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

Reference
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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 3 of a multi-part deliverable covering Network Integration Testing between SIP and ISDN/PSTN network signalling protocols, as identified below:

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

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

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

1 Scope

This document specifies the Test Suite Structure and Test Purposes (TSS&TP) for Network Integration Testing (NIT) to verify the overall compatibility of SIP networks. For SIP and SDP specific terminology, reference shall be made to ES 283 003 [1] and RFC 3261 [2] respectively".

2 References

References are either specific (identified by date of publication and/or edition number or version number) or non-specific.

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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] 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]".

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

[3] Void.

[4] Void.

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

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

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

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

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

[10] ISO/IEC 9646-7 (1994): "Conformance testing methodology and framework - Part 7: Implementation Conformance Statement etworking (TISPAN); PSTN/ISDN simulation services: Conference (CONF); Protocol specification".

[11] 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)".

[12] 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)".

[13] Void.

[14] 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]".

[15] 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]".

[16] 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]".

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

[18] ETSI TS 183 028: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Common Basic Communication procedures; Protocol specification".

[19] ETSI TS 183 007: "Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR) PSTN/ISDN Simulation Services".

[20] ETSI TS 183 005: "Telecommunications and Internet converged Services and Protocols for Advanced IETF RFC 3204 (2001), MIME media types for ISDNand QSIG Objects".

[21] 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".

2.2 Informative references

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

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

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

3 Definitions and abbreviations

3.1 Definitions

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

For SIP and SDP specific terminology, reference shall be made to RFC 3261 [2] and RFC 2327 [22] respectively.

SIP precondition: Indicates the support of the SIP "precondition procedure" as defined in RFC 3312 [23].

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

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 (all parts) [5] to [10].

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

3.2 Conventions for representation of SIP/SDP information

1) All letters of SIP method names are capitalised.

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 [2]), in plain text without surrounding angle brackets.

EXAMPLE 3: Request-URI, the userinfo 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.

3.3 Abbreviations

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

GW	GateWay
I	Inopportune
IUT	Implementation Under Test
MCU	Multipoint Control Unit
MGCF	Media Gateway Control Function
MSI	Manufacturer Specific Information
PDU	Protocol Data Unit
PER	Packed Encoding Rules
PICS	Protocol Implementation Conformance Statement
PIXIT	Protocol Implementation eXtra Information for Testing
PSA	PSase A : Call setup signalling procedures
PSE	PSase E : Call termination signalling procedures
RAS	Registration, Admission and Status
RCF	Register Confirm
REG	REGistration
RRJ	Register Reject
RRQ	Register Request
S	Syntactically invalid
STA	STAtus
TE	Terminal
TP	Test Purpose
TSS	Test Suite Structure
UCF	Unregistration ConFirm
UE	User Equipment
URJ	Unregistration ReJect
URQ	Unregistration ReQuest
V	Valid

4 Test Suite Structure (TSS)

4.1 SIP-SIP

C – Plane / U – Plane			
Basic_Call	Successful		SS___XX___xx
		Codec negotiation	SS___CN___xx
		UPDATE	SS___XX__UP__xx
		Unsuccessful	SS___XX__Uxx
Supplementary Services			
		OIP/OIR	SS___XXSS_OIPxx
			SS___XXSS_OIRxx
		TIP/TIR	SS___XXSS_TIPxx
		HOLD	SS___XXSS_CHxx
		CFU	SS___XXSS_CFUxx
		CFB	SS___XXSS_CFBxx
		CFNR	SS___XXSS_CFNRxx
		CFNL	SS___XXSS_CHFNLxx
		CONF	SS___XXSS_CONFxx

5 Numbering Scheme

5.1 General description

Pos. 1: Network of the A-Subscriber

Pos. 2: Network of the B-Subscriber

Pos. 3: Network of the C-Subscriber

Pos. 4: Network of the D-Subscriber

Pos. 5: Network of the E-Subscriber

The following Network Codes apply:

_: No such network used (used e.g. for C-Subscriber in successful A to B Calls)

(underscore makes it easier to read the name)

P: PSTN

I: ISDN

S: SIP

(Extensions will be added when needed)

Pos. 6 and 7: Bearer- or Teleservice involved

XX: Defined per PIXIT value

NOTE: TSIs may be appropriate for Test Purposes (provided the Test Purpose states for wSIch Bearer- and/or Tele Services it should be tested). It is however NOT appropriate for Test Cases since it would be detrimental to Test Automation.

SP: Speech

AU: 3,1 kHz Audio

UD: UDI

UT: UDI/TA

CN: Codec negotiation

DT: DTMF

UP: UPDATE Method

Pos. 8&9:

__: No Supplementary Services Involved / Successful

_U: No Supplementary Services Involved / Unsuccessful

SS: Supplementary Services Involved

SI: Supplementary Services interaction

SN: Nonsymmetrical Supplementary Services Involved

ST: Supplementary Services transparent

5.2 Basic Call

Speech	IS___XX__XX

1	2	3	4	5	6	7	8	9	10	11
I	S	_	_	_	S	P	_	_	x	x

5.3 Supplementary Services

CLIP	IS___XXSSCLIP XX

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
I	S	_	_	_	X	X	S	S	C	L	I	P	x	X

6 Test purposes

The registration and application usage procedures in the ATS shall be compliant to RFC 3261 [2] and ES 283 003 [1] (modified TS 124 229 [11]). The validation of the registration procedure is out of scope of the present document and is part of the Preamble used in the test cases.

The registration conformance tests based on TS 124 229 [11] are contained in TS 134 229-1 [12]

The preconditions mechanism shall be supported by the UE in case of supporting IMS.

The handling of preconditions at the originating or /and terminating UE (MGCF in case if interworking) is described in the following table.

PIXIT Values			
	UE (MGCF) originating case	**UE (MGCF) terminating case**	
VA	precondition" option-tag in the Supported header	local resource reservation is **required** at the terminating UE	local resource reservation is **not required** by the terminating UE and the terminating UE supports the precondition mechanism
VA_1	"precondition" option-tag in the Supported header	the terminating UE shall make use of the precondition mechanism	
VA_2.1	"precondition" option-tag in the Supported headerd and required resources at the originating netwotk are **not** reserved	the terminating UE shall make use of the precondition mechanism	
VA_2.2	"precondition" option-tag in the Supported headerd and required resources at the originating network are **not** reserved		the terminating UE shall use the precondition mechanism
VA_3.1	"precondition" option-tag in the Supported header and required local resources at the originating network	the terminating UE **shall** make use of the precondition mechanism	
VA_3.2	"precondition" option-tag in the Supported header and required local resources at the originating network		the required local resources at the originating UE and the terminating UE are available, the terminating UE may use the precondition mechanism
VA_4.1	INVITE request does not include the "precondition" option-tag in the Supported header	the terminating UE shall not make use of the precondition mechanism.	
VA_4.2	INVITE request does not include the "precondition" option-tag in the Supported header		the terminating UE shall not make use of the precondition mechanism.

Dial string parameters options

	To header field- UE originated
VA_5.1	sip: dialled digits@homehostportion;user=dialstring
VA_5.2	sip: dialled digits@homehostportion;user=phone
VA_5.3	sip: dialed digits; phone-context=<"+"CC>@homehostportion;user=phone
VA_5.3	sip: dialed digits; phone-context=<"+"CC+NDC>@homehostportion;user=phone

	Request-URI
VA_6.1	E164 Address (format "+"CC+NDC+SN) (e.g. as User info in SIP URI with user= phone, or as tel URI)

6.1 Void

6.2 Void

6.3 Void

6.4 Void

6.5 Test purposes for SIP-SIP

6.5.1 Test purposes for SIP-SIP, Basic call, Successful

SS___XX__01	**NGN reference to:** **RFC 3261 [2]** **TS 124 229 [11] / ES 283 003 [1]**
TSS reference:	SIP-SIP/Basic_call/Successful.
Selection criteria:	
Test purpose:	Ensure that call establishment and the correct handling and mapping of the SDP parameters of the INVITE message defined as : TYPE_SDP is performed correctly. Ensure that in the active call state the voice/data transfer on the media channels is performed correctly (e.g. testing QoS parameters). In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in table 1 applies. The call is released from the **called** user.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel and precondition.
Comments:	A) SDP pre-condition not requested **SIP** **SUT** **SIP** INVITE → → INVITE 100 Trying ← 180 Ringing ← ← 180 Ringing 200 OK INVITE ← ← 200 OK INVITE ACK → → ACK BYE ← ← BYE 200 OK BYE → → 200 OK BYE C) pre-condition and 100 rel INVITE → → INVITE 100 Trying ← ← 183 Session Progress SDP Authorize QoS resource at P-SCCF 183 Session Progress SDP ← PRACK → → PRACK Resource Reservation at UE1 200 OK PRACK ← ← 200 OK PRACK UPDATE → → UPDATE 200 OK UPDATE ← ← 200 OK UPDATE 180 Ringing ← ← 180 Ringing PRACK → → PRACK 200 OK PRACK ← ← 200 OK PRACK 200 OK INVITE ← ← 200 OK INVITE ACK → → ACK BYE ← ← BYE 200 OK BYE → → 200 OK BYE

SS___XX__02	**NGN reference to:** **RFC 3261 [2]** **TS 124 229 [11] / ES 283 003 [1]**	
TSS reference:	SIP-SIP/Basic_call/Successful.	
Selection criteria:		
Test purpose:	Ensure that call establishment and the correct handling and mapping of the SDP parameters of the INVITE message defined as : TYPE_SDP is performed correctly. Ensure that in the active call state the voice/data transfer on the media channels is performed correctly (e.g. testing QoS parameters). In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in table 1 applies. The call is released from the **calling** user.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel and precondition.	
Comments:		

A) SDP pre-condition not requested

SIP		**SUT**		**SIP**
INVITE	→		→	INVITE
100 Trying	←			
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE
C) pre-condition and 100 rel				
INVITE	→		→	INVITE
100 Trying	←			
			←	183 Session Progress SDP
		Authorize QoS resource at P-SCCF		
183 Session Progress SDP	←			
PRACK	→		→	PRACK
		Resource Reservation at UE1		
200 OK PRACK	←		←	200 OK PRACK
UPDATE	→		→	UPDATE
200 OK UPDATE	←		←	200 OK UPDATE
180 Ringing	←		←	180 Ringing
PRACK	→		→	PRACK
200 OK PRACK	←		←	200 OK PRACK
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS___XX__03	**NGN reference to:** **RFC 3261 [2]** **TS 124 229 [11] / ES 283 003 [1]**	
TSS reference:	SIP-SIP/Basic_call/Successful.	
Selection criteria:		
Test purpose:	Ensure that call establishment and the correct if the called user answers with a 180 Ringing message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters). In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in table 2 applies.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel and precondition	