suite Structure
V	Valid

4 Test Suite Structure (TSS)

4.1 Test Suite Structure (TSS)

4.1.1 ISDN-SIP

C - Plane / U - Plane Basic_Call	Successful	Voice	IS_XX_xx
		Codec negotiation	IS_CN_xx
		Update Tests	IS_XX_UP_xx
		DTMF	IS_DTMF_xx
		UDI	IS_UD_xx
	Unsuccessful		IS_XX_Uxx
C - Plane Supplementary Services		CLIP	IS_XXSSCLIPxx
		CLIR	IS_XXSSCLIRxx
		COLP/COLR (TIP/TIR)	IS_XXSSCOLPxx
		CFU	ISI_XXSSCFUxx
			ISS_XXSSCFUxx
		CFB	ISI_XXSSCFBxx
			ISS_XXSSCFBxx
		CFNR	ISI_XXSSCFNRxx
			ISS_XXSSCFNRx
		CFNL	ISS_XXSSCFNLxx
		3PTY	ISI_XXSS3PTYxx
			ISS_XXSS3PTYxx
		HOLD	ISI_XXSSHOLDxx
		CONF	IS_XXSSCONFxx

4.1.2 SIP-ISDN

C - Plane / U - Plane Basic_Call	Successful	3,1 kHz audio	SI_AU_xx
		Codec negotiation	SI_XX_CN_xx
		DTMF	SI_XX_DT_xx
		UDI	SI_UD_xx
	Unsuccessful		SI_XX _Uxx
C - Plane Supplementary Services		CLIP	SI_XXSSOIPxx
		CLIR	SI_XXSSOIRxx
		COLP/COLR (TIP/TIR)	SI_XXSSCOLPxx
		CFU	SIS_XXSSCFUxx
			SII_XXSSCFUxx
		CFB	SIS_XXSSCFBxx
			SII_XXSSCFBxx
		CFNR	SIS_XXSSCFNRxx
			SII_XXSSCFNRxx
		3PTY	SII_XXSS3PTYXX
			SIS_XXSS3PTYXX
		TP	SI_XXSSTPxx
		CUG	SI_XXSSCUGxx
		HOLD	SI_XXSSHOLDxx
		CONF	SI_XXSSCONFxx
		CW	SI_XXSSCWxx
		ACR	SI_XXSSACRxx

4.1.3 PSTN-SIP

C - Plane / U - Plane Basic_Call	Successful		PS_AU_Xxx
	Unsuccessful		PS_AU_Uxx
C - Plane Supplementary Services		CLIP	PS_XXSSCLIPxx
		CLIR	PS_XXSSCLIRxx
		CFU	PSP_XXSSCFUxx
			PSS_XXSSCFUxx
		CFB	PSP_XXSSCFBxx
			PSS_XXSSCFBxx
		CFNR	PSP_XXSSCFNRxx
			PSS_XXSSCFNRxx
		CFNL	PSP_XXSSCFNLxx