International Telecommunication Union

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

Q.4014.2

(04/2019)

SERIES Q: SWITCHING AND SIGNALLING, AND ASSOCIATED MEASUREMENTS AND TESTS

Testing specifications – Testing specifications for SIP-IMS

PSTN/ISDN terminal equipment using IP multimedia core network subsystem; Conformance testing – Part 2: Test suite structure and test purposes

Recommendation ITU-T Q.4014.2

ITU-T

ITU-T Q-SERIES RECOMMENDATIONS

SWITCHING AND SIGNALLING, AND ASSOCIATED MEASUREMENTS AND TESTS

SIGNALLING IN THE INTERNATIONAL MANUAL SERVICE	Q.1–Q.3
INTERNATIONAL AUTOMATIC AND SEMI-AUTOMATIC WORKING	Q.4–Q.59
FUNCTIONS AND INFORMATION FLOWS FOR SERVICES IN THE ISDN	Q.60–Q.99
CLAUSES APPLICABLE TO ITU-T STANDARD SYSTEMS	Q.100–Q.119
SPECIFICATIONS OF SIGNALLING SYSTEMS No. 4, 5, 6, R1 AND R2	Q.120–Q.499
DIGITAL EXCHANGES	Q.500–Q.599
INTERWORKING OF SIGNALLING SYSTEMS	Q.600–Q.699
SPECIFICATIONS OF SIGNALLING SYSTEM No. 7	Q.700–Q.799
Q3 INTERFACE	Q.800–Q.849
DIGITAL SUBSCRIBER SIGNALLING SYSTEM No. 1	Q.850–Q.999
PUBLIC LAND MOBILE NETWORK	Q.1000–Q.1099
INTERWORKING WITH SATELLITE MOBILE SYSTEMS	Q.1100–Q.1199
INTELLIGENT NETWORK	Q.1200–Q.1699
SIGNALLING REQUIREMENTS AND PROTOCOLS FOR IMT-2000	Q.1700–Q.1799
SPECIFICATIONS OF SIGNALLING RELATED TO BEARER INDEPENDENT CALL CONTROL (BICC)	Q.1900–Q.1999
BROADBAND ISDN	Q.2000–Q.2999
SIGNALLING REQUIREMENTS AND PROTOCOLS FOR THE NGN	Q.3000–Q.3709
SIGNALLING REQUIREMENTS AND PROTOCOLS FOR SDN	Q.3710–Q.3899
TESTING SPECIFICATIONS	Q.3900–Q.4099
Testing specifications for next generation networks	Q.3900–Q.3999
Testing specifications for SIP-IMS	**Q.4000–Q.4039**
Testing specifications for Cloud computing	Q.4040–Q.4059
Testing specifications for IMT-2020 and IoT	Q.4060–Q.4099
SIGNALLING REQUIREMENTS AND PROTOCOLS FOR IMT-2020	Q.5000–Q.5049
COMBATING COUNTERFEITING AND STOLEN ICT DEVICES	Q.5050–Q.5069

For further details, please refer to the list of ITU-T Recommendations.

Recommendation ITU-T Q.4014.2

PSTN/ISDN terminal equipment using IP multimedia core network subsystem; Conformance testing – Part 2: Test suite structure and test purposes

Summary

Recommendation ITU-T Q.4014.2 is part 2 of the testing specifications of the terminal equipment used in the IMS-based public switched telephone network/integrated services digital network (PSTN/ISDN) emulation subsystem based on the media gateway control protocol, the session initiation protocol and the associated session description protocol.

The Recommendation specifies the test suite structure and test purposes (TSS and TP) to test PSTN/ISDN terminal equipment using Internet protocol (IP) multimedia core network subsystem.

History

Edition	Recommendation	Approval	Study Group	Unique ID*
1.0	ITU-T Q.4014.2	2019-04-29	11	11.1002/1000/13886

Keywords

IAD, IMS, PES, SDP, SIP, testing, TP, TSS, user side

* To access the Recommendation, type the URL http://handle.itu.int/ in the address field of your web browser, followed by the Recommendation's unique ID. For example, http://handle.itu.int/11.1002/1000/11830-en.

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications, information and communication technologies (ICTs). The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

Compliance with this Recommendation is voluntary. However, the Recommendation may contain certain mandatory provisions (to ensure, e.g., interoperability or applicability) and compliance with the Recommendation is achieved when all of these mandatory provisions are met. The words "shall" or some other obligatory language such as "must" and the negative equivalents are used to express requirements. The use of such words does not suggest that compliance with the Recommendation is required of any party.

INTELLECTUAL PROPERTY RIGHTS

ITU draws attention to the possibility that the practice or implementation of this Recommendation may involve the use of a claimed Intellectual Property Right. ITU takes no position concerning the evidence, validity or applicability of claimed Intellectual Property Rights, whether asserted by ITU members or others outside of the Recommendation development process.

As of the date of approval of this Recommendation, ITU had not received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementers are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database at http://www.itu.int/ITU-T/ipr/.

Table of Contents

Recommendation ITU-T Q.4014.2

PSTN/ISDN terminal equipment using IP Multimedia core network subsystem; Conformance testing – Part 2: Test suite structure and test purposes

1 Scope

This Recommendation specifies the test suite structure and test purposes to test PSTN/ISDN terminal equipment using IP multimedia core network subsystem.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

[ITU-T H.248.1] Recommendation ITU-T H.248.1 (2013), *Gateway control protocol: Version 3*.

[ITU-T Q.4014.1] Recommendation ITU-T Q.4014.1 (2019), *PSTN/ISDN terminal equipment using IP Multimedia core network subsystem; Conformance testing – Part 1: PICS.*

[ISO/IEC 9646-1] Recommendation ISO/IEC 9646-1, *Information technology – Open systems interconnection – Conformance testing methodology and framework – Part 1: General concepts.*

[ETSI TS 124 147] Recommendation ETSI TS 124 147 (2015-01), *Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Conferencing using the IP Multimedia (IM) Core Network (CN) subsystem; Stage 3*.

[ETSI TS 124 229] Recommendation ETSI TS 124 229 (2019), *Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3 (3GPP TS 24.229 Release 10).*

[ETSI TS 124 605] Recommendation ETSI TS 124 605 (2013-01), *Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Conference (CONF) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.605 version 10.1.0 Release 10).*

[ETSI TS 124 608] Recommendation ETSI TS 124 608 (2013-07), *Digital cellular telecommunications system (Phase 2+);Universal Mobile Telecommunications System (UMTS);LTE;Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.608 version 10.1.0 Release 10).*

[ETSI TS 124 628] Recommendation ETSI TS 124 628 (2013), *Digital cellular telecommunications system (Phase 2+);Universal Mobile Telecommunications System (UMTS);LTE;Common Basic Communication procedures using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.628 version 10.4.0 Release 10).*

[ETSI TS 129 163] Recommendation ETSI TS 129 163 (2019), *Digital cellular telecommunications system (Phase 2+) (GSM);Universal Mobile Telecommunications System (UMTS); LTE; Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks.*

[ETSI TS 183 036] Recommendation ETSI TS 183 036 (2012), *Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); ISDN/SIP interworking; Protocol specification.*

[ETSI TS 183 043] Recommendation ETSI TS 183 043 (2011), *Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS-based PSTN/ISDN Emulation; Stage 3 specification.*

[IETF RFC 768] Recommendation IETF RFC RFC 768 (1980), *User Datagram Protocol.*

[IETF RFC 793] Recommendation IETF RFC RFC 793 (1981), *Transmission Control Protocol.*

[IETF RFC 2327] Recommendation IETF RFC 2327 (1998), *SDP: Session Description Protocol.*

[IETF RFC 2805] Recommendation IETF RFC 2805 (2000), *Media Gateway Control Protocol Architecture and Requirements.*

[IETF RFC 3261] Recommendation IETF RFC 3261 (2002), *SIP: Session Initiation Protocol.*

[IETF RFC 3262] Recommendation IETF RFC 3262 (2002), *Integration of Resource Management and Session Initiation Protocol (SIP).*

[IETF RFC 3312] Recommendation IETF RFC 3312 (2002), *Reliability of Provisional Responses in the Session Initiation Protocol (SIP).*

[IETF RFC 3323] Recommendation IETF RFC 3325 (2002), *A Privacy Mechanism for the Session Initiation Protocol (SIP).*

[IETF RFC 3325] Recommendation IETF RFC 3325 (2002), *Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks.*

[IETF RFC 5009] Recommendation IETF RFC 5009 (2007), *Private Header (P-Header) Extension to Session Initiation Protocol (SIP) for Authorization of Early Media.*

[IETF RFC 6080] Recommendation IETF RFC 3312 (2011), *A Framework for Session Initiation Protocol User Agent Profile Delivery.*

[IETF RFC 6140] Recommendation IETF RFC 6140 (2011), *Registration for Multiple Phone Numbers in the Session Initiation Protocol (SIP).*

3 Definitions

3.1 Terms defined elsewhere

This Recommendation uses the following terms defined elsewhere:

3.1.1 access gateway [ETSI TS 183 043]: Gateway device that interworks a significant number of analogue lines/ISDN accesses (directly or via an V5 Access Network) to a packet network and is located at the operator's premises.

3.1.2 loose coupling [ETSI TS 183 043]: On-hook and flash-hook are analyzed in the AGCF/VGW; much like a simulation endpoint would operate.

3.1.3 media gateway (MGW) [ETSI TS 183 043]: Gateway device acting at the media/transport plane, providing the functions of an MGF.

3.1.4 media gateway controller (MGC) [ITU-T H.248.1]: Controls the parts of the call state that pertain to connection control for media channels in an MG.

3.1.5 residential gateway [ETSI TS 183 043]: Gateway device that interworks a small number of analogue lines/ISDN accesses.

3.1.6 tight coupling [ETSI TS 183 043]: On-hook and flash-hook are interpreted by the AS.

3.1.7 voice over IP gateway (VGW) [ETSI TS 183 043]: SIP-based gateway device that implements both a media gateway function and a media gateway controller function as defined in IETF RFC 2805 and supports the provision of voice based services to analogue lines/ISDN accesses.

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

3PTY	Three-Party Service
ACR	Anonymous Communication Rejection
AGCF	Access Gateway Control Function
AGW	Access Gateway
AOC-D	Advice of Charge During the call
AOC-E	Advice of Charge at the End of the call
AOC-S	Advice of Charge at call Set-up time
AS	Application Server
BC	Bearer Capability information element
BRI	Basic Rate Interface
CCBS	Call Completion on Busy Subscriber
CCNR	Call Completion on No Reply
CD	Call Deflection
CFB	Call Forwarding on Busy
CFU	Call Forwarding Unconditional
CLIP	Calling Line Identification Presentation
COLP	Connected Line identification Presentation
CSCF	Call Session Control Function

CUG	Closed User Group
CW	Communication Waiting
FQDN	Fully Qualified Domain Name
HLC	High Layer Compatibility
HOLD	call HOLD
IAD	Integrated Access Device
IMS	IP Multimedia Subsystem
IP	Internet Protocol
ISDN	Integrated Services Digital Network
IUT	Implementation Under Test
MCID	Malicious Communication Identification
MGC	Media Gateway Controller
MGF	Media Gateway Function
MIME	Multimodal Internet Mail Extensions
MWI	Message Waiting Indication
OIP	Originating Identification Presentation
OIR	Originating Identification
PICS	Protocol Implementation Conformance Statement
POTS	Plain Old Telephone Service
PSTN	Public Switched Telephone Network
S-CSCF	Serving CSCF
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SOC	Switching Order Command
SUT	System Under Test
TIP	Terminating Identification Presentation
TIR	Terminating Identification Restriction
TP	Test Purposes
TSS	Test Suite Structure
UA	User Agent
UE	User Equipment
URI	Uniform Resource Identifier
URN	User Requirements Notation
VGW	Voice over IP Gate Way
XML	Extensible Markup Language

5 Conventions

– <reference specification type> is "protocol";

– <reference specification id> is "ETSI TS 183 043";

– <reference specification title> is "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS-based PSTN/ISDN Emulation; Stage 3 specification";

– <reference specification description> is "IMS-based PSTN/ISDN Emulation; Stage 3 specification".

6 Test suite structure

Table 6-1 – Test suite structure

Registration	Initial_registration	TP_101_xxx
	User-initiated_re-registration	TP_102_xxx
	User-initiated_deregistration	TP_103_xxx
	Network-initiated_deregistration	TP_104_xxx
	Subscription_and_notification	TP_105_xxx
Call initiation_UE-originating_case	Orig_Establishment_of_an_early_dialogue	TP_201_xxx
	Orig_Establishment_of_a_confirmed_dialogue	TP_202_xxx
	Orig_Release_initiated_by_the_terminating_user	TP_203_xxx
	Orig_Release_initiated_by_the_terminating_user	TP_204_xxx
	Orig_Timers	TP_205_xxx
	Orig_Abnormal_situations	TP_206_xxx
Call initiation_UE_terminating_case	Term_Establishment_of_an_early_dialogue	TP_301_xxx
	Term_Establishment_of_a_confirmed_dialogue	TP_302_xxx
	Term_Release_initiated_by_the_originating_user	TP_303_xxx
	Term_Release_initiated_by_the_terminating_user	TP_304_xxx
	Term_Timers	TP_305_xxx
	Term_Abnormal_situations	TP_306_xxx
Emergency_service		TP_401_xxx
Supplementary_Service_control	OIP_OIR	TP_501_xxx
	TIP_TIR	TP_502_xxx
	HOLD	TP_503_xxx
	CDIV	TP_504_xxx
	3PTY	TP_505_xxx
	CUG	TP_506_xxx
	CW	TP_507_xxx
	TP	TP_508_xxx
	ECT	TP_509_xxx
	UUS	TP_510_xxx
	SUB	TP_511_xxx
	MCID	TP_512_xxx
	MWI	TP_513_xxx
	CCBS_CCNR	TP_514_xxx

7 Test purposes

7.1 Introduction

For each test requirement, a test purpose (TP) is defined.

All protocol implementation conformance statement (PICS) items referred to in this clause are as specified in [ITU-T Q.4014.1] unless indicated otherwise by another numbered reference.

7.1.1 TP naming convention

TPs are numbered, starting at 001, within each group. Groups are organized according to the test suite structure (TSS). Additional references are added to identify the actual test suite and whether or not it applies to the network (see Table 7.1.1).

Table 7.1.1 – TP identifier naming convention scheme

–	**Identifier: TP_<group>_<nnn>**			
–	**<group>**	**=**	**group 3 digit field representing group reference according to TSS**	
–	**<nnn>**	**=**	**sequential number**	**(001-999)**

7.1.2 Test strategy

As the base standards [ETSI TS 124 229] and [ETSI TS 183 036] contain no explicit requirements for testing, the TPs were generated as a result of an analysis of the base standard and the PICS specification [ITU-T Q.4014.1]. The criteria applied include the following:

– Whether or not a test case can be built from the TP is not considered.

7.2 Procedures at the user equipment

7.2.1 Registration and authentication

7.2.1.1 Initial registration

TSS Initial_registration	TP_101_001	Reference clause 8.1.3.5 and 22.2 of [IETF RFC 3261]	Selection expression
Test purpose *REGISTER request have to be authorized receipt of 401* Ensure that the SUT sends a REGISTER request, receives a 401 Unauthorized and the WWW- Authenticate header is present. The SUT repeats the REGISTER request. The Authorization header is present including 'username', 'realm', 'nonce', 'digest-uri' and 'response' HTTP parameters.			
SIP header values **401** WWW-Authenticate: **REGISTER 2** Authorization: username[proper value];realm=[proper value];nonce=[proper value];digest-uri=[proper value]			
Message flow **End device** / **Test equipment** **Make end device available** → REGISTER 1 ← 401 Unauthorized → REGISTER 2 ← 200 OK (REGISTER) **Apply post test routine**			

TSS Initial_registration	TP_101_002	Reference section 8.1.3.5 and 22.2 of [IETF RFC 3261]	Selection expression

Test purpose

REGISTER request have to be authorized, receipt of 407

Ensure that the SUT sends a REGISTER request, receives a 407 Proxy Authentication Required and the Proxy-Authenticate header is present. The SUT repeats the REGISTER request. The Proxy-Authorization header is present including 'username', 'realm', 'nonce', 'digest-uri' and 'response' HTTP parameters.

SIP header values

407

Proxy-Authenticate:

REGISTER 2

Proxy-Authorization: username=[proper value];realm=[proper value];nonce=[proper value];digest-uri=[proper value]

Message flow

End device		Test equipment
Make end device available		
	➔	REGISTER 1
	←	407 Proxy Authentication Required
	➔	REGISTER 2
	←	200 OK (REGISTER)

Apply post test routine

TSS Initial_registration	TP_101_003	Reference section 8.1.3.5 and 22.2 of [IETF RFC 3261]	Selection expression

Test purpose

REGISTER request have to be authorized, receipt of 407 the Cseq is incremented

Ensure that the SUT sends a REGISTER request, receives a 407 Proxy Authentication Required and the Proxy-Authenticate header is present. The SUT repeats the REGISTER request. The Proxy-Authorization header is present and the Cseq header value is incremented.

SIP header values

REGISTER 1

Call-ID [any value]

Cseq x REGISTER

REGISTER 2

Call-ID [same value as in REGISTER 1]

Cseq x+1 REGISTER

Message flow

End device		Test equipment
Make end device available		
	➔	REGISTER 1
	←	407 Proxy Authentication Required
	➔	REGISTER 2
	←	200 OK (REGISTER)

Apply post test routine

TSS Initial_registration	TP_101_004	Reference section 10.2 and 22 of [IETF RFC 3261]	Selection expression

Test purpose

Request line in the REGISTER request

Ensure that the SUT sends a REGISTER request to its registrar and there is no 'userpart' in the SIP-URI of the Request-line.

SIP header values

REGISTER

Request-line: sip:[hostpart=domain of location service]

Message flow

End device		Test equipment
Make end device available		
	→	REGISTER 1
	←	401 Unauthorized
	→	REGISTER 2
	←	200 OK (REGISTER)

Apply post test routine

TSS Initial_registration	TP_101_005	Reference section 10.3 and 22 of [IETF RFC 3261]	Selection expression

Test purpose

Successful final response to the REGISTER request

Ensure that the SUT receives a 200 OK Final response to the REGISTER request sent to its registrar and the Contact header and an 'expires' parameter in the Contact header.

SIP header values

200 OK (REGISTER)

Contact: sip:[any registered URI];expires=[any value]

Message flow

End device		Test equipment
Make end device available		
	→	REGISTER 1
	←	401 Unauthorized
	→	REGISTER 2
	←	200 OK (REGISTER)

Apply post test routine

TSS Initial_registration	TP_101_006	Reference section 10.2.6 of [IETF RFC 3261]	Selection expression PICS 5.2/1

Test purpose

Successful registration using a preconfigured registrar address

Ensure that the SUT sends a REGISTER request addressed to a pre-configured registrar without a 'userpart' in the Request URI.

<table>
<tr><td>SIP header values
REGISTER
Request-line: sip:[hostpart=preconfigured registrar address]</td></tr>
<tr><td>Message flow
End device | Test equipment
Make end device available
→ REGISTER 1
← 401 Unauthorized
→ REGISTER 2
← 200 OK (REGISTER)
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Initial_registration</td><td>TP_101_007</td><td>Reference
section 10.2.6 of
[IETF RFC 3261]</td><td>Selection expression
PICS 5.2/2</td></tr>
<tr><td colspan="4">Test purpose
Successful registration using the 'multicast' mechanism

Ensure that the SUT sends a REGISTER request to the "all SIP servers" multicast address "sip.mcast.net" without a 'userpart'.</td></tr>
<tr><td colspan="4">SIP header values
REGISTER
Request-line: sip: <sip.mcast.net>
OR
Request-line: sip: <224.0.1.75></td></tr>
<tr><td colspan="4">Message flow
End device | Test equipment
Make end device available.
→ REGISTER
← 401 Unauthorized
→ REGISTER
← 200 OK (REGISTER)
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Initial_registration</td><td>TP_101_008</td><td>Reference
section 10.2 of [IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
Successful registration To header

Ensure that the SUT sends a REGISTER request to the registrar and the To header with the address of the registrant and the type of the To header is 'sip'</td></tr>
<tr><td colspan="4">SIP header values
REGISTER
To: sip:[address of registrant]</td></tr>
</table>

Message flow **End device** **Test equipment** **Make end device available.** → REGISTER ← 401 Unauthorized → REGISTER ← 200 OK (REGISTER) **Apply post test routine**

TSS Initial_registration	TP_101_009	Reference section 10.2 of [IETF RFC 3261]	Selection expression
Test purpose *Successful registration From header* Ensure that the SUT sends a REGISTER request to the registrar and the 'From' header is set to the same value as the To header.			
SIP header values **REGISTER** To: sip:[address of registrant] From: sip:[address of registrant]			
Message flow **End device** **Test equipment** **Make end device available.** → REGISTER ← 401 Unauthorized → REGISTER ← 200 OK (REGISTER) **Apply post test routine**			

TSS Initial_registration	TP_101_010	Reference subclause 5.1.1.2.1 of [ETSI TS 124 229]	Selection expression PICS 5.2/1
Test purpose *Successful registration Via header for UDP* Ensure that the SUT sends a REGISTER request to the registrar and the 'Via' header contains a sent-by field containing the IP address or fully qualified domain name (FQDN) of the UE and the port number where the UE expects to receive the response to this request when UDP is used.			
SIP header values **REGISTER** Via: SIP/2.0/UDP [IP Address]:[Port number];branch=z9hG4bK[branch]; sent-by=[IP Address]:[Port number]			

Message flow **End device** **Test equipment** **Make end device available.** ➔ REGISTER ← 401 Unauthorized ➔ REGISTER ← 200 OK (REGISTER) **Apply post test routine**

TSS Initial_registration	TP_101_011	Reference subclause 5.1.1.2.1 of [ETSI TS 124 229]	Selection expression PICS 5.2/2
Test purpose *Successful registration Via header for TCP* Ensure that the SUT sends a REGISTER request to the registrar and the 'Via' header contains a "rport" header field parameter with no value in the Via header field when TCP is used.			
SIP header values **REGISTER** Via: SIP/2.0/UDP [IP Address]:[Port number];branch=z9hG4bK[branch];rport			
Message flow **End device** **Test equipment** **Make end device available.** ➔ REGISTER ← 401 Unauthorized ➔ REGISTER ← 200 OK (REGISTER) **Apply post test routine**			

TSS Initial_registration	TP_101_012	Reference subclause 5.1.1.2.1 of [ETSI TS 124 229]	Selection expression
Test purpose *Successful registration option-tag 'path' in the Supported header* Ensure that the SUT sends a REGISTER request to the registrar and the 'Supported' header contains the option-tag "path".			
SIP header values **REGISTER** Supported: path			
Message flow **End device** **Test equipment** Make end device available. ➔ REGISTER ← 401 Unauthorized ➔ REGISTER ← 200 OK (REGISTER) **Apply post test routine**			

TSS Initial_registration	TP_101_013	Reference subclause 5.1.1.2.1 of [ETSI TS 124 229]	Selection expression PICS 5.2/3

Test purpose

Successful registration option-tag 'gruu' in the Supported header

Ensure that the SUT sends a REGISTER request to the registrar and the 'Supported' header contains the option-tag "gruu" if the SUT supports GRUU.

SIP header values

REGISTER

Supported: gruu

Message flow

End device		Test equipment
Make end device available.		
	→	REGISTER
	←	401 Unauthorized
	→	REGISTER
	←	200 OK (REGISTER)

Apply post test routine

TSS Initial_registration	TP_101_014	Reference subclause 5.1.1.2.1 of [ETSI TS 124 229]	Selection expression PICS 5.2/4 and 5.2

Test purpose

Successful registration option-tag 'gin' in the Require header

Ensure that the SUT sends a REGISTER request to the registrar and the 'Require' header contains the option-tag "gin" if the SUT performs the functions of an external attached network.

SIP header values

REGISTER

Require: gin

Message flow

End device		Test equipment
Make end device available.		
	→	REGISTER
	←	401 Unauthorized
	→	REGISTER
	←	200 OK (REGISTER)

Apply post test routine

TSS Initial_registration	TP_101_015	Reference subclause 5.1.1.2.1 of [ETSI TS 124 229]	Selection expression PICS 5.2/4 and 5.2

Test purpose

Successful registration option-tag 'gin' in the Proxy-Require header

Ensure that the SUT sends a REGISTER request to the registrar and the 'Proxy-Require' header contains the option-tag "gin" if the SUT performs the functions of an external attached network.

SIP header values **REGISTER** Proxy-Require: gin		
Message flow		
End device		**Test equipment**
Make end device available.		
	→	REGISTER
	←	401 Unauthorized
	→	REGISTER
	←	200 OK (REGISTER)
	Apply post test routine	

TSS Initial_registration	TP_101_016	Reference section 10.2 of [IETF RFC 3261]	Selection expression
Test purpose *Successful registration. No new registration before successful final response* Ensure that the SUT does not send a new REGISTER request before it has received a successful final response to its previous REGISTER request or before expiration of the re-attempt timeout.			
SIP header values **REGISTER**			
Message flow			

End device		**Test equipment**
Make end device available.		
	→	REGISTER
	←	401 Unauthorized
	→	REGISTER
	Timeout initial request	
	→	REGISTER
	←	200 OK (REGISTER)
	Apply post test routine	

TSS Initial_registration	TP_101_017	Reference section 10.2 of [IETF RFC 3261]	Selection expression
Test purpose *Successful registration increments the Cseq in a new request* Ensure that the SUT sends a new REGISTER request and increments the Cseq header and uses the same Call-ID value as in the initial REGISTER request.			

SIP header values **REGISTER 1** Call-ID [any value] Cseq x REGISTER **REGISTER 2** Call-ID [same value as in REGISTER 1] Cseq x+1 REGISTER
Message flow **End device** **Test equipment** **SUT is already registered** → REGISTER 1 ← 401 Unauthorized → REGISTER 2 ← 200 OK (REGISTER) **Apply post test routine**

TSS Initial_registration	TP_101_018	Reference section 10.2.4 of [IETF RFC 3261]	Selection expression
Test purpose *Successful registration. Refreshing of registration* Ensure that the SUT refreshes each Binding given in the expires parameter of the Contact header received in the previous 200 OK REGISTER final response. The Call-ID is the same as in the previous REGISTER request and the Cseq header value is incremented.			
SIP header values **REGISTER 1** Call-ID [any value] Cseq x REGISTER **REGISTER 2** Call-ID [same value as in REGISTER 1] Cseq x+1 REGISTER			
Message flow **End device** **Test equipment** **SUT is already registered** → REGISTER ← 401 Unauthorized → REGISTER 2 ← 200 OK (REGISTER) **Apply post test routine**			

<table>
<tr><td>TSS
Initial_registration</td><td>TP_101_019</td><td>Reference
section 10.2.2 of [IETF RFC 3261]</td><td>Selection expression
PICS 5.2/6</td></tr>
<tr><td colspan="4">Test purpose
Successful registration removing a Binding

Ensure that the SUT is able to deregister a particular Binding by sending a REGISTER request and the expires parameter for that Binding is set to '0' or the Expires header is set to '0'.</td></tr>
<tr><td colspan="4">SIP header values
REGISTER
Contact: sip:[any registered URI];expires=0
OR
Contact: *
Expires: 0</td></tr>
<tr><td colspan="4">Message flow
End device Test equipment
SUT is already registered
→ REGISTER
← 401 Unauthorized
→ REGISTER
← 200 OK (REGISTER)
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Initial_registration</td><td>TP_101_020</td><td>Reference
section 10.2.1 of [IETF RFC 3261]</td><td>Selection expression
PICS 5.2/7</td></tr>
<tr><td colspan="4">Test purpose
The SUT gets it registered contacts

Ensure that the SUT, in order to get its registered contacts, sends a REGISTER request to its registrar without Contact header.</td></tr>
<tr><td colspan="4">SIP header values
REGISTER
[without Contact header]</td></tr>
<tr><td colspan="4">Message flow
End device Test equipment
Make end device available
→ REGISTER
← 200 OK (REGISTER)
Apply post test routine</td></tr>
</table>

TSS Initial_registration	TP_101_021	Reference Annex A and section 17.1.2.2 of [IETF RFC 3261]	Selection expression PICS 5.2/1

Test purpose

REGISTER request is repeated if no response is received

Ensure that the SUT or the IUT, having sent a REGISTER request, repeats its request after timer E set to T1 value expires, if an unreliable transport is used.

SIP header values

Message flow

End device		**Test equipment**
	➔	REGISTER
Start timer E		
Tomeout timer E		
	➔	REGISTER

Apply post test routine

TSS Initial_registration	TP_101_022	Reference Annex A and section 17.1.3 of [IETF RFC 3261]	Selection expression PICS 5.2/1

Test purpose

REGISTER request is repeated if no response is received

Ensure that the SUT, having sent a REGISTER request twice, repeats its request after timer E set to the MIN(2*T1,T2) value expires, if an unreliable transport is used.

SIP header values

Message flow

End device		**Test equipment**
	➔	REGISTER
Start timer E=T1		
Tomeout timer E		
	➔	REGISTER
Start timer E=2*T1		
Tomeout timer E		
	➔	REGISTER

Apply post test routine

TSS Initial_registration	TP_101_023	Reference Annex A and section 17.1.3 of [IETF RFC 3261]	Selection expression PICS 5.2/1

Test purpose

REGISTER request is repeated if no response is received

Ensure that the SUT, having sent a REGISTER request three times, repeats its request after timer E set to the MIN(4*T1,T2) value expires, If an unreliable transport is used.

SIP header values

Message flow **End device** → **Test equipment** ➔ REGISTER **Start timer E=T1** **Tomeout timer E** ➔ REGISTER **Start timer E=2*T1** **Tomeout timer E** ➔ REGISTER **Start timer E=4*T1** **Tomeout timer E** ➔ REGISTER **Apply post test routine**

TSS Initial_registration	TP_101_024	Reference Annex A and section 17.1.3 of [IETF RFC 3261]	Selection expression PICS 5.2/2 and 5.2/12
Test purpose *The REGISTER is not repeated after timer F expires* Ensure that the SUT does not repeat a REGISTER request, after timer F set to 64*T1 expires, If an unreliable transport is used.			
SIP header values			
Message flow **End device** **Test equipment** ➔ REGISTER Start timer F=64*T1 ➔ REGISTER ➔ REGISTER ➔ REGISTER Timeout timer F **Apply post test routine**			

TSS Initial_registration	TP_101_025	Reference Annex A and section 17.1.3 of [IETF RFC 3261]	Selection expression
Test purpose *REGISTER request in the Proceeding state is repeated if no response is received* Ensure that the SUT, when a REGISTER client transaction is in the Proceeding state, repeats its REGISTER request after timer E set to T1 value expires.			
SIP header values			

Message flow **End device**   **Test equipment** → REGISTER ← 100 Trying **Start timer E=T1** **Tomeout timer E** → REGISTER **Apply post test routine**

TSS Initial_registration	TP_101_026	Reference Annex A and section 17.1.3 of [IETF RFC 3261]	Selection expression
Test purpose *REGISTER request in the Proceeding state is repeated if no response is received* Ensure that the SUT, when a REGISTER client transaction is in the Proceeding state, repeats its REGISTER request after timer E set to T1 value expires.			
SIP header values			
Message flow **End device** **Test equipment** → REGISTER ← 100 Trying **Start timer E=T1** **Tomeout timer E** → REGISTER ← 100 Trying **Start timer E=T2** **Tomeout timer E** → REGISTER **Apply post test routine**			

TSS Initial_registration	TP_101_027	Reference Annex A and section 17.1.3 of [IETF RFC 3261]	Selection expression PICS 5.2/12
Test purpose *REGISTER request in the Proceeding state is not repeated if no response is received* Ensure that the SUT, when a REGISTER client transaction is in the Proceeding state and REGISTER request have already been repeated in this state, repeats its REGISTER request after timer E set to T2 value expires.			
SIP header values			

Message flow		
End device		**Test equipment**
	→	REGISTER
	←	100 Trying
Start timer F=64*T1		
	→	REGISTER
	←	100 Trying
	.....	
	→	REGISTER
	←	100 Trying
Timeout timer F		
	Apply post test routine	

TSS Initial_registration	TP_101_028	Reference subclause 5.1.1.2.1 of [ETSI TS124 229]	Selection expression
Test purpose *Receipt of a 305 (Use Proxy) response to the unprotected REGISTER request* Ensure that the SUT, on receipt of a 305 (Use Proxy) response to the unprotected REGISTER request, select a P-CSCF address from the address list, which is different from the previously used address, and perform the procedures for initial registration.			
SIP header values			
Message flow			

End device		**Test equipment**
	→	REGISTER
	←	305 Use Proxy
	→	REGISTER
	Apply post test routine	

TSS Initial_registration	TP_101_029	Reference subclause 5.1.1.2.1 of [ETSI TS124 229]	Selection expression
Test purpose *Receipt of a 423 (Interval Too Brief) response to the unprotected REGISTER request* Ensure that the SUT, on receipt of a 423 (Interval Too Brief) response to the unprotected REGISTER request, send another REGISTER request populating the registration expiration interval value with an expiration timer of at least the value received in the Min-Expires header field of the 423 (Interval Too Brief) response.			
SIP header values 423: Min-Expires: <new expires value> INVITE2 Expires: <new expires value>			

Message flow
End device → REGISTER ← 423 Interval Too Brief → REGISTER2 **Test equipment**
Apply post test routine

TSS Initial_registration	TP_101_030	Reference subclause 5.1.1.2.1 of [ETSI TS124 229]	Selection expression
Test purpose *Receipt of a 408 (Request Timeout) response to the unprotected REGISTER request* Ensure that the SUT, on receipt of a 408 (Request Timeout) response to the unprotected REGISTER request attempt to perform initial registration again.			
SIP header values			
Message flow **End device** / **Test equipment** → REGISTER ← 408 Request Timeout → REGISTER **Apply post test routine**			

TSS Initial_registration	TP_101_031	Reference subclause 5.1.1.2.1of [ETSI TS124 229]	Selection expression
Test purpose *Receipt of a 500 (Server Internal Error) response to the unprotected REGISTER request* Ensure that the SUT, on receipt of a 500 (Server Internal Error) response to the unprotected REGISTER request attempt to perform initial registration again.			
SIP header values			
Message flow **End device** / **Test equipment** → REGISTER ← 500 Server Internal Error → REGISTER2 **Apply post test routine**			

TSS Initial_registration	TP_101_032	Reference subclause 5.1.1.2.1 of [ETSI TS124 229]	Selection expression
Test purpose *Receipt of a 504 (Server Time-Out) response to the unprotected REGISTER request* Ensure that the SUT, on receipt of a 504 (Server Time-Out) response to the unprotected REGISTER request attempt to perform initial registration again.			
SIP header values			

Message flow **End device** **Test equipment** → REGISTER ← 504 Server Time-Out → REGISTER **Apply post test routine**

TSS Initial_registration	TP_101_033	Reference Annex A and section 17.1.3 of [IETF RFC 3261]	Selection expression
Test purpose *Receipt of a 600 (Busy Everywhere) response to the unprotected REGISTER request* Ensure that the SUT, on receipt of a 600 (Busy Everywhere) response to the unprotected REGISTER request attempt to perform initial registration again.			
SIP header values			
Message flow **End device** **Test equipment** → REGISTER ← 600 Busy Everywhere → REGISTER **Apply post test routine**			

TSS	TP_101_034	Reference Annex A and section 17.1.3 of [IETF RFC 3261]	Selection expression
Test purpose *Receipt of a* 403 (Forbidden) *response to the unprotected REGISTER request* Ensure that the SUT, on receipt of a 403 (Forbidden) response to the unprotected REGISTER request, considers the registration to have failed.			
SIP header values			
Message flow **End device** **Test equipment** → REGISTER ← 403 Forbidden **Apply post test routine**			

TSS Initial_registration	TP_101_035	Reference subclause 5.1.1.3 of [ETSI TS 124 229]	Selection expression PICS VGW

Test purpose

Subscription to the registration-state event package

Ensure that the SUT, on receipt of a 200 OK REGISTER response, subscribes to the 'reg' event package. The SUT sends a SUBSCRIBE request to the resource to which the UE wants to be subscribed to.

SIPheader values

SUBSCRIBE

Request URI: SIP URI which is the default public user identity used for subscription

From: [default public user identity used for subscription]

To: [default public user identity used for subscription]

Event: reg

Expires: 600000

200 OK SUBSCRIBE

Expires: [any value <=600000]

NOTIFY

Event: reg

Subscription-State: active; expires=<any value>

Message flow

End device		**Test equipment**
	→	REGISTER
	←	401 Unauthorized
	→	REGISTER
	←	200 OK REGISTER
CASE A	→	SUBSCRIBE
	←	2xx OK SUBSCRIBE
	←	NOTIFY
	→	200 OK NOTIFY
CASE B	→	SUBSCRIBE
	←	407 Proxy Authentication Required
	→	SUBSCRIBE
	←	2xx OK SUBSCRIBE
	←	NOTIFY
	→	200 OK NOTIFY

Apply post test routine

<table>
<tr><td>TSS
Initial_registration</td><td>TP_101_036</td><td>Reference
subclause 5.1.1.3 of
[ETSI TS124 229]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
Re-subscription to the registration-state event package

Ensure that the SUT, after the registration state event is subscribed and the expires time is reached, starts the re-subscription procedure. The SUT sends a new SUBSCRIPTION request to re-subscribe the registration state event.</td></tr>
<tr><td colspan="4">SIPheader values
SUBSCRIBE
Request URI: SIP URI which is the default public user identity used for subscription
From: [default public user identity used for subscription]
To: [default public user identity used for subscription]
Event: reg
Expires: 600000

200 OK SUBSCRIBE
Expires: [any value <=600000]

NOTIFY
Event: reg
Subscription-State : active; expires=<any value></td></tr>
<tr><td colspan="4">Message flow
End device Test equipment
SUT is already registered
CASE A → SUBSCRIBE
← 2xx OK SUBSCRIBE
← NOTIFY
→ 200 OK NOTIFY

CASE B → SUBSCRIBE
← 407 Proxy Authentication Required
→ SUBSCRIBE
← 2xx OK SUBSCRIBE
← NOTIFY
→ 200 OK NOTIFY
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Initial_registration</td><td>TP_101_037</td><td>Reference
subclause 5.1.1.5A and
5.1.1.4.1of
[ETSI TS124 229]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
Network-initiated re-authentication

Ensure that the SUT, upon receipt of a NOTIFY to a subscribed registration state event, starts the re-authentication procedures.</td></tr>
</table>

SIPheader values **NOTIFY** Event: reg Subscription-State : active Contact: <sip:[registered contact address]>;event= shortened;expires=[any value] **REGISTER** Request-line: sip:[hostpart=domain of location service] Call-ID [any value] Cseq x REGISTER From: sip:[address of registrant] To: sip:[address of registrant] Expires: 600000
Message flow **End device** **Test equipment** **SUT is already registered** ← NOTIFY → 200 OK NOTIFY → REGISTER ← 401 Unauthorized → REGISTER ← 200 OK (REGISTER) **Apply post test routine**

7.2.1.2 User-initiated re-registration and registration of an additional public user identity

<table>
<tr><td>TSS
User-initiated_re-registration</td><td>TP_102_001</td><td>Reference
subclause 5.1.1.4.1 of
[ETSI TS124 229]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
User initiated reregistration

Ensure that the SUT, after the SUT is registered, starts the reregistration procedure of a previously registered public user identity.</td></tr>
<tr><td colspan="4">SIP: Header values
REGISTER 1
Request-line: sip:[hostpart=domain of location service]
Call-ID [any value]
Cseq x REGISTER
From: sip:[address of registrant]
To: sip:[address of registrant]
Expires: 600000</td></tr>
</table>

Message flow **End device** **Test equipment** **SUT is already registered** The SUT has determined that a continued registration is not required → REGISTER ← 401 Unauthorized → REGISTER ← 200 OK REGISTER **Apply post test routine**

TSS User-initiated_re-registration	TP_102_002	Reference subclause 5.1.1.4.1 of [ETSI TS124 229]	Selection expression PICS 5.2/3
Test purpose *User initiated reregistration – GRUU supported* Ensure that the SUT, after the SUT is registered, starts the reregistration procedure of a previously registered public user identity.			
SIPheader values **REGISTER 1** Supported: path,gruu			
Message flow **End device** **Test equipment** **SUT is already registered** The SUT has determined that a continued registration is not required → REGISTER ← 401 Unauthorized → REGISTER ← 200 OK REGISTER **Apply post test routine**			

TSS User-initiated_re-registration	TP_102_003	Reference subclause 5.1.1.4.1 of [ETSI TS124 229]	Selection expression PICS 5.2/4
Test purpose *User initiated reregistration - multiple registrations is supported* Ensure that the SUT, after the SUT is registered, starts the reregistration procedure of a previously registered public user identity.			
SIPheader values **REGISTER 1** Supported: path,outbound			

Message flow **End device** **Test equipment** **SUT is already registered** The SUT has determined that a continued registration is not required → REGISTER ← 401 Unauthorized → REGISTER ← 200 OK REGISTER **Apply post test routine**

TSS User-initiated_re-registration	TP_102_004	Reference subclause 5.1.1.4.1 of [ETSI TS124 229]	Selection expression PICS 5.2/4 and 5.2/5
Test purpose *User initiated reregistration - functions of an external attached network is supported* Ensure that the SUT, after the SUT is registered, starts the reregistration procedure of a previously registered public user identity.			
SIPheader values **REGISTER 1** Require: gin			
Message flow **End device** **Test equipment** **SUT is already registered** The SUT has determined that a continued registration is not required → REGISTER ← 401 Unauthorized → REGISTER ← 200 OK REGISTER **Apply post test routine**			

7.2.1.3 User-initiated deregistration

TSS User-initiated_deregistration	TP_103_001	Reference subclause 5.1.1.6.1 of [ETSI TS124 229]	Selection expression
Test purpose *User initiated deregistration – general procedure* Ensure that the SUT, after the SUT is registered, starts the deregistration procedure of a previously registered public user identity.			

SIPheader values **REGISTER 1** Request-line: sip:[hostpart=domain of location service] From: sip:[address of registrant] To: sip:[address of registrant] Expires: 0
Message flow **End device** **Test equipment** **SUT is already registered** **The SUT has determined that a deregistration is required** → REGISTER ← 401 Unauthorized → REGISTER ← 200 OK REGISTER **Apply post test routine**

TSS User-initiated_deregistration	TP_103_002	Reference subclause 5.1.1.6.1 of [ETSI TS124 229]	Selection expression PICS 5.2/3
Test purpose *User initiated deregistration – GRUU supported* Ensure that the SUT, after the UE is registered, starts the deregistration procedure of a previously registered public user identity when GRUU is supported.			
SIPheader values **REGISTER 1** Supported: path,gruu Expires: 0			
Message flow **End device** **Test equipment** **SUT is already registered** **The SUT has determined that a deregistration is required** → REGISTER ← 401 Unauthorized → REGISTER ← 200 OK REGISTER **Apply post test routine**			

TSS User-initiated_deregistration	TP_103_003	Reference subclause 5.1.1.6.1 of [ETSI TS124 229]	Selection expression PICS 5.2/4
Test purpose *User initiated deregistration - multiple registrations is supported* Ensure that the SUT, after the UE is registered, starts the deregistration procedure of a previously registered public user identity when multiple registration is supported.			
SIPheader values **REGISTER 1** Supported: path,outbound Expires: 0			
Message flow **End device** **Test equipment** **SUT is already registered** **The SUT has determined that a deregistration is required** → REGISTER ← 401 Unauthorized → REGISTER ← 200 OK REGISTER **Apply post test routine**			

TSS User-initiated_deregistration	TP_103_004	Reference subclause 5.1.1.6.1 of [ETSI TS124 229]	Selection expression PICS 5.2/4 and 5.2/5
Test purpose *User initiated deregistration - functions of an external attached network is supported* Ensure that the SUT, after the UE is registered, starts the deregistration procedure of a previously registered public user identity when external attached network is performed.			
SIPheader values **REGISTER 1** Require: gin Expires: 0			
Message flow **End device** **Test equipment** **SUT is already registered** **The SUT has determined that a deregistration is required** → REGISTER ← 401 Unauthorized → REGISTER ← 200 OK REGISTER **Apply post test routine**			

7.2.1.4 Network-initiated deregistration

<table>
<tr><td>TSS
Network-initiated_deregistration</td><td>TP_104_001</td><td>Reference
subclause 5.1.1.7 of
[ETSI TS124 229]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
Network-initiated deregistration

Ensure that the SUT, upon receipt of a NOTIFY to a subscribed registration state event, starts the deregistration procedures.</td></tr>
<tr><td colspan="4">SIPheader values
NOTIFY
Event: reg
Subscription-State: terminated
Contact: <sip:[registered contact address]>;event= shortened;expires=[any value]

REGISTER 1
Request-line: sip:[hostpart=domain of location service]
Call-ID [any value]
Cseq x REGISTER
From: sip:[address of registrant]
To: sip:[address of registrant]
Expires: 0</td></tr>
<tr><td colspan="4">Message flow
End device Test equipment
SUT is already registered
← NOTIFY
→ 200 OK NOTIFY

→ REGISTER
← 401 Unauthorized
→ REGISTER
← 200 OK (REGISTER)
Apply post test routine</td></tr>
</table>

7.2.1.5 Subscription and notification

<table>
<tr><td>TSS
Subscription_and_notification</td><td>TP_105_001</td><td>Reference
subclause 5.3.1of
[ETSI TS183 043]</td><td>Selection expression
PICS 5.3/1</td></tr>
<tr><td colspan="4">Test purpose
Subscription to the User Agent Profile Delivery

Ensure that the SUT sends a SUBCRIBE to the network to subscribe to the User Agent Profile Delivery. The SUT receives a dial-tone-management XML element.</td></tr>
</table>

SIPheader values

SUBCRIBE

Event: ua-profile;profile-type=user ; vendor="[any reasonable value]";model="[any reasonable value]";version="[any reasonable value]"

NOTIFY

Event: ua-profile;profile-type=user ; vendor="[any reasonable value]";model="[any reasonable value]";version="[any reasonable value]"

<dial-tone-management>
<dial-tone-pattern>standard-dial-tone</dial-tone-pattern> (optional)
<dial-tone-pattern>special-condition-tone</dial-tone-pattern> (optional)
<dial-tone-pattern>message-waiting-tone</dial-tone-pattern> (optional)
</dial-tone-management>

Message flow

End device		**Test equipment**
	SUT is already registered	
	→	SUBCRIBE
	←	200 OK (SUBCSRIBE)
	←	NOTIFY
	→	200 OK (NOTIFY)
	Apply post test routine	

TSS Subscription_and_notification	TP_105_002	Reference subclause 5.3.1 of [ETSI TS183 043]	Selection expression PICS 5.3/1

Test purpose

Subscription to the Message waiting indication

Ensure that the SUT sends a SUBCRIBE to the network to subscribe to the Message Waiting indication service. The SUT receives in a NOTIFY request a MIME body with new reports about waiting messages.

SIPheader values

SUBCRIBE

Accept: application/simple-message-summary

Expires: <a valid value>

Event: message-summary

NOTIFY

Event: message-summary

Subscription-State: active

Content-Type: application/simple-message-summary

MIME body:

Messages-Waiting: yes

Message-Account: sip:served_user@Server

Voice-Message: 4/1 (2/0)

Video-Message: 3/1 (1/0)

Fax-Message: 2/1 (0/1)

Message flow

End device		Test equipment
	SUT is already registered	
	→	SUBCRIBE
	←	200 OK (SUBCSRIBE)
	←	NOTIFY
	→	200 OK (NOTIFY)
	Apply post test routine	

TSS Subscription_and_notification	TP_105_003	Reference subclause 5.3.1 of [ETSI TS183 043]	Selection expression PICS 5.3/1

Test purpose

Subscription to the Message waiting indication rejected with 503

Ensure that the SUT sends a SUBCRIBE to the network to subscribe to the Message Waiting indication service. The SUT receives a 503 (Service Unavailable) response to a SUBSCRIBE request containing a Retry-After header. The SUT does not automatically reattempt the request until after the period indicated by the Retry-After header contents

SIPheader values

SUBCRIBE

Accept: application/simple-message-summary

Expires: <a valid value>

Event: message-summary

503

Retry-After: 20

Message flow

End device		Test equipment
	SUT is already registered	
	→	SUBCRIBE
	←	200 OK (SUBCSRIBE)
	←	503 Service Unavailable
Wait for the time indicated in the Retry-After header		
	→	SUBCRIBE
	←	200 OK (SUBCSRIBE)
	Apply post test routine	

7.2.2 Call initiation – UE-originating case

7.2.2.1 Establishment of an early dialogue

7.2.2.1.1 SIP basic procedures

TSS Orig_Establishment_of_an_early_dialogue	TP_201_001	Reference section 8.1.1 of [IETF RFC 3261]	Selection expression

Test purpose

Sending of INVITE request containing all mandatory headers

Ensure that the SUT is able to sent an INVITE request containing all mandatory SIP headers:

SIP Header values INVITE To From CSeq Call-ID Max-Forwards Contact Via
Message flow **End device** **Test equipment** **Interworking POTS** Off hook Dial number → INVITE ← 100 Trying **ISDN interworking** SETUP → INVITE ← 100 Trying **Apply post test routine**

TSS Orig_Establishment_of_an_early_dialogue	TP_201_002	Reference section 8.1.1. of [IETF RFC 3261]	Selection expression
Test purpose *The Request line is set to the same URI value of the To header* Ensure that the IUT is able to establish a session and the Request line URI is set to the same value as the To header URI			
SIP header values INVITE sip: [Request URI] SIP/2.0 To: < sip: [Request URI]>			
Message flow **End device** **Test equipment** **Interworking POTS** Off hook Dial number → INVITE ← 100 Trying **ISDN interworking** SETUP → INVITE ← 100 Trying **Apply post test routine**			

<table>
<tr><td>TSS
Orig_Establishment_of_an_early_dialogue</td><td>TP_201_003</td><td>Reference
section 8.1.1.2 of [IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
The To header does not contain a 'tag' parameter

Ensure that the IUT is able to establish a session and the 'To' header in the INVITE request does not contain the 'tag' parameter</td></tr>
<tr><td colspan="4">SIP header values
INVITE
To: <To header URI></td></tr>
<tr><td colspan="4">Message flow
End device | Test equipment
Interworking POTS
Off hook
Dial number → INVITE
← 100 Trying

ISDN interworking
SETUP → INVITE
← 100 Trying
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Orig_Establishment_of_an_early_dialogue</td><td>TP_201_004</td><td>Reference
section 8.1.1.3 of [IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
The From header contains a 'tag' parameter

Ensure that the IUT is able to establish a session and the 'From' header in the INVITE request contains the 'tag' parameter</td></tr>
<tr><td colspan="4">SIP header values
INVITE
From: <To header URI>;tag=[any value]</td></tr>
<tr><td colspan="4">Message flow
End device | Test equipment
Interworking POTS
Off hook
Dial number → INVITE
← 100 Trying

ISDN interworking
SETUP → INVITE
← 100 Trying
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Orig_Establishment_of_an_early_dialogue</td><td>TP_201_005</td><td>Reference
section 8.1.1.5 of
[IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
The CSeq header contains the method INVITE

Ensure that the IUT is able to establish a session and the INVITE includes a CSeq header and the method parameter is set to 'INVITE'.</td></tr>
<tr><td colspan="4">SIP header values
INVITE
CSeq: <any number> INVITE</td></tr>
<tr><td colspan="4">Message flow
End device | Test equipment
Interworking POTS
Off hook
Dial number → INVITE
← 100 Trying

ISDN interworking
SETUP → INVITE
← 100 Trying
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Orig_Establishment_of_an_early_dialogue</td><td>TP_201_006</td><td>Reference
section 8.1.1.6 of
[IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
Max-Forwards initial value

Ensure that the IUT is able to establish a session and the INVITE contains a Max-Forwards header set to the initial value '70'</td></tr>
<tr><td colspan="4">SIP header values
INVITE
Max-Forwards: 70</td></tr>
<tr><td colspan="4">Message flow
End device | Test equipment
Interworking POTS
Off hook
Dial number → INVITE
← 100 Trying

ISDN interworking
SETUP → INVITE
← 100 Trying
Apply post test routine</td></tr>
</table>

TSS Orig_Establishment_of_an_early_dialogue	TP_201_007	Reference section 8.1.1.7 of [IETF RFC 3261]	Selection expression

Test purpose

Coding of Via header in the INVITE

Ensure that the IUT is able to establish a session and the INVITE contains a Via header with a protocol name set to SIP, a protocol version set to 2.0 and a branch parameter set to a value beginning with "z9hG4bK".

SIP header values

INVITE

Via:SIP 2.0 <transport> <IP address>;branch= z9hG4bK.......

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	100 Trying
ISDN interworking		
SETUP	→	INVITE
	←	100 Trying

Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_008	Reference section 13.2.1 of [IETF RFC 3261]	Selection expression

Test purpose

Allow and Supported header in the INVITE

Ensure that the IUT is able to establish a session and the INVITE contains the Allow and Supported header

SIP header values

INVITE

Allow: <any extensions>

Supported: <any methods>

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	100 Trying
ISDN interworking		
SETUP	→	INVITE
	←	100 Trying

Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_009	Reference sections 8, 8.1.3.2, 13.2.2.1 and Figure 5 of [IETF RFC 3261]	Selection expression
Test purpose *100 received, the client enters the Proceeding state* Ensure that the IUT is able to establish a session and the INVITE client transaction is in the Calling state, on receipt of a Trying (100 Trying) response enters in the Proceeding state.			
SIP header values			
Message flow			

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	100 Trying
ISDN interworking		
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying
Apply post test routine		

TSS Orig_Establishment_of_an_early_dialogue	TP_201_010	Reference sections 8, 8.1.3.2, 13.2.2.1 and Figure 5 of [IETF RFC 3261]	Selection expression
Test purpose *183 received, the client enters the Proceeding state* Ensure that the IUT is able to establish a session and the INVITE client transaction is in the Calling state, on receipt of a Session Progress (183 Session Progress) response enters in the Proceeding state.			
SIP header values			
Message flow			

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	183 Session Progress
ISDN interworking		
SETUP	→	INVITE
CALL PROCEEDING	←	183 Session Progress
Apply post test routine		

TSS Orig_Establishment_of_an _early_dialogue	TP_201_011	Reference sections 8, 8.1.3.2, 13.2.2.1, and Figure 5 of [IETF RFC 3261]	Selection expression
Test purpose *180 received, the client enters the Proceeding state* Ensure that the IUT is able to establish a session and the INVITE client transaction is in the Calling state, on receipt of a Ringing (180 Ringing) response enters in the Proceeding state.			
SIP header values			
Message flow			

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	180 Ringing
ISDN interworking		
SETUP	→	INVITE
ALERTING	←	180 Ringing

Apply post test routine

TSS Orig_Establishment_of_an _early_dialogue	TP_201_012	Reference sections 8, 8.1.3.2, 13.2.2.1 and Figure 5 of [IETF RFC 3261]	Selection expression
Test purpose *198 received, the client enters the Proceeding state* Ensure that the IUT is able to establish a session and the INVITE client transaction is in the Calling state, on receipt of a Unknown (198 Unknown) response enters in the Proceeding state.			
SIP header values			
Message flow			

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	198 Unknown
ISDN interworking		
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	198 Unknown

Apply post test routine

TSS	TP_201_013	Reference	Selection expression
Orig_Establishment_of_an_early_dialogue		sections 8, 8.1.3.2, 13.2.2.1 and Figure 5 [IETF RFC 3261]	

Test purpose

100 received, the client remains the Proceeding state

Ensure that the IUT is able to establish a session and the INVITE client transaction is in the Proceeding state, on receipt of a Trying (100 Trying) response stays in the Proceeding state.

SIP header values

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	100 Trying
ISDN interworking		
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying

Apply post test routine

TSS	TP_201_014	Reference	Selection expression
Orig_Establishment_of_an_early_dialogue		section 20.14 and 13.2.1 of [IETF RFC 3261]	

Test purpose

Content-Length header and SDP in the initial INVITE

Ensure that the IUT is able to establish a session and the INVITE request contains the Content-Length header set to the size of the body in the message that contains the session description.

SIP header values

INVITE

Content-Length: <any value>

SDP

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	100 Trying
ISDN interworking		
SETUP	→	INVITE
	←	100 Trying

Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_015	Reference section 20.15 and 13.2.1 of [IETF RFC 3261]	Selection expression
Test purpose *Content-Type header and SDP in the initial INVITE* Ensure that the IUT is able to establish a session and the INVITE request contains the Content-Type header in the message that contains the session description.			
SIP header values INVITE Content-Type: application/sdp SDP			
Message flow **End device** / **Test equipment** **Interworking POTS** Off hook Dial number → INVITE ← 100 Trying **ISDN interworking** SETUP → INVITE ← 100 Trying **Apply post test routine**			

TSS Orig_Establishment_of_an_early_dialogue	TP_201_016	Reference section 20.14 and 13.2.1 of [IETF RFC 3261]	Selection expression PICS 5.1.2/2
Test purpose *Content-Length header and SDP in the initial INVITE and TCP* Ensure that the IUT is able to establish a session and the INVITE request contains the Content-Length header set to the size of the body in the message that contains the session description and a reliable transport (TCP) is used.			
SIP header values INVITE Content-Type: application/sdp SDP			
Message flow **End device** / **Test equipment** **Interworking POTS** Off hook Dial number → INVITE ← 100 Trying **ISDN interworking** SETUP → INVITE ← 100 Trying **Apply post test routine**			

<table>
<tr><td>TSS
Orig_Establishment_of_an_early_dialogue</td><td>TP_201_017</td><td>Reference
section 8.1.3.5 and 22.2 of [IETF RFC 3261]</td><td>Selection expression on</td></tr>
<tr><td colspan="4">Test purpose
The CSeq in the repeated INVITE is incremented

Ensure that the IUT is able to establish a session and the INVITE request, on receipt of an Unauthorized (401 Unauthorized) response including a WWW-Authenticate header, repeats its INVITE request with an Authorization header and with an incremented Cseq value.</td></tr>
<tr><td colspan="4">SIP header values
INVITE1:
CSeq: <value > INVITE
INVITE2:
Authorization:
CSeq: <value +1> INVITE</td></tr>
<tr><td colspan="4">Message flow</td></tr>
<tr><td colspan="2">End device</td><td></td><td>Test equipment</td></tr>
<tr><td colspan="4">Interworking POTS</td></tr>
<tr><td colspan="2">Off hook</td><td></td><td></td></tr>
<tr><td colspan="2">Dial number</td><td>→</td><td>INVITE 1</td></tr>
<tr><td colspan="2"></td><td>←</td><td>401 Unauthorized</td></tr>
<tr><td colspan="2"></td><td>→</td><td>ACK</td></tr>
<tr><td colspan="2"></td><td>→</td><td>INVITE 2</td></tr>
<tr><td colspan="4">ISDN interworking</td></tr>
<tr><td colspan="2">SETUP</td><td>→</td><td>INVITE 1</td></tr>
<tr><td colspan="2"></td><td>←</td><td>401 Unauthorized</td></tr>
<tr><td colspan="2"></td><td>→</td><td>ACK</td></tr>
<tr><td colspan="2"></td><td>→</td><td>INVITE 2</td></tr>
<tr><td colspan="4">Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Orig_Establishment_of_an_early_dialogue</td><td>TP_201_018</td><td>Reference
section 8.1.3.5 and 22.2 of [IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
Authorization header present in the repeated INVITE

Ensure that the IUT is able to establish a session and the INVITE request, on receipt of a 401 Unauthorized response including a WWW-Authenticate header, repeats its INVITE request with an Authorization header including proper values for username, realm nonce, digest-uri and response HTTP parameters.</td></tr>
<tr><td colspan="4">SIP header values
INVITE2:
Authorization: Digest username="=<any value>" , realm="=<any value>",nonce="=<any value>",uri="=<any value>", qop=auth, cnonce="=<any value>",response="=<any value>",algorithm=<any value></td></tr>
</table>

Message flow **End device** — **Test equipment** **Interworking POTS** Off hook Dial number → INVITE1 ← 401 Unauthorized → ACK → INVITE2 **ISDN interworking** SETUP → INVITE ← 401 Unauthorized → ACK → INVITE2 **Apply post test routine**

TSS Orig_Establishment_of_an_early_dialogue	TP_201_019	Reference section 8.1.3.5 and 22.2 of [IETF RFC 3261]	Selection expression
Test purpose *The CSeq in the repeated INVITE is incremented* Ensure that the IUT is able to establish a session and the INVITE request, on receipt of a 407 Proxy Authentication Required response including a Proxy-Authenticate header, repeats its INVITE request with an Proxy-Authorization header and with an incremented Cseq value.			
SIP header values INVITE1: CSeq: <value > INVITE INVITE2: Proxy-Authenticate: CSeq: <value +1> INVITE			
Message flow **End device** — **Test equipment** **Interworking POTS** Off hook Dial number → INVITE ← 407 Proxy Authentication Required → ACK → INVITE2 **ISDN interworking** SETUP → INVITE ← 407 Proxy Authentication Required → ACK → INVITE2 **Apply post test routine**			

<table>
<tr><td>TSS
Orig_Establishment_of_an_early_dialogue</td><td>TP_201_020</td><td>Reference
section 8.1.3.5 and 22.2 of [IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
Proxy-Authorization header present in the repeated INVITE

Ensure that the IUT is able to establish a session and the INVITE request, on receipt of an 407 Proxy Authentication Required response including a Proxy-Authenticate header, repeats its INVITE request with an Proxy-Authorization header including proper values for username, realm nonce, digest-uri and response HTTP parameters.</td></tr>
<tr><td colspan="4">SIP header values
INVITE2:
Proxy-Authorization: Digest username="=<any value>" , realm="=<any value>",nonce="=<any value>",uri="=<any value>", qop=auth, cnonce="=<any value>",response="=<any value>",algorithm=<any value></td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><td>End device</td><td></td><td>Test equipment</td></tr>
<tr><td>Interworking POTS</td><td></td><td></td></tr>
<tr><td>Off hook</td><td></td><td></td></tr>
<tr><td>Dial number</td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td>←</td><td>407 Proxy Authentication Required</td></tr>
<tr><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td>→</td><td>INVITE2</td></tr>
<tr><td>ISDN interworking</td><td></td><td></td></tr>
<tr><td>SETUP</td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td>←</td><td>407 Proxy Authentication Required</td></tr>
<tr><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td>→</td><td>INVITE2</td></tr>
</table>
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Orig_Establishment_of_an_early_dialogue</td><td>TP_201_021</td><td>Reference
subclause 5.1.3.1 of [ETSI TS124 229]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
Accept header with support of SDP in the initial INVITE

Ensure that the IUT is able to generate an initial INVITE request, the UE shall include the Accept header field with "application/sdp".</td></tr>
<tr><td colspan="4">SIP header values
INVITE2:
Accept: < application/sdp ></td></tr>
</table>

Message flow **End device** — **Test equipment** **Interworking POTS** Off hook Dial number → INVITE ← 407 Proxy Authentication Required → ACK → INVITE2 **ISDN interworking** SETUP → INVITE ← 407 Proxy Authentication Required → ACK → INVITE2 **Apply post test routine**

TSS Orig_Establishment_of_an_early_dialogue	TP_201_022	Reference subclause 5.1.3.1 of [ETSI TS124 229]	Selection expression PICS 5.2/14
Test purpose *Precondition support in the initial INVITE* Ensure that the IUT is able to generating an initial INVITE request with indication of support of resource reservation, the SUT shall include the Support header field with "precondition" and "100rel".			
SIP header values INVITE2: Supported: precondition,100rel SDP . a=curr:qos local none a=curr:qos remote none a=des:qos mandatory/optional local sendrecv a=des:qos mandatory/optional remote sendrecv			
Message flow **End device** — **Test equipment** **Interworking POTS** Off hook Dial number → INVITE ← 407 Proxy Authentication Required → ACK → INVITE2 **ISDN interworking** SETUP → INVITE ← 407 Proxy Authentication Required → ACK → INVITE2 **Apply post test routine**			

TSS Orig_Establishment_of_an_early_dialogue	TP_201_023	Reference subclause 5.1.3.1 of [ETSI TS124 229]	Selection expression PICS 5.2/14
Test purpose *Successful resource reservation procedure* Ensure that the IUT is able to complete the resource reservation procedure. The (reliable) 183 contains in the SDP the instruction to reserve the quality of service on the SUT side. The SUT indicates that resource is reserved in the UPDATE request and the remote side confirms the resource reservation in the 200 OK UPDATE.			
SIP header values INVITE2: Supported: precondition,100rel SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory/optional local sendrecv a=des:qos mandatory/optional remote sendrecv 183: Require: precondition,100rel SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory/optional local sendrecv a=des:qos mandatory/optional remote sendrecv a=conf:qos remote sendrecv UPDATE SDP a=curr:qos local sendrecv a=curr:qos remote none a=des:qos mandatory/optional local sendrecv a=des:qos mandatory/optional remote sendrecv 200 OK UPDATE SDP a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory/optional local sendrecv a=des:qos mandatory/optional remote sendrecv			

Message flow		
End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
		183 Session progress
		PRACK
		200 OK PRACK
		UPDATE
		200 OK UPDATE
ISDN interworking		
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
		183 Session progress
		PRACK
		200 OK PRACK
		UPDATE
		200 OK UPDATE
Apply post test routine		

TSS Orig_Establishment_of_an_early_dialogue	TP_201_024	Reference subclause 5.1.3.1 of [ETSI TS124 229]	Selection expression PICS 5.2/14
Test purpose *Successful resource reservation procedure* Ensure that the IUT upon receiving a 421 (Extension Required) response to an initial INVITE request in which the precondition mechanism was not used, including the "precondition" option-tag in the Require header field, the SUT sends a new INVITE request and the Supported header contains the precondition option tag.			

SIP header values

INVITE3:

Supported: precondition,100rel

SDP a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory/optional local sendrecv
a=des:qos mandatory/optional remote sendrecv

183:

Require: precondition,100rel

SDP a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory/optional local sendrecv
a=des:qos mandatory/optional remote sendrecv
a=conf:qos remote sendrecv

UPDATE/PRACK

SDP a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory/optional local sendrecv
a=des:qos mandatory/optional remote sendrecv

200 OK UPDATE/200 OK PRACK

SDP a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory/optional local sendrecv
a=des:qos mandatory/optional remote sendrecv

Message flow		
End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
	←	421 Extension Required
	→	ACK
	→	INVITE3
	←	183 Session progress
	→	PRACK
	←	200 OK PRACK
	→	UPDATE
	←	200 OK UPDATE
ISDN interworking		
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
	←	421 Extension Required
	→	ACK
	→	INVITE3
	←	183 Session progress
	→	PRACK
	←	200 OK PRACK
	→	UPDATE
	←	200 OK UPDATE
	Apply post test routine	

TSS Orig_Establishment_of_an_early_dialogue	TP_201_025	Reference subclause 5.1.3.1 of [ETSI TS124 229]	Selection expression PICS 5.2/14
Test purpose *SUT wishes to receive early media authorization indications* Ensure that the IUT, if it wishes to receive early media authorization indications, the SUT inserts the P-Early-Media header field with the "supported" parameter to the INVITE request.			
SIP header values INVITE2: P-Early-Media: supported			

Message flow		
End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
ISDN interworking		
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
Apply post test routine		

TSS Orig_Establishment_of_an_early_dialogue	TP_201_026	Reference section 4 of [IETF RFC 3262]	Selection expression PICS 5.2/15
Test purpose *SUT supports the reliable provisional response procedure* Ensure that the IUT supports the reliable provisional response procedure, the "supported" header set to '100rel' is present in the INVITE request. Upon a 18x provisional response containing a Require header set to '100rel' and an RSeq header, the SUT sends a PRACK request containing the RAck header. The response-num is equal to the value in the received RSeq header. The CSeq-num is equal to the CSeq value in the sent INVITE. The Method parameter is set to 'INVITE'.			
SIP header values INVITE2: Supported: 100rel CSeq: 2 INVITE 183: Require: 100rel RSeq: 3 PRACK RAck: 3 2 INVITE			

Message flow		
End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
CASE A	←	183 Session Progress
	→	PRACK
	←	200 OK (PRACK)
CASE B	←	183 Session Progress
	→	PRACK
	←	200 OK (PRACK)
ISDN interworking		
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
CASE A	←	183 Session Progress
	→	PRACK
	←	200 OK (PRACK)
CASE B	←	183 Session Progress
	→	PRACK
	←	200 OK (PRACK)
	Apply post test routine	

TSS Orig_Establishment_of_an_early_dialogue	TP_201_027	Reference subclause 5.1.3.1 of [ETSI TS124 229] Section 6 and 8 of [IETF RFC 5009]	Selection expression PICS5.2/16
Test purpose *SUT presents received early media to the user. 183 received, no P-Early-Media header present.* Ensure that the IUT upon receiving a 183 Session Progress provisional response without a P-Early-Media header field and SDP answer is received, the SUT presents the received early media to the user. Forward early media is not gated.			
SIP header values INVITE2: P-Early-Media: supported SDP offer 183: SDP answer			

Message flow		
End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
	←	183 Session Progress
Media	←	**Media**
Media	→	
ISDN interworking		
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
	←	183 Session Progress
Media	←	**Media**
Media	→	
Apply post test routine		

TSS Orig_Establishment_of_an_early_dialogue	TP_201_028	Reference subclause 5.1.3.1 of [ETSI TS124 229] section 6 and 8 of [IETF RFC 5009]	Selection expression PICS5.2/16
Test purpose *SUT presents received early media to the user. 180 received, no P-Early-Media header present.* Ensure that the IUT upon receiving a 180 Ringing provisional response without a P-Early-Media header field and an SDP answer is received, the SUT presents the received early media to the user. Forward early media is not gated.			
SIP header values INVITE2: P-Early-Media: supported SDP offer 183: SDP answer			

Message flow		
End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
	←	183 Session Progress
Media	←	**Media**
Media	→	
ISDN interworking		
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
	←	183 Session Progress
Media	←	**Media**
Media	→	
Apply post test routine		

TSS Orig_Establishment_of_an_early_dialogue	TP_201_029	Reference clause 5.1.3.1 of [ETSI TS124 229] section 6 and 8 of [IETF RFC 5009]	Selection expression PICS5.2/16
Test purpose *SUT presents received early media to the user. 183 received, P-Early-Media header present.* Ensure that the IUT upon receiving a 180 Ringing provisional response with a P-Early-Media header field indicating authorized early media and SDP answer is received, the SUT presents the received early media to the user. In case of P-Early-Media sendrecv, early media generated by the user is presented to the network.			
SIP header values INVITE2: P-Early-Media: supported SDP offer 180: P-Early-Media: [sendonly]/[sendrecv] SDP answer			

Message flow		
End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
	←	180 Ringing
CASE A sendonly		
Media	←	**Media**
Media	→	
CASE A sendrecv		
Media	←	**Media**
Media	→	**Media**
ISDN interworking		
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
	←	180 Ringing
CASE A sendonly		
Media	←	**Media**
Media	→	
CASE A sendrecv		
Media	←	**Media**
Media	→	**Media**
	Apply post test routine	

TSS Orig_Establishment_of_an_early_dialogue	TP_201_030	Reference subclause 5.1.3.1 of [ETSI TS124 229] section 6 sand 8 of [IETF RFC 5009]	Selection expression PICS5.2/16
Test purpose *SUT presents received early media to the user. 183 received, P-Early-Media header present.* Ensure that the IUT upon receiving a 183 Session Progress provisional response with a P-Early-Media header field indicating authorized early media and SDP answer is received, the SUT presents the received early media to the user. In case of P-Early-Media sendrecv, early media generated by the user is presented to the network.			

SIP header values INVITE2: P-Early-Media: supported SDP offer 183: P-Early-Media: [sendonly]/[sendrecv] SDP answer

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
	←	183 Session Progress
CASE A sendonly		
Media	←	**Media**
Media	→	
CASE A sendrecv		
Media	←	**Media**
Media	→	**Media**
ISDN interworking		
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
	←	183 Session Progress
CASE A sendonly		
Media	←	**Media**
Media	→	
CASE A sendrecv		
Media	←	**Media**
Media	→	**Media**

Apply post test routine

TSS Orig_Establishment_of_an _early_dialogue	TP_201_031	Reference subclause 5.1.3.1 of [ETSI TS124 229] section 6 and 8 [IETF RFC 5009]	Selection expression PICS5.2/16

Test purpose

SUT does not present early media to the user. 183 received, no P-Early-Media header present.

Ensure that the IUT upon receiving a 183 Session Progress provisional response without receipt of a P-Early-Media header field, the SUT does not present early media to the user if no RTP is received from the network. Forward early media is not gated.

SIP header values

INVITE2:

P-Early-Media: supported

SDP offer

183:

SDP answer yes/no

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
	←	183 Session Progress
ISDN interworking		
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
	←	183 Session Progress

Apply post test routine

TSS Orig_Establishment_of_an _early_dialogue	TP_201_032	Reference subclause 5.1.3.1 of [ETSI TS124 229] section 6 and 8 of [IETF RFC 5009]	Selection expression PICS5.2/16

Test purpose

SUT does not present early media to the user. 183 received, P-Early-Media header present.

Ensure that the IUT upon receiving a 183 Session Progress provisional response a P-Early-Media header field is received, value sendonly or sendrecv, the SUT does not present early media to the user if no RTP is received from the network. Forward early media is not gated.

SIP header values INVITE2: P-Early-Media: supported SDP offer 183: P-Early-Media: [sendonly]/[sendrecv] SDP answer yes/no

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
	←	183 Session Progress
ISDN interworking		
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
	←	183 Session Progress

Apply post test routine

TSS Orig_Establishment_of_an _early_dialogue	TP_201_033	Reference subclause 5.1.3.1 of [ETSI TS124 229] section 6 and 8 of [IETF RFC 5009]	Selection expression PICS5.2/16

Test purpose

SUT does not present early media to the user. 183 received, P-Early-Media header present.

Ensure that the IUT upon receiving a 183 Session Progress provisional response a P-Early-Media header field is received, value inactive or recvonly, the SUT does not presents early media to the user. Forward early media is not gated.

SIP header values

INVITE2:

P-Early-Media: supported

SDP offer

183:

P-Early-Media: [sendonly]/[sendrecv]

SDP answer yes/no

Message flow **End device** **Test equipment** **Interworking POTS** Off hook Dial number → INVITE ← 407 Proxy Authentication Required → ACK → INVITE2 ← 183 Session Progress **ISDN interworking** SETUP → INVITE ← 407 Proxy Authentication Required → ACK → INVITE2 ← 183 Session Progress **Apply post test routine**

TSS Orig_Establishment_of_an_early_dialogue	TP_201_034	Reference subclause 5.1.3.1 of [ETSI TS124 229] section 6 and 8 of [IETF RFC 5009]	Selection expression PICS5.2/16
Test purpose *SUT local ring back tone is presented to the user. 183 received, P-Early-Media header not present.* Ensure that the IUT upon receiving a 183 Session Progress provisional response a P-Early-Media header field is not received, the SUT play ringing tones for the caller, indicating that the callee is being alerted. Forward early media is not gated.			
SIP header values INVITE2: P-Early-Media: supported SDP offer 183: SDP answer yes/no			
Message flow **End device** **Test equipment** **Interworking POTS** Off hook Dial number → INVITE ← 407 Proxy Authentication Required → ACK → INVITE2 ← 183 Session Progress **ISDN interworking** SETUP → INVITE ← 407 Proxy Authentication Required → ACK → INVITE2 ← 183 Session Progress **Apply post test routine**			

<table>
<tr><td>TSS
Orig_Establishment_of_an_early_dialogue</td><td>TP_201_035</td><td>Reference
subclause 5.1.3.1 of [ETSI TS124 229]
section 6 and 8 of [IETF RFC 5009]</td><td>Selection expression
PICS5.2/16</td></tr>
<tr><td colspan="4">Test purpose
SUT local ring back tone is presented to the user. 183 received, P-Early-Media header value sendonly or sendrecv present.

Ensure that the IUT upon receiving a 183 Session Progress provisional response a P-Early-Media header field is received value sendonly or sendrecv, the SUT plays ringing tones for the caller, indicating that the callee is being alerted if no media is received from the network. Forward early media is not gated.</td></tr>
<tr><td colspan="4">SIP header values
INVITE2:
P-Early-Media: supported
SDP offer

183:
P-Early-Media: [sendonly]/[sendrecv]
SDP answer yes/no</td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><th>End device</th><th></th><th>Test equipment</th></tr>
<tr><td>Interworking POTS</td><td></td><td></td></tr>
<tr><td>Off hook</td><td></td><td></td></tr>
<tr><td>Dial number</td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td>←</td><td>407 Proxy Authentication Required</td></tr>
<tr><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td>→</td><td>INVITE2</td></tr>
<tr><td></td><td>←</td><td>183 Session Progress</td></tr>
<tr><td>Local Ring back Tone</td><td>←</td><td></td></tr>
<tr><td></td><td></td><td></td></tr>
<tr><td>ISDN interworking</td><td></td><td></td></tr>
<tr><td>SETUP</td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td>←</td><td>407 Proxy Authentication Required</td></tr>
<tr><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td>→</td><td>INVITE2</td></tr>
<tr><td></td><td>←</td><td>183 Session Progress</td></tr>
<tr><td>Local Ring back Tone</td><td>←</td><td></td></tr>
</table>
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Orig_Establishment_of_an_early_dialogue</td><td>TP_201_036</td><td>Reference
subclause 5.1.3.1 of [ETSI TS124 229]
section 6 and 8 of [IETF RFC 5009]</td><td>Selection expression
PICS5.2/16</td></tr>
<tr><td colspan="4">Test purpose
SUT local ring back tone is presented to the user. 183 received, P-Early-Media header value sendonly or sendrecv present.

Ensure that the IUT upon receiving a 183 Session Progress provisional response a P-Early-Media header field is received value recvonly or inactive, the SUT plays ringing tones for the caller, indicating that the callee is being alerted. Forward early media is not gated.</td></tr>
</table>

SIP header values INVITE2: P-Early-Media: supported SDP offer 183: P-Early-Media: [recvonly]/[inactive] SDP answer yes/no
Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
	←	183 Session Progress
Local Ring back Tone	←	
ISDN interworking		
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
	←	183 Session Progress
Local Ring back Tone	←	

Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_037	Reference subclause 6.1.2 of [ETSI TS124 229]	Selection expression
Test purpose *Handling of SDP offer* Ensure that the IUT when sending an INVITE request contains a SDP offer and at least one media description. This SDP offer shall reflect the calling user's terminal capabilities and user preferences for the session.			
SIP header values INVITE2: SDP offer			
Message flow			

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2

ISDN interworking
SETUP → INVITE
← 407 Proxy Authentication Required
→ ACK
→ INVITE2
Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_038	Reference subclause 6.1.2 of [ETSI TS124 229]	Selection expression
Test purpose *Handling of SDP answer* Ensure that the IUT Upon generating the SDP offer for an INVITE request generated after receiving a 488 (Not Acceptable Here) response, the SDP offer contains a subset of the allowed media types, codecs and other parameters from the SDP message bodies of all 488 (Not Acceptable Here) responses so far received for the same session establishment attempt (i.e. a set of INVITE requests used for the same session establishment). For each media line, the SUT shall order the codecs in the SDP offer according to the order of the codecs in the SDP message bodies of the 488 (Not Acceptable Here) responses.			
SIP header values INVITE2: SDP offer1 488 SDP answer1 INVITE3: SDP offer2 equal SDP answer1			
Message flow **End device** **Test equipment** **Interworking POTS** Off hook Dial number → INVITE ← 407 Proxy Authentication Required → ACK → INVITE2 ← 488 (Not Acceptable Here → ACK → INVITE3 **ISDN interworking** SETUP → INVITE ← 407 Proxy Authentication Required → ACK → INVITE2 ← 488 Not Acceptable Here → ACK → INVITE3 **Apply post test routine**			

TSS Orig_Establishment_of_an_early_dialogue	TP_201_039	Reference	Selection expression

Test purpose

SUT terminates the forked not confirmed dialogue with a BYE request.

Ensure that the IUT when a final answer is received for one of the (forked) early dialogues, the SUT proceeds to set up the SIP session. An ACK is sent to acknowledge the confirmed dialogue and a BYE is sent to terminate the remaining early dialogues.

SIP header values

INVITE2:

P-Early-Media: supported

SDP offer

180 1:

P-Early-Media: [recvonly]/[inactive]

SDP answer yes/no

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
	←	180 Ringing1
	←	180 Ringing2
	←	200 OK (INVITE)2
	→	ACK
	→	BYE1
	←	200 OK (BYE)
ISDN interworking		
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
	←	180 Ringing1
	←	180 Ringing2
	←	200 OK (INVITE)2
	→	ACK
	→	BYE1
	←	200 OK (BYE)

Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_040	Reference section 8.1.1 of [IETF RFC 3261]	Selection expression

Test purpose

Retry of INVITE request after 503 with Retry-After header received

Ensure that the SUT Upon receiving a 503 (Service Unavailable) response to an initial INVITE request containing a Retry-After header does not automatically reattempt the request until after the period indicated by the Retry-After header contents.

SIP header values

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
	←	503 Service Unavailable
	→	ACK
Wait for the time indicated in the Retry-After header		
	→	INVITE
ISDN interworking		
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
	←	503 Service Unavailable
	→	ACK
Wait for the time indicated in the Retry-After header		
	→	INVITE

Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_041	Reference subclause 5.3.1.5.1 of [ETSI TS183 043]	Selection expression PICS 5.2/19

Test purpose

Sending of a P-Preferred-Identity or From header field

Ensure that the SUT is able to include the default identity associated with the analogue termination where the off-hook event was detected a P-Preferred-Identity header field or the From header.

SIP header values

INVITE:

P-Preferred-Identity: {default user identity}

From: {default user identity}

Message flow		
End device		**Test equipment**
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
Apply post test routine		

TSS Orig_Establishment_of_an_early_dialogue	TP_201_042	Reference subclause 6.3.2.4 of [ETSI TS 183 043]	Selection expression
Test purpose *Modifying SDP in early dialogue* Ensure that the SUT is able to receive an UPDATE request to modify the SDP in early dialogue. A 200 OK UPDATE with SDP answer is sent.			
SIP header values 183/180 SDP answer 1 UPDATE SDP offer 2 200 OK (UPDATE) SDP answer 2			

Message flow		
End device		**Test equipment**
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
	←	183/180
	←	UPDATE
	→	200 OK UPDATE
Apply post test routine		

TSS Orig_Establishment_of_an_early_dialogue	TP_201_043	Reference subclause 5.3.1.4 of [ETSI TS183 043]	Selection expression PICS 5.3/2
Test purpose *Charging procedure with AOC-S* Ensure that the SUT is able to receive charging information for a communication in the call setup phase. After sending the initial INVITE request, a 18x reliable provisional response is received containing a aoc-extended/ aoc-s XML element, a PRACK is sent.			

SIP header values

18x:

Require: 100rel

Content-Type: application/vnd.etsi.aoc+xml

```
<aoc-extended>
    <aoc-s>
        <charged-items>
            <basic>
                <price-time>
                    <currency-amount>1</currency-amount>
                    <charging-type>normal-charging</charging-type>
                    <granularity>
                        <time-unit>1</time-unit>
                        <scale>ten-seconds</scale>
                    </granularity>
                </price-time>
            </basic>
        </charged-items>
    </aoc-s>
</aoc-extended>
```

Message flow

End device		**Test equipment**
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
	←	18x
	→	PRACK
	←	200 OK (PRACK)

Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_044	Reference subclause 5.3.1.4 of [ETSI TS183 043]	Selection expression PICS 5.3/2

Test purpose

Charging procedure with AOC-D

Ensure that the SUT is able to receive charging information for a communication in the active call phase. After the communication is in the confirmed state, a NOTIFY is received containing a aoc-extended/ aoc-d XML element, a 200 OK (INFO) is sent.

SIP header values INFO: Content-Type: application/vnd.etsi.aoc+xml <aoc-extended>   <aoc-d>   <charging-info>subtotal</charging-info>   < recorded-charges>   <recorded-currency-units>   <currency-id>UNIT</currency-id>   <currency-amount>2</currency-amount>   </recorded-currency-units>   </ recorded-charges>   <billing-id> normal-charging </billing-id>   </ aoc-d > </aoc-extended>

Message flow

End device		Test equipment
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
Ringing tone	←	180 Ringing
Conversation	←	200 OK (INVITE)
	→	ACK
Update of charging info	←	INFO
	→	200 OK (INFO)

Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_045	Reference subclause 5.3.1.4 of [ETSI TS183 043]	Selection expression PICS 5.3/2

Test purpose

Charging procedure with AOC-E

Ensure that the SUT is able to receive charging information after a communication is terminated. When a communication is terminated, a BYE or a 200 OK (BYE) is received containing a aoc-extended XML element.

SIP header values

BYE/200 OK:

Content-Type: application/vnd.etsi.aoc+xml

<aoc-extended>
< aoc-e >
< recorded-charges>
<recorded-currency-units>
<currency-id>UNIT</currency-id>
<currency-amount>2</currency-amount>
</recorded-currency-units>
</ recorded-charges>
<billing-id> normal-charging </billing-id>
</ aoc-e >
</aoc-extended>

Message flow

End device		**Test equipment**
	A communication is already established	
CASE A		
Update of charging info in 200 OK (BYE)	→	BYE
	←	200 OK (BYE)
CASE B		
Update of charging info in BYE	←	BYE
	→	200 OK (BYE)

7.2.2.1.2 Test purposes for POTS

TSS Orig_Establishment_of_an_early_dialogue	TP_201_101	Reference subclause 4.9.2.1 of [ETSI TS 124 229]	Selection expression PICS 5.1.1/2 and 5.3/4

Test purpose

Overlap dialling using the multiple-INVITE method

Ensure that the SUT sends an additional INVITE request if an 404 or 484 final response is received on the previous sent INVITE request and the user dials further digits. The INVITE request contains the previous sent digits and the current dialled digit. The From header and the Call-ID are the same as used in the initial INVITE request.

SIP header values INVITE1 Request Line Dialled number1 From: [value initial invite] Call-ID: [value initial invite] INVITE2 Request Line Dialled number1 + dialled digit1 From: [value initial invite] Call-ID: [value initial invite] INVITE3 Request Line Dialled number1+ dialled digit1 + dialled digit2 From: [value initial invite] Call-ID: [value initial invite]

Message flow

End device		Test equipment
Off hook		
Dial number1		
	→	INVITE1
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE1
	←	404/484
	→	ACK
Dial digit1		
	→	INVITE2
	←	404/484
	→	ACK
Dial digit2		
	→	INVITE3
Ringing tone	←	180

Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_102	Reference subclause 4.9.2.1 of [ETSI TS 124 229]	Selection expression PICS 5.1.1/2 and 5.3/5

Test purpose

Overlap dialling using the in-dialogue method

Ensure that the SUT sends an additional INVITE request if an 404 or 484 final response is received on the previous sent INVITE request and the user dials further digits. The INVITE request contains the previous sent digits and the current dialled digit. The From header and the Call-ID are the same as used in the initial INVITE request.

Ensure that when a 183 without an SDP is received additional dialled digits are transferred in an INFO request.

SIP header values INVITE1 Request Line Dialled number1 From: [value initial invite] Call-ID: [value initial invite] INVITE2 Request Line Dialled number1 + dialled digit1 From: [value initial invite] Call-ID: [value initial invite] INFO Request Line Dialled number1+ dialled digit1 From: [value initial invite] Call-ID: [value initial invite] Content-Type: application/session-info Content-Disposition : signal ; handling=optional SubsequentDigit : [Dial digit2]

Message flow

End device		Test equipment
Off hook		
Dial number1		
	→	INVITE1
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE1
	←	404/484
	→	ACK
Dial digit1		
	→	INVITE2
	←	183 Session Progress
Dial digit2		
	→	INFO
	←	200 OK INFO
Ringing tone	←	180

Apply post test routine

7.2.2.1.3 Test purposes for ISDN

TSS Orig_Establishment_of_an _early_dialogue	TP_201_201	Reference subclause 5.3.1.5.3 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/2 and 5.4/1
Test purpose *Indication of support of the PSTN XML schema* Ensure that the SUT supports the PSTN XML schema, the initial INVITE request contains the application/vnd.etsi.pstn+xml tag in the Accept header.			
SIP header values INVITE: Accept: application/vnd.etsi.pstn+xml			
Message flow			

End device		Test equipment
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying

Apply post test routine

TSS Orig_Establishment_of_an _early_dialogue	TP_201_202	Reference subclause D.1.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2
Test purpose SETUP without called party number IE received Ensure that on receipt of a SETUP message and the Information Transfer Capability value for the BC is set to 'speech' or 'audio 3 kBit/s' and no called party number Information Element is present, no INVITE request is sent. The AGCF/VGW sends a SETUP ACKNOWLEDGE message and a Progress Indicator IE is present the Progress description is set to 8.			
SIP header values			
Message flow			

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying

Apply post test routine

TSS			
Orig_Establishment_of_an_early_dialogue	TP_201_203	Reference subclause D.1.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.3/4

Test purpose

Overlap sending using the multiple INVITE method

Ensure that an initial INVITE is sent after a called party number was received in a SETUP followed by several digits in INFO messages. All digits collected from the SETUP and INFO messages are present in the Request line of the INVITE request. The sending of the INVITE request depends on the configuration of the digit map.

Ensure that after sending of the initial INVITE the SUT is able to process a received **404 Not Found**. The final response is not relayed to the DSS1 user equipment. An additional received INFO is sent in a new INVITE request. The Request URI of all additionally sent INVITE requests contain all collected digits received in the SETUP and INFO messages.

SIP header values

INVITE1:

Request URI (digits INFO 1 to 3)

INVITE2:

Request URI (digits INFO 1 to 4)

INVITE3:

Request URI (digits INFO 1 to 5)

DSS1 parameter values

SETUP: Called party number

INFO: Called party number

Message flow

End device		Test equipment
SETUP		
SETUP ACKNOWLEDGE		
INFO1		
INFO2		
INFO3	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE1
	←	404 Not Found
	→	ACK
INFO4	→	INVITE2
	←	404 Not Found
	→	ACK
INFO5	→	INVITE3
ALERTING	←	180 Ringing

Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_204	Reference subclause D.1.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.3/4

Test purpose

Overlap sending using the multiple INVITE method

Ensure that an initial INVITE is sent after a called party number was received in a SETUP followed by several digits in INFO messages. All digits collected from the SETUP and INFO messages are present in the Request line of the INVITE request. The sending of the INVITE request depends on the configuration of the digit map.

Ensure that after sending of the initial INVITE the SUT is able to process a received **484 Address Incomplete**. The final response is not relayed to the DSS1 user equipment. An additional received INFO is sent in a new INVITE request. The Request URI of all additionally sent INVITE requests contain all collected digits received in the SETUP and INFO messages.

SIP header values

INVITE1:

Request URI (digits INFO 1 to 3)

INVITE2:

Request URI (digits INFO 1 to 4)

INVITE3:

Request URI (digits INFO 1 to 5)

DSS1 parameter values

SETUP: Called party number

INFO: Called party number

Message flow

End device		Test equipment
SETUP		
SETUP ACKNOWLEDGE		
INFO1		
INFO2		
INFO3	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE1
	←	484 Address Incomplete
	→	ACK
INFO4	→	INVITE2
	←	484 Address Incomplete
	→	ACK
INFO5	→	INVITE3
ALERTING	←	180 Ringing

Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_205	Reference subclause D.1.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.3/4

Test purpose

Sending of initial INVITE based on timeout Tinfo

Ensure that an initial INVITE is sent after a called party number was received in a SETUP followed by several digits in INFO messages. All digits collected from the SETUP and INFO messages are present in the Request line of the INVITE request. The INVITE is sent after the timeout of timer Tinfo

Ensure that after sending of the initial INVITE the SUT is able to process a received 404 Not Found or 484 Address Incomplete. The final response is not relayed to the DSS1 user equipment. An additional received INFO is sent in a new INVITE request. The Request URI of all additionally sent INVITE requests contain all collected digits received in the SETUP and INFO messages.

SIP header values

INVITE1:
Request URI (digits INFO 1 to 3)

INVITE2:
Request URI (digits INFO 1 to 4)

INVITE3:
Request URI (digits INFO 1 to 5)

DSS1 parameter values

SETUP: Called party number
INFO: Called party number

Message flow

End device		Test equipment
SETUP		
SETUP ACKNOWLEDGE		
	Start Tinfo	
INFO1	Start Tinfo	
INFO2	Start Tinfo	
INFO3	Start Tinfo	
	Timeout Tinfo	
	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE1
	←	404/484
	→	ACK
INFO4	→	INVITE2
	←	404/484
	→	ACK
INFO5	→	INVITE3
ALERTING	←	180 Ringing
	Apply post test routine	

<table>
<tr><td>TSS
Orig_Establishment_of_an_early_dialogue</td><td>TP_201_206</td><td>Reference
subclause 5.1.1.1 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 and 5.4/1</td></tr>
<tr><td colspan="4">Test purpose
INVITE is sent after receipt of "#" character in the called party number, XML sendingCompleteIndication is contained

Ensure that after initiating the normal incoming call establishment procedures, determining the end of address signalling and selecting to route the call to the IMS domain, the originating SUT sends the initial INVITE.
The end of address is detemined by receipt of a "#" character Sending complete Information Element as a sending complete indication.
– The INVITE contains the XML sendingCompleteIndication element</td></tr>
<tr><td colspan="4">SIP header values
INVITE:
<?xml version="1.0" encoding="utf-8"?>
<sendingCompleteIndication /></td></tr>
<tr><td colspan="4">DSS1 parameter values
SETUP: Called Party number contained the "#" character
or
Sending complete</td></tr>
<tr><td colspan="4">Message flow
<table><tr><th>End device</th><th></th><th>Test equipment</th></tr><tr><td>SETUP</td><td>→</td><td>INVITE</td></tr><tr><td>SETUP ACKNOWLEDGE</td><td>←</td><td>407 Proxy Authentication Required</td></tr><tr><td></td><td>→</td><td>ACK</td></tr><tr><td></td><td>→</td><td>INVITE</td></tr></table>Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Orig_Establishment_of_an_early_dialogue</td><td>TP_201_207</td><td>Reference
subclause 5.1.1.1 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2</td></tr>
<tr><td colspan="4">Test purpose
En-bloc sending is used, the complete called party number is contained in the called party number

Ensure that if the SETUP message contains the complete called number information,the called party number information is included in the Called party number information element possibly completed by the Called party subaddress information element.
The SUT sends a CALL PROCEEDING message. This acknowledges the SETUP message and indicates that the call is being processed and that no further address information is expected.</td></tr>
<tr><td colspan="4">SIP header values
INVITE: Request URI Called party number digits
To: Called party number digits; isub= Called paery subaddress; isub-encoding=nsap-ia5</td></tr>
<tr><td colspan="4">DSS1 Parameter values
SETUP: Called party number: Complete number information
Called party subaddress</td></tr>
<tr><td colspan="4">Message flow
<table><tr><th>End device</th><th></th><th>Test equipment</th></tr><tr><td>SETUP</td><td>→</td><td>INVITE</td></tr><tr><td>SETUP ACKNOWLEDGE</td><td>←</td><td>407 Proxy Authentication Required</td></tr><tr><td></td><td>→</td><td>ACK</td></tr><tr><td></td><td>→</td><td>INVITE</td></tr></table>Apply post test routine</td></tr>
</table>

TSS Orig_Establishment_of_an_early_dialogue	TP_201_208	Reference subclause 5.1.1.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

One bearer capability information element is received. Mapped into the XML BearerCapability element

Ensure that the whole bearer capability information element, as received in the SETUP message, shall be mapped to the PSTN XML bearer capability body in SIP. A bearer capability information element is received. The Information transfer capability value is indicated in Table 7.2.2.1.4-1.

SIP header values

INVITE:

<?xml version="1.0" encoding="utf-8"?>

PSTN

    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>ITC_value<
<   BCoctet4
            TransferMode>00<
            InformationTransferRate>10000<
        BCoctet5
            Layer1Identification>01<
            UserInfoLayer1Protocol>00011<

DSS1 parameter values

SETUP:

bearer capability Information transfer capability = ITC_value

Message flow

End device		**Test equipment**
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE

Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_209	Reference subclause 5.1.1.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

Two bearer capability information element are received. Mapped into two XML BearerCapability element

Ensure that if two bearer capability information elements are received and the information transfer capability of the first BC is set to speech or 3,1 kHz audio and the information transfer capability of the second BC is set to unrestricted digital information with tones and announcements, two XML BearerCapability elements are contained in the PSTN XML attachment.

- The InformationTransferCabability of the first is set to '00000' or '10000'.
- The InformationTransferCabability of the second is set to '10001'.

SIP header values

INVITE:

<?xml version="1.0" encoding="utf-8"?>

PSTN

BearerCapability
BCoctet3
CodingStandard>00<
InformationTransferCabability>**00000**<
or
InformationTransferCabability>**10000**<
BCoctet4
TransferMode>00<
InformationTransferRate>10000<

BCoctet5

Layer1Identification>01<
UserInfoLayer1Protocol>00011<

BearerCapability
BCoctet3
CodingStandard>00<
InformationTransferCabability>**10001**<
BCoctet4
TransferMode>00<
InformationTransferRate>10000<

BCoctet5

Layer1Identification>01<
UserInfoLayer1Protocol>00011<

SDP: m line contains as the first codec CLEARMODE and as the second codec a G.711 codec

DSS1 parameter values

SETUP: First Bearer Capability Information transfer capability = Speech or 3.1 kHz audio
Second Bearer Capability Information transfer capability = Unrestricted digital information with tones/announcements

Message flow

End device		**Test equipment**
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE

Apply post test routine

Table 7.2.2.1.4-1 – Mapping of bearer capability to PSTN XML BearerCapability

ITC_value	BC Information transfer capability	XML InformationTransferCabability
ITC_VA_1	Speech	'00000'
ITC_VA_2	3,1 kHz audio	'10000'
ITC_VA_3	unrestricted digital information	'01000'

TSS Orig_Establishment_of_an_early_dialogue	TP_201_210	Reference subclause 5.1.1.1.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

Mapping of Progress Indicator

Ensure that the Progress Indicator received in the SETUP is mapped into the PSTN XML ProgressIndicator value as indicated in Table 7.2.2.1.4-2. An additionally ProgressIndicator element is is contained and the ProgressDescription value is set to '0000110'

SIP header values

INVITE:

<?xml version="1.0" encoding="utf-8"?>

PSTN

    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>yyyy<
        ProgressOctet4
            ProgressDescription>0000110<
    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>0000<
        ProgressOctet4
        ProgressDescription>PI_value<

DSS1 parameter values

SETUP:

Progress Indicator Coding standard ='00', Location ='0000', Progress description='0000yyyy'

Message flow

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE

Apply post test routine

Table 7.2.2.1.4-2 – Mapping of progress indicator information element to PSTN XML ProgressIndicator

PI_value	DSS1 progress indicator value	XML ProgressIndicator ProgressDescription
PI_VA_1	Call is not end-to-end 5.1.1/2; further call progress information may be available in-band	'0000001'
PI_VA_2	Destination address is non-5.1.1/2	'0000010'
PI_VA_3	Origination address is non-5.1.1/2	'0000011'
PI_VA_4	Call has returned to the 5.1.1/2	'0000100'
PI_VA_5	Interworking has occurred and has resulted in a telecommunication service change	'0000101'
PI_VA_6	In-band information or an appropriate pattern is now available	'0001000'

<table>
<tr><td>TSS
Orig_Establishment_of_an_early_dialogue</td><td>TP_201_211</td><td>Reference
subclause 5.1.1.1.3 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 and 5.4/1</td></tr>
<tr><td colspan="4">Test purpose
No Progress Indicator received

Ensure that if no progress indicator was received in the a ProgressIndicator element is is contained and the ProgressDescription value is set to '0000110'</td></tr>
<tr><td colspan="4">SIP header values
INVITE:
<?xml version="1.0" encoding="utf-8"?>
PSTN
ProgressIndicator
ProgressOctet3
CodingStandard>00<
Location>yyyy<
ProgressOctet4
ProgressDescription>0000110<
ProgressIndicator
ProgressOctet3
CodingStandard>00<
Location>0000<
ProgressOctet4
ProgressDescription>PI_value<</td></tr>
<tr><td colspan="4">DSS1 parameter values
SETUP:</td></tr>
<tr><td colspan="4">Message flow</td></tr>
<tr><td colspan="2">End device</td><td></td><td>Test equipment</td></tr>
<tr><td colspan="2">SETUP</td><td>→</td><td>INVITE</td></tr>
<tr><td colspan="2">SETUP ACKNOWLEDGE</td><td>←</td><td>407 Proxy Authentication Required</td></tr>
<tr><td colspan="2"></td><td>→</td><td>ACK</td></tr>
<tr><td colspan="2"></td><td>→</td><td>INVITE</td></tr>
<tr><td colspan="4">Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Orig_Establishment_of_an_early_dialogue</td><td>TP_201_212</td><td>Reference
clause 5.1.1.1.3 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 and 5.4/1</td></tr>
<tr><td colspan="4">Test purpose
Mapping of high layer compatibility information element

Ensure that the high layer compatibility information element received in the SETUP is mapped into the PSTN XML HighLayerCompatibility</td></tr>
<tr><td colspan="4">SIP header values
INVITE:
<?xml version="1.0" encoding="utf-8"?>
PSTN
HighLayerCompatibility
HLOctet3
CodingStandard>00<
Interpretation>100<
PresentationMethod>01<
HLOctet4
HighLayerCharacteristics>HLC_value<</td></tr>
</table>

DSS1 parameter values

SETUP:

High layer compatibility Coding standard='00', Interpretation='100', High layer characteristics identification=HLC_value as indicated in Table 7.2.2.1.4-3

Message flow

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE

Apply post test routine

TSS	TP_201_213	Reference	Selection expression
Orig_Establishment_of_an _early_dialogue		subclause 5.3.1.5.3 of [ETSI TS 183 043]	PICS 5.1.1/2 and 5.4/1

Test purpose

Two high layer compatibility information elements received

Ensure that the first high layer compatibility information element received in the SETUP is mapped into the first PSTN XML HighLayerCompatibility and the second High layer compatibility information element received in the SETUP is mapped into the second PSTN XML HighLayerCompatibility

SIP header values

INVITE:

<?xml version="1.0" encoding="utf-8"?>

PSTN

    HighLayerCompatibility
        HLOctet3
            CodingStandard>00<
            Interpretation>100<
            PresentationMethod>01<
        HLOctet4
            HighLayerCharacteristics>0000001<
    HighLayerCompatibility
        HLOctet3
            CodingStandard>00<
            Interpretation>100<
            PresentationMethod>01<
        HLOctet4
      HighLayerCharacteristics>HLC_value<

DSS1 Parameter values

SETUP:

First High layer compatibility Coding standard='00', Interpretation='100', High layer characteristics identification= Telephony

Second High layer compatibility Coding standard='00', Interpretation='100', High layer characteristics identification=HLC_value as indicated in Table 7.2.2.1.4-3

Message flow

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE

Apply post test routine

Table 7.2.2.1.4-3 – Mapping of high layer compatibility information element to PSTN XML HighLayerCharacteristic

HLC_value	DSS1 High layer characteristics identification	XML HighLayerCharacteristic
HLC_VA_1	Telephony	'0000001'
HLC_VA_2	Facsimile Group 2/3	'0000100'
HLC_VA_3	Facsimile Group 4 Class I	'0100001'
HLC_VA_4	Facsimile service Group 4, Classes II ad III	'0100100'
HLC_VA_5	Syntax based Videotex	'0110010'
HLC_VA_6	International Videotex interworking via gateways or interworking units	'0110011'
HLC_VA_7	Telex service	'0110101'
HLC_VA_8	FTAM application	'1000010'
HLC_VA_9	Videotelephony	'1100000'

TSS Orig_Establishment_of_an_early_dialogue	TP_201_214	Reference 5.1.1.2.1, 5.1.1.2.1.1/ [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1
Test purpose *180 received, No PSTN XML body present, PI 1 is sent* Ensure that on receipt of a 180 (Ringing) and a PSTN XML body is not present, an ALERTING message is sent to the calling user equipment that contains a Progress Indicator Information Element and the Progress description is set to value 1.			
SIP header values			
DSS1 parameter values ALERTING: Progress Indicator value 1			
Message flow			

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 Ringing

Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_215	Reference subclauses 5.1.1.2.1 and 5.1.1.2.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1
Test purpose *180 received, PSTN XML body present two XML PI set to value 7 and x an ALERTING is sent PI value x* Ensure that on receipt of a 180 (Ringing) a PSTN XML body is present contains two ProgressIndicator elements set to x and 7, an ALERTING is sent to the calling user equipment and a Progress Indicator Information Element is present, the Progress description is set to the value PI_VA as indicated in Table 7.2.2.1.4-4			

SIP header values

180 (Ringing): XML body

<?xml version="1.0" encoding="utf-8"?>

PSTN
    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>yyyy<
        ProgressOctet4
            ProgressDescription>PI_VA<
    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>yyyy<
        ProgressOctet4
            ProgressDescription>0000111<

DSS1 Parameter values

ALERTING: Progress Indicator value x

Message flow

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 Ringing

Apply post test routine

Table 7.2.2.1.4-4 – Mapping of progress indicator information element to PSTN XML ProgressIndicator

PI_value	DSS1 progress indicator value	XML ProgressIndicator ProgressDescription
PI_VA_1	Call is not end-to-end 5.1.1/2; further call progress information may be available in-band	'0000001'
PI_VA_2	Destination address is non-5.1.1/2	'0000010'
PI_VA_3	Origination address is non-5.1.1/2	'0000011'
PI_VA_4	Call has returned to the 5.1.1/2	'0000100'
PI_VA_5	Interworking has occurred and has resulted in a telecommunication service change	'0000101'
PI_VA_6	In-band information or an appropriate pattern is now available	'0001000'

TSS Orig_Establishment_of_an_early_dialogue	TP_201_216	Reference subclauses 5.1.1.2.1 and 5.1.1.2.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

180 received, PSTN XML body present XML PI set to value 7

Ensure that on receipt of the first 180 (Ringing) and a PSTN XML body present the XML PI is set to value 7, an ALERTING message is sent to the calling user equipment does not contain a Progress Indicator Information Element.

SIP header values

180 (Ringing): XML body

<?xml version="1.0" encoding="utf-8"?>

PSTN

  ProgressIndicator

    ProgressOctet3

      CodingStandard>00<

      Location>yyyy<

    ProgressOctet4

      ProgressDescription>0000111<

DSS1 Parameter values

ALERTING: No Progress Indicator present

Message flow

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 Ringing

Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_217	Reference subclauses 5.1.1.2.1 and 5.1.1.2.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

Subsequent 180 received, No PSTN XML body present, PI 1 is sent.

Ensure that on receipt of a subsequent 180 (Ringing) and a PSTN XML body is not present, a PROGRESS message is sent to the calling user equipment that contains a Progress Indicator Information Element and the Progress description is set to value 1 if a PROGRESS message was sent before that contains Progress Indicator value 4.

SIP header values

180 (Ringing) 1: no XML body present
180 (Ringing) 2: XML body
<?xml version="1.0" encoding="utf-8"?>
PSTN
    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>yyyy<
        ProgressOctet4
            ProgressDescription>0000111<
180 (Ringing) 3 no XML body present

DSS1 Parameter values

PROGRESS 1: Progress Indicator value 4
PROGRESS 2: Progress Indicator value 1

Message flow

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 Ringing1
PROGRESS 1	←	180 Ringing2
PROGRESS 2	←	180 Ringing3

Apply post test routine

TSS	TP_201_218	Reference	Selection expression
Orig_Establishment_of_an_early_dialogue		subclause 5.1.1.2.1 of [ETSI TS 183 036]	PICS 5.1.1/2 and 5.4/1

Test purpose

Subsequent 180 received and the P-Early-Media header set to sendonly, a PROGRESS message is sent.

Ensure that on receipt of a subsequent 180 (Ringing) that contains a P-Early-Media header set to sendonly, the SUT the Progress Indicator Information Element is present in the sent PROGRESS message and the Progress description is set to value 8.

SIP header values

180 (Ringing) 2: P-Early-Media: sendonly

DSS1 Parameter values

PROGRESS: Progress Indicator value 8

Message flow

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 Ringing1
PROGRESS	←	180 Ringing2

Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_219	Reference subclauses 5.1.1.2.1 and 5.1.1.2.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

Subsequent 180 received, PSTN XML body present two XML PI set to value 7 and x a PROGRESS is sent PI value x included

Ensure that on receipt of a subsequent 180 (Ringing) a PSTN XML body is present, which contains two Progress Indicator elements set to x and 7, a PROGRESS message is sent to the calling user equipment. A Progress Indicator Information Element is present, the Progress description is set to the value PI_VA as indicated in Table 7.2.2.1.4-5 and a second Progress Indicator Information Element is present, the Progress description is set to the value 4 if a 180 (Ringing) was received before and no PSTN XML body was present.

SIP header values

180 (Ringing) 1: no XML body present

180 (Ringing) 2: XML body

<?xml version="1.0" encoding="utf-8"?>

PSTN
    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location<yyyy>
        ProgressOctet4
            ProgressDescription>PI_value<
    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>yyyy<
        ProgressOctet4
            ProgressDescription>0000111<

DSS1 Parameter values

PROGRESS: Progress Indicator value x and a Progress Indicator value 4 Call has returned to the 5.1.1/2

Message flow

End device		**Test equipment**
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 Ringing1
PROGRESS	←	180 Ringing2

Apply post test routine

Table 7.2.2.1.4-5 – Mapping of progress indicator information element to PSTN XML ProgressIndicator

PI_value	DSS1 Progress Indicator value	XML ProgressIndicator ProgressDescription
PI_VA_1	Call is not end-to-end 5.1.1/2; further call progress information may be available in-band	'0000001'
PI_VA_2	Destination address is non-5.1.1/2	'0000010'
PI_VA_3	Origination address is non-5.1.1/2	'0000011'
PI_VA_4	Call has returned to the 5.1.1/2	'0000100'
PI_VA_5	Interworking has occurred and has resulted in a telecommunication service change	'0000101'
PI_VA_6	In-band information or an appropriate pattern is now available	'0001000'

<table>
<tr><td>TSS
Orig_Establishment_of_an_early_dialogue</td><td>TP_201_220</td><td>Reference
subclauses 5.1.1.2.1 and 5.1.1.2.1.1 of [ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.4/1</td></tr>
<tr><td colspan="4">Test purpose
180 received, PSTN XML body present XML PI set to value 7

Ensure that on receipt of a subsequent 180 (Ringing) and a PSTN XML body present the XML PI is set to value 7 and PROGRESS is sent toward the calling user equipment and a Progress Indicator Information Element is present the Progress description is set to the value 4 if a 180 (Ringing) was received without a PSTN XML body.</td></tr>
<tr><td colspan="4">SIP header values
180 (Ringing) 1: no XML body present
180 (Ringing) 2: XML body
<?xml version="1.0" encoding="utf-8"?>
PSTN
    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>yyyy<
        ProgressOctet4
            ProgressDescription>0000111<</td></tr>
<tr><td colspan="4">DSS1 Parameter values
ALERTING: No Progress Indicator present
PROGRESS: Progress Indicator value 4 Call has returned to the 5.1.1/2</td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><th>End device</th><th></th><th>Test equipment</th></tr>
<tr><td>SETUP</td><td>→</td><td>INVITE</td></tr>
<tr><td>SETUP ACKNOWLEDGE</td><td>←</td><td>407 Proxy Authentication Required</td></tr>
<tr><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>ALERTING</td><td>←</td><td>180 Ringing1</td></tr>
<tr><td>PROGRESS</td><td>←</td><td>180 Ringing2</td></tr>
</table>
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Orig_Establishment_of_an_early_dialogue</td><td>TP_201_221</td><td>Reference
subclause 5.1.1.2.1.2 of [ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 and 5.4/1</td></tr>
<tr><td colspan="4">Test purpose
Receiving of PSTN XML HighLayerCompatibility fallback information an ALERTING is sent

Ensure that on receipt of a 180 (Ringing) that contains a PSTN XML HighLayerCompatibility element that indicates fallback, the SUT sends an ALERTING message to the calling user equipment that does not contain a High Layer Compatibility Information Element.</td></tr>
<tr><td colspan="4">SIP header values
180 (Ringing): XML body
<?xml version="1.0" encoding="utf-8"?>
PSTN
    HighLayerCompatibility
        HLOctet3
            CodingStandard>00<
            Interpretation>100<
            PresentationMethod>01<
        HLOctet4
            HighLayerCharacteristics>HLC_value<</td></tr>
<tr><td colspan="4">DSS1 parameter values
SETUP: First Bearer Capability Information transfer capability = Speech or 3.1 kHz audio
Second Bearer Capability Information transfer capability = Unrestricted digital information with tones/announcements
ALERTING:</td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><th>End device</th><th></th><th>Test equipment</th></tr>
<tr><td>SETUP</td><td>→</td><td>INVITE</td></tr>
<tr><td>SETUP ACKNOWLEDGE</td><td>←</td><td>407 Proxy Authentication Required</td></tr>
<tr><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>ALERTING</td><td>←</td><td>180 Ringing</td></tr>
</table>
Apply post test routine</td></tr>
</table>

Table 7.2.2.1.4-5 – Mapping of high layer compatibility information element to PSTN XML HighLayerCharacteristic

HLC_value	DSS1 high layer characteristics identification	XML HighLayerCharacteristic
HLC_VA_1	Telephony	'0000001'
HLC_VA_2	Facsimile Group 2/3	'0000100'
HLC_VA_3	Facsimile Group 4 Class I	'0100001'
HLC_VA_4	Facsimile service Group 4, Classes II ad III	'0100100'
HLC_VA_5	Syntax based Videotex	'0110010'
HLC_VA_6	International Videotex interworking via gateways or interworking units	'0110011'
HLC_VA_7	Telex service	'0110101'
HLC_VA_8	FTAM application	'1000010'
HLC_VA_9	Videotelephony	'1100000'

TSS Orig_Establishment_of_an _early_dialogue	TP_201_222	Reference subclause 5.1.1.2.1.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

Receiving of PSTN XML BearerCapability fallback information an ALERTING is sent.

Ensure that on receipt of a 180 (Ringing) that contains a PSTN XML BearerCapability element that indicates fallback, the SUT sends an ALERTING message to the calling user equipment that does not contain a Bearer Capability Information Element.

SIP header values

180 (Ringing): XML body

<?xml version="1.0" encoding="utf-8"?>

PSTN
- BearerCapability
  - BCoctet3
    - CodingStandard>00<
    - InformationTransferCabability>00000<
  - BCoctet4
    - TransferMode>00<
    - InformationTransferRate>10000<
  - BCoctet5>
    - Layer1Identification>01<
    - UserInfoLayer1Protocol>00011<

DSS1 Parameter values

SETUP: First Bearer Capability Information transfer capability = Speech or 3.1 kHz audio
Second Bearer Capability Information transfer capability = Unrestricted digital information with tones/announcements

ALERTING:

Message flow

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 Ringing

Apply post test routine

TSS Orig_Establishment_of_an _early_dialogue	TP_201_223	Reference subclause 5.1.1.2.1.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

No fallback information received, CLEARMODE codec not the preferred codec in the 180

Ensure that on receipt of 180 (Ringing) and the CLEARMODE codes is not the first codec in the codec list, the SUT sends an ALERTING toward the calling user equipment and fallback is not indicated. No BC is sent in the ALERTING.

SIP header values
DSS1 Parameter values SETUP: First Bearer Capability Information transfer capability = Speech or 3.1 kHz audio Second Bearer Capability Information transfer capability = Unrestricted digital information with tones/announcements ALERTING:
Message flow

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 Ringing

Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_224	Reference subclause 5.1.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

Receipt of first 183 with P-Early-Media header CALL PROCEEDING is sent

Ensure that on receipt of the first 183 (Session Progress) with P-Early-Media header, the SUT does not send a local ringing tone to the calling user. The early media is passed through. A CALL PROCEEDING message is sent to the calling user contains a Progress Indicator Information Element and the Progress description is set to value 8.

SIP header values

183 (Session Progress): P-Early-Media: sendonly, SDP

DSS1 Parameter values

CALL PROCEEDING: Progress Indicator value 8

Message flow

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
CALL PROCEEDING	←	183 (Session Progress)

Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_225	Reference subclause 5.1.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.2/16

Test purpose

Receipt of 183 with P-Early-Media header PROGRESS is sent

Ensure that on receipt of a 183 (Session Progress) with P-Early-Media header, the SUT does not send a local ringing tone to the calling user. The early media is passed through. A PROGRESS message is sent to the calling user contains a Progress Indicator Information Element and the Progress description is set to value 8.

SIP header values

183 (Session Progress): P-Early-Media: sendonly, SDP

DSS1 Parameter values

PROGRESS: Progress Indicator value 8

Message flow

End device		**Test equipment**
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 (Ringing)
PROGRESS	←	183 (Session Progress)

Apply post test routine

TSS	TP_201_226	Reference	Selection expression
Orig_Establishment_of_an_early_dialogue		subclause 5.1.1.2.2.1 of [ETSI TS 183 036]	PICS 5.1.1/2 and 5.4/1

Test purpose

183 received, PSTN XML body present two XML PI set to value 7 and x CALL PROCEEDING is sent PI value x

Ensure that on receipt of a 183 (Session Progress), a PSTN XML body is present that contains two ProgressIndicator elements set to x and 7, a CALL PROCEEDING is sent to the calling user equipment and a Progress Indicator Information Element is present, the Progress description is set to the value PI_VA as indicated in Table 7.2.2.1.4-6

SIP header values

180 (Ringing): XML body

<?xml version="1.0" encoding="utf-8"?>

PSTN

    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>yyyy<
        ProgressOctet4
            ProgressDescription>PI_VA<
    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>yyyy<
        ProgressOctet4
            ProgressDescription>0000111<

DSS1 Parameter values

CALL PROCEEDING: Progress Indicator value x

Message flow

End device		**Test equipment**
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
CALL PROCEEDING	←	183 (Session Progress)

Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_227	Reference subclause 5.1.1.2.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1
Test purpose *183 received, PSTN XML body present XML PI set to value 7* Ensure that on receipt of a subsequent 183 (Session Progress) and a PSTN XML body is present, the XML PI is set to value 7 a CALL PROCEEDING is sent toward the calling user equipment, and a Progress Indicator Information Element is present. The Progress description is set to the value 4 if a 180 (Ringing) was received without a PSTN XML body.			
SIP header values 180 (Ringing): XML body <?xml version="1.0" encoding="utf-8"?> PSTN ProgressIndicator ProgressOctet3 CodingStandard>00< Location>yyyy< ProgressOctet4 ProgressDescription>0000111<			
DSS1 Parameter values CALL PROCEEDING: Progress Indicator not present			
Message flow			
End device		**Test equipment**	
SETUP	→	INVITE	
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required	
	→	ACK	
	→	INVITE	
CALL PROCEEDING	←	183 (Session Progress)	
Apply post test routine			

TSS Orig_Establishment_of_an_early_dialogue	TP_201_228	Reference subclause 5.1.1.2.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.2/16
Test purpose *183 received and the P-Early-Media header set to sendonly, a PROGRESS message is sent.* Ensure that on receipt of a 183 (Session Progress) that contains a P-Early-Media header set to sendonly, the SUT the Progress Indicator Information Element is present in the sent PROGRESS message and the Progress description is set to value 8.			
SIP header values 183 (Session Progress): P-Early-Media: sendonly			
DSS1 Parameter values ALERTING: No Progress Indicator present PROGRESS: Progress Indicator value 8			

Message flow		
End device		**Test equipment**
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 Ringing1
PROGRESS	←	183 Session Progress
	Apply post test routine	

TSS Orig_Establishment_of_an_early_dialogue	TP_201_229	Reference subclause 5.1.1.2.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1
Test purpose *183 received, PSTN XML body present two XML PI set to value 7 and x a PROGRESS is sent PI value x* Ensure that on receipt of a 183 (Session Progress) that contains a P-Early-Media header set to sendonly, in the SUT the Progress Indicator Information Element is present in the sent PROGRESS message and the Progress description is set to value 8.			
SIP header values 180 (Ringing): <?xml version="1.0" encoding="utf-8"?> PSTN ProgressIndicator ProgressOctet3 CodingStandard>00< Location>yyyy< ProgressOctet4 ProgressDescription>0000111< 183 (Session Progress): XML body <?xml version="1.0" encoding="utf-8"?> PSTN ProgressIndicator ProgressOctet3 CodingStandard>00< Location>yyyy< ProgressOctet4 ProgressDescription>PI_VA< ProgressIndicator ProgressOctet3 CodingStandard>00< Location>yyyy< ProgressOctet4 ProgressDescription>0000111<			
DSS1 Parameter values ALERTING: No Progress Indicator present PROGRESS: Progress Indicator value 8			
Comments The Location value is don't care. The order of the received ProgressIndicator elements is accidental			

Message flow		
End device		**Test equipment**
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 Ringing1
PROGRESS	←	183 Session Progress
	Apply post test routine	

TSS Orig_Establishment_of_an_early_dialogue	TP_201_230	Reference subclause 5.1.1.2.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

183 received, PSTN XML body present two XML PI set to value 7 and x a PROGRESS is sent PI value x

Ensure that on receipt of a 183 (Session Progress), a PSTN XML body is present, which contains two ProgressIndicator elements set to x and 7, a PROGRESS is sent to the calling user equipment and a Progress Indicator Information Element is present, the Progress description is set to the value PI_VA as indicated in Table 7.2.2.1.4-6. In the previous 180 (Ringing) was a PSTN XML body.

SIP header values

180 (Ringing):

```
<?xml version="1.0" encoding="utf-8"?>
PSTN
    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>yyyy<
        ProgressOctet4
            ProgressDescription>0000111<
```

183 (Session Progress): XML body

```
<?xml version="1.0" encoding="utf-8"?>
PSTN
    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>yyyy<
        ProgressOctet4
            ProgressDescription>PI_VA<
    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>yyyy<
        ProgressOctet4
            ProgressDescription>0000111<
```

DSS1 Parameter values

PROGRESS: Progress Indicator value x

Comments

The Location value is don't care. The order of the received ProgressIndicator elements is accidental

Message flow

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 Ringing
PROGRESS	←	183 Session Progress

Apply post test routine

TSS Orig_Establishment_of_an_early_dialogue	TP_201_231	Reference subclause 5.1.1.2.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1

Test purpose

183 received, PSTN XML body present two XML PI set to value 7 and x a PROGRESS is sent PI value x and value 4

Ensure that on receipt of a 183 (Session Progress) a PSTN XML body is present that contains two ProgressIndicator elements set to x and 7, a PROGRESS is sent to the calling user equipment and a Progress Indicator Information Element is present, the Progress description is set to the value PI_VA as indicated in Table 7.2.2.1.4-6 and in addition a second Progress Indicator Information Element is present, the Progress description is set to the value 4. In the previous 180 (Ringing) there was no PSTN XML body.

SIP header values

180 (Ringing):

183 (Session Progress): XML body

```
<?xml version="1.0" encoding="utf-8"?>
PSTN
   ProgressIndicator
      ProgressOctet3
         CodingStandard>00<
         Location>yyyy<
      ProgressOctet4
         ProgressDescription>PI_VA<
   ProgressIndicator
      ProgressOctet3
         CodingStandard>00<
         Location>yyyy<
      ProgressOctet4
         ProgressDescription>0000111<
```

DSS1 Parameter values

PROGRESS: Progress Indicator value x and 4

Comments

The Location value is don't care. The order of the received ProgressIndicator elements is accidental

Message flow

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 Ringing
PROGRESS	←	183 Session Progress

Apply post test routine

<table>
<tr><td>TSS</td><td>TP_201_232</td><td>Reference
subclause 5.1.1.2.2.1 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 and 5.4/1</td></tr>
<tr><td colspan="4">Test purpose
183 received, PSTN XML body present XML PI set to value 7

Ensure that on receipt of the 183 (Session Progress) and if a PSTN XML body is present and the XML PI is set to value 7, a PROGRESS message is sent to the calling user equipment. A Progress Indicator Information Element is present, the Progress description is set to the value 4.</td></tr>
<tr><td colspan="4">SIP header values
183 (Session Progress):
<?xml version="1.0" encoding="utf-8"?>
    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>yyyy<
        ProgressOctet4
            ProgressDescription>0000111<</td></tr>
<tr><td colspan="4">DSS1 Parameter values
PROGRESS: Progress Indicator value x and 4</td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><th>End device</th><th></th><th>Test equipment</th></tr>
<tr><td>SETUP</td><td>→</td><td>INVITE</td></tr>
<tr><td>SETUP ACKNOWLEDGE</td><td>←</td><td>407 Proxy Authentication Required</td></tr>
<tr><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>ALERTING</td><td>←</td><td>180 Ringing</td></tr>
<tr><td>PROGRESS</td><td>←</td><td>183 Session Progress</td></tr>
</table>
Apply post test routine</td></tr>
</table>

Table 7.2.2.1.4-6 – Mapping of progress indicator information element to PSTN XML ProgressIndicator

PI_value	DSS1 Progress Indicator value	XML ProgressIndicator ProgressDescription
PI_VA_1	Call is not end-to-end 5.1.1/2; further call progress information may be available in-band	'0000001'
PI_VA_2	Destination address is non-5.1.1/2	'0000010'
PI_VA_3	Origination address is non-5.1.1/2	'0000011'
PI_VA_4	Call has returned to the 5.1.1/2	'0000100'
PI_VA_5	Interworking has occurred and has resulted in a telecommunication service change	'0000101'
PI_VA_6	In-band information or an appropriate pattern is now available	'0001000'

TSS	TP_201_233	Reference clause 5.1.1.2.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

Receiving of PSTN XML BearerCapability fallback information in a 183 a CALL PROCEEDING is sent.

Ensure that on receipt of a 183 (Session Progress) that contains a PSTN XML BearerCapability element that indicates fallback, the SUT sends a CALL PROCEEDING message to the calling user equipment that does not contain a Bearer Capability Information Element.

SIP header values

183 (Session Progress):

<?xml version="1.0" encoding="utf-8"?>
BearerCapability
BCoctet3
CodingStandard>00<
InformationTransferCabability>00000<
BCoctet4
TransferMode>00<
InformationTransferRate>10000<
BCoctet5>
Layer1Identification>01<
UserInfoLayer1Protocol>00011<

DSS1 Parameter values

SETUP: First Bearer Capability Information transfer capability = Speech or 3.1 kHz audio
Second Bearer Capability Information transfer capability = Unrestricted digital information with tones/announcements

CALL PROCEEDING

Message flow

End device		**Test equipment**
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
CALL PROCEEDING	←	183 Session Progress

Apply post test routine

TSS	TP_201_234	Reference subclause 5.3.1.5.3 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

Receiving of PSTN XML BearerCapability fallback information in a 183 a PROGRESS is sent.

Ensure that on receipt of a 183 (Session Progress) that contains a PSTN XML BearerCapability element that indicates fallback, the SUT sends a PROGRESS message to the calling user equipment that does not contain a Bearer Capability Information Element.

SIP header values

183 (Session Progress):

<?xml version="1.0" encoding="utf-8"?>

BearerCapability

BCoctet3

CodingStandard>00<

InformationTransferCabability>00000<

BCoctet4

TransferMode>00<

InformationTransferRate>10000<

BCoctet5>

Layer1Identification>01<

UserInfoLayer1Protocol>00011<

DSS1 Parameter values

SETUP: First Bearer Capability Information transfer capability = Speech or 3.1 kHz audio

Second Bearer Capability Information transfer capability = Unrestricted digital information with tones/announcements

PROGRESS:

Message flow

End device		**Test equipment**
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
PROGRESS	←	183 Session Progress

Apply post test routine

TSS	TP_201_235	Reference subclause 5.1.1.2.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

Receiving of PSTN XML HighLayerCompatibility fallback information a CALL PROCEEDING is sent.

Ensure that on receipt of a 183 (Session Progress) that contains a PSTN XML HighLayerCompatibility element according to Table 7.2.2.1.4-5 and a ProgressIndicator value set to 5 that indicates fallback, the SUT sends a CALL PROCEEDING message to the calling user equipment that does not contain a High Layer Compatibility Information Element.

SIP header values

183 (Session Progress): PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

  HighLayerCompatibility

    HLOctet3

      CodingStandard>00<

      Interpretation>100<

      PresentationMethod>01<

    HLOctet4

      HighLayerCharacteristics>HLC_value<

  ProgressIndicator

    ProgressOctet3

      CodingStandard>00<

      Location>yyyy<

    ProgressOctet4

      ProgressDescription>0000101<

DSS1 Parameter values

SETUP: First Bearer Capability Information transfer capability = Speech or 3.1 kHz audio

Second Bearer Capability Information transfer capability = Unrestricted digital information with tones/announcements

CALL PROCEEDING:

Message flow

End device		**Test equipment**
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
CALL PROCEEDING	←	183 Session Progress

Apply post test routine

TSS	TP_201_236	Reference subclause 5.1.1.2.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

Receiving of PSTN XML HighLayerCompatibility fallback information a PROGRESS is sent.

Ensure that on receipt of a 183 (Session Progress) that contains a PSTN XML HighLayerCompatibility element according to Table 7.2.2.1.4-5 and a ProgressIndicator value set to 5 that indicates fallback, the SUT sends a PROGRESS message to the calling user equipment that does not contain a High Layer Compatibility Information Element.

SIP header values

183 (Session Progress): PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

HighLayerCompatibility

HLOctet3

CodingStandard>00<

Interpretation>100<

PresentationMethod>01<

HLOctet4

HighLayerCharacteristics>HLC_value<

ProgressIndicator

ProgressOctet3

CodingStandard>00<

Location>yyyy<

ProgressOctet4

ProgressDescription>0000101<

DSS1 Parameter values

SETUP: First Bearer Capability Information transfer capability = Speech or 3.1 kHz audio

Second Bearer Capability Information transfer capability = Unrestricted digital information with tones/announcements

PROGRESS:

Message flow

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 (Ringing)
PROGRESS:	←	183 Session Progress

Apply post test routine

TSS	TP_201_237	Reference subclause 5.1.1.2.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

No fallback information received, CLEARMODE codec not the preferred codec in the 183.

Ensure that on receipt of 183 (Session Progress) and the CLEARMODE codes is not the first codec in the codec list, the SUT sends a CALL PROCEEDING toward the calling user equipment and fallback is not indicated. No BC is sent in the CALL PROCEEDING

SIP header values

DSS1 Parameter values

SETUP: First Bearer Capability Information transfer capability = Speech or 3.1 kHz audio

Second Bearer Capability Information transfer capability = Unrestricted digital information with tones/announcements

CALL PROCEEDING

Message flow		
End device		**Test equipment**
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
CALL PROCEEDING	←	183 Session Progress
Apply post test routine		

TSS	TP_201_238	Reference subclause 5.1.1.2.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

No fallback information received, CLEARMODE codec not the preferred codec in the 183

Ensure that on receipt of 183 (Session Progress) and the CLEARMODE codes is not the first codec in the codec list, the SUT sends a PROGRESS toward the calling user equipment and fallback is not indicated. No BC is sent in the PROGRESS

SIP header values

183 (Session Progress): CLEARMODE not the first codec

DSS1 Parameter values

SETUP: First Bearer Capability Information transfer capability = Speech or 3.1 kHz audio

Second Bearer Capability Information transfer capability = Unrestricted digital information with tones/announcements

PROGRESS

Message flow		
End device		**Test equipment**
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 (Ringing)
PROGRESS	←	183 Session Progress
Apply post test routine		

7.2.2.2 Establishment of a confirmed dialogue

7.2.2.2.1 SIP basic procedures

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_001	Reference sections 12.2.1.1, 13.2.2.4, 17.1.1.3 and Figure 5 of [IETF RFC 3261]	Selection expression
Test purpose *200 OK received, an ACK is sent* Ensure that when an early dialogue is not established that on receipt of a Success (200 OK) response an ACK request is sent.			
SIP header values			
Message flow			

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	200 OK INVITE
	→	ACK
ISDN interworking		
CASE A		
SETUP	→	INVITE
CONNECT	←	200 OK INVITE
	→	ACK
CASE B		
SETUP	→	INVITE
CONNECT	←	200 OK INVITE
CONNECT ACKNOWLEDGE	→	ACK
Apply post test routine		

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_002	Reference sections 12.2.1.1, 13.2.2.4, 17.1.1.3 and Figure 5 of [IETF RFC 3261]	Selection expression
Test purpose *200 OK received in the Proceeding state an ACK is sent* Ensure that when the client transaction in the **Proceeding state** that on receipt of a Success (200 OK) response an ACK request is sent.			
SIP header values			

Message flow **End device** **Test equipment** **Interworking POTS** Off hook Dial number → INVITE ← 100 Trying ← 200 OK INVITE → ACK **ISDN interworking** **CASE A** SETUP → INVITE SETUP ACKNOWLEDGE ← 100 Trying CONNECT ← 200 OK INVITE → ACK **CASE B** SETUP → INVITE SETUP ACKNOWLEDGE ← 100 Trying CONNECT ← 200 OK INVITE CONNECT ACKNOWLEDGE → ACK **Apply post test routine**

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_003	Reference sections 12.2.1.1, 13.2.2.4, and 5 Figure 17.1.1.3 of [IETF RFC 3261]	Selection expression
Test purpose *200 OK received in an early dialogue an ACK issent* Ensure that when an **early dialogue** is established that on receipt of a Success (200 OK) response an ACK request is sent.			
SIP header values			
Message flow **End device** **Test equipment** **Interworking POTS** Off hook Dial number → INVITE ← 180 Ringing ← 200 OK INVITE → ACK **ISDN interworking** **CASE A** SETUP → INVITE ALERTING ← 180 Ringing CONNECT ← 200 OK INVITE → ACK			

CASE B SETUP → INVITE ALERTING ← 180 Ringing CONNECT ← 200 OK INVITE CONNECT ACKNOWLEDGE → ACK **Apply post test routine**

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_004	Reference section 8.1.3.3 and Figure 5 of [IETF RFC 3261]	Selection expression
Test purpose *200 OK containing two Via header discarded* Ensure that when the client transaction is in the **Proceeding state** that on receipt of a Success (200 OK) response with more than one Via header value no ACK request is sent, discards the response.			
SIP header values 200 OK INVITE Via:SIP 2.0 <transport> <IP address>;branch= <any branch value> Via:SIP 2.0 <transport> <IP address>;branch= <any branch value>			
Message flow **End device** **Test equipment** **Interworking POTS** Off hook Dial number → INVITE ← 100 Trying ← 200 OK INVITE **ISDN interworking** SETUP → INVITE SETUP ACKNOWLEDGE ← 100 Trying CONNECT ← 200 OK INVITE **Apply post test routine**			

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_005	Reference section 12.2.1.1, 13.2.2.4, 17.1.1.3 and Figure 5 of [IETF RFC 3261]	Selection expression
Test purpose *ACK is sent with the same CSeq as in the initial INVITE* Ensure that when the client transaction is in the **Calling state** that on receipt of a Success (200 OK) response an ACK request is sent with the same CSeq sequence number as in the original INVITE request and the CSeq method field value set to "ACK".			
SIP header values INVITE 1 CSeq: <value invite> INVITE ACK 1 CSeq: <value invite> ACK			

Message flow
End device / **Test equipment**
Interworking POTS
Off hook
Dial number → INVITE 1
← 200 OK INVITE
→ ACK 1
ISDN interworking
CASE A
SETUP → INVITE 1
CONNECT ← 200 OK INVITE
→ ACK 1
CASE B
SETUP → INVITE 1
CONNECT ← 200 OK INVITE
CONNECT ACKNOWLEDGE → ACK 1
Apply post test routine

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_006	Reference sections 12.2.1.1, 13.2.2.4, 17.1.1.3 and Figure 5 of [IETF RFC 3261]	Selection expression
Test purpose *ACK is sent with the same To header filed as received in the 200 OK* Ensure that when the client transaction is in the **Calling state** that on receipt of a Success (200 OK) response an ACK request is sent with the To header set to the same value as in the received final response.			
SIP header values 200 OK INVITE To: <any URI>;tag=<any 200 OK tag value> ACK To: <any URI>;tag=<any 200 OK tag value>			
Message flow **End device** / **Test equipment** **Interworking POTS** Off hook Dial number → INVITE ← 200 OK INVITE → ACK 1 **ISDN interworking** **CASE A** SETUP → INVITE CONNECT ← 200 OK INVITE → ACK 1			

CASE B		
SETUP	→	INVITE
CONNECT	←	200 OK INVITE
CONNECT ACKNOWLEDGE	→	ACK 1

Apply post test routine

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_007	Reference section 12.2.1.1, 13.2.2.4, Figure 5 and 17.1.1.3 of [IETF RFC 3261]	Selection expression

Test purpose

ACK is sent with the same To header filed as received in the 180

Ensure that when the client transaction is in the **Proceeding state** that on receipt of a Success (200 OK) response an ACK request is sent with the To header set to the same value as in the received provisional response.

SIP header values

200 OK INVITE

To: <any URI>;tag=<any 200 OK tag value>

ACK

To: <any URI>;tag=<any 200 OK tag value>

Message flow

End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	180 Ringing 1
	←	200 OK INVITE 1
	→	ACK 1
ISDN interworking		
CASE A		
SETUP	→	INVITE
ALERTING	←	180 Ringing 1
CONNECT	←	200 OK INVITE 1
	→	ACK 1
CASE B		
SETUP	→	INVITE
ALERTING	←	180 Ringing 1
CONNECT	←	200 OK INVITE 1
CONNECT ACKNOWLEDGE	→	ACK 1

Apply post test routine

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_008	Reference sections 12.2.1.1, 13.2.2.4, 17.1.1.3 and Figure 5 of [IETF RFC 3261]	Selection expression

Test purpose

ACK is sent without a To tag

Ensure that when the client transaction is in the **Calling state** that on receipt of a Success (200 OK) response including a To header without tag an ACK request is sent with a To header without tag.

SIP header values

200 OK INVITE
  To: <any URI>

ACK
  To: <any URI>

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	200 OK INVITE 1
	→	ACK 1
ISDN interworking		
CASE A		
SETUP	→	INVITE
CONNECT	←	200 OK INVITE 1
	→	ACK 1
CASE B		
SETUP	→	INVITE
CONNECT	←	200 OK INVITE 1
CONNECT ACKNOWLEDGE	→	ACK 1

Apply post test routine

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_009	Reference sections 12.2.1.1, 13.2.2.4, 17.1.1.2 and Figure 5 of [IETF RFC 3261]	Selection expression

Test purpose

ACK is sent for each received 200 OK

Ensure that when the client transaction is in the **Calling state** that on receipt of Success (200 OK) responses differing only on the tag in the To header, an ACK request is sent with a To header identical to the one received for each received Success (200 OK) responses.

SIP header values

200 OK INVITE 1

To: <any URI>;tag=<any tag 1>

ACK 1

To: <any URI>;tag=<any tag 1>

200 OK INVITE 2

To: <any URI>;tag=<any tag 2>

ACK 2

To: <any URI>;tag=<any tag 2>

Message flow

End device		**Test equipment**
Interworking POTS		
Off hook	→	INVITE
Dial number	←	200 OK INVITE 1
	→	ACK 1
	←	200 OK INVITE 2
	→	ACK 2
ISDN interworking		
CASE A		
SETUP	→	INVITE
CONNECT	←	200 OK INVITE 1
	→	ACK 1
	←	200 OK INVITE 2
	→	ACK 2
CASE B		
SETUP	→	INVITE
CONNECT	←	200 OK INVITE 1
CONNECT ACKNOWLEDGE	→	ACK 1
	←	200 OK INVITE 2
	→	ACK 2

Apply post test routine

TSS	TP_202_010	Reference	Selection expression
Orig_Establishment_of_a_confirmed_dialogue		section 12.2.1.1, 13.2.2.4, Figure 5 and 17.1.1.3 of [IETF RFC 3261]	

Test purpose

ACK is sent with the same Session identifier as in the sent INVITE

Ensure that when the client transaction is in the **Calling state** that on receipt of a Success (200 OK) response an ACK request is sent with the same Call-ID and From headers as in the original INVITE request.

SIP header values INVITE 1 Call-ID: <any invite Call ID> From: <any invite URI>;tag=<any invite tag> ACK 1 Call-ID: <any invite Call ID> From: <any invite URI>;tag=<any invite tag>
Message flow

End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE 1
	←	200 OK INVITE
	→	ACK 1
ISDN interworking		
CASE A		
SETUP	→	INVITE 1
CONNECT	←	200 OK INVITE
	→	ACK 1
CASE B		
SETUP	→	INVITE 1
CONNECT	←	200 OK INVITE
CONNECT ACKNOWLEDGE	→	ACK 1
	Apply post test routine	

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_011	Reference sections 12.2.1.1, 13.2.2.4, 17.1.1.3 and Figure 5 of [IETF RFC 3261]	Selection expression
Test purpose *ACK is sent to the address indicated in the 200 OK Contact header* Ensure, when the client transaction is in the **Calling state,** that on receipt of a Success (200 OK) response with no Record-Route header set, an ACK request is sent with the Request-URI set to the Contact URI included in the received final response and with no Route header set.			
SIP header values 200 OK Contact: <any 200 ok contact URI> ACK ACK sip: <any 200 ok contact URI> SIP 2.0			

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	200 OK INVITE 1
	→	ACK 1
ISDN interworking		
CASE A		
SETUP	→	INVITE
CONNECT	←	200 OK INVITE 1
	→	ACK 1
CASE B		
SETUP	→	INVITE
CONNECT	←	200 OK INVITE 1
CONNECT ACKNOWLEDGE	→	ACK 1

Apply post test routine

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_012	Reference sections 12.2.1.1, 13.2.2.4 and Figure 5 of [IETF RFC 3261]	Selection expression

Test purpose

ACK is sent to the address indicated in the Record-Route header with 'lr' parameter

Ensure that when the client transaction is in the **Calling state** that on receipt of a Success (200 OK) response with a Record-Route header set to a list in which the last element contains lr parameter, an ACK request is sent with the Request-URI set to the Contact URI and a Route header set to the list in a reverse order of the Record-Route included in the received final response.

SIP header values

200 OK 1

Record-Route: <any 200 ok route URI1;lr> ,<any 200 ok route URI2;lr>

Contact: <any 200 ok contact URI>

ACK 1

ACK sip: <any 200 ok route URI> SIP 2.0

Route: <any 200 ok route URI2;lr> ,<any 200 ok route URI1;lr>

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	200 OK INVITE
	→	ACK 1
ISDN interworking		

CASE A

SETUP	→	INVITE
CONNECT	←	200 OK INVITE 1
	→	ACK 1

CASE B

SETUP	→	INVITE
CONNECT	←	200 OK INVITE 1
CONNECT ACKNOWLEDGE	→	ACK 1

Apply post test routine

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_013	Reference section 12.2.1.1, 13.2.2.4, Figure 5 and 17.1.1.3 of [IETF RFC 3261]	Selection expression

Test purpose

ACK is sent to the address indicated in the Record-Route header without 'lr' parameter

Ensure that when the client transaction is in the **Calling state** that on receipt of a Success (200 OK) response with a Record-Route header set to a list in which the last element does not contain lr parameter, an ACK request is sent with the Request-URI set to this element and a Route header set to the remainder list in a reverse order of the received Record-Route appended with the received Contact URI.

SIP header values

200 OK

Record-Route: <any 200 ok route URI1>,<any 200 ok route URI2;lr>

Contact: <any 200 ok contact URI>

ACK

ACK sip: <any 200 ok route URI1> SIP 2.0

Route: <any 200 ok route URI2;lr>,<any 200 ok contact URI>

Message flow

End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	200 OK INVITE 1
	→	ACK 1

ISDN interworking

CASE A

SETUP	→	INVITE
CONNECT	←	200 OK INVITE 1
	→	ACK 1

CASE B

SETUP	→	INVITE
CONNECT	←	200 OK INVITE 1
CONNECT ACKNOWLEDGE	→	ACK 1

Apply post test routine

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_014	Reference section 13.2.1 of [IETF RFC 3261]	Selection expression

Test purpose

SDP offer sent in the initial INVITE request

Ensure that while the client is establishing a call, a unique session description offer is sent in the INVITE request to answer the initial offers given in the final 2XX response.

SIP header values

INVITE
  SDP
    'm' line
    'a' line
200 OK INVITE
  SDP
    'm' line
    'a' line

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	200 OK INVITE
ISDN interworking		
SETUP	→	INVITE
	←	200 OK INVITE

Apply post test routine

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_015	Reference section 13.2.1 of [IETF RFC 3261]	Selection expression

Test purpose

SDP answer sent in the initial ACK request

Ensure that while the client is establishing a call, a unique session description offer is sent in the ACK request to answer the initial offers given in the final 2XX response.

SIP header values

200 OK 1
  SDP
    'm' line
    'a' line

ACK 1
  SDP
    'm' line
    'a' line

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	200 OK INVITE 1
	→	ACK 1
ISDN interworking		
CASE A		
SETUP	→	INVITE
CONNECT	←	200 OK INVITE 1
	→	ACK 1
CASE B		
SETUP	→	INVITE
CONNECT	←	200 OK INVITE 1
CONNECT ACKNOWLEDGE	→	ACK 1

Apply post test routine

TSS	TP_202_016	Reference	Selection expression
Orig_Establishment_of_a_confirmed_dialogue		sections 13.2.2.3, 17, 17.1.1.2, 17.1.1.3. and Figure 5 of [IETF RFC 3261]	

Test purpose

Update Record-Route header

Ensure that when the client transaction is in the **Calling state** that on receipt of a Success (200 OK) response, with a different Record-Route as in previous response, but with the same Via branch parameter and CSeq header method as in the INVITE request, an ACK request is sent with a Route header set according to this new Record-Route.

SIP header values

200 OK 2
    Record-Route: <any 200 ok route>
ACK 2
    Route: <any 200 ok route>

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE 1
	←	200 OK INVITE 1
	→	ACK 1
	←	200 OK INVITE 2
	→	ACK 2
ISDN interworking		
SETUP	→	INVITE 1
CONNECT	←	200 OK INVITE 1
CONNECT ACKNOWLEDGE	→	ACK 1
	←	200 OK INVITE 2
	→	ACK 2

Apply post test routine

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_017	Reference section 20.14 and 13.2.1 of [IETF RFC 3261]	Selection expression

Test purpose

Content-Length header present in the initial INVITE request

Ensure that when the client transaction is establishing a call, a Content-Length header set to the size of the body in the message that contains the session description is sent.

SIP header values

INVITE
    Content-Length: <any value>

    SDP
        'm' line
        'a' line

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
ISDN interworking		
SETUP	→	INVITE

Apply post test routine

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_018	Reference section 20.14 and 13.2.1 of [IETF RFC 3261]	Selection expression

Test purpose

Content Type header indicated SDP

Ensure when the client transaction is establishing a call, a Content-Type header in the message that contains the session description is sent.

SIP header values

INVITE
    Content-Type: application/sdp

    SDP
        'm' line
        'a' line

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	200 OK INVITE 1
	→	ACK 1
ISDN interworking		

CASE A		
SETUP	→	INVITE
CONNECT	←	200 OK INVITE 1
	→	ACK 1
CASE B		
SETUP	→	INVITE
CONNECT	←	200 OK INVITE 1
CONNECT ACKNOWLEDGE	→	ACK 1
Apply post test routine		

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_019	Reference subclause 6.3.2.4 of [ETSI TS 183 043]	Selection expression
Test purpose *Modifying SDP in confirmed dialogue* Ensure that the SUT is able to receive an UPDATE request to modify the SDP in confirmed dialogue. A 200 OK UPDATE with SDP answer is sent.			
SIP header values INVITE/UPDATE SDP offer 2 200 OK (INVITE/UPDATE) SDP answer 2			
Message flow			

End device		**Test equipment**
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
	←	180 Ringing
		200 IK INVITE
		ACK
CASE A		
	←	INVITE
	→	200 OK INVITE
	←	ACK
CASE B		
	←	UPDATE
	→	200 OK UPDATE
Apply post test routine		

7.2.2.2.3 Test purposes for ISDN

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_201	Reference subclause 5.1.1.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

200 OK received PSTN XML ProgressIndicator present value x, a CONNECT is sent PI x

Ensure that on receipt of a 200 OK (INVITE) where a PSTN XML body is present, ProgressIndicator value is set to PI_VA, a CONNECT message is sent to the calling user equipment and a Progress Indicator Information Element is included, the Progress description value is set to PI_VA as described in Table 7.2.2.2.3-1.

SIP header values

200 OK (INVITE): XML body

<?xml version="1.0" encoding="utf-8"?>

PSTN
  ProgressIndicator
    ProgressOctet3
      CodingStandard>00<
      Location>yyyy<
    ProgressOctet4
      ProgressDescription>PI_VA<
  ProgressIndicator
    ProgressOctet3
      CodingStandard>00<
      Location>yyyy<
    ProgressOctet4
      ProgressDescription>0000111<

DSS1 Parameter values

CONNECT:

Message flow

End device		Test equipment
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 (Ringing)
CONNECT	←	200 OK (INVITE)
	→	ACK

Apply post test routine

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_202	Reference subclause 5.1.1.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

200 OK received a PSTN XML ProgressIndicator value x present, a CONNECT is sent PI x

Ensure that on receipt of a 200 OK (INVITE) where a PSTN XML body is present, ProgressIndicator value is set to PI_VA, a CONNECT message is sent to the calling user equipment and a Progress Indicator Information Element is included, the Progress description value is set to PI_VA. An additional Progress Indicator Information Element is present and the Progress description value is set to value 4 if a Progress Indicator value 1 was sent before in a ALERTING message as described in Table 7.2.2.2.3-1.

SIP header values

180 (Ringing): no PSTN XML body present

200 OK (INVITE): XML body

<?xml version="1.0" encoding="utf-8"?>

PSTN
    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>yyyy<
        ProgressOctet4
            ProgressDescription>PI_VA<
    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>yyyy<
        ProgressOctet4
            ProgressDescription>0000100<

DSS1 Parameter values

CONNECT:

Message flow

End device		Test equipment
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 (Ringing)
CONNECT	←	200 OK (INVITE)
	→	ACK

Apply post test routine

TSS	TP_202_203	Reference	Selection expression
Orig_Establishment_of_a_confirmed_dialogue		subclause 5.1.1.3 of [ETSI TS 183 036]	PICS 5.1.1/2 and 5.4/1

Test purpose

200 OK received PSTN XML ProgressIndicator present value 7, a CONNECT is sent.

Ensure that on receipt of a 200 OK (INVITE) where a PSTN XML body is present, ProgressIndicator value is set to 7 and a CONNECT message is sent to the calling user equipment as described in Table 7.2.2.2.3-1.

SIP header values

180 (Ringing): PSTN XML body present (PI #7)

200 OK (INVITE): XML body

<?xml version="1.0" encoding="utf-8"?>

PSTN
    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>yyyy<
        ProgressOctet4
            ProgressDescription>PI_VA<

DSS1 Parameter values
CONNECT:

Message flow

End device		Test equipment
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 (Ringing)
CONNECT	←	200 OK (INVITE)
	→	ACK

Apply post test routine

TSS	TP_202_204	Reference	Selection expression
Orig_Establishment_of_a_confirmed_dialogue		subclause 5.1.1.3 of [ETSI TS 183 036]	PICS 5.1.1/2 and 5.4/1

Test purpose

200 OK received PSTN XML ProgressIndicator present value 7, a CONNECT is sent.

Ensure that on receipt of a 200 OK (INVITE) where a PSTN XML body is present, ProgressIndicator value is set to 7 and a CONNECT message is sent to the calling user equipment as described in Table 7.2.2.2.3-1.

SIP header values

180 (Ringing): PSTN XML body present (PI #7)
200 OK (INVITE): XML body
<?xml version="1.0" encoding="utf-8"?>
PSTN
  ProgressIndicator
    ProgressOctet3
      CodingStandard>00<
      Location>yyyy<
    ProgressOctet4
      ProgressDescription>PI_VA<

DSS1 Parameter values
CONNECT:

Message flow

End device		Test equipment
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 (Ringing)
CONNECT	←	200 OK (INVITE)
	→	ACK

Apply post test routine

<table>
<tr><td>TSS
Orig_Establishment_of_a_confirmed_dialogue</td><td>TP_202_205</td><td>Reference
subclause 5.1.1.3 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 and 5.4/1</td></tr>
<tr><td colspan="4">Test purpose
200 OK received PSTN XML ProgressIndicator present value 7, a CONNECT is sent.

Ensure that on receipt of a 200 OK (INVITE) where a PSTN XML body is present, ProgressIndicator value is set to 7 and a CONNECT message is sent to the calling user equipment. A Progress Indicator Information Element is present the Progress description value is set to value 4 if a Progress Indicator value 1 was sent before in a ALERTING message.</td></tr>
<tr><td colspan="4">SIP header values
180 (Ringing): PSTN XML body present (PI #7)
200 OK (INVITE): XML body
<?xml version="1.0" encoding="utf-8"?>
PSTN
    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>yyyy<
        ProgressOctet4
            ProgressDescription>0000111<</td></tr>
<tr><td colspan="4">DSS1 Parameter values
ALERTING: PI #1
CONNECT: PI #4</td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><th>End device</th><th></th><th>Test equipment</th></tr>
<tr><td>SETUP</td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td>←</td><td>407 Proxy Authentication Required</td></tr>
<tr><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>ALERTING</td><td>←</td><td>180 (Ringing)</td></tr>
<tr><td>CONNECT</td><td>←</td><td>200 OK (INVITE)</td></tr>
<tr><td></td><td>→</td><td>ACK</td></tr>
</table>
Apply post test routine</td></tr>
</table>

Table 7.2.2.2.3-1 – Mapping of progress indicator information element to PSTN XML ProgressIndicator

PI_value	DSS1 Progress Indicator value	XML ProgressIndicator ProgressDescription
PI_VA_1	Call is not end-to-end 5.1.1/2; further call progress information may be available in-band	'0000001'
PI_VA_2	Destination address is non-5.1.1/2	'0000010'
PI_VA_3	Origination address is non-5.1.1/2	'0000011'
PI_VA_4	Call has returned to the 5.1.1/2	'0000100'
PI_VA_5	Interworking has occurred and has resulted in a telecommunication service change	'0000101'
PI_VA_6	In-band information or an appropriate pattern is now available	'0001000'

TSS Orig_Establishment_of_a _confirmed_dialogue	TP_202_206	Reference subclause 5.1.1.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

PSTN XML body LowLayerCompatibility received in 200 OK. A CONNECT is sent LLC is present

Ensure that on receipt of a LowLayerCompatibility PSTN XML element present in a 200 OK INVITE, a DSS1 CONNECT message is sent to the calling user equipment and a HLC IE is present as described in Table 7.2.2.2.3-2.

SIP header values

200 OK (INVITE): PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

    LowLayerCompatibility>
        LLOctet3>
            CodingStandard>00<
            InformationTransferCapability>ITC_VA<
        LLOctet4>
            TransferMode>00<
            InformationTransferRate>10000<

DSS1 Parameter values

CONNECT: HLC

Message flow

End device		Test equipment
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 (Ringing)
CONNECT	←	200 OK (INVITE)
	→	ACK

Apply post test routine

Table 7.2.2.2.3-2 – Mapping of low layer compatibility to PSTN XML LowLayerCompatibility

ITC_value	LLC information transfer capability	XML LLC InformationTransferCabability
ITC_VA_1	Speech	'00000'
ITC_VA_2	3,1 kHz audio	'10000'
ITC_VA_3	Unrestricted digital info	'01001'
ITC_VA_3	7 kHz audio	'10001'

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_207	Reference subclause 5.1.1.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

PSTN XML body BearerCapability received in 200 OK CONNECT is sent BC present

Fallback connection type indicated in the SETUP. Ensure that on receipt of a 200 OK (INVITE) that contains a PSTN XML BearerCapability indicating fallback, a CONNECT message is sent to the calling user equipment and a Bearer Capability Information Element is present set to the value of the received BearerCapability XML element as described in Table 5.2.1.3-5.

SIP header values

200 OK (INVITE): PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

  BearerCapability

    BCoctet3

      CodingStandard>00<

      InformationTransferCabability>ITC_VA<

    BCoctet4

      TransferMode>00<

      InformationTransferRate>10000<

    BCoctet5>

      Layer1Identification>01<

      UserInfoLayer1Protocol>00011<

  ProgressIndicator

    ProgressOctet3

      CodingStandard>00<

      Location>yyyy<

    ProgressOctet4

      ProgressDescription>0000111<

SDP:

DSS1 Parameter values

SETUP: First Bearer Capability Information transfer capability = Speech or 3.1 kHz audio

Second Bearer Capability Information transfer capability = Unrestricted digital information with tones/announcements

CONNECT: BC

Message flow

End device		Test equipment
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 (Ringing)
CONNECT	←	200 OK (INVITE)
	→	ACK

Apply post test routine

Table 7.2.2.2.3-3 – Mapping of bearer capability to PSTN XML BearerCapability

ITC_value	BC information transfer capability	XML InformationTransferCabability	SDP m - line: first stated codec
ITC_VA_1	Speech	'00000'	0 or 8
ITC_VA_2	3,1 kHz audio	'10000'	0 or 8
ITC_VA_3	Unrestricted digital inf. W/tone/ann	'10001'	CLEARMODE

<table>
<tr><td>TSS
Orig_Establishment_of_a_confirmed_dialogue</td><td>TP_202_208</td><td>Reference
subclause 5.1.1.3 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 and 5.4/1</td></tr>
<tr><td colspan="4">Test purpose
200 OK received no PSTN XML ProgressIndicator present, a CONNECT BC included is sent.

Fall-back connection type indicated in the SETUP. Ensure that on receipt of a 200 OK (INVITE) and no PSTN XML body is present, a CONNECT message is sent to the calling user equipment. A Bearer Capability Information Element is present set to the value speech or audio 3.1 kHz</td></tr>
<tr><td colspan="4">SIP header values
200 OK (INVITE):
no XML body present

SDP:</td></tr>
<tr><td colspan="4">DSS1 Parameter values
SETUP: First Bearer Capability Information transfer capability = Speech or 3.1 kHz audio
Second Bearer Capability Information transfer capability = Unrestricted digital information with tones/announcements

CONNECT: BC speech or audio 3.1 kHz</td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><td>End device</td><td></td><td>Test equipment</td></tr>
<tr><td>SETUP</td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td>←</td><td>407 Proxy Authentication Required</td></tr>
<tr><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>ALERTING</td><td>←</td><td>180 (Ringing)</td></tr>
<tr><td>CONNECT</td><td>←</td><td>200 OK (INVITE)</td></tr>
<tr><td></td><td>→</td><td>ACK</td></tr>
</table>
Apply post test routine</td></tr>
</table>

TSS Orig_Establishment_of_a_confirmed_dialogue	TP_202_209	Reference subclause 5.1.1.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

Fallback information received as PSTN XML body HLC in the 200 a CONNECT contains the fall-back information

Fallback connection type indicated in the SETUP. Ensure that the SUT on receipt of a HighLayerCapability value set to HLC_VA and ProgressIndicator No. 5 PSTN XML body, a CONNECT message is sent to the calling user equipment. The CONNECT message contains a HLC Information Element derived from the received PSTN XML HighLayerCapability and a Progress Indicator Information Element derived form the received PSTN XML HighLayerCompatibility as described in Table 7.2.2.2.3-4.

SIP header values

<?xml version="1.0" encoding="utf-8"?>

PSTN
- **HighLayerCompatibility**
  - HLOctet3
    - CodingStandard>00<
    - Interpretation>100<
    - PresentationMethod>01<
  - HLOctet4
    - HighLayerCharacteristics>HLC_VA<
- **ProgressIndicator**
  - ProgressOctet3
    - CodingStandard>00<
    - Location>yyyy<
  - ProgressOctet4
    - ProgressDescription>0000101<
- **ProgressIndicator**
  - ProgressOctet3
    - CodingStandard>00<
    - Location>yyyy<
  - ProgressOctet4
    - ProgressDescription>0000111<

DSS1 Parameter values

SETUP: First Bearer Capability Information transfer capability = Speech or 3.1 kHz audio
Second Bearer Capability Information transfer capability = Unrestricted digital information with tones/announcements

CONNECT: BC speech or audio 3.1 kHz

Message flow

End device		Test equipment
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ALERTING	←	180 (Ringing)
CONNECT	←	200 OK (INVITE)
	→	ACK

Apply post test routine

Table 7.2.2.2.3-4 – Mapping of high layer compatibility information element to PSTN XML HighLayerCharacteristic

HLC_value	DSS1 high layer characteristics identification	XML HighLayerCharacteristic
HLC_VA_1	Telephony	'0000001'
HLC_VA_2	Facsimile Group 2/3	'0000100'
HLC_VA_3	Facsimile Group 4 Class I	'0100001'
HLC_VA_4	Facsimile service Group 4, Classes II ad III	'0100100'
HLC_VA_5	Syntax based Videotex	'0110010'
HLC_VA_6	International Videotex interworking via gateways or interworking units	'0110011'
HLC_VA_7	Telex service	'0110101'
HLC_VA_8	FTAM application	'1000010'
HLC_VA_9	Videotelephony	'1100000'

7.2.2.3 Call release

7.2.2.3.1 Release initiated by the originating user

7.2.2.3.1.1 SIP basic procedures

TSS Orig_Orig_Release_initiated_by_the_terminating_user	TP_203_001	Reference section 12.2.1.1 and 15 of [IETF RFC 3261]	Selection expression

Test purpose

To header in BYE is used from the last final response

Ensure that once a dialogue has been established the IUT sends a BYE request to release it where a To header is set to the same value as in the last received final response.

SIP header values

200 OK
 To: <any 200 ok To URI; tag=<any 200ok tag>
BYE
 To: <any 200 ok To URI; tag=<any 200ok tag>

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	180 Ringing
	←	200 OK INVITE
	→	ACK
On hook	→	BYE
	←	200 OK BYE
ISDN interworking		
CASE A		
SETUP	→	INVITE
ALERTING	←	180 Ringing
CONNECT	←	200 OK INVITE
	→	ACK
DISCONNECT	→	BYE
RELEASE	←	200 OK BYE
RELEASE COMPLETE	→	

<table>
<tr><td>TSS
Orig_Orig_Release_initiated
_by_the_terminating_user</td><td>TP_203_002</td><td>Reference
section 12.2.1.1 and 15 of
[IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
BYE is sent without To tag

Ensure that if a dialog had been established with a final response in which the 'tag' in the To header was omitted, to release it the IUT sends a BYE request with an identical To header without 'tag' value.</td></tr>
<tr><td colspan="4">SIP header values
200 OK
To: <any 200 ok To URI>
BYE
To: <any 200 ok To URI></td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><td>End device</td><td></td><td>Test equipment</td></tr>
<tr><td>Interworking POTS</td><td></td><td></td></tr>
<tr><td>Off hook</td><td></td><td></td></tr>
<tr><td>Dial number</td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td>→</td><td>ACK</td></tr>
<tr><td>On hook</td><td>→</td><td>BYE</td></tr>
<tr><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td>ISDN interworking</td><td></td><td></td></tr>
<tr><td>SETUP</td><td>→</td><td>INVITE</td></tr>
<tr><td>CONNECT</td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td>→</td><td>ACK</td></tr>
<tr><td>DISCONNECT</td><td>→</td><td>BYE</td></tr>
<tr><td>RELEASE</td><td>←</td><td>200 OK BYE</td></tr>
<tr><td>RELEASE COMPLETE</td><td>→</td><td></td></tr>
</table></td></tr>
</table>

<table>
<tr><td>TSS
Orig_Orig_Release_initiated
_by_the_terminating_user</td><td>TP_203_003</td><td>Reference
section 12.2.1.1 and 15 of
[IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
To header in BYE is used from the last final response

Ensure that the IUT, once a dialogue has been established to release, it sends a BYE request with a To header set to the same value as in the last received final response.</td></tr>
<tr><td colspan="4">SIP header values
200 OK
To: <any 200 ok To URI; tag=<any 200ok tag>
BYE
To: <any 200 ok To URI; tag=<any 200ok tag></td></tr>
</table>

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	180 Ringing
	←	200 OK INVITE
	→	ACK
On hook	→	BYE
	←	200 OK BYE
ISDN interworking		
SETUP	→	INVITE
ALERTING	←	180 Ringing
CONNECT	←	200 OK INVITE
	→	ACK
DISCONNECT	→	BYE
RELEASE	←	200 OK BYE
RELEASE COMPLETE	→	

TSS			
TSS Orig_Orig_Release_initiated_by_the_terminating_user	TP_203_004	Reference section 12.2.1.1 and 15 of [IETF RFC 3261]	Selection expression

Test purpose

Call-ID is set to the value as in the original INVITE

Ensure that the IUT, once a dialogue has been established to release, it sends a BYE request with the same Call-ID, From headers as in the original INVITE message.

SIP header values

200 OK
    Call-ID: <any 200 ok Call-ID value>
BYE
    Call-ID: <any 200 ok Call-ID value>

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	180 Ringing
	←	200 OK INVITE
	→	ACK
On hook	→	BYE
	←	200 OK BYE
ISDN interworking		
SETUP	→	INVITE
ALERTING	←	180 Ringing
CONNECT	←	200 OK INVITE
	→	ACK
DISCONNECT	→	BYE
RELEASE	←	200 OK BYE
RELEASE COMPLETE	→	

TSS			
TSS Orig_Orig_Release_initiated_by_the_terminating_user	TP_203_005	Reference section 12.2.1.1 of [IETF RFC 3261]	Selection expression

Test purpose

CSeq in BYE is incremented

Ensure that the IUT, once a dialogue has been established to release, it sends a BYE request with an increment of one CSeq value, a method field in the CSeq header set to "BYE".

SIP header values

INVITE
    CSeq: <any INVITE sequence number> INVITE
BYE
    CSeq: <any INVITE sequence number> BYE

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	180 Ringing
	←	200 OK INVITE
	→	ACK
On hook	→	BYE
	←	200 OK BYE
ISDN interworking		
SETUP	→	INVITE
ALERTING	←	180 Ringing
CONNECT	←	200 OK INVITE
	→	ACK
DISCONNECT	→	BYE
RELEASE	←	200 OK BYE
RELEASE COMPLETE	→	

TSS Orig_Orig_Release_initiated_by_the_terminating_user	TP_203_006	Reference section 12.2.1.1 and 15 of [IETF RFC 3261]	Selection expression

Test purpose

BYE is sent to the address indicated in the 200 OK Contact header

Ensure that the IUT, once a dialogue has been established with a Success (200 OK) response including no Record Route header set, to release it, sends a BYE request with the Request URI set to the Contact URI included in the received final response and with no Route header set.

SIP header values

200 OK
    Contact: <any 200 ok contact value>
BYE: <any 200 ok contact value> SIP/2.0

Message flow		
End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	180 Ringing
	←	200 OK INVITE
	→	ACK
On hook	→	BYE
	←	200 OK BYE
ISDN interworking		
SETUP	→	INVITE
ALERTING	←	180 Ringing
CONNECT	←	200 OK INVITE
	→	ACK
DISCONNECT	→	BYE
RELEASE	←	200 OK BYE
RELEASE COMPLETE	→	

TSS Orig_Orig_Release_initiated_by_the_terminating_user	TP_203_007	Reference section 12.2.1.1 and 15 of [IETF RFC 3261]	Selection expression

Test purpose

BYE is sent to the address indicated in the Record-Route header with 'lr' parameter

Ensure that the IUT, once a dialogue has been established with a Success (200 OK) response including a Record-Route header set to a list in which the last element contains lr parameter, to release the call, sends a BYE request with the Request-URI set to the Contact URI and a Route header set to the list in a reverse order of the Record-Route included in the received final response.

SIP header values

200 OK

Record-Route: <any 200 ok route URI1;lr> ,<any 200 ok route URI2;lr>

Contact: <any 200 ok contact URI>

BYE

BYE sip: <any 200 ok contact URI2> SIP 2.0

Route: <any 200 ok route URI2;lr> ,<any 200 ok route URI1;lr>

Message flow		
End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	180 Ringing
	←	200 OK INVITE
	→	ACK
On hook	→	BYE
	←	200 OK BYE
ISDN interworking		
SETUP	→	INVITE
ALERTING	←	180 Ringing
CONNECT	←	200 OK INVITE
	→	ACK
DISCONNECT	→	BYE
RELEASE	←	200 OK BYE
RELEASE COMPLETE	→	

TSS Orig_Orig_Release_initiated _by_the_terminating_user	TP_203_008	Reference section 12.2.1.1 and 15 of [IETF RFC 3261]	Selection expression
Test purpose *Route set in BYE is derived from the Record-Route headers in revers order* Ensure that the IUT, once a dialogue has been established with a Success (200 OK) response including a Record-Route header set to a list in which the last element does not contain lr parameter, to release the call, sends an BYE request with the Request-URI set to this element and a Route header set to the remainder list in a reverse order of the received Record-Route appended with the received Contact URI.			
SIP header values 200 OK Record-Route: <any 200 ok route URI1>,<any 200 ok route URI2;lr> Contact: <any 200 ok contact URI> BYE BYE sip: <any 200 ok route URI1> SIP 2.0 Route: <any 200 ok route URI2;lr>,<any 200 ok contact URI>			

Message flow **End device** **Test equipment** **Interworking POTS** Off hook Dial number → INVITE ← 180 Ringing ← 200 OK INVITE → ACK On hook → BYE ← 200 OK BYE **ISDN interworking** SETUP → INVITE ALERTING ← 180 Ringing CONNECT ← 200 OK INVITE → ACK DISCONNECT → BYE RELEASE ← 200 OK BYE RELEASE COMPLETE →

TSS Orig_Orig_Release_initiated_by_the_terminating_user	TP_203_009	Reference section 12.2.1.1 and 15 of [IETF RFC 3261]	Selection expression
Test purpose *Session in confirmed dialogue is terminated by sending a BYE message* Ensure that the IUT, when a dialogue has been established, having sent a BYE request, on receipt of a (200 OK) response considers the session and the dialogue terminated.			
SIP header values			
Message flow **End device** **Test equipment** **Interworking POTS** Off hook Dial number → INVITE 1 ← 180 Ringing ← 200 OK INVITE → ACK On hook → BYE Start timer K ← 200 OK BYE Timeout timer K ← BYE → 481 Call Leg Does Not Exist			

ISDN interworking SETUP → → INVITE 1 ALERTING ← ← 180 Ringing CONNECT ← ← 200 OK INVITE → ACK DISCONNECT → Start timer K → BYE RELEASE ← ← 200 OK BYE RELEASE COMPLETE → Timeout timer K ← BYE → 481 Call Leg Does Not Exist

TSS Orig_Orig_Release_initiated_by_the_terminating_user	TP_203_010	Reference section 12.2.1.1 and 15.1.1 of [IETF RFC 3261]	Selection expression
Test purpose *BYE received after 200 OK BYE was sent* Ensure that the IUT, when a dialogue has been established, having sent a BYE request, on receipt of a Call Leg/Transaction Does Not Exist (481 Call Leg/Transaction Does Not Exist) considers the session and the dialogue terminated.			
SIP header values			
Message flow **End device** **Test equipment** **Interworking POTS** Off hook Dial number → INVITE 1 ← 180 Ringing ← 200 OK INVITE → ACK On hook → BYE Start timer K ← 200 OK BYE Timeout timer K ← BYE → 481 Call Leg Does Not Exist **ISDN interworking** SETUP → → INVITE 1 ALERTING ← ← 180 Ringing CONNECT ← ← 200 OK INVITE → ACK DISCONNECT → Start timer K → BYE RELEASE ← ← 200 OK BYE RELEASE COMPLETE → Timeout timer K ← BYE → 481 Call Leg Does Not Exist			

TSS Orig_Orig_Release_initiated _by_the_terminating_user	TP_203_011	Reference section 15 and 9.1, Figure 5 of [IETF RFC 3261]	Selection expression

Test purpose

Session in early dialogue is terminated by sending a CANCEL or BYE message

Ensure that the IUT, having received a Trying (100 Trying) response to its INVITE request, to give up the call, sends a CANCEL/BYE request.

SIP header values

Message flow

End device			Test equipment
Interworking POTS			
Off hook			
Dial number		→	INVITE
		←	100 Trying
On hook		→	CANCEL
		←	200 OK CANCEL
		←	487 Request Terminated
		→	ACK
ISDN interworking			
SETUP	→	→	INVITE
		←	100 Trying
RELEASE	→	→	CANCEL/BYE
RELEASE COMPLETE	←	←	200 OK CANCEL/BYE
		←	487 Request Terminated
		→	ACK

TSS Orig_Orig_Release_initiated _by_the_terminating_user	TP_203_012	Reference section 15 and 9.1, Figure 5 of [IETF RFC 3261]	Selection expression

Test purpose

Header in the CANCEL or BYE message

Ensure that the IUT, having received a Trying (100 Trying) response to its INVITE request, to give up the call, sends a CANCEL/BYE request with the same Request-URI, Call-ID, From, To headers as in the original INVITE message.

SIP header values

INVITE
INVITE sip: <invite request-line>
From: sip:<invite from uri>;tag=<from tag>
To: sip: sip:<invite to uri>
Call-ID: <invite call-id>

CANCEL
INVITE sip: <invite request-line>
From: sip:<invite from uri>;tag=<from tag>
To: sip: sip:<invite to uri>
Call-ID: <invite call-id>

Message flow

End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	100 Trying
On hook	→	CANCEL/BYE
	←	200 OK CANCEL/BYE
	←	487 Request Terminated
	→	ACK

ISDN interworking			
SETUP	→	→	INVITE
		←	100 Trying
RELEASE	→	→	CANCEL/BYE
RELEASE COMPLETE	←	←	200 OK CANCEL/BYE
		←	487 Request Terminated
		→	ACK

TSS Orig_Orig_Release_initiated_by_the_terminating_user	TP_203_013	Reference section 15, 9.1 and Figure 5 of [IETF RFC 3261]	Selection expression

Test purpose

CSeq method parameter in CANCEL or BYE

Ensure that the IUT, having received a Trying (100 Trying) response to its INVITE request, sends a CANCEL request with the same numeric part of CSeq as in the original INVITE message with a method field in the CSeq header set to "CANCEL/BYE ".

SIP header values

INVITE

Call-ID: <invite cseq#> INVITE

CANCEL

Call-ID: <invite cseq#> CANCEL

Message flow

End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	100 Trying
On hook	→	CANCEL/BYE
	←	200 OK CANCEL/BYE
	←	487 Request Terminated
	→	ACK

ISDN interworking			
SETUP	→	→	INVITE
		←	100 Trying
RELEASE	→	→	CANCEL/BYE
RELEASE COMPLETE	←	←	200 OK CANCEL/BYE
		←	487 Request Terminated
		→	ACK

<table>
<tr><td>TSS
Orig_Orig_Release_initiated_by_the_terminating_user</td><td>TP_203_014</td><td>Reference
section 15, 9.1 and Figure 5 of [IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
CANCEL or BYE contains the Via header from the initial INVITE request

Ensure that the IUT, having received a Trying (100 Trying) response to its INVITE request, sends a CANCEL/BYE request with a single Via header value matching the top Via value of the Via header of the original INVITE message.</td></tr>
<tr><td colspan="4">SIP header values
INVITE
Via: SIP/2.0/<any via protocol> <any via uri>;branch=<any via branch

CANCEL
Via: SIP/2.0/<any via protocol> <any via uri>;branch=<any via branch</td></tr>
<tr><td colspan="4">Message flow</td></tr>
<tr><td colspan="2">End device</td><td colspan="2">Test equipment</td></tr>
<tr><td colspan="4">Interworking POTS</td></tr>
<tr><td>Off hook</td><td></td><td></td><td></td></tr>
<tr><td>Dial number</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td>←</td><td>100 Trying</td></tr>
<tr><td>On hook</td><td></td><td>→</td><td>CANCEL/BYE</td></tr>
<tr><td></td><td></td><td>←</td><td>200 OK CANCEL/BYE</td></tr>
<tr><td></td><td></td><td>←</td><td>487 Request Terminated</td></tr>
<tr><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td colspan="4">ISDN interworking</td></tr>
<tr><td>SETUP</td><td>→</td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td>←</td><td>100 Trying</td></tr>
<tr><td>RELEASE</td><td>→</td><td>→</td><td>CANCEL/BYE</td></tr>
<tr><td>RELEASE COMPLETE</td><td>←</td><td>←</td><td>200 OK CANCEL/BYE</td></tr>
<tr><td></td><td></td><td>←</td><td>487 Request Terminated</td></tr>
<tr><td></td><td></td><td>→</td><td>ACK</td></tr>
</table>

<table>
<tr><td>TSS
Orig_Orig_Release_initiated_by_the_terminating_user</td><td>TP_203_015</td><td>Reference
section 15, 9.1 and Figure 5 of [IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
The CANCEL or BYE does not contain a Require or Proxy Require header

Ensure that the IUT, having received a Trying (100 Trying) response to its INVITE request, to give up the call, sends a CANCEL/BYE request without Require or Proxy-Require header.</td></tr>
<tr><td colspan="4">SIP header values
CANCEL
Require: [not present]
Proxy-Require: [not present]</td></tr>
</table>

Message flow

End device			Test equipment
Interworking POTS			
Off hook			
Dial number		→	INVITE
		←	100 Trying
On hook		→	CANCEL/BYE
		←	200 OK CANCEL/BYE
		←	487 Request Terminated
		→	ACK
ISDN interworking			
SETUP	→	→	INVITE
		←	100 Trying
RELEASE	→	→	CANCEL/BYE
RELEASE COMPLETE	←	←	200 OK CANCEL/BYE
		←	487 Request Terminated
		→	ACK

TSS	TP_203_016	Reference	Selection expression
Orig_Orig_Release_initiated_by_the_terminating_user		section 15, 9.1 and Figure 5 of [IETF RFC 3261]	

Test purpose

A CANCEL or BYE is sent after 180 was received

Ensure that the IUT, having received a 180 Ringing provisional response to its INVITE request, to give up the call, sends a CANCEL/BYE request.

Message flow

End device			Test equipment
Interworking POTS			
Off hook			
Dial number		→	INVITE
		←	180 Ringing
On hook		→	CANCEL/BYE
		←	200 OK CANCEL/BYE
		←	487 Request Terminated
		→	ACK
ISDN interworking			
SETUP	→	→	INVITE
		←	180 Ringing
RELEASE	→	→	CANCEL/BYE
RELEASE COMPLETE	←	←	200 OK CANCEL/BYE
		←	487 Request Terminated
		→	ACK

TSS Orig_Orig_Release_initiated_by_the_terminating_user	TP_203_017	Reference section 15, 9.1 and Figure 5 of [IETF RFC 3261]	Selection expression

Test purpose

A CANCEL or BYE is sent after 181 was received

Ensure that the IUT, having received a 181 Call Is Being Forwarded provisional response to its INVITE request, to give up the call, sends a CANCEL/BYE request.

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	Call Is Being Forwarded
On hook	→	CANCEL/BYE
	←	200 OK CANCEL/BYE
	←	487 Request Terminated
	→	ACK

End device			Test equipment
ISDN interworking			
SETUP	→	→	INVITE
		←	180 Ringing
RELEASE	→	→	CANCEL/BYE
RELEASE COMPLETE	←	←	200 OK CANCEL/BYE
		←	487 Request Terminated
		→	ACK

TSS Orig_Orig_Release_initiated_by_the_terminating_user	TP_203_018	Reference section 15, 9.1 and Figure 5 of [IETF RFC 3261]	Selection expression

Test purpose

A CANCEL or BYE is sent after 182 was received

Ensure that the IUT, having received a 182 Queued provisional response to its INVITE request, to give up the call, sends a CANCEL/BYE request.

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	182 Queued
On hook	→	CANCEL/BYE
	←	200 OK CANCEL/BYE
	←	487 Request Terminated
	→	ACK

ISDN interworking SETUP ➔ ➔ INVITE 🡐 182 Queued RELEASE ➔ ➔ CANCEL/BYE RELEASE COMPLETE 🡐 🡐 200 OK CANCEL/BYE 🡐 487 Request Terminated ➔ ACK

TSS Orig_Orig_Release_initiated_by_the_terminating_user	TP_203_019	Reference section 15, 9.1 and Figure 5 of [IETF RFC 3261]	Selection expression
Test purpose *A CANCEL or BYE is sent after 183 was received* Ensure that the IUT, having received a 183 Session Progress provisional response to its INVITE request, to give up the call, sends a CANCEL/BYE request.			
Message flow **End device** **Test equipment** **Interworking POTS** Off hook Dial number ➔ INVITE 🡐 183 Session Progress On hook ➔ CANCEL/BYE 🡐 200 OK CANCEL/BYE 🡐 487 Request Terminated ➔ ACK **ISDN interworking** SETUP ➔ ➔ INVITE 🡐 183 Session Progress RELEASE ➔ ➔ CANCEL/BYE RELEASE COMPLETE 🡐 🡐 200 OK CANCEL/BYE 🡐 487 Request Terminated ➔ ACK			

TSS Orig_Orig_Release_initiated_by_the_terminating_user	TP_203_020	Reference section 15, 9.1 and Figure 5 of [IETF RFC 3261]	Selection expression
Test purpose *A CANCEL or BYE is sent a 200 OK INVITE was received at the same time* Ensure that the IUT, having received a Trying (100 Trying) response to its INVITE request, to give up the call having sent a CANCEL/BYE request, on receipt of a 2XX response to the original INVITE sends an ACK request.			
SIP header values			

Message flow **End device** — **Test equipment** **Interworking POTS** Off hook Dial number → INVITE ← 100 Trying On hook → CANCEL/BYE ← 200 OK INVITE → ACK → BYE ← 200 OK BYE **ISDN interworking** SETUP → → INVITE ← 100 Trying RELEASE → → CANCEL/BYE RELEASE COMPLETE ← ← 200 OK INVITE → ACK → BYE ← 200 OK BYE

7.2.2.3.2 Release initiated by the terminating user

7.2.2.3.2.1 SIP basic procedures

TSS Orig_Release_initiated_by_the_terminating_user	TP_204_001	Reference section 15, 9.1 and Figure 5 of [IETF RFC 3261]	Selection expression PICS 5.1.1/1
Test purpose *A BYE is received, a 200 OK BYE is sent* Ensure that the IUT, in the confirmed state when a BYE is received, a 200 OK BYE is sent. The connection is terminated.			
SIP header values			
Message flow **End device** — **Test equipment** Off hook Dial number → INVITE ← 407 Proxy Authentication Required → ACK → INVITE ← 180 Ringing Conversation ← 200 OK INVITE → ACK Terminated ← BYE → 200 OK BYE			

TSS Orig_Release_initiated_by_the_terminating_user	TP_204_002	Reference section 13.2.2 of [IETF RFC 3261]	Selection expression PICS 5.1.1/1
Test purpose *Receipt of a final response before a provisional response was received* Ensure that a final response as indicated in Table 7.2.2.3.2-1 is received after an early dialogue was established, a 180 Ringing was received and the SUT sends an ACK message.			
SIP header values			
Message flow			
End device		**Test equipment**	
Interworking POTS			
Off hook			
Dial number	→	INVITE	
	←	407 Proxy Authentication Required	
	→	ACK	
	→	INVITE	
	←	180 Ringing	
	←	SIP_FINAL_RESPONSE	
	→	ACK	

TSS Orig_Release_initiated_by_the_terminating_user	TP_204_003	Reference section 13.2.2 of [IETF RFC 3261]	Selection expression PICS 5.1.1/1
Test purpose *Receipt of a final response after a181 Call Is Being Forwarded provisional response was received* Ensure that a final response as indicated in Table 7.2.2.3.2-1 is received after an early dialogue was established, a 181 Call Is Being Forwarded was received and the SUT sends an ACK message.			
SIP header values			
Message flow			
End device		**Test equipment**	
Interworking POTS			
Off hook			
Dial number	→	INVITE	
	←	407 Proxy Authentication Required	
	→	ACK	
	→	INVITE	
	←	181 Call Is Being Forwarded	
	→	SIP_FINAL_RESPONSE	
	→	ACK	

TSS Orig_Release_initiated_by_the_terminating_user	TP_204_004	Reference section 13.2.2 of [IETF RFC 3261]	Selection expression PICS 5.1.1/1
Test purpose *Receipt of a final response after a 182 Queued provisional response was received* Ensure that a final response as indicated in Table 7.2.2.3.2-1 is received after an early dialogue was established, a 182 Queued was received and the SUT sends an ACK message.			
SIP header values			
Message flow **End device** — **Test equipment** **Interworking POTS** Off hook Dial number → INVITE ← 407 Proxy Authentication Required → ACK → INVITE ← 182 Queued → SIP_FINAL_RESPONSE → ACK			

TSS Orig_Release_initiated_by_the_terminating_user	TP_204_005	Reference section 13.2.2 of [IETF RFC 3261]	Selection expression PICS 5.1.1/1
Test purpose *Receipt of a final response after a 183 Session Progress provisional response was received* Ensure that a final response as indicated in Table 7.2.2.3.2-1 is received after an early dialogue was established, a 183 Session Progress was received, the SUT sends an ACK message.			
SIP header values			
Message flow **End device** — **Test equipment** **Interworking POTS** Off hook Dial number → INVITE ← 407 Proxy Authentication Required → ACK → INVITE ← 183 Session Progress → SIP_FINAL_RESPONSE → ACK			

Table 7.2.2.3.2-1 – 3xx/4xx/5xx/6xx final response

CAUSE_VA	←DISCONNECT (cause value)	←SIP_FINAL_RESPONSE
CAUSE_VA_01	127 (interworking unspecified)	400 Bad Request
CAUSE_VA_02	127 (interworking unspecified)	401 Unauthorized
CAUSE_VA_03	127 (interworking unspecified)	402 Payment Required
CAUSE_VA_04	127 (interworking unspecified)	403 Forbidden
CAUSE_VA_05	1 (Unallocated number)	404 Not Found
CAUSE_VA_06	127 (interworking unspecified)	405 Method Not Allowed
CAUSE_VA_07	127 (interworking unspecified)	406 Not Acceptable
CAUSE_VA_08	127 (interworking unspecified)	407 Proxy authentication required
CAUSE_VA_09	127 (interworking unspecified)	408 Request Timeout
CAUSE_VA_10	22 (Number changed)	410 Gone
CAUSE_VA_11	127 (interworking unspecified)	413 Request Entity too long
CAUSE_VA_12	127 (interworking unspecified)	414 Request-URI too long
CAUSE_VA_13	127 (interworking unspecified)	415 Unsupported Media type
CAUSE_VA_14	127 (interworking unspecified)	416 Unsupported URI scheme
CAUSE_VA_15	127 (interworking unspecified)	420 Bad Extension
CAUSE_VA_16	127 (interworking unspecified)	421 Extension required
CAUSE_VA_17	127 (interworking unspecified)	423 Interval Too Brief
CAUSE_VA_18	24 (call rejected due to ACR supplementary service)	433 Anonymity Disallowed
CAUSE_VA_19	20 Subscriber absent	480 Temporarily Unavailable
CAUSE_VA_20	127 (interworking unspecified)	481 Call/Transaction does not exist
CAUSE_VA_21	127 (interworking unspecified)	482 Loop detected
CAUSE_VA_22	127 (interworking unspecified)	483 Too many hops
CAUSE_VA_23	28 (Invalid Number format)	484 Address Incomplete
CAUSE_VA_24	127 (interworking unspecified)	485 Ambiguous
CAUSE_VA_25	17 (User busy)	486 Busy Here
CAUSE_VA_26	127 (Interworking unspecified) or not interworked. (Note 1)	487 Request terminated
CAUSE_VA_27	127 (interworking unspecified)	488 Not acceptable here
CAUSE_VA_28	127 (interworking unspecified)	493 Undecipherable
CAUSE_VA_29	127 (interworking unspecified)	500 Server Internal error
CAUSE_VA_30	127 (interworking unspecified)	501 Not implemented
CAUSE_VA_31	127 (interworking unspecified)	502 Bad Gateway
CAUSE_VA_32	127 (interworking unspecified)	503 Service Unavailable
CAUSE_VA_33	127 (interworking unspecified)	504 Server timeout
CAUSE_VA_34	127 (interworking unspecified)	505 Version not supported

Table 7.2.2.3.2-1 – 3xx/4xx/5xx/6xx final response

CAUSE_VA	←DISCONNECT (cause value)	←SIP_FINAL_RESPONSE
CAUSE_VA_35	127 (interworking unspecified)	513 Message too large
CAUSE_VA_36	127 (interworking unspecified)	580 Precondition failure
CAUSE_VA_37	17 (User busy)	600 Busy Everywhere
CAUSE_VA_38	21 (Call rejected)	603 Decline
CAUSE_VA_39	1 (unallocated number)	604 Does not exist anywhere
CAUSE_VA_40	127 (interworking unspecified)	606 Not acceptable

7.2.2.3.2.3 Test purposes for ISDN

TSS Orig_Release_initiated_by _the_terminating_user	TP_204_101	Reference subclause 5.1.1.4 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2

Test purpose

Receipt of BYE, a DSS1 DISCONNECT is sent cause value 16

Ensure that on receipt of a BYE request a DSS1 DISCONNECT is sent to the calling user equipment. The cause value of the DISCONNECT is set to '16'. 'The location of the cause IE is set to '1010' network beyond interworking point.

SIP header values

DSS1 Parameter values

DISCONNECT: Cause IE cause value=16

Message flow

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
		→	ACK
DISCONNECT	←	←	BYE
RELEASE	→	→	200 OK BYE
RELEASE COMPLETE	→		

TSS Orig_Release_initiated_by _the_terminating_user	TP_204_102	Reference subclause 5.1.1.4 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2

Test purpose

Receipt of a final response before a provisional response was received Reason header not included.

Ensure that when a final response is received before an early dialogue was established and if no Reason header is included,a DISCONNECT is sent to the calling user equipment. The case value of the DISCONNECT is derived according the mapping in Table 7.2.2.3.2-1. The location of the cause IE is set to '1010' network beyond interworking point.

SIP header values
DSS1 Parameter values DISCONNECT: Cause IE cause value = CAUSE_VA
Message flow

End device			**Test equipment**
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
DISCONNECT	←	←	SIP_FINAL_RESPONSE
RELEASE	→	→	ACK
RELEASE COMPLETE	→		

TSS Orig_Release_initiated_by _the_terminating_user	TP_204_103	Reference subclause 5.1.1.4 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2
Test purpose *Receipt of a final response after a 180 provisional response was received Reason header not included.* Ensure that when a final response is received after a 180 Ringing is received to establish an early dialogue and a no Reason header is included, a DISCONNECT is sent to the calling user equipment. The case value of the DISCONNECT is derived according to the mapping in Table 7.2.2.3.2-1. The location of the cause IE is set to '1010' network beyond interworking point.			
SIP header values			
DSS1 Parameter values DISCONNECT: Cause IE cause value = CAUSE_VA			
Message flow			

End device			**Test equipment**
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 Ringing
DISCONNECT	←	←	SIP_FINAL_RESPONSE
RELEASE	→	→	ACK
RELEASE COMPLETE	→		

TSS Orig_Release_initiated_by _the_terminating_user	TP_204_104	Reference subclause 5.1.1.4 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2
Test purpose *Receipt of a final response after a 181 provisional response was received Reason header not included.* Ensure that when a final response is received after a 181 Call Is Being Forwarded is received to establish an early dialogue and a no Reason header is included, a DISCONNECT is sent to the calling user equipment. The case value of the DISCONNECT is derived according to the mapping Table 7.2.2.3.2-1. The location of the cause IE is set to '1010' network beyond interworking point			
SIP header values			

DSS1 Parameter values DISCONNECT: Cause IE cause value = CAUSE_VA
Message flow

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
NOTIFY	←	←	181 Call Is Being Forwarded
DISCONNECT	←	←	SIP_FINAL_RESPONSE
RELEASE	→	→	ACK
RELEASE COMPLETE	→		

TSS Orig_Release_initiated_by_the_terminating_user	TP_204_105	Reference subclause 5.1.1.4 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2
Test purpose *Receipt of a final response after a 182 provisional response was received Reason header not included.* Ensure that when a final response is received after a 182 Queued is received to establish an early dialogue and no Reason header is included, a DISCONNECT is sent to the calling user equipment. The case value of the DISCONNECT is derived according to the mapping Table 7.2.2.3.2-1. The location of the cause IE is set to '1010' network beyond interworking point.			
SIP header values			
DSS1 Parameter values DISCONNECT: Cause IE cause value = CAUSE_VA			
Message flow			

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
NOTIFY/ CALL PROCEEDING	←	←	182 Queued
DISCONNECT	←	←	SIP_FINAL_RESPONSE
RELEASE	→	→	ACK
RELEASE COMPLETE	→		

TSS Orig_Release_initiated_by_the_terminating_user	TP_204_106	Reference subclause 5.1.1.4 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2
Test purpose *Receipt of a final response after a 183 provisional response was received Reason header not included.* Ensure that when a final response is received after a 183 Session Progress is received to establish an early dialogue and a no Reason header is included, a DISCONNECT is sent to the calling user equipment. The case value of the DISCONNECT is derived according to the mapping Table 7.2.2.3.2-1. The location of the cause IE is set to '1010' network beyond interworking point.			
SIP header values			

DSS1 Parameter values DISCONNECT: Cause IE cause value = CAUSE_VA
Message flow

End device			**Test equipment**
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
CALL PROCEEDING	←	←	183 Session Progress
DISCONNECT	←	←	SIP_FINAL_RESPONSE
RELEASE	→	→	ACK
RELEASE COMPLETE	→		

TSS Orig_Release_initiated_by _the_terminating_user	TP_204_107	Reference subclause 5.1.1.4 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2
Test purpose *DISCONNECT received, a BYE is sent* Ensure that the IUT, while the dialogue is in a confirmed state, of an ISDN DISCONNECT message sends a BYE request.			
SIP header values			
DSS1 Parameter values DISCONNECT: Cause IE cause value=<appropriate value>			
Message flow			

End device			**Test equipment**
Interworking POTS			
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
	←	←	
DISCONNECT	←	←	SIP_FINAL_RESPONSE
RELEASE	→	→	ACK
RELEASE COMPLETE	→		

7.2.2.4 Timers

TSS Orig_Timers	TP_205_001	Reference section 17.1.1.1 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/1
Test purpose *UDP: INVITE is repeated if timer T1 expires* Ensure that the IUT, when an INVITE client transaction is in the Calling state repeats its INVITE request on the timeout condition of timer A set with a value of T1 if an unreliable transport (UDP) is used.			

Message flow

End device				Test equipment
Interworking POTS				
Off hook		Start timer A	→	INVITE
		1*T1		
Dial number		Timeout timer A	→	INVITE
ISDN interworking				
SETUP	→	Start timer A	→	INVITE
		1*T1		
		Timeout timer A	→	INVITE

Apply post test routine

TSS Orig_Timers	TP_205_002	Reference section 17.1.1.1 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/2

Test purpose

TCP: INVITE is not repeated if timer T1 expires

Ensure that the IUT, when an INVITE client transaction is in the Calling state, does not repeat its INVITE request on the timeout condition of timer A set with a value of T1 if a reliable transport (TCP) is used.

Message flow

End device				Test equipment
Interworking POTS				
Off hook				
Dial number		Start timer A	→	INVITE
		1*T1		
		Timeout timer A		
ISDN interworking				
SETUP	→	Start timer A	→	INVITE
		1*T1		
		Timeout timer A		

Apply post test routine

TSS Orig_Timers	TP_205_003	Reference section 17.1.1.1 and Annex A/[IETF RFC 3261]	Selection expression PICS 5.1.2/1

Test purpose

UDP: INVITE is retransmitted if timer T1 expires repeatedly

Ensure that the IUT, when an INVITE client transaction is in the Calling state having already repeated its INVITE, wait for a timer A set with a value of 2*T1 before sending it again if an unreliable transport (UDP) is used.

Message flow
End device **Test equipment**
Interworking POTS
Off hook
Dial number — Start timer A 1*T1 ➔ INVITE
Timeout timer A Start timer A 2*T1 ➔ INVITE
Timeout timer A ➔ INVITE
ISDN interworking
SETUP ➔ Start timer A 1*T1 ➔ INVITE
Timeout timer A Start timer A 2*T1 ➔ INVITE
Timeout timer A ➔ INVITE
Apply post test routine

TSS Orig_Timers	TP_205_004	Reference section 17.1.1.1 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/1
Test purpose *UDP: the INVITE is retransmitted with intervals that double after each transmission* Ensure that the IUT, when an INVITE client transaction is in the Calling state, retransmits its INVITE request with intervals that double after each transmission if an unreliable transport (UDP) is used.			
Message flow **End device** **Test equipment** **Interworking POTS** Off hook Dial number — Start timer A 1*T1 ➔ INVITE Timeout timer A Start timer A 2*T1 ➔ INVITE Timeout timer A Start timer A 4*T1 ➔ INVITE Timeout timer A Start timer A 32*T1 ➔ INVITE Timeout timer A Start timer A 64*T1 ➔ INVITE			

ISDN interworking SETUP ➔ Start timer A 1*T1 ➔ INVITE Timeout timer A Start timer A 2*T1 ➔ INVITE Timeout timer A Start timer A 4*T1 ➔ INVITE Timeout timer A Start timer A 32*T1 ➔ INVITE Timeout timer A Start timer A 64*T1 ➔ INVITE **Apply post test routine**

TSS Orig_Timers	TP_205_005	Reference section 17.1.1.1 and Annex A of [IETF RFC 3261]	Selection expression
Test purpose *Timer B expires no ACK is sent* Ensure that the IUT, when an INVITE client transaction is in the Calling state, when timer B set to a value of 64*T1 expires, does not send an ACK.			
Message flow **End device** **Test equipment** **Interworking POTS** Off hook Dial number Start timer A Start timer B ➔ INVITE Timeout timer B 🡐 200 OK INVITE **ISDN interworking** SETUP ➔ Start timer A Start timer B ➔ INVITE Timeout timer B 🡐 200 OK INVITE			

TSS Orig_Timers	TP_205_006	Reference section 17.1.1.1 and Annex A/[IETF RFC 3261]	Selection expression
Test purpose *Timer B expires, Transaction is in terminated state* Ensure that the IUT, when an INVITE client transaction is in the Calling state, when timer B set to a value of 64*T1 expires, considers the transaction terminated.			

Message flow			
End device			**Test equipment**
Interworking POTS			
Off hook			
Dial number		Start timer A Start timer B	→ INVITE
		Timeout timer B	
			← BYE
			→ 481 Call/Transaction Does Not Exists
			→ ACK
ISDN interworking			
SETUP	→	Start timer A Start timer B	→ INVITE
		Timeout timer B	
			← BYE
			→ 481 Call/Transaction Does Not Exists
			→ ACK

TSS Orig_Timers	TP_205_007	Reference section 17.1.1.1 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/1
Test purpose *100 received, the INVITE is not repeated* Ensure that the IUT, when an INVITE client transaction is in the Proceeding state, does not repeat its INVITE request.			
Message flow			
End device			**Test equipment**
Interworking POTS			
Off hook			
Dial number			→ INVITE
			← 100 Trying
ISDN interworking	→		
SETUP			→ INVITE
			← 100 Trying
Apply post test routine			

TSS Orig_Timers	TP_205_008	Reference section 17.1.1.1 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/1

Test purpose

486 received, timer D is started ACK is repeated

Ensure that the IUT, when an INVITE client transaction is in the Completed state, on receipt of final responses that matches the transaction, still answer with an ACK request until timer D set to at least 32 second expires if an unreliable transport is used.

Message flow

End device				**Test equipment**
Interworking POTS				
Off hook				
Dial number			→	INVITE
		Start timer D	←	486 Busy Here
			→	ACK
			←	486 Busy Here
			→	ACK
ISDN interworking				
SETUP	→		→	INVITE
		Start timer D	←	486 Busy Here
			→	ACK
			←	486 Busy Here
			→	ACK

TSS Orig_Timers	TP_205_009	Reference section 17.1.1.1 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/2

Test purpose

486 received, timer D is started ACK is not repeated

Ensure that the IUT, when an INVITE client transaction is in the Completed state, on receipt of a final response that matches the transaction, does not repeat its ACK request if a reliable transport is used.

Message flow

End device				**Test equipment**
Interworking POTS				
Off hook				
Dial number			→	INVITE
		Start timer D	←	486 Busy Here
			→	ACK
			←	486 Busy Here

ISDN interworking

SETUP	→		→ INVITE
		Start timer D	← 486 Busy Here
			→ ACK
			← 486 Busy Here

TSS Orig_Timers	TP_205_010	Reference section 17.1.1.1 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/1

Test purpose

486 repeated Via with different values, ACK is not sent

Ensure that the IUT, when an INVITE client transaction is in the Completed state, on receipt of new final responses with different Via branch parameter value, does not repeat its ACK request until timer D set to at least 32 second expires if an unreliable transport is used.

SIP header values

486 1

Via: <any via URI>;branch=<any branch value 1>

486 2

Via: <any via URI>;branch=<any branch value 2>

Message flow

End device			**Test equipment**
Interworking POTS			
Off hook			
Dial number			→ INVITE
		Start timer D	← 486 Busy Here 1
			→ ACK
			← 486 Busy Here 2
ISDN interworking			
SETUP	→		→ INVITE
		Start timer D	← 486 Busy Here 1
			→ ACK
			← 486 Busy Here 2

Apply post test routine

TSS Orig_Timers	TP_205_011	Reference section 13.2.2.4 and Annex A of [IETF RFC 3261]	Selection expression

Test purpose

200 OK received, ACK is sent

Ensure that the IUT, when an INVITE client transaction has been in the Terminated state, on receipt of a retransmitted Success (200 OK) response sends an ACK request until 64*T1 duration expires.

Message flow

End device			**Test equipment**
Interworking POTS			
Off hook			
Dial number		→	INVITE
	Start timer 64*T1	←	200 OK INVITE
		→	ACK
		←	200 OK INVITE
		→	ACK
ISDN interworking			
SETUP	→	→	INVITE
	Start timer 64*T1	←	200 OK INVITE
		→	ACK
		←	200 OK INVITE
		→	ACK

Apply post test routine

TSS Orig_Timers	TP_205_012	Reference section 13.2.2.4 and Annex A of [IETF RFC 3261]	Selection expression

Test purpose

*Timeout 64*T1 ACK is not repeated*

Ensure that the IUT, when an INVITE client transaction has been in the Terminated state, after 64*T1 duration expires, on receipt of a retransmitted Success (200 OK) responsedoes not send an ACK request.

Message flow

End device			**Test equipment**
Interworking POTS			
Off hook			
Dial number		→	INVITE
	Start timer 64*T1	←	200 OK INVITE
		→	ACK
	Timeout 64*T1		
		←	200 OK INVITE

ISDN interworking			
SETUP	→		→ INVITE
		Start timer 64*T1	← 200 OK INVITE
			→ ACK
		Timeout 64*T1	
			← 200 OK INVITE
		Apply post test routine	

TSS Orig_Timers	TP_205_013	Reference section 17.1.2.2 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/1

Test purpose

BYE was sent start timer E; on timeout timer E the BYE is repeated

Ensure that the IUT, having sent a BYE request on an established dialog, repeats its request after timer E set to T1 value expires if an unreliable transport is used.

Message flow

End device			**Test equipment**
Interworking POTS			
Off hook			
Dial number			→ INVITE
			← 180 Ringing
			← 200 OK INVITE
			→ ACK
On hook		Start timer E	→ BYE
		Timeout timer E	→ BYE
ISDN interworking			
SETUP	→		→ INVITE
ALERTING	←		← 180 Ringing
CONNECT	←		← 200 OK INVITE
			→ ACK
DISCONNECT	→	Start timer E	→ BYE
		Timeout timer E	→ BYE

TSS Orig_Timers	TP_205_014	Reference section 17.1.2.2 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/1

Test purpose

BYE is repeated after timeout 2 x timer E

Ensure that the IUT, having twicetransmitted a BYE request on an established dialogue, repeats its request after timer E set to the MIN(2*T1,T2) value expires if an unreliable transport is used.

SIP header values

Message flow

End device				Test equipment
Interworking POTS				
Off hook				
Dial number			→	INVITE
			←	180 Ringing
			←	200 OK INVITE
			→	ACK
On hook		Start timer E	→	BYE
		Timeout timer E Start timer E 2*T1 +T2	→	BYE
		Timeout timer E	→	BYE
ISDN interworking				
SETUP	→		→	INVITE
ALERTING	←		←	180 Ringing
CONNECT	←		←	200 OK INVITE
			→	ACK
DISCONNECT	→	Start timer E	→	BYE
		Timeout timer E Start timer E 2*T1 +T2	→	BYE
		Timeout timer E	→	BYE

TSS Orig_Timers	TP_205_015	Reference section 17.1.2.2 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/1

Test purpose

BYE is repeated after timeout 3 x timer E

Ensure that the IUT, having transmitted three times a BYE request on an established dialogue, repeats its request after timer E set to the MIN(4*T1,T2) value expires, if an unreliable transport is used.

SIP header values

Message flow

End device			Test equipment
Interworking POTS			
Off hook			
Dial number		→	INVITE
		←	180 Ringing
		←	200 OK INVITE
		→	ACK
On hook	Start timer E	→	BYE
	Timeout timer E Start timer E 2*T1 +T2	→	BYE
	Timeout timer E Start timer E 4*T1+T2	→	BYE
	Timeout timer E	→	BYE

ISDN interworking			
SETUP	→		→ INVITE
ALERTING	←		← 180 Ringing
CONNECT	←		← 200 OK INVITE
			→ ACK
DISCONNECT	→	Start timer E	→ BYE
		Timeout timer E Start timer E 2*T1 +T2	→ BYE
		Timeout timer E Start timer E 4*T1+T2	→ BYE
		Timeout timer E	→ BYE
		Apply post test routine	

TSS Orig_Timers	TP_205_016	Reference section 17.1.2.2 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/1
Test purpose *Timeout timer F, the BYE is not repeated* Ensure that the IUT does not repeat a BYE request on an established dialogue after timer F set to 64*T1 expires, if an unreliable transport is used.			
SIP header values			
Message flow			
End device			**Test equipment**
Interworking POTS			
Off hook			
Dial number			→ INVITE
			← 180 Ringing
			← 200 OK INVITE
			→ ACK
On hook		Start timer F	→ BYE
		Timeout timer F	
ISDN interworking			
SETUP	→		→ INVITE
ALERTING	←		← 180 Ringing
CONNECT	←		← 200 OK INVITE
			→ ACK
DISCONNECT	→	Start timer F	→ BYE
		Timeout timer F	

TSS Orig_Timers	TP_205_017	Reference section 17.1.2.2 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.2/18

Test purpose

BYE is sent and timer E is started; on timeout timer E, the BYE is repeated

Ensure that the IUT, when a BYE client transaction is in the **Proceeding state**, repeats its BYE request after timer E set to T1 value expires.

SIP header values

Message flow

End device				**Test equipment**
Interworking POTS				
Off hook				
Dial number			→	INVITE
			←	180 Ringing
On hook		Start timer E	→	BYE
		Timeout timer E	→	BYE
ISDN interworking				
SETUP	→		→	INVITE
ALERTING	←		←	180 Ringing
DISCONNECT	→	Start timer E	→	BYE
		Timeout timer E	→	BYE

TSS Orig_Timers	TP_205_018	Reference section 17.1.2.2 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.2/18

Test purpose

BYE is repeated after timeout 2 x timer E

Ensure that the IUT, when a BYE client transaction is in the **Proceeding state** and BYE request have been already repeated in this state, repeats its BYE request after timer E set to T2 value expires.

SIP header values

Message flow

End device				**Test equipment**
Interworking POTS				
Off hook				
Dial number			→	INVITE
			←	180 Ringing
On hook		Start timer E	→	BYE
		Timeout timer E Start timer E T2	→	BYE
		Timeout timer E	→	BYE
ISDN interworking				
SETUP	→		→	INVITE
ALERTING	←	Start timer E	→	BYE
DISCONNECT	→	Timeout timer E Start timer E T2	→	BYE
		Timeout timer E	→	BYE

TSS Orig_Timers	TP_205_019	Reference section 17.1.2.2 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.2/18

Test purpose

BYE is nit repeated after timeout timer F

Ensure that the IUT, when a BYE client transaction is in the **Proceeding state**, does not repeat a BYE request on an established dialogue, after timer F set to 64*T1 expires.

SIP header values

Message flow

End device				Test equipment
Interworking POTS				
Off hook				
Dial number			→	INVITE
			←	180 Ringing
On hook		Start timer F	→	BYE
		Timeout timer F		
ISDN interworking				
SETUP	→		→	INVITE
ALERTING	←		←	180 Ringing
DISCONNECT	→	Start timer F	→	BYE
		Timeout timer F		

TSS Orig_Timers	TP_205_020	Reference section 17.1.2.2 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.2/18

Test purpose

Timeout timer F, the BYE is not repeated

Ensure that the IUT, when a BYE client transaction is in the **Trying state**, considers the transaction terminated after timer F set to 64*T1 duration expires without receiving any final response.

SIP header values

Message flow

End device			Test equipment
Interworking POTS			
Off hook			
Dial number		→	INVITE
On hook	Start timer F	→	BYE
	Timeout timer F		

ISDN interworking			
SETUP	→		→ INVITE
RELEASE	→	Start timer F	→ BYE
		Timeout timer F	

7.2.2.5 Abnormal situations

TSS Orig_Abnormal_situations	TP_206_001	Reference sections 13.2.2.3, 17.1.1.2, 17.1.1.3 and Figure 5, of [IETF RFC 3261]	Selection expression
Test purpose *404 received in calling state; setting of Call-ID, From headers and Request-URI* Ensure that when the client transaction is in the **Calling state** that on receipt of a Not Found (404 Not Found) response, an ACK request is sent with the same Call-ID, From headers and Request-URI as in the original INVITE request and the same Tag in the To header as in this response.			
SIP header values INVITE 1 sip: [Request URI] SIP/2.0 Call-ID: <any invite Call ID> From: <any invite URI>;tag=<any invite tag> ACK 1 sip: [Request URI] SIP/2.0 Call-ID: <any invite Call ID> From: <any invite URI>;tag=<any invite tag>			
Message flow			

End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE 1
Busy tone	←	404 Not Found
On hook	→	ACK 1
ISDN interworking		
SETUP	→	INVITE 1
DISCONNECT	←	404 Not Found 1
RELEASE	→	ACK 1
RELEASE COMPLETE	←	

TSS Orig_Abnormal_situations	TP_206_002	Reference sections 13.2.2.3, 17.1.1.2, 17.1.1.3 and Figure 5 of [IETF RFC 3261]	Selection expression
Test purpose *404 received in proceeding state; setting of Call-ID, From headers and Request-URI* Ensure that when the client transaction is in the **Proceeding state**, that on receipt of a Not Found (404 Not Found) response,an ACK request is sent with the same Call-ID, From headers and Request-URI as in the original INVITE request and the same Tag in the To header as in this response.			

SIP header values

INVITE 1 sip: [Request URI] SIP/2.0
    Call-ID: <any invite Call ID>
    From: <any invite URI>;tag=<any invite tag>

404 1
    To: <any 404 URI>;tag=<any 404 tag>

ACK 1 sip: [Request URI] SIP/2.0
    Call-ID: <any invite Call ID>
    From: <any invite URI>;tag=<any invite tag>
    To: <any 404 URI>;tag=<any 404 tag>

Message flow

End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE 1
	←	183 Session Progress
Busy tone	←	404 Not Found
On hook	→	ACK 1
ISDN interworking		
SETUP	→	INVITE 1
CALL PROCEEDING	←	183 Session Progress
DISCONNECT	←	404 Not Found 1
RELEASE	→	ACK 1
RELEASE COMPLETE	←	

TSS Orig_Abnormal_situations	TP_206_003	Reference sections 13.2.2.3, 17.1.1.2, 17.1.1.3 and Figure 5 of [IETF RFC 3261]	Selection expression

Test purpose

410 received in calling state; setting of Call-ID, From headers and Request-URI

Ensure that when the client transaction is in the **Calling state** that on receipt of a Gone (410 Gone) response, an ACK request is sent with the same Call-ID, From headers and Request-URI as in the original INVITE request and the same Tag in the To header as in this response.

SIP header values

INVITE 1 sip: [Request URI] SIP/2.0
    Call-ID: <any invite Call ID>
    From: <any invite URI>;tag=<any invite tag>

410 1
    To: <any 410 URI>;tag=<any 410 tag>

ACK 1 sip: [Request URI] SIP/2.0
    Call-ID: <any invite Call ID>
    From: <any invite URI>;tag=<any invite tag>
    To: <any 410 URI>;tag=<any 410 tag>

Message flow		
End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE 1
	←	410 Gone 1
	→	ACK 1
ISDN interworking		
SETUP	→	INVITE 1
DISCONNECT	←	410 Gone 1
RELEASE	→	ACK 1
RELEASE COMPLETE	←	
Apply post test routine		

TSS Orig_Abnormal_situations	TP_206_004	Reference sections 13.2.2.3, 17.1.1.2, 17.1.1.3 and Figure 5 of [IETF RFC 3261]	Selection expression
Test purpose *480 received in calling state; setting of Call-ID, From headers and Request-URI* Ensure that when the client transaction is in the **Calling state** that on receipt of a Temporarily Unavailable (480 Temporarily Unavailable) response, an ACK request is sent with the same Call-ID, From headers and Request-URI as in the original INVITE request and the same Tag in the To header as in this response.			
SIP header values INVITE 1 sip: [Request URI] SIP/2.0 Call-ID: <any invite Call ID> From: <any invite URI>;tag=<any invite tag> 480 1 To: <any 480 URI>;tag=<any 480 tag> ACK 1 sip: [Request URI] SIP/2.0 Call-ID: <any invite Call ID> From: <any invite URI>;tag=<any invite tag> To: <any 480 URI>;tag=<any 480 tag>			

Message flow		
End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE 1
	←	480 Temporarily Unavailable
	→	ACK 1
ISDN interworking		
SETUP	→	INVITE 1
DISCONNECT	←	480 Temporarily Unavailable
RELEASE	→	ACK 1
RELEASE COMPLETE	←	

TSS Orig_Abnormal_situations	TP_206_005	Reference sections 13.2.2.3, 17.1.1.2, 17.1.1.3 and Figure 5 of [IETF RFC 3261]	Selection expression

Test purpose

486 received in calling state; setting of Call-ID, From headers and Request-URI

Ensure that when the client transaction is in the **Calling state** that, on receipt of a Busy Here (486 Busy Here) response an ACK request is sent with the same Call-ID, From headers and Request-URI as in the original INVITE request and the same Tag in the To header as in this response.

SIP header values

INVITE 1 sip: [Request URI] SIP/2.0
  Call-ID: <any invite Call ID>
  From: <any invite URI>;tag=<any invite tag>

486 1
  To: <any 486 URI>;tag=<any 486 tag>

ACK 1 sip: [Request URI] SIP/2.0
  Call-ID: <any invite Call ID>
  From: <any invite URI>;tag=<any invite tag>
  To: <any 486 URI>;tag=<any 486 tag>

Message flow

End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE 1
	←	486 Busy Here
	→	ACK 1
ISDN interworking		
SETUP	→	INVITE 1
DISCONNECT	←	486 Busy Here
RELEASE	→	ACK 1
RELEASE COMPLETE	←	

TSS Orig_Abnormal_situations	TP_206_006	Reference sections 13.2.2.3, 17.1.1.2, 17.1.1.3 and Figure 5 of [IETF RFC 3261]	Selection expression

Test purpose

500 received in calling state; setting of Call-ID, From headers and Request-URI

Ensure that when the client transaction is in the **Calling state** and that on receipt of a Server Internal Error (500 Server Internal Error) response, an ACK request is sent with the same Call-ID, From headers and Request-URI as in the original INVITE request, and the same Tag in the To header as in this response.

SIP header values

INVITE 1 sip: [Request URI] SIP/2.0
  Call-ID: <any invite Call ID>
  From: <any invite URI>;tag=<any invite tag>

500 1
  To: <any 500 URI>;tag=<any 500 tag>

ACK 1 sip: [Request URI] SIP/2.0
  Call-ID: <any invite Call ID>
  From: <any invite URI>;tag=<any invite tag>
  To: <any 500 URI>;tag=<any 500 tag>

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE 1
	←	500 Server Internal Error
	→	ACK 1
ISDN interworking		
SETUP	→	INVITE 1
DISCONNECT	←	500 Server Internal Error
RELEASE	→	ACK 1
RELEASE COMPLETE	←	

TSS	TP_206_007	Reference	Selection expression
Orig_Abnormal_situations		sections 13.2.2.3, 17.1.1.2, 17.1.1.3 and Figure 5 of [IETF RFC 3261]	

Test purpose

600 and 500 with different branch parameter value received in calling state; only one ACK is sent

Ensure that when the client transaction is in the **Calling state,** that on receipt of a Busy Everywhere (600 Busy Everywhere) and a Server Internal Error (500 Server Internal Error) responses with different branch parameter value on the top Via header, send only one ACK request.

SIP header values

600
  Via: SIP 2.0 <transport><any via URI >; branch=<any branch value 1>

500
  Via: SIP 2.0 <transport><any via URI >; branch=<any branch value 2>

ACK 1 sip: [Request URI] SIP/2.0

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE 1
	←	600 Busy Everywhere
	←	500 Server Internal Error
	→	ACK 1
ISDN interworking		
SETUP	→	INVITE 1
DISCONNECT	←	600 Busy Everywhere
	←	500 Server Internal Error
RELEASE	→	ACK 1
RELEASE COMPLETE	←	

TSS	TP_206_008	Reference	Selection expression
Orig_Abnormal_situations		sections 13.2.2.3, 17.1.1.2, 17.1.1.3 and Figure 5 of [IETF RFC 3261]	

Test purpose

603 received an ACK is sent

Ensure that when the client transaction is in the **Calling state,** that on receipt of a Decline (603 Decline) response an ACK request is sent with the same Call-ID, From headers and Request-URI as in the original INVITE request and the same Tag in the To header as in this response.

SIP header values

INVITE 1 sip: [Request URI] SIP/2.0
  Call-ID: <any invite Call ID>
  From: <any invite URI>;tag=<any invite tag>

603
  To: <any 603 URI>;tag=<any 603 tag>

ACK 1 sip: [Request URI] SIP/2.0
  Call-ID: <any invite Call ID>
  From: <any invite URI>;tag=<any invite tag>
  To: <any 603 URI>;tag=<any 603 tag>

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE 1
	←	603 Decline
	→	ACK 1
ISDN interworking		
SETUP	→	INVITE 1
DISCONNECT	←	603 Decline
RELEASE	→	ACK 1
RELEASE COMPLETE	←	

<table>
<tr><td>TSS
Orig_Abnormal_situations</td><td>TP_206_009</td><td>Reference
sections 13.2.2.3, 17.1.1.2, 17.1.1.3 and Figure 5 of [IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
603 received. The same branch parameter value is sent in the ACK as received in the initial INVITE request

Ensure that when the client transaction is in the Calling state, that on receipt of a Busy Everywhere (600 Busy Everywhere) and a Server Internal Error (500 Server Internal Error) responses with different branch parameter value on the top Via header, send only one ACK request.</td></tr>
<tr><td colspan="4">SIP header values
INVITE:
Via: SIP 2.0 <transport><any via URI >; branch=<any invite branch value>

ACK:
Via: SIP 2.0 <transport><any via URI >; branch=< any invite branch value ></td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><td>End device</td><td></td><td>Test equipment</td></tr>
<tr><td>Interworking POTS</td><td></td><td></td></tr>
<tr><td>Off hook</td><td></td><td></td></tr>
<tr><td>Dial number</td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td>←</td><td>603 Decline</td></tr>
<tr><td></td><td>→</td><td>ACK</td></tr>
<tr><td>ISDN interworking</td><td></td><td></td></tr>
<tr><td>SETUP</td><td>→</td><td>INVITE</td></tr>
<tr><td>DISCONNECT</td><td>←</td><td>603 Decline</td></tr>
<tr><td>RELEASE</td><td>→</td><td>ACK</td></tr>
<tr><td>RELEASE COMPLETE</td><td>←</td><td></td></tr>
</table></td></tr>
</table>

<table>
<tr><td>TSS
Orig_Abnormal_situations</td><td>TP_206_010</td><td>Reference
sections 8.1.3.2, 13.2.2.3, 17.1.1.2, 17.1.1.3 and Figure 5 of [IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
Unknown unsuccessful final response received.

Ensure that when the client transaction is in the Calling state, that on receipt of an Unknown (699 Unknown) response, an ACK request is sent with the same Call-ID, From headers and Request-URI as in the original INVITE request and the same Tag in the To header as in this response.</td></tr>
</table>

SIP header values INVITE: INVITE sip: [invite Request URI] SIP/2.0 Call-ID: <any invite Call ID> From: <any invite URI>;tag=<any invite tag> 699: To: <any URI>;tag=<any 699 tag value> ACK: INVITE sip: [invite Request URI] SIP/2.0 Call-ID: <any invite Call ID> From: <any invite URI>;tag=<any invite tag> To: <any URI>;tag=<any 699 tag value>

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE 1
	←	699 Unknown
	→	ACK 1
ISDN interworking		
SETUP	→	INVITE 1
DISCONNECT	←	699 Unknown
RELEASE	→	ACK 1
RELEASE COMPLETE	←	

TSS Orig_Abnormal_situations	TP_206_011	Reference sections 12.2.1.1, 13.2.2.3, 17.1.1.2, 17.1.1.3 and Figure 5 of [IETF RFC 3261]	Selection expression

Test purpose

Not acceptable SDP answer. Call is released.

Ensure when the IUT is establishing a call on receipt of in 2XX a not acceptable session description, sends an ACK request immediately followed by a BYE request.

SIP header values

200 OK
SDP
'm' line <not acceptable codec>
'a' line

Message flow		
End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	200 OK INVITE
	→	ACK
	→	BYE
	←	200 OK BYE
ISDN interworking		
SETUP	→	INVITE
CONNECT	←	200 OK INVITE
	→	ACK
DISCONNECT	→	BYE
RELEASE	←	200 OK BYE
RELEASE COMPLETE	←	

TSS Orig_Abnormal_situations	TP_206_012	Reference section 13.2.1 of [IETF RFC 3261]	Selection expression

Test purpose

Route header in the ACK is sent with the address of the last received Record-Route header.

The IUT having already received a 2XX final response to its INVITE request, ensure that on receipt of a Success (200 OK) response, with a different Record-Route as in previous response, but with the same Via branch parameter and CSeq header method as in the INVITE request, sends an ACK request with a Route header set according to this new Record-Route.

SIP header values

180:

Record-Route: <sip: record_route_180;lr>

200:

Record-Route: <sip: record_route_200;lr>

ACK:

Route: <sip: record_route_200;lr>

Message flow		
End device		**Test equipment**
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	180 Ringing
	←	200 OK INVITE
	→	ACK

Apply post test routine

ISDN interworking		
SETUP	→	INVITE
ALERTING		180 Ringing
CONNECT	←	200 OK INVITE
	→	ACK
	Apply post test routine	

7.2.3 Call initiation – UE terminating case

7.2.3.1 Establishment of an early dialogue

TSS Term_Establishment_of_an_early_dialogue	TP_301_001	Reference sections 8, 8.2, and 13.3.1.1 of [IETF RFC 3261]	Selection expression
Test purpose *The IUT sends a 180 Ringing* Ensure that the IUT on receipt of an INVITE request, sends a provisional (180 Ringing) response.			

Message flow

Test equipment				**End device**
Interworking POTS				
INVITE	→			
180 Ringing	←			Ringing
ISDN interworking				
INVITE	→		→	SETUP
180 Ringing	←		←	ALERTING
		Apply post test routine		

TSS Term_Establishment_of_an_early_dialogue	TP_301_002	Reference sections 8, 8.2, and 13.3.1.1 of [IETF RFC 3261]	Selection expression
Test purpose *The IUT sends a 200 OK INVITE* Ensure that the IUT on receipt of an INVITE request, sends a Success (200 OK) response.			

Message flow

Test equipment				**End device**
Interworking POTS				
INVITE	→			Ringing
180 Ringing	←			
200 OK INVITE	←			Off hook
ACK	→			
ISDN interworking				
INVITE	→		→	SETUP
200 OK INVITE	←		←	CONNECT
ACK	→			
		Apply post test routine		

TSS Term_Establishment_of_ an_early_dialogue	TP_301_003	Reference section 8 and 8.2.6.1 of [IETF RFC 3261]	Selection expression
Test purpose *The Timestamp header value in the response is increased* Ensure that the IUT on receipt of an INVITE request with a Timestamp header, when it answers with a provisional response Trying (100 Trying), set a Timestamp header with an increased value of the received Timestamp in its response.			
SIP header values INVITE: Timestamp: <timestamp_value> 100: Timestamp: <timestamp_value + any value>			
Message flow			
Test equipment		**End device**	
Interworking POTS			
INVITE	➔	Ringing	
100 Trying	←		
ISDN interworking			
INVITE	➔	➔ SETUP	
100 Trying	←		
Apply post test routine			

TSS Term_Establishment_of_ an_early_dialogue	TP_301_004	Reference section 13.2.1 and 13.3.1 of [IETF RFC 3261]	Selection expression
Test purpose *Initial offer in the 200 OK INVITE, initial answer in the ACK* Ensure that the IUT on receipt of an INVITE request including no message body, includes in its first 2xx response an initial offer session description.			
SIP header values 200 OK: SDP m – line offer a – line offer ACK SDP m – line answer a – line answer			

Message flow
Test equipment   **End device**
Interworking POTS
INVITE → Ringing
200 OK INVITE ← On hook
ACK →
ISDN interworking
INVITE → → SETUP
200 OK INVITE ← ← CONNECT
ACK →
Apply post test routine

TSS Term_Establishment_of_an_early_dialogue	TP_301_005	Reference section 13.2.1 and 13.3.1 of [IETF RFC 3261]	Selection expression
Test purpose *Initial offer in the initial INVITE, initial answer in the 200 OK INVITE* Ensure that the IUT on receipt of an INVITE request including an initial offer session description in its message body, includes the answer in its first 2xx response in a session description.			
SIP header values INVITE: SDP m – line offer a – line offer 200 OK: SDP m – line answer a – line answer			
Message flow **Test equipment** **End device** **Interworking POTS** INVITE → Ringing 200 OK INVITE ← On hook ACK → **ISDN interworking** INVITE → → SETUP 200 OK INVITE ← ← CONNECT ACK → **Apply post test routine**			

<table>
<tr><td>TSS
Term_Establishment_of_
an_early_dialogue</td><td>TP_301_006</td><td>Reference
section 8.2.6.2 of
[IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
From, Call-ID, CSeq and Via headers copy from the INVITE

Ensure that the IUT on receipt of an INVITE request, sends a provisional (101-199) response including the headers From, Call-ID, CSeq and Via headers copy from the INVITE request.</td></tr>
<tr><td colspan="4">SIP header values
INVITE:
From: <from_value_invite>
Call-ID: <callid_value_invite>
CSeq: <cseq_value_invite>
Via: <via_value_invite>

180:
From: <from_value_invite>
Call-ID: <callid_value_invite>
CSeq: <cseq_value_invite>
Via: <via_value_invite></td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><td>Test equipment</td><td></td><td></td><td>End device</td></tr>
<tr><td>Interworking POTS</td><td></td><td></td><td></td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>Ringing</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td></td></tr>
<tr><td colspan="4">Apply post test routine</td></tr>
<tr><td>ISDN interworking</td><td></td><td></td><td></td></tr>
<tr><td>INVITE</td><td>→</td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td>←</td><td>ALERTING</td></tr>
<tr><td colspan="4">Apply post test routine</td></tr>
</table></td></tr>
</table>

<table>
<tr><td>TSS
Term_Establishment_of_
an_early_dialogue</td><td>TP_301_007</td><td>Reference
sections 8.2.6.2, 12.2.2 and
13.3.1 of [IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
To tag is sent in the response

Ensure that the IUT on receipt of an INVITE request with no TAG set on the To header, sends a provisional (101-199) response including the same URI and an additional TAG for the To header.</td></tr>
<tr><td colspan="4">SIP header values
INVITE:
To: <sip:<i>to-uri</i>>

180:
To: <sip:<i>to-uri</i>>;tag=to_tag</td></tr>
</table>

Message flow			
Test equipment			**End device**
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
	Apply post test routine		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
	Apply post test routine		

TSS Term_Establishment_of_an_early_dialogue	TP_301_008	Reference section 8.2.6.2 of [IETF RFC 3261]	Selection expression
Test purpose *To tag in the INVITE request* Ensure that the IUT on receipt of an INVITE request with a TAG set on the To header, either: • sends a provisional (101-199) response including the same URI and the same TAG for the To header (recommended for robustness), • or reject the INVITE request with a Call/Transaction does not exist (481 Call/Transaction does not exist).			
SIP header values INVITE: To: <sip:*to_uri_value*>;tag=*to_tag_value* 180: To: <sip:*to_uri_value*>;tag=*to_tag_value*			

Message flow			
Test equipment			**End device**
Interworking POTS			
INVITE	→		Ringing
CASE A			
180 Ringing	←		
	Apply post test routine		
CASE B			
481 Call/Transaction does not exist	←		
ACK	→		
ISDN interworking			
INVITE	→	→	SETUP
CASE A			
180 Ringing	←	←	ALERTING
	Apply post test routine		
CASE B			
481 Call/Transaction does not exist	←		
ACK	→		

<table>
<tr><td>TSS
Term_Establishment_of_
an_early_dialogue</td><td>TP_301_009</td><td>Reference
section 12.1.1 of
[IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
Contact header in the response.

Ensure that the IUT on receipt of an INVITE request, sends a provisional (101-199) response including a single Contact header.</td></tr>
<tr><td colspan="4">SIP header values
180:
    Contact: <sip: <i>contact_value</i>></td></tr>
<tr><td colspan="4">Message flow
Test equipment   End device
Interworking POTS
INVITE → Ringing
180 Ringing ←
Apply post test routine

ISDN interworking
INVITE → → SETUP
180 Ringing ← ← ALERTING
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Term_Establishment_of_
an_early_dialogue</td><td>TP_301_010</td><td>Reference
section 12.1.1 of
[IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
Record-Route header copied from the INVITE request into the response

Ensure that the IUT on receipt of an INVITE request including a Record-Route header, sends a provisional (101-199) response including a Record-Route header copy from the INVITE request, in the same order.</td></tr>
<tr><td colspan="4">SIP header values
INVITE:
    Record-Route: <sip:<i>invite_record_route</i>>;lr

180:
    Record-Route: <sip:<i>invite_record_route</i>>;lr</td></tr>
<tr><td colspan="4">Message flow
Test equipment   End device
Interworking POTS
INVITE → Ringing
180 Ringing ←
Apply post test routine

ISDN interworking
INVITE → → SETUP
180 Ringing ← ← ALERTING
Apply post test routine</td></tr>
</table>

TSS Term_Establishment_of_ an_early_dialogue	TP_301_011	Reference section 12.1.1of [IETF RFC 3261]	Selection expression

Test purpose

Record-Route header with unknown parameter

Ensure that the IUT on receipt of an INVITE request including a Record-Route header with parameters that it does not understand, sends a provisional (101-199) response including a Record-Route header copy from the INVITE request, with the unknown parameters.

SIP header values

INVITE:

Record-Route: <sip:*record-route_value_invite*;unknown=etsi>

180:

Record-Route: <sip:record-route_value_invite;unknown=etsi>

Message flow

Test equipment			**End device**
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
	Apply post test routine		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
	Apply post test routine		

TSS Term_Establishment_of_ an_early_dialogue	TP_301_012	Reference section 12.1.1 of [IETF RFC 3261]	Selection expression

Test purpose

From header without "tag"

Ensure that the IUT on receipt of an INVITE request including From header without tag, sends a provisional (101-199) response including a From header without tag.

SIP header values

INVITE:

From: <sip:*from_value_invite*>

180:

From: <sip:*from_value_invite*>

Message flow

Test equipment			**End device**
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
	Apply post test routine		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
	Apply post test routine		

TSS Term_Establishment_of_an_early_dialogue	TP_301_013	Reference section 17.2.1 and 17.2.3 of [IETF RFC 3261]	Selection expression

Test purpose

Additional identical INVITE received, the previous sent response is repeated

Ensure that the IUT when a server INVITE transaction is in the Proceeding state, on receipt of an INVITE request, including a Via header set with the same branch parameter and sent-by value in the topmost list value, repeats its last response.

SIP header values

INVITE1:

Via: *any_invite1_via_value*;branch=*any_invite1_branch_value*
From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value*
To: *any_invite1_to_value*
Call-ID: *any_invite1_call-id_value*
CSeq: *any_invite1_cseq*

INVITE2:

Via: *any_invite1_via_value*;branch=*any_invite1_branch_value*
From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value*
To: *any_invite1_to_value*
Call-ID: *any_invite1_call-id_value*
CSeq: *any_invite1_cseq*

Message flow

Test equipment			End device
Interworking POTS			
INVITE1	→		Ringing
180 Ringing	←		
INVITE2	→		
180 Ringing	←		
	Apply post test routine		
ISDN interworking			
INVITE1	→	→	SETUP
180 Ringing	←	←	ALERTING
INVITE2	→		
180 Ringing	←		
	Apply post test routine		

TSS Term_Establishment_of_an_early_dialogue	TP_301_014	Reference section 17.2.1 and 17.2.3 of [IETF RFC 3261]	Selection expression

Test purpose

Additional identical INVITE and Via header without branch parameter received, the previous sent response is repeated

Ensure that the IUT when a server INVITE transaction is in the Proceeding state, on receipt of an INVITE request, including a Via header set with no branch parameter but with the Request-URI, To tag, From tag, Call-ID, CSeq and top Via identical as in the first INVITE request, repeats its last response.

SIP header values INVITE1: Via: *any_invite1_via_value*;branch= *z9hG4bK any_invite1_branch_value* From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq* INVITE2: Via: *any_invite1_via_value*;branch= z9hG4bK_*any_invite1_branch_value* **Via: *any_invite2_via_value*** From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq*
Message flow

Test equipment			**End device**
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
INVITE2	→		
180 Ringing	←		
	Apply post test routine		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
INVITE2	→		
180 Ringing	←		
	Apply post test routine		

TSS Term_Establishment_of_an_early_dialogue	TP_301_015	Reference section 17.2.1 and 17.2.3 of [IETF RFC 3261]	Selection expression
Test purpose *Additional identical INVITE and Via header without magic cookie in the branch parameter received, the previous sent response is repeated* Ensure that the IUT when a server INVITE transaction is in the Proceeding state, on receipt of an INVITE request, including a Via header set with a branch parameter without the magic cookie "z9hG4bK" but with the Request-URI, To tag, From tag, Call-ID, CSeq and top Via identical as in the first INVITE request, repeats its last response.			

SIP header values INVITE1: Via: *any_invite1_via_value*;branch= *z9hG4bK any_invite1_branch_value* From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq* INVITE2: Via: *any_invite1_via_value*;branch= *z9hG4bK any_invite1_branch_value* Via: *any_invite1_via_value*;branch= *any_invite2_branch_value* From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq*

Message flow

Test equipment			End device
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
INVITE2	→		
180 Ringing	←		
	Apply post test routine		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
INVITE2	→		
180 Ringing	←		
	Apply post test routine		

TSS Term_Establishment_of_an_early_dialogue	TP_301_016	Reference section 22.2 of [IETF RFC 3261]	Selection expression
Test purpose *INVITE without Authorization header received a 401 is sent containing a WWW-Authenticate header* Ensure that the IUT on receipt of an INVITE request not including an Authorization header field, sends an Unauthorized (401 Unauthorized) response, containing a WWW-Authenticate header.			
SIP header values 401: WWW-Authenticate:			

Message flow **Test equipment** **End device** **Interworking POTS** INVITE → 401 Unauthorized ← ACK → **Apply post test routine** **ISDN interworking** INVITE → 401 Unauthorized ← ACK → **Apply post test routine**

TSS Term_Establishment_of_an_early_dialogue	TP_301_017	Reference section 22.2 of [IETF RFC 3261]	Selection expression
Test purpose *INVITE without Authorization header received a 401 is sent containing a WWW-Authenticate header with proper values* Ensure that the IUT on receipt of an INVITE request not including an Authorization header field, sends an Unauthorized (401 Unauthorized) response, containing a WWW-Authenticate header including proper value for realm and nonce HTTP parameters.			
SIP header values 401: WWW-Authenticate: Digest realm="[*any value*]",nonce="[*any value*]",algorithm=MD5,qop="auth"			
Message flow **Test equipment** **End device** **Interworking POTS** INVITE → 401 Unauthorized ← ACK → **Apply post test routine** **ISDN interworking** INVITE → 401 Unauthorized ← ACK → **Apply post test routine**			

TSS Term_Establishment_of_ an_early_dialogue	TP_301_018	Reference section 22.2 of [IETF RFC 3261]	Selection expression

Test purpose

INVITE with valid Authorization header received, a 200 OK INVITE is sent

Ensure that the IUT having sent an Unauthorized (401 Unauthorized) response to an INVITE request, on receipt of a INVITE request including a valid Authorization header field, sends a Success (200 OK) response.

SIP header values

401:

WWW-Authenticate: Digest realm="[*any value*]",nonce="[*any value*]",algorithm=MD5,qop="auth"

INVITE 2

Authorization: Digest username="[any value]", realm="[any value]", nonce="[any value]", uri="sip:tel0.ver.sul.t-online.de", qop=auth, nc=[any value], cnonce="[any value]", response="[any value]", algorithm=MD5

Message flow

Test equipment			End device
Interworking POTS			
INVITE 1	→		
401 Unauthorized	←		
ACK	→		
INVITE 2	→		Ringing
200 OK INVITE	←		Off hook
ACK	→		

Apply post test routine

Test equipment			End device
ISDN interworking			
INVITE	→		
401 Unauthorized	←		
ACK	→		
INVITE 2	→	→	SETUP
200 OK INVITE	←	←	CONNECT
ACK	→		

Apply post test routine

TSS Term_Establishment_of_ an_early_dialogue	TP_301_019	Reference subclause 5.1.4.1 of [ETSI TS 124 229]	Selection expression PICS: 5.2/14 and 5.2/15

Test purpose

INVITE with support for preconditions received. Preconditions supported.

Ensure that upon the IUT receiving an INVITE request containing the precondition and the100rel option-tag in the Supported header, the SUT sends a (reliable) 183 that contains, in the SDP, the instruction to reserve the quality of service on the remote side and the required preconditions and a reliable transport of provisional responses. The remote entity indicates that the resource is reserved in the UPDATE request and the SUT confirms the resource reservation in the 200 OK UPDATE. The user is alerted.

SIP header values INVITE: Supported: precondition,100rel SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory/optional local sendrecv a=des:qos mandatory/optional remote sendrecv 183: Require: precondition,100rel SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory/optional local sendrecv a=des:qos mandatory/optional remote sendrecv a=conf:qos remote sendrecv UPDATE SDP a=curr:qos local sendrecv a=curr:qos remote none a=des:qos mandatory/optional local sendrecv a=des:qos mandatory/optional remote sendrecv 200 OK UPDATE SDP a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory/optional local sendrecv a=des:qos mandatory/optional remote sendrecv

Message flow

Test equipment				End device
Interworking POTS				
INVITE	→			
183 Session Progress	←			
PRACK	→			
200 OK PRACK	←			
UPDATE	→			
200 OK UPDATE	←			
180 Ringing	←			Ringing
		Apply post test routine		
ISDN interworking				
INVITE	→			
183 Session Progress	←			
PRACK	→			
200 OK PRACK	←			
UPDATE	→			
200 OK UPDATE	←			
			→	SETUP
180 Ringing	←		←	ALERTING
		Apply post test routine		

TSS Term_Establishment_of_ an_early_dialogue	TP_301_020	Reference subclause 5.1.4.1 of [ETSI TS 124 229]	Selection expression PICS: NOT 5.2/14
Test purpose *INVITE with support for preconditions received. Preconditions supported.* Ensure that the upon the IUT receiving an INVITE request containing the precondition and the100rel option-tag in the Supported header,the SUT sends a (reliable) 183 session Progress or 180 Ringing and the SDP that does not contain information that preconditions are supported. A Require header requesting preconditions is not included.			
SIP header values INVITE: Supported: precondition,100rel SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory/optional local sendrecv a=des:qos mandatory/optional remote sendrecv 183/180:			
Message flow			

Test equipment			End device
Interworking POTS			
INVITE	→		
CASE A			
183 Session Progress	←		
CASE B			
180 Ringing	←		Ringing
	Apply post test routine		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
	Apply post test routine		

TSS Term_Establishment_of_ an_early_dialogue	TP_301_021	Reference subclause 5.1.4.1 of [ETSI TS 124 229]	Selection expression PICS: 5.2/14 AND 5.2/15
Test purpose *INVITE without SDP offer received.* Ensure that the IUT upon receiving an INVITE request without an SDP offer and transport of reliable provisional responses, send a reliable 183 Session Progress containing an SDP offer. The SDP answer is received in an UPDATE request or ACK request			

SIP header values
INVITE:
    Supported: 100rel
 	
183
SDP offer
 	
CASE A
ACK
    SDP answer
 	
CASE B
UPDATE
    SDP answer

Message flow

Test equipment			End device
Interworking POTS			
INVITE	→		
183 Session Progress	←		Ringing
PRACK	→		
200 OK PRACK	←		
CASE A			
200 OK INVITE	←		Off hook
ACK	→		
CASE B			
UPDATE	→		
200 OK UPDATE	←		
	Apply post test routine		
ISDN interworking			
INVITE	→	→	SETUP
183 Session Progress	←	←	ALERTING
PRACK	→		
200 OK PRACK	←		
CASE A			
200 OK INVITE	←	←	CONNECT
ACK	→		
CASE B			
UPDATE	→		
200 OK UPDATE	←		
	Apply post test routine		

TSS Term_Establishment_of_an_early_dialogue	TP_301_022	Reference subclause 6.3.2.4 of [ETSI TS 183 043]	Selection expression

Test purpose

INVITE without SDP offer received.

Ensure that the IUT upon receipt of an INVITE request without an SDP offer and transport of reliable provisional responses, send a reliable 183 Session Progress containing an SDP offer. The SDP answer is received in an UPDATE request or ACK request

SIP header values

INVITE:

Supported: 100rel

183

SDP offer

CASE A

ACK

SDP answer

CASE B

UPDATE

SDP answer

Message flow

Test equipment			End device
Interworking POTS			
INVITE	→		
183 Session Progress	←		Ringing
PRACK	→		
200 OK PRACK	←		
CASE A			
200 OK INVITE	←		Off hook
ACK	→		
CASE B			
UPDATE	→		
200 OK UPDATE	←		
	Apply post test routine		
ISDN interworking			
INVITE	→	→	SETUP
183 Session Progress	←	←	ALERTING
PRACK	→		
200 OK PRACK	←		

CASE A			
200 OK INVITE	🡐	🡐	CONNECT
ACK	🡒		
CASE B			
UPDATE	🡒		
200 OK UPDATE	🡐		
	Apply post test routine		

TSS Term_Establishment_of_an_early_dialogue	TP_301_023	Reference subclause 6.3.2.4 of [ETSI TS 183 043]	Selection expression
Test purpose *Modifying SDP in early dialogue* Ensure that the SUT is able to receive an UPDATE request to modify the SDP in early dialogue. A 200 OK UPDATE with SDP answer is sent.			
SIP header values 183/180 SDP answer 1 UPDATE SDP offer 2 200 OK (UPDATE) SDP answer 2			
Message flow			
Test equipment			**End device**
Interworking POTS			
INVITE	🡒		
180/183	🡐		Ringing
UPDATE	🡒		
200 OK UPDATE	🡐		
	Apply post test routine		
ISDN interworking			
INVITE	🡒	🡒	SETUP
180 Ringing	🡐	🡐	ALERTING
UPDATE	🡒		
200 OK UPDATE	🡐		
	Apply post test routine		

7.2.3.1.2 Test purposes for ISDN

<table>
<tr><td>TSS
Term_Establishment_of_
an_early_dialogue</td><td>TP_301_101</td><td>Reference
subclause 5.1.2.1 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.4/1</td></tr>
<tr><td colspan="4">Test purpose
A BearerCapability speech is included in the PSTN XML element

Ensure that when a PSTN XML element is included in a received INVITE request and the BearerCapability value is set to ITC_VA as indicated in Table 7.2.3.1-1, a SETUP is sent where the Bearer Capability is derived from the received PSTM XML BearerCapability.</td></tr>
<tr><td colspan="4">SIP header values
INVITE: PSTN XML MIME body
<?xml version="1.0" encoding="utf-8"?>
PSTN
    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>ITC_value<
        BCoctet4
            TransferMode>00<
            InformationTransferRate>10000<
        BCoctet5
            Layer1Identification>01<
            UserInfoLayer1Protocol>00011<
If ITC_value = '01000' the SDP m line contains CLEARMODE as preferred codec</td></tr>
<tr><td colspan="4">DSS1 Parameter values
SETUP: Bearer Capability = PSTM XML BearerCapability</td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><th colspan="2">Test equipment</th><th colspan="2">End device</th></tr>
<tr><td>INVITE</td><td>→</td><td>→</td><td>SETUP</td></tr>
<tr><td>100 Trying</td><td>←</td><td></td><td></td></tr>
</table>
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Term_Establishment_of_
an_early_dialogue</td><td>TP_301_102</td><td>Reference
subclause 5.1.2.1 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.4/1</td></tr>
<tr><td colspan="4">Test purpose
A BearerCapability speech and a BearerCapability UDI with tones/annount. is included in the PSTN XML element

Ensure that AGCF/VGW is able to send two Bearer Capability IE in the SETUP to the called user equipment in the same order as received in the PSTN BearerCapability XML. The first PSTN BearerCapability element is sent in the first Bearer Capability IE in the SETUP and the second PSTN BearerCapability element is sent in the second Bearer Capability IE in the SETUP.</td></tr>
</table>

SIP header values

INVITE: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN
BearerCapability
BCoctet3
CodingStandard>00<
InformationTransferCabability>00000<
or
InformationTransferCabability>10000<
BCoctet4
TransferMode>00<
InformationTransferRate>10000<
BCoctet5
Layer1Identification>01<
UserInfoLayer1Protocol>00011<
BearerCapability
BCoctet3
CodingStandard>00<
InformationTransferCabability>10001<
BCoctet4
TransferMode>00<
InformationTransferRate>10000<
BCoctet5
Layer1Identification>01<
UserInfoLayer1Protocol>00011<

SDP: m line contains as the first codec CLEARMODE and as the second codec an ITU-T G.711 codec

DSS1 Parameter values

SETUP: First Bearer Capability Information transfer capability = Speech or 3.1 kHz audio
Second Bearer Capability Information transfer capability = Unrestricted digital information with tones/announcements

Message flow

Test equipment			End device
INVITE	→	→	SETUP
100 Trying	←		

Apply post test routine

Table 7.2.3.1-1 – Mapping of PSTN XML BearerCapability to Bearer Capability

ITC_value	XML InformationTransferCabability	BC Information transfer capability
ITC_VA_1	'00000'	Speech
ITC_VA_2	'10000'	3,1 kHz audio
ITC_VA_3	'01000'	unrestricted digital information

<table>
<tr><td>TSS
Term_Establishment_of_
an_early_dialogue</td><td>TP_301_103</td><td>Reference
subclause 5.1.2.1 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2</td></tr>
<tr><td colspan="4">Test purpose
INVITE received, no PSTN XML instance present

Ensure that if an INVITE request is received and no PSTN BearerCapability XML element is present, the Bearer Capability IE is set according Table 7.2.3.1-2.</td></tr>
<tr><td colspan="4">SIP header values
INVITE: SDP m line = SDP_VA</td></tr>
<tr><td colspan="4">DSS1 Parameter values
SETUP: Bearer Capability</td></tr>
<tr><td colspan="4">Message flow
Test equipment   End device
INVITE → → SETUP
100 Trying ←
Apply post test routine</td></tr>
</table>

Table 7.2.3.1-2 – Coding of from SDP: SIP to BC DSS1

SDP_VA		**m= line**		**a= line**	**Bearer Capability IE**	
	<media>	**<transport>**	**<fmt-list>**	**Rtpmap:<dynamic-PT> <encoding name>/<clock rate>/encoding parameters>**	**Information transport capability**	**User information Layer 1 Protocol Indicator**
SDP_VA_01	Audio	RTP/AVP	0	N/A	*"3.1 kHz audio"*	*"G.711 A-law"*
SDP_VA_02	Audio	RTP/AVP	0	N/A	*"3.1 kHz audio"*	*"G.711 µ-law"*
SDP_VA_03	Audio	RTP/AVP	Dynamic PT	rtpmap:<dynamic-PT> PCMU/8000	*"3.1 kHz audio"*	*"G.711 A-law"*
SDP_VA_04	Audio	RTP/AVP	Dynamic PT	rtpmap:<dynamic-PT> PCMU/8000	*"3.1 kHz audio"*	*"G.711 µ-law"*
SDP_VA_05	Audio	RTP/AVP	8	N/A	*"3.1 kHz audio"*	*"G.711 A-law"*
SDP_VA_06	Audio	RTP/AVP	8	N/A	*"3.1 kHz audio"*	*"G.711 µ-law"*
SDP_VA_07	Audio	RTP/AVP	Dynamic PT	rtpmap:<dynamic-PT> PCMA/8000	*"3.1 kHz audio"*	*"G.711 A-law"*
SDP_VA_08	Audio	RTP/AVP	Dynamic PT	rtpmap:<dynamic-PT> PCMA/8000	*"3.1 kHz audio"*	*"G.711 µ-law"*
SDP_VA_09	Audio	RTP/AVP	Dynamic PT,	rtpmap:<dynamic-PT> CLEARMODE/8000	*"Unrestricted digital inf. W/tone/ann."*)	
SDP_VA_10	Audio	RTP/AVP	Dynamic PT	Rtpmap:<dynamic-PT> CLEARMODE/8000	*"Unrestricted digital information"*	
SDP_VA_11	Image	Udptl	t38	Based on [ITU-T T.38]	*"3.1 kHz audio"*	*"G.711 A-law"*
SDP_VA_12	Image	Tcptl	t38	Based on [ITU-T T.38]	*"3.1 kHz audio"*	*"G.711 A-law"*
SDP_VA_13	Image	Udptl	t38	Based on [ITU-T T.38]	*"3.1 kHz audio"*	*"G.711 µ-law"*
SDP_VA_14	Image	Tcptl	t38	Based on [ITU-T T.38]	*"3.1 kHz audio"*	*"G.711 µ-law"*

TSS Term_Establishment_of_ an_early_dialogue	TP_301_104	Reference subclause 5.1.2.1/ [ETSI TS 183 036]	Selection expression PICS 5.1.1/2
Test purpose *INVITE received, no PSTN XML instance present* Ensure that if an INVITE request is received and no PSTN BearerCapability XML element is present, the High Layer Compatibility IE if present is set according Table 7.2.3.1-3.			
SIP header values INVITE: SDP m line = SDP_VA			
DSS1 Parameter values SETUP: High Layer Compatibility, High Layer Characteristics Identificatio value			
Message flow			

Test equipment			**End device**
INVITE	→	→	SETUP
100 Trying	←		
	Apply post test routine		

Table 7.2.3.1-3 – Coding of from SDP: SIP to HLC DSS1

SDP_VA		**m= line**		**a= line**	**HLC parameter (optional)**
	<media>	**<transport>**	**<fmt-list>**	**Rtpmap:<dynamic-PT> <encoding name>/<clock rate>/encoding parameters>**	**High Layer Characteristics Identification**
SDP_VA_01	Image	Udptl	t38	Based on ITU-T T.38	*"Facsimile Group 2/3"*
SDP_VA_02	Image	Tcptl	t38	Based on ITU-T T.38	*"Facsimile Group 2/3"*
SDP_VA_03	Image	Udptl	t38	Based on ITU-T T.38	*"Facsimile Group 2/3"*
SDP_VA_04	Image	Tcptl	t38	Based on ITU-T T.38	*"Facsimile Group 2/3"*

TSS Term_Establishment_of_ an_early_dialogue	TP_301_105	Reference subclause 5.1.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1
Test purpose *Mapping of PSTN XML ProgressIndicator into DSS1 Progress Indicator IE* Ensure that on receipt of an INVITE request and the PSTN XML contains the ProgressIndicator, a SETUP is sent. A Progress Indicator IE is present derived from the received ProgressIndicator PI_value according to Table 7.2.3.1-4.			

SIP header values

INVITE:

PSTM XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

ProgressIndicator

ProgressOctet3

CodingStandard>00<

Location>yyyy<

ProgressOctet4

ProgressDescription>0000110<

ProgressIndicator

ProgressOctet3

CodingStandard>00<

Location>0000<

ProgressOctet4

ProgressDescription>PI_value<

DSS1 Parameter values

SETUP: Progress Indicator Coding standard ='00', Location ='0000', Progress description= PI_value

Message flow

Test equipment			**End device**
INVITE	➔	➔	SETUP
100 Trying	🡐		

Apply post test routine

TSS Term_Establishment_of_an_early_dialogue	TP_301_106	Reference subclause 5.1.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1

Test purpose

Receipt of PSTN XML ProgressIndicator value 6

Ensure that on receipt of an INVITE request and the PSTN XML contains the ProgressIndicator value set to 6, a SETUP is sent and no Progress Indicator IE is present.

SIP header values

INVITE:

PSTM XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

ProgressIndicator

ProgressOctet3

CodingStandard>00<

Location>yyyy<

ProgressOctet4

ProgressDescription>0000110<

DSS1 Parameter values

Message flow

Test equipment			**End device**
INVITE	➔	➔	SETUP
100 Trying	🡐		

Apply post test routine

TSS Term_Establishment_of_ an_early_dialogue	TP_301_107	Reference subclause 5.1.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1
Test purpose *INVITE request no PSTN XML instance present.* Ensure that on receipt of an INVITE request and no PSTN is present, a SETUP is sent. A Progress Indicator IE is present and the Progress Description value is set to 1.			
SIP header values			
DSS1 Parameter values			
Message flow			
Test equipment			**End device**
INVITE	→	→	SETUP
100 Trying	←		
Apply post test routine			

Table 7.2.3.1-4 – Mapping of PSTN XML ProgressIndicator to DSS1 Progress Indicator information element

PI_value	XML ProgressIndicator ProgressDescription	DSS1 Progress Indicator value
PI_VA_1	'0000001'	Call is not end-to-end 5.1.1/2; further call progress information may be available in-band
PI_VA_2	'0000010'	Destination address is non-5.1.1/2
PI_VA_3	'0000011'	Origination address is non-5.1.1/2
PI_VA_4	'0000100'	Call has returned to the 5.1.1/2
PI_VA_5	'0000101'	Interworking has occurred and has resulted in a telecommunication service change
PI_VA_6	'0001000'	In-band information or an appropriate pattern is now available

TSS Term_Establishment_of_ an_early_dialogue	TP_301_108	Reference subclause 5.1.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1
Test purpose *Mapping of PSTN XML LowLayerCompatibility into DSS1 Low Layer Compatibility IE* Ensure that on receipt of an INVITE request and the PSTN XML contains the LowLayerCompatibility, a SETUP is sent. A Low Layer Compatibility IE is present derived from the received LowLayerCompatibility ITC_value according to Table 7.2.3.1-5.			

SIP header values

INVITE:

<?xml version="1.0" encoding="utf-8"?>

PSTN

    LowLayerCompatibility>
        LLOctet3>
            CodingStandard>00<
            InformationTransferCapability>ITC_VA<
        LLOctet4>
            TransferMode>00<
            InformationTransferRate>10000<

DSS1 Parameter values

SETUP: Low Layer Compatibility

Message flow

Test equipment			End device
INVITE	➔	➔	SETUP
100 Trying	🡐		

Apply post test routine

Table 7.2.3.1-5 – Mapping of PSTN XML LowLayerCompatibility to DSS1 low layer compatibility

ITC_value	XML LLC InformationTransferCabability	LLC Information transfer capability
ITC_VA_1	'00000'	Speech
ITC_VA_2	'10000'	3,1 kHz audio
ITC_VA_3	'01001'	Unrestricted digital info
ITC_VA_3	'10001'	7 kHz audio

TSS Term_Establishment_of_an_early_dialogue	TP_301_109	Reference subclause 5.1.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1

Test purpose

Mapping of PSTN XML HighLayerCompatibility into DSS1 High Layer Compatibility IE

Ensure that on receipt of an INVITE request and the PSTN XML contains the HighLayerCompatibility, a SETUP is sent. A High Layer Compatibility IE is present derived from the received HighLayerCompatibility HLC_value according Table 7.2.3.1-6.

SIP header values

INVITE:

PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

    HighLayerCompatibility
        HLOctet3
            CodingStandard>00<
            Interpretation>100<
            PresentationMethod>01<
        HLOctet4
            HighLayerCharacteristics>HLC_value<

DSS1 Parameter values SETUP: High Layer Compatibility
Message flow **Test equipment** **End device** INVITE → → SETUP 100 Trying ← **Apply post test routine**

TSS Term_Establishment_of_an_early_dialogue	TP_301_110	Reference subclause 5.1.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1
Test purpose *Two High layer compatibility information elements received* Ensure that the first PSTN XML HighLayerCompatibility received in the INVITE request is mapped into the first High layer compatibility information element in the sent SETUP and the second PSTN XML HighLayerCompatibility received in the INVITE request is mapped into the second High layer compatibility information element in the sent SETUP			
SIP header values INVITE: PSTN XML MIME body <?xml version="1.0" encoding="utf-8"?> PSTN HighLayerCompatibility HLOctet3 CodingStandard>00< Interpretation>100< PresentationMethod>01< HLOctet4 HighLayerCharacteristics>0000001< HighLayerCompatibility HLOctet3 CodingStandard>00< Interpretation>100< PresentationMethod>01< HLOctet4 HighLayerCharacteristics>HLC_value<			
DSS1 parameter values SETUP: **First high layer compatibility** Coding standard='00', Interpretation='100', High layer characteristics identification= Telephony **Second high layer compatibility** Coding standard='00', Interpretation='100', High layer characteristics identification=HLC_value as indicated in Table 7.2.3.1-6			
Message flow **Test equipment** **End device** INVITE → → SETUP 100 Trying ← **Apply post test routine**			

Table 7.2.3.1-6 – Mapping of PSTN XML HighLayerCharacteristic to DSS1 High layer compatibility information element

HLC_value	XML HighLayerCharacteristic	DSS1 High layer characteristics identification
HLC_VA_1	'0000001'	Telephony
HLC_VA_2	'0000100'	Facsimile Group 2/3
HLC_VA_3	'0100001'	Facsimile Group 4 Class I
HLC_VA_4	'0100100'	Facsimile service Group 4, Classes II ad III
HLC_VA_5	'0110010'	Syntax based Videotex
HLC_VA_6	'0110011'	International Videotex interworking via gateways or interworking units
HLC_VA_7	'0110101'	Telex service
HLC_VA_8	'1000010'	FTAM application
HLC_VA_9	'1100000'	Videotelephony

<table>
<tr><td>TSS
Term_Establishment_of_
an_early_dialogue</td><td>TP_301_111</td><td>Reference
subclause 5.1.2.1/
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.4/1</td></tr>
<tr><td colspan="4">Test purpose
CALL PROCEEDING received Progress Indicator is present

Ensure that on receipt of a DSS1 CALL PROCEEDING and a Progress Indicator value PI_value, as described in Table 7.2.3.1-4, is present, a 183 Session Progress is sent.</td></tr>
<tr><td colspan="4">SIP header values</td></tr>
<tr><td colspan="4">DSS1 Parameter values
CALL PROCEEDING: Progress Indicator Progress Description = PI_value</td></tr>
<tr><td colspan="4">Message flow
Test equipment   End device
INVITE →   → SETUP
100 Trying ←
183 Session Progress ←   ← CALL PROCEEDING
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Term_Establishment_of_
an_early_dialogue</td><td>TP_301_112</td><td>Reference
subclause 5.1.2.2 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.4/1</td></tr>
<tr><td colspan="4">Test purpose
CALL PROCEEDING received Progress Indicator is present

Ensure that on receipt of a DSS1 CALL PROCEEDING, and a Progress Indicator value PI_value is present, as described in Table 7.2.3.1-4, a 183 Session Progress is sent. The 183 Session Progress contains a PSTN XML ProgressIndicator ProgressDescription = PI_value. An additional ProgressIndicator element with value = 7 is present.</td></tr>
</table>

SIP header values

183 Session Progress:

PSTM XML MIME body

```
<?xml version="1.0" encoding="utf-8"?>
PSTN
    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>yyyy<
        ProgressOctet4
            ProgressDescription>0000111<
    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>0000<
        ProgressOctet4
            ProgressDescription>PI_value<
```

DSS1 Parameter values

CALL PROCEEDING: Progress Indicator Progress Description = PI_value

Message flow

Test equipment			End device
INVITE	→	→	SETUP
100 Trying	←		
183 Session Progress	←	←	CALL PROCEEDING

Apply post test routine

TSS Term_Establishment_of_an_early_dialogue	TP_301_113	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1

Test purpose

CALL PROCEEDING received Progress Indicator is not present

Ensure that on receipt of a DSS1 CALL PROCEEDING and no Progress Indicator is present, a 183 Session Progress is sent. The 183 Session Progress contains a PSTN XML ProgressIndicator ProgressDescription = 7.

SIP header values

183 Session Progress:

PSTM XML MIME body

```
<?xml version="1.0" encoding="utf-8"?>
PSTN
    ProgressIndicator
        ProgressOctet3
            CodingStandard>00<
            Location>yyyy<
        ProgressOctet4
            ProgressDescription>0000111<
```

DSS1 Parameter values

CALL PROCEEDING:

Message flow
Test equipment   **End device**
INVITE → → SETUP
100 Trying ←
183 Session Progress ← ← CALL PROCEEDING
Apply post test routine

TSS Term_Establishment_of_ an_early_dialogue	TP_301_114	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.2/4
Test purpose *183 is sent after the SUT has determined independently of access indications that the complete called party number has been received* Ensure that the AGCF/VGW is able to send a 183 Session Progress independently of access indications that the complete called party number has been received.			
SIP header values			
DSS1 Parameter values			
Message flow **Test equipment** **End device** INVITE → → SETUP 100 Trying ← 183 Session Progress ← ← CALL PROCEEDING **Apply post test routine**			

TSS Term_Establishment_of_ an_early_dialogue	TP_301_115	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.2/16
Test purpose *The SUT sends a P-Early-Media header if an INVITE (audio) is received* Ensure that on receipt of an INVITE request containing a PSTN XML BearerCapability set to speech or audio 3 kBit/s, a P-Early-Media header is sent in a 183 Session Progress response authorize early-media if a CALL PROCEEDING message is received from the terminating user equipment.			

SIP header values
INVITE: PSTN XML MIME body
P-Erly-Media: supported
<?xml version="1.0" encoding="utf-8"?>
PSTN
  BearerCapability
    BCoctet3
      CodingStandard>00<
      InformationTransferCabability>00000<
      or
      InformationTransferCabability>10000<
    BCoctet4
      TransferMode>00<
      InformationTransferRate>10000<
    BCoctet5
      Layer1Identification>01<
      UserInfoLayer1Protocol>00011<
 	
183 Session Progress: P-Early-Media: <appropriate value>
SDP answer

DSS1 Parameter values
Message flow

Test equipment			End device
INVITE	→	→	SETUP
100 Trying	←		
183 Session Progress	←	←	CALL PROCEEDING
	Apply post test routine		

TSS Term_Establishment_of_an_early_dialogue	TP_301_116	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1 and 5.1.3/1
Test purpose *Handling of fallback information BC speech (only applicable at T reference point)* Ensure that the called user is able to indicate the Fallback to speech if the user equipment is **not able to support the UDI/TA**. The AGCF/VGW receives a CALL PROCEEDING. A Bearer Capability IE is present and the Information Transfer Capability indicator is set to speech and a Progress Indicator IE value 5 is present. A 183 Session Progress is sent to the calling user equipment, the PSTN XML BearerCapability is present and the InformationTransferCabability element is set to speech and a ProgressIndicator element is present set to value 5. The first stated codec in the SDP answer is not equal CLEARMODE			

SIP header values

INVITE: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

BearerCapability

BCoctet3

CodingStandard>00<

InformationTransferCabability>00000<

BearerCapability

BCoctet3

CodingStandard>00<

InformationTransferCabability>10001<

183 Session Progress: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

BearerCapability

BCoctet3

CodingStandard>00<

InformationTransferCabability>00000<

ProgressIndicator

ProgressOctet4

ProgressDescription>0000101<

SDP: m= audio xxxx RTP/AVP 8

DSS1 Parameter values

CALL PROCEEDING: Bearer Capability: Information Transfer Capability = speech, Progress Indicator Progress description = 5

Message flow

Test equipment			End device
INVITE	→	→	SETUP
100 Trying	←		
183 Session Progress	←	←	CALL PROCEEDING

Apply post test routine

TSS Term_Establishment_of_an_early_dialogue	TP_301_117	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1 and 5.1.3/1

Test purpose

Handling of fallback information BC audio (only applicable at T reference point)

Ensure that the called user is able to indicate the Fallback to 3.1 kHz audio if the user equipment is **not able to support the UDI/TA**. The AGCF/VGW receives a CALL PROCEEDING. A Bearer Capability IE is present and the Information Transfer Capability indicator is set to 3.1 kHz audio and a Progress Indicator IE value 5 is present. A 183 Session Progress is sent to the calling user equipment, the PSTN XML BearerCapability is present and the InformationTransferCabability element is set to 3.1 kHz audio and a ProgressIndicator element is present set to value 5. The first stated codec in the SDP answer is not equal to CLEARMODE.

SIP header values

INVITE: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>
PSTN
    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>10000<
    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>10001<

183 Session Progress: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>
PSTN
    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>10000<
    ProgressIndicator
        ProgressOctet4
            ProgressDescription>0000101<

SDP: m= audio xxxx RTP/AVP 8

DSS1 Parameter values

CALL PROCEEDING: Bearer Capability: Information Transfer Capability = speech, Progress Indicator Progress description = 5

Message flow

Test equipment			End device
INVITE	→	→	SETUP
100 Trying	←		
183 Session Progress	←	←	CALL PROCEEDING

Apply post test routine

TSS Term_Establishment_of_an_early_dialogue	TP_301_118	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1 and 5.1.3/1

Test purpose

Handling of fallback information HLC (only applicable at T reference point)

Ensure that the called user is able to indicate the Fallback if the user equipment is not able to support the UDI/TA. The AGCF/VGW receives a CALL PROCEEDING. The High Layer Compatibility IE is present. A 183 Session Progress is sent to the calling user equipment, the PSTN XML HighLayerCompatibility is mapped from the DSS1 High Layer Compatibility IE.

SIP header values **INVITE**: PSTN XML MIME body <?xml version="1.0" encoding="utf-8"?> PSTN **BearerCapability** BCoctet3 CodingStandard>00< InformationTransferCabability>00000< **BearerCapability** BCoctet3 CodingStandard>00< InformationTransferCabability>10001< **183 Session Progress**: HighLayerCompatibility
DSS1 Parameter values CALL PROCEEDING: High Layer Compatibility High Layer Characteristics <appropriate value>
Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
100 Trying	←		
183 Session Progress	←	←	CALL PROCEEDING
	Apply post test routine		

TSS	TP_301_119	Reference	Selection expression
Term_Establishment_of_an_early_dialogue		subclause 5.1.2.2 of [ETSI TS 183 036]	PICS 5.1.1/2 AND NOT 5.4/1
Test purpose *PROGRESS received Progress Indicator is present* Ensure that on receipt of a DSS1 PROGRESS and a Progress Indicator value PI_value is present, as described in Table 7.2.3.1-4, a 183 Session Progress is sent.			
SIP header values			
DSS1 Parameter values PROGRESS: Progress Indicator Progress Description = PI_value			
Message flow			

Test equipment			**End device**
INVITE	→	→	SETUP
100 Trying	←		
183 Session Progress	←	←	CALL PROCEEDING
183 Session Progress	←	←	PROGRESS
	Apply post test routine		

TSS Term_Establishment_of_an_early_dialogue	TP_301_120	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

PROGRESS received Progress Indicator is present

Ensure that on receipt of a DSS1 PROGRESS and a Progress Indicator value PI_value is present, as described in Table 7.2.3.1-4, a 183 Session Progress is sent. The 183 Session Progress contains a PSTN XML ProgressIndicator ProgressDescription = PI_value. An additional ProgressIndicator element withvalue = 7 is present.

SIP header values

183 Session Progress2

PSTM XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

  ProgressIndicator

    ProgressOctet3

      CodingStandard>00<

      Location>yyyy<

    ProgressOctet4

      ProgressDescription>**0000111**<

  ProgressIndicator

    ProgressOctet3

      CodingStandard>00<

      Location>0000<

    ProgressOctet4

      ProgressDescription>**PI_value**<

DSS1 Parameter values

PROGRESS: Progress Indicator Progress Description = PI_value

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
100 Trying	←		
183 Session Progress1	←	←	CALL PROCEEDING
183 Session Progress2	←	←	PROGRESS

Apply post test routine

TSS Term_Establishment_of_an_early_dialogue	TP_301_121	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1

Test purpose

PROGRESS received Progress Indicator is not present

Ensure that on receipt of a DSS1 PROGRESS and if no Progress Indicator is present, a 183 Session Progress is sent. The 183 Session Progress contains a PSTN XML ProgressIndicator ProgressDescription = 7

SIP header values

183 Session Progress2

PSTM XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

ProgressIndicator

ProgressOctet3

CodingStandard>00<

Location>yyyy<

ProgressOctet4

ProgressDescription>**0000111**<

DSS1 Parameter values

PROGRESS:

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
100 Trying	←		
183 Session Progress1	←	←	CALL PROCEEDING
183 Session Progress2	←	←	PROGRESS

Apply post test routine

TSS Term_Establishment_of_an_early_dialogue	TP_301_122	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1 and 5.2/16

Test purpose

The SUT sends a P-Early-Media header if an INVITE (audio) is received

Ensure that on receipt of an INVITE request containing a PSTN XML BearerCapability set to speech or audio 3 kBit/s, a P-Early-Media header is sent in a 183 Session Progress response authorize early-media if a CALL PROCEEDING message is received from the terminating user equipment.

SIP header values

INVITE: PSTN XML MIME body

P-Erly-Media: supported

<?xml version="1.0" encoding="utf-8"?>

PSTN

BearerCapability

BCoctet3

CodingStandard>00<

InformationTransferCabability>00000<

or

InformationTransferCabability>10000<

BCoctet4

TransferMode>00<

InformationTransferRate>10000<

BCoctet5

Layer1Identification>01<

UserInfoLayer1Protocol>00011<

183 Session Progress2: P-Early-Media: <appropriate value>

SDP answer

DSS1 Parameter values PROGRESS:

Message flow

Test equipment			End device
INVITE	→	→	SETUP
100 Trying	←		
183 Session Progress1	←	←	CALL PROCEEDING
183 Session Progress2	←	←	PROGRESS

Apply post test routine

TSS Term_Establishment_of_an_early_dialogue	TP_301_123	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/1

Test purpose

Handling of fallback information BC speech (only applicable at T reference point)

Ensure that the called user is able to indicate the Fallback to speech if the user equipment is not able to support the UDI/TA. The AGCF/VGW receives a PROGRESS. A Bearer Capability IE is present and the Information Transfer Capability indicator is set to speech and a Progress Indicator IE value 5 is present. A 183 Session Progress is sent to the calling user equipment, the PSTN XML BearerCapability is present and the InformationTransferCabability element is set to speech and a ProgressIndicator element is present and set to value 5. The first stated codec in the SDP answer is not equal CLEARMODE.

SIP header values

INVITE: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN
    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>10000<
    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>10001<
        BCoctet4

183 Session Progress2: HighLayerCompatibility

DSS1 Parameter values

PROGRESS: Bearer Capability: Information Transfer Capability = speech, Progress Indicator Progress description = 5

Message flow

Test equipment			End device
INVITE	→	→	SETUP
100 Trying	←		
183 Session Progress1	←	←	CALL PROCEEDING
183 Session Progress2	←	←	PROGRESS

Apply post test routine

TSS Term_Establishment_of_an_early_dialogue	TP_301_124	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/1

Test purpose

Handling of fallback information HLC (only applicable at T reference point)

Ensure that the called user is able to indicate the Fallback if the user equipment is not able to support the UDI/TA. The AGCF/VGW receives a PROGRESS. The High Layer Compatibility IE is present. A 183 Session Progress is sent to the calling user equipment, the PSTN XML HighLayerCompatibility is mapped from the DSS1 High Layer Compatibility IE.

SIP header values

INVITE: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>
PSTN
    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>10000<
    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>10001<
        BCoctet4

183 Session Progress2: HighLayerCompatibility

DSS1 Parameter values

PROGRESS: Bearer Capability: Information Transfer Capability = speech, Progress Indicator Progress description = 5

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
100 Trying	←		
183 Session Progress1	←	←	CALL PROCEEDING
183 Session Progress2	←	←	PROGRESS

Apply post test routine

TSS Term_Establishment_of_an_early_dialogue	TP_301_125	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND NOT 5.4/1

Test purpose

ALERTING received Progress Indicator is present

Ensure that on receipt of a DSS1 ALERTING, and if a Progress Indicator value PI_value as described in Table 7.2.3.1-4 is present, a 180 Ringing is sent.

SIP header values

DSS1 Parameter values

ALERTING: Progress Indicator Progress Description = **PI_value**

Message flow

Test equipment			End device
INVITE	→	→	SETUP
100 Trying	←		
180 Ringing	←	←	ALERTING

Apply post test routine

TSS Term_Establishment_of_ an_early_dialogue	TP_301_126	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1

Test purpose

ALERTING received Progress Indicator is not present

Ensure that on receipt of a DSS1 ALERTING and if no Progress Indicator is present, a 180 Ringing is sent. The 180 Ringing contains a PSTN XML ProgressIndicator ProgressDescription = 7

SIP header values

180 Ringing:

PSTM XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

  ProgressIndicator

    ProgressOctet3

      CodingStandard>00<

      Location>yyyy<

    ProgressOctet4

      ProgressDescription>**0000111**<

DSS1 Parameter values

ALERTING:

Message flow

Test equipment			End device
INVITE	→	→	SETUP
100 Trying	←		
180 Ringing	←	←	ALERTING

Apply post test routine

TSS Term_Establishment_of_ an_early_dialogue	TP_301_127	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.2/16

Test purpose

ALERTING received Progress Indicator is not present P-Early-Media header is present

Ensure that on receipt of a DSS1 ALERTING and if no Progress Indicator is present, a 180 Ringing is sent. The 180 Ringing contains a PSTN XML ProgressIndicator ProgressDescription = 7 and a PSTN XML ProgressIndicator ProgressDescription = 8 and the P-Early-Media authorize the early-media

SIP header values

INVITE:

P-Early-Media: supported

180 Ringing:

P-Early-Media: <appropriate value>

PSTM XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

- ProgressIndicator
  - ProgressOctet3
    - CodingStandard>00<
    - Location>yyyy<
  - ProgressOctet4
    - ProgressDescription>0000111<
- ProgressIndicator
  - ProgressOctet3
    - CodingStandard>00<
    - Location>yyyy<
  - ProgressOctet4
    - ProgressDescription>0001000<

SDP answer

DSS1 Parameter values

ALERTING:

Message flow

Test equipment			End device
INVITE	→	→	SETUP
100 Trying	←		
180 Ringing	←	←	ALERTING

Apply post test routine

TSS	TP_301_128	Reference	Selection expression
Term_Establishment_of_an_early_dialogue		subclause 5.1.2.2 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/1

Test purpose

ALERTING received Progress Indicator is present

Ensure that on receipt of a DSS1 ALERTING and a Progress Indicator value PI_value as described in Table 7.2.3.1-4 is present, a 180 Ringing is sent. The 180 Ringing contains a PSTN XML ProgressIndicator ProgressDescription = PI_value. An additional ProgressIndicator element with value = 7 is present.

SIP header values

180 Ringing:

PSTM XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

- **ProgressIndicator**
  - ProgressOctet3
    - CodingStandard>00<
    - Location>yyyy<
  - ProgressOctet4
    - ProgressDescription>**0000111**<
- **ProgressIndicator**
  - ProgressOctet3
    - CodingStandard>00<
    - Location>0000<
  - ProgressOctet4
    - ProgressDescription>**PI_value**<

DSS1 Parameter values

ALERTING: Progress Indicator Progress Description = **PI_value**

Message flow

Test equipment			End device
INVITE	→	→	SETUP
100 Trying	←		
180 Ringing	←	←	ALERTING

Apply post test routine

TSS Term_Establishment_of_an_early_dialogue	TP_301_129	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/1

Test purpose

Handling of fallback information BC speech (only applicable at T reference point)

Ensure that the called user is able to indicate the Fallback to speech if the user equipment is not able to support the UDI/TA. The AGCF/VGW receives an ALERTING. A Bearer Capability IE is present and the Information Transfer Capability indicator is set to speech and a Progress Indicator IE value 5 is present. A 180 Ringing is sent to the calling user equipment, the PSTN XML BearerCapability is present and the InformationTransferCabability element is set to speech and a ProgressIndicator element that is present is set to value 5. The first stated codec in the SDP answer is not equal to CLEARMODE.

SIP header values

INVITE: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>
PSTN
BearerCapability
BCoctet3
CodingStandard>00<
InformationTransferCabability>00000<
BearerCapability
BCoctet3
CodingStandard>00<
InformationTransferCabability>10001<

180 Ringing: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>
PSTN
BearerCapability
BCoctet3
CodingStandard>00<
InformationTransferCabability>00000<
BCoctet4
ProgressIndicator
ProgressOctet4
ProgressDescription>0000101<

SDP: m= audio xxxx RTP/AVP 8

DSS1 Parameter values

ALERTING: Bearer Capability: Information Transfer Capability = **speech**, Progress Indicator Progress description = **5**

Message flow

Test equipment			End device
INVITE	→	→	SETUP
100 Trying	←		
180 Ringing	←	←	ALERTING

Apply post test routine

TSS Term_Establishment_of_an_early_dialogue	TP_301_130	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/1

Test purpose

Handling of fallback information HLC (only applicable at T reference point)

Ensure that the called user is able to indicate the Fallback if the user equipment is not able to support the UDI/TA. The AGCF/VGW receives an ALERTING. The High Layer Compatibility IE is present. A 180 Ringing is sent to the calling user equipment, the PSTN XML HighLayerCompatibility is mapped from the DSS1 High Layer Compatibility IE.

SIP header values

INVITE: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

  BearerCapability

    BCoctet3

      CodingStandard>00<

      InformationTransferCabability>00000<

  BearerCapability

    BCoctet3

      CodingStandard>00<

      InformationTransferCabability>10001<

180 Ringing: HighLayerCompatibility

DSS1 Parameter values

ALERTING: Bearer Capability: Information Transfer Capability = **speech**, Progress Indicator Progress description = **5**

Message flow

Test equipment			End device
INVITE	➔	➔	SETUP
100 Trying	🡐		
180 Ringing	🡐	🡐	ALERTING

Apply post test routine

TSS Term_Establishment_of_an_early_dialogue	TP_301_131	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/1

Test purpose

Handling of fallback information PI (only applicable at T reference point)

Ensure that the called user is able to indicate the Fallback if the user equipment is not able to support the UDI/TA. The AGCF/VGW receives an ALERTING. The Progress Indicator IE value 5 is present. A 180 Ringing is sent to the calling user equipment, the PSTN XML ProgressIndicator is mapped from the DSS1 Progress Indicator IE.

SIP header values

INVITE: PSTN XML MIME body

```
<?xml version="1.0" encoding="utf-8"?>
PSTN
    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>00000<
        BCoctet4
            TransferMode>00<
            InformationTransferRate>10000<
        BCoctet5
            Layer1Identification>01<
            UserInfoLayer1Protocol>00011<
    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>10001<
        BCoctet4
            TransferMode>00<
            InformationTransferRate>10000<
        BCoctet5
            Layer1Identification>01<
            UserInfoLayer1Protocol>00011<
```

180 Ringing: ProgressIndicator ProgressDescription 5

DSS1 Parameter values

ALERTING: Bearer Capability: Information Transfer Capability = **speech**, Progress Indicator Progress description = **5**

Message flow

Test equipment			End device
INVITE	→	→	SETUP
100 Trying	←		
180 Ringing	←	←	ALERTING

Apply post test routine

TSS Term_Establishment_of_an_early_dialogue	TP_301_132	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND NOT 5.4/1

Test purpose

PROGRESS received Progress Indicator is present

Ensure that on receipt of a DSS1 PROGRESS and a Progress Indicator value PI_value, as described in Table 7.2.3.1-4, is present, a 183 Session Progress is sent.

SIP header values

DSS1 Parameter values

PROGRESS: Progress Indicator Progress Description = **PI_value**

Message flow

Test equipment			End device
INVITE	➔	➔	SETUP
180 Ringing	🡐	🡐	ALERTING
183 Session Progress	🡐	🡐	PROGRESS

Apply post test routine

TSS Term_Establishment_of_an_early_dialogue	TP_301_133	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1

Test purpose

PROGRESS received Progress Indicator is present

Ensure that on receipt of a DSS1 PROGRESS and a Progress Indicator value PI_value is present as described in Table 7.2.3.1-4, a 183 Session Progress is sent. The 183 Session Progress contains a PSTN XML ProgressIndicator ProgressDescription = PI_value. An additional ProgressIndicator element is present value = 7

SIP header values

183 Session Progress:

PSTM XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

  ProgressIndicator

    ProgressOctet3

      CodingStandard>00<

      Location>yyyy<

    ProgressOctet4

      ProgressDescription>**0000111**<

  ProgressIndicator

    ProgressOctet3

      CodingStandard>00<

      Location>0000<

    ProgressOctet4

      ProgressDescription>**PI_value**<

DSS1 Parameter values

PROGRESS: Progress Indicator Progress Description = **PI_value**

Message flow

Test equipment			End device
INVITE	➔	➔	SETUP
180 Ringing	🡐	🡐	ALERTING
183 Session Progress	🡐	🡐	PROGRESS

Apply post test routine

TSS Term_Establishment_of_an_early_dialogue	TP_301_134	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1

Test purpose

PROGRESS received Progress Indicator is not present

Ensure that on receipt of a DSS1 PROGRESS and if no Progress Indicator is present, a 183 Session Progress is sent. The 183 Session Progress contains a PSTN XML ProgressIndicator ProgressDescription = 7

SIP header values

183 Session Progress:

PSTM XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

ProgressIndicator

ProgressOctet3

CodingStandard>00<

Location>yyyy<

ProgressOctet4

ProgressDescription>**0000111**<

DSS1 Parameter values

PROGRESS:

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
183 Session Progress	←	←	PROGRESS

Apply post test routine

TSS Term_Establishment_of_an_early_dialogue	TP_301_135	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.2/16

Test purpose

The SUT sends a P-Early-Media header if an INVITE (audio) is received

Ensure that on receipt of an INVITE request containing a PSTN XML BearerCapability set to **speech** or **audio 3 kBit/s**, a P-Early-Media header is sent in a 183 Session Progress response authorize early-media if a CALL PROCEEDING message is received from the terminating user equipment.

SIP header values

INVITE: PSTN XML MIME body

P-Early-Media: supported

<?xml version="1.0" encoding="utf-8"?>

PSTN

BearerCapability

BCoctet3

CodingStandard>00<

InformationTransferCabability>**00000**<

or

InformationTransferCabability>**10000**<

BCoctet4

TransferMode>00<

InformationTransferRate>10000<

BCoctet5

Layer1Identification>01<

UserInfoLayer1Protocol>00011<

183 Session Progress: P-Early-Media: <appropriate value>

DSS1 Parameter values

PROGRESS:

Message flow

Test equipment			End device
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
183 Session Progress	←	←	PROGRESS

Apply post test routine

TSS	TP_301_136	Reference	Selection expression
Term_Establishment_of_an_early_dialogue		subclause 5.1.2.2 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/1

Test purpose

Handling of fallback information BC speech (only applicable at T reference point)

Ensure that the called user is able to indicate the Fallback to speech if the user equipment is not able to support the UDI/TA. The AGCF/VGW receives a PROGRESS. A Bearer Capability IE is present and the Information Transfer Capability indicator is set to speech and a Progress Indicator IE value 5 is present. A 183 Session Progress is sent to the calling user equipment, the PSTN XML BearerCapability is present and the InformationTransferCabability element is set to speech, and a ProgressIndicator element set to value 5 is present. The first stated codec in the SDP answer is not equal to CLEARMODE.

SIP header values

INVITE: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>
PSTN
    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>**00000**<
    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>**10001**<

183 Session Progress: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>
PSTN
    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>**00000**<
    ProgressIndicator
        ProgressOctet4
            ProgressDescription>0000101<
SDP: m= audio xxxx RTP/AVP 8

DSS1 Parameter values

PROGRESS: Bearer Capability: Information Transfer Capability = speech, Progress Indicator Progress description = 5

Message flow

Test equipment			End device
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
183 Session Progress	←	←	PROGRESS

Apply post test routine

TSS Term_Establishment_of_an_early_dialogue	TP_301_137	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/1

Test purpose

Handling of fallback information BC audio (only applicable at T reference point)

Ensure that the called user is able to indicate the Fallback to 3,1 kHz audio if the user equipment is not able to support the UDI/TA. The AGCF/VGW receives a PROGRESS. A Bearer Capability IE is present and the Information Transfer Capability indicator is set to 3,1 kHz audio and a Progress Indicator IE value 5 is present. A 183 Session Progress is sent to the calling user equipment, the PSTN XML BearerCapability is present and the InformationTransferCabability element is set to 3,1 kHz audio and a ProgressIndicator element is present set to value 5. The first stated codec in the SDP answer is not equal to CLEARMODE.

SIP header values

INVITE: PSTN XML MIME body
<?xml version="1.0" encoding="utf-8"?>
PSTN
    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>**10000**<
    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>**10001**<

183 Session Progress: PSTN XML MIME body
<?xml version="1.0" encoding="utf-8"?>
PSTN
    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>**10000**<
    ProgressIndicator
        ProgressOctet4
            ProgressDescription>**0000101**<
SDP: m= audio xxxx RTP/AVP

DSS1 Parameter values

PROGRESS: Bearer Capability: Information Transfer Capability = 3,1 kHz audio, Progress Indicator Progress description = 5

Message flow

Test equipment			End device
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
183 Session Progress	←	←	PROGRESS

Apply post test routine

TSS Term_Establishment_of_ an_early_dialogue	TP_301_138	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/1

Test purpose

Handling of fallback information HLC (only applicable at T reference point)

Ensure that the called user is able to indicate the Fallback if the user equipment is not able to support the UDI/TA. The AGCF/VGW receives a PROGRESS. The High Layer Compatibility IE is present. A 183 Session Progress is sent to the calling user equipment, the PSTN XML HighLayerCompatibility is mapped from the DSS1 High Layer Compatibility IE.

SIP header values

INVITE: PSTN XML MIME body

```
<?xml version="1.0" encoding="utf-8"?>
PSTN
   BearerCapability
      BCoctet3
         CodingStandard>00<
         InformationTransferCabability>10000<
      BCoctet4
         TransferMode>00<
         InformationTransferRate>10000<
      BCoctet5
         Layer1Identification>01<
         UserInfoLayer1Protocol>00011<
   BearerCapability
      BCoctet3
         CodingStandard>00<
         InformationTransferCabability>10001<
      BCoctet4
         TransferMode>00<
         InformationTransferRate>10000<
      BCoctet5
         Layer1Identification>01<
         UserInfoLayer1Protocol>00011<
```

183 Session Progress: HighLayerCompatibility

DSS1 Parameter values

PROGRESS: High Layer Compatibility High Layer Characteristics <appropriate value>

Message flow

Test equipment			**End device**
INVITE	➔	➔	SETUP
180 Ringing	←	←	ALERTING
183 Session Progress	←	←	PROGRESS

Apply post test routine

7.2.3.2 Establishment of a confirmed dialogue

<table>
<tr><td>TSS
Term_Establishment_of_
a_confirmed_dialogue</td><td>TP_302_001</td><td>Reference
[IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
From, Call-ID, CSeq and Via headers copy from the INVITE

Ensure that the IUT on receipt of an INVITE request, sends a Success (200 OK) including the headers From, Call-ID, CSeq and Via headers copy from the INVITE request.</td></tr>
<tr><td colspan="4">SIP header values
INVITE:
From: <from_value_invite>
Call-ID: <callid_value_invite>
CSeq: <cseq_value_invite>
Via: <via_value_invite>

200 OK:
From: <from_value_invite>
Call-ID: <callid_value_invite>
CSeq: <cseq_value_invite>
Via: <via_value_invite></td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><th>Test equipment</th><th></th><th></th><th>End device</th></tr>
<tr><td>Interworking POTS</td><td></td><td></td><td></td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>Ringing</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>Off hook</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td></tr>
<tr><td colspan="4">Apply post test routine</td></tr>
<tr><td>ISDN interworking</td><td></td><td></td><td></td></tr>
<tr><td>INVITE</td><td>→</td><td>→</td><td>SETUP</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td>←</td><td>CONNECT</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td></tr>
<tr><td colspan="4">Apply post test routine</td></tr>
</table></td></tr>
</table>

<table>
<tr><td>TSS
Term_Establishment_of_
a_confirmed_dialogue</td><td>TP_302_002</td><td>Reference
sections 8.2.6.2, 12.2.2 and 13.3.1 of
[IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
To tag is sent in the response

Ensure that the IUT on receipt of an INVITE request with no TAG set on the To header, sends a Success (200 OK) response including the same URI and an additional TAG for the To header.</td></tr>
</table>

SIP header values

INVITE:

To: <sip:*to-uri*>

200 OK:

To: <sip:*to-uri*>;tag=to_tag

Message flow

Test equipment			End device
Interworking POTS			
INVITE	→		Ringing
200 OK INVITE	←		Off hook
ACK	→		
	Apply post test routine		
ISDN interworking			
INVITE	→	→	SETUP
200 OK INVITE	←	←	CONNECT
ACK	→		
	Apply post test routine		

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_003	Reference section 8.2.6.2 of [IETF RFC 3261]	Selection expression

Test purpose

To tag in the INVITE request

Ensure that the IUT on receipt of an INVITE request with a TAG set on the To header, either:

– sends a Success (200 OK) response including the same URI and the same TAG for the To header (recommended for robustness),

– or reject the INVITE request with a Call/Transaction does not exist (481 Call/Transaction does not exist).

SIP header values

INVITE:

To: <sip:*to_uri_value*>;tag=*to_tag_value*

200 OK:

To: <sip:*to_uri_value*>;tag=*to_tag_value*

Message flow

Test equipment		End device
Interworking POTS		
INVITE	→	Ringing
CASE A		
200 OK INVITE	←	Off hook
ACK	→	
	Apply post test routine	
CASE B		
481 Call/Transaction does not exist	←	
ACK	→	

ISDN interworking			
INVITE	→	→	SETUP
CASE A			
200 OK INVITE	←	←	CONNECT
ACK	→		
	Apply post test routine		
CASE B			
481 Call/Transaction does not exist	←		
ACK	→		

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_004	Reference section 12.1.1 of [IETF RFC 3261]	Selection expression

Test purpose

Contact header in the response.

Ensure that the IUT on receipt of an INVITE request, sends a Success (200 OK) response including a single Contact header.

SIP header values

200 OK:

Contact: <sip: *contact_value*>

Message flow

Test equipment			End device
Interworking POTS			
INVITE	→		Ringing
200 OK INVITE	←		Off hook
ACK	→		
	Apply post test routine		
ISDN interworking			
INVITE	→	→	SETUP
200 OK INVITE	←	←	CONNECT
ACK	→		
	Apply post test routine		

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_005	Reference section 12.1.1 of [IETF RFC 3261]	Selection expression

Test purpose

Record-Route header copied from the INVITE request into the response

Ensure that the IUT on receipt of an INVITE request including a Record-Route header, sends a Success (200 OK) response including a Record-Route header copy from the INVITE request, in the same order.

SIP header values INVITE: Record-Route: <sip:*invite_record_route*>;lr 200 OK: Record-Route: <sip:*invite_record_route*>;lr
Message flow **Test equipment** **End device** **Interworking POTS** INVITE → Ringing 200 OK INVITE ← Off hook ACK → **Apply post test routine** **ISDN interworking** INVITE → → SETUP 200 OK INVITE ← ← CONNECT ACK → **Apply post test routine**

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_006	Reference section 12.1.1 of [IETF RFC 3261]	Selection expression
Test purpose *Record-Route header with unknown parameter* Ensure that the IUT on receipt of an INVITE request including a Record-Route header with parameters that it does not understand, sends a Success (200 OK) response including a Record-Route header copy from the INVITE request, with the unknown parameters.			
SIP header values INVITE: Record-Route: <sip:record-route_value_invite;unknown=etsi> 200 OK: Record-Route: <sip:record-route_value_invite;unknown=etsi>			
Message flow **Test equipment** **End device** **Interworking POTS** INVITE → Ringing 200 OK INVITE ← Off hook ACK → **Apply post test routine** **ISDN interworking** INVITE → → SETUP 200 OK INVITE ← ← CONNECT ACK → **Apply post test routine**			

<table>
<tr><td>TSS
Term_Establishment_of_
a_confirmed_dialogue</td><td>TP_302_007</td><td>Reference
section 12.1.1 of
[IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
From header without "tag"

Ensure that the IUT on receipt of an INVITE request including From header without tag, sends a Success (200 OK) response including a From header without tag.</td></tr>
<tr><td colspan="4">SIP header values
INVITE:
From: <sip:<i>from_value_invite</i>>

200 OK:
From: <sip:<i>from_value_invite</i>></td></tr>
<tr><td colspan="4">Message flow
Test equipment — End device
Interworking POTS
INVITE → Ringing
200 OK INVITE ← Off hook
ACK →
Apply post test routine

ISDN interworking
INVITE → → SETUP
200 OK INVITE ← ← CONNECT
ACK →
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Term_Establishment_of_
a_confirmed_dialogue</td><td>TP_302_008</td><td>Reference
section 13.3.1.4 of
[IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
Allow and a Supported headers sent in response

Ensure that the IUT having received an INVITE request, sends a Success (200 OK) including an Allow and a Supported headers.</td></tr>
<tr><td colspan="4">SIP header values
200 OK:
Allow:
Supported:</td></tr>
<tr><td colspan="4">Message flow
Test equipment — End device
Interworking POTS
INVITE → Ringing
200 OK INVITE ← Off hook
ACK →
Apply post test routine</td></tr>
</table>

ISDN interworking		
INVITE	→	→ SETUP
200 OK INVITE	←	← CONNECT
ACK	→	
	Apply post test routine	

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_009	Reference section 6.3.2.4 of [ETSI TS 183 043]	Selection expression

Test purpose

Modifying SDP in confirmed dialogue

Ensure that the SUT is able to receive an UPDATE request to modify the SDP in confirmed dialogue. A 200 OK UPDATE with SDP answer is sent.

SIP header values

200 OK:

Allow:

Supported:

Message flow

Test equipment		**End device**
Interworking POTS		
INVITE	→	Ringing
180/183	←	
200 OK INVITE	←	Off hook
ACK	→	
CASE A		
INVITE	→	
200 OK INVITE	←	
ACK	→	
CASE B		
UPDATE	→	
200 OK UPDATE	←	
	Apply post test routine	
ISDN interworking		
INVITE	→	→ SETUP
180 Ringing	←	← ALERTING
200 OK INVITE	←	← CONNECT
ACK	→	
CASE A		
INVITE	→	
200 OK INVITE	←	
ACK	→	

CASE B		
UPDATE	→	
200 OK UPDATE	←	
	Apply post test routine	

7.2.3.2.2 Test purposes for ISDN

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_101	Reference subclause 5.1.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1

Test purpose

Sending of 200 OK (INVITE) PSTN XML ProgressIndicator is present

Ensure that on receipt of a DSS1 CONNECT message a SIP 200 OK (INVITE) is sent to the calling user equipment. Ensure that if a Progress Indicator IE is present in the CONNECT message, a PSTN XML ProgressIndicator is present in the 200 OK (INVITE) the ProgressDescription is derived from the Progress Description indicator as indicated in Table 7.2.3.2.2-1. Also, an additional PSTN XML ProgressIndicator is present with the ProgressDescription value set to 7

SIP header values

200 OK (INVITE):

PSTM XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN
- ProgressIndicator
  - ProgressOctet3
    - CodingStandard>00<
    - Location>yyyy<
  - ProgressOctet4
    - ProgressDescription>**0000111**<
- ProgressIndicator
  - ProgressOctet3
    - CodingStandard>00<
    - Location>0000<
  - ProgressOctet4
    - ProgressDescription>**PI_value**<

DSS1 Parameter values

CONNECT: Progress Indicator Progress Description **PI_value**

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→	→	CONNECT ACK
	Apply post test routine		

Table 7.2.3.2.2-1 – Mapping of DSS1 Progress Indicator information to PSTN XML ProgressIndicator element

PI_value	XML ProgressIndicator ProgressDescription	DSS1 Progress Indicator value
PI_VA_1	'0000001'	Call is not end-to-end 5.1.1/2; further call progress information may be available in-band
PI_VA_2	'0000010'	Destination address is non-5.1.1/2
PI_VA_3	'0000011'	Origination address is non-5.1.1/2
PI_VA_4	'0000100'	Call has returned to the 5.1.1/2
PI_VA_5	'0000101'	Interworking has occurred and has resulted in a telecommunication service change
PI_VA_6	'0001000'	In-band information or an appropriate pattern is now available

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_102	Reference subclause 5.1.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1

Test purpose

Sending of 200 OK (INVITE) PSTN XML LowLayerCompatibility is present

Ensure that on receipt of a DSS1 CONNECT message a SIP 200 OK (INVITE) is sent to the calling user equipment. Ensure that if a Low Layer Compatibility IE is present in the CONNECT message, a PSTN XML LowLayerCompatibility is present in the 200 OK (INVITE) the InformationTransferCapability is derived from the Information Transfer Capability indicator as indicated in Table 7.2.3.2.2-2.

SIP header values

200 OK (INVITE):

PSTM XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

LowLayerCompatibility>

LLOctet3>

CodingStandard>00<

InformationTransferCapability>**ITC_VA**<

LLOctet4>

TransferMode>00<

InformationTransferRate>10000<

DSS1 Parameter values

CONNECT: Low Layer Compatibility Information Transfer Capability **ITC_value**

Message flow

Test equipment			End device
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→	→	CONNECT ACK

Apply post test routine

Table 7.2.3.2.2-2 – DSS1 Low Layer Compatibility to Mapping of PSTN XML LowLayerCompatibility

ITC_value	LLC Information transfer capability	XML LLC InformationTransferCabability
ITC_VA_1	Speech	'00000'
ITC_VA_2	3,1 kHz audio	'10000'
ITC_VA_3	Unrestricted digital info	'01001'
ITC_VA_3	7 kHz audio	'10001'

<table>
<tr><td>TSS
Term_Establishment_of_
a_confirmed_dialogue</td><td>TP_302_103</td><td>Reference
subclause 5.1.2.3 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.4/1</td></tr>
<tr><td colspan="4">Test purpose
Sending of 200 OK (INVITE) PSTN XML HighLayerCompatibility is present

Ensure that on receipt of a DSS1 CONNECT message a SIP 200 OK (INVITE) is sent to the calling user equipment. Ensure that if a High Layer Compatibility IE is present in the CONNECT message, a PSTN XML HighLayerCompatibility is present in the 200 OK (INVITE) the HighLayerCharacteristics element is derived from the High Layer Characteristics indicator as indicated in Table 7.2.3.2.2-3.</td></tr>
<tr><td colspan="4">SIP header values
200 OK (INVITE):
<?xml version="1.0" encoding="utf-8"?>
PSTN
    HighLayerCompatibility
        HLOctet3
            CodingStandard>00<
            Interpretation>100<
            PresentationMethod>01<
        HLOctet4
            HighLayerCharacteristics>HLC_value<</td></tr>
<tr><td colspan="4">DSS1 Parameter values
CONNECT: High Layer Compatibility High Layer Characteristics HLC_value</td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><td>Test equipment</td><td></td><td></td><td>End device</td></tr>
<tr><td>INVITE</td><td>→</td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td>←</td><td>ALERTING</td></tr>
<tr><td>200 OK (INVITE)</td><td>←</td><td>←</td><td>CONNECT</td></tr>
<tr><td>ACK</td><td>→</td><td>→</td><td>CONNECT ACK</td></tr>
</table>
Apply post test routine</td></tr>
</table>

Table 7.2.3.2.2-3 – Mapping of DSS1 High layer compatibility information element to PSTN XML HighLayerCharacteristic

HLC_value	DSS1 High layer characteristics identification	XML HighLayerCharacteristic
HLC_VA_1	Telephony	'0000001'
HLC_VA_2	Facsimile Group 2/3	'0000100'
HLC_VA_3	Facsimile Group 4 Class I	'0100001'
HLC_VA_4	Facsimile service Group 4, Classes II ad III	'0100100'
HLC_VA_5	Syntax based Videotex	'0110010'
HLC_VA_6	International Videotex interworking via gateways or interworking units	'0110011'
HLC_VA_7	Telex service	'0110101'
HLC_VA_8	FTAM application	'1000010'
HLC_VA_9	Videotelephony	'1100000'

<table>
<tr><td>TSS
Term_Establishment_of_a_confirmed_dialogue</td><td>TP_302_104</td><td>Reference
subclause 5.1.2.3 of [ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.4/1</td></tr>
<tr><td colspan="4">Test purpose
Sending of 200 OK (INVITE) PSTN XML BearerCapability is present

Ensure that on receipt of a DSS1 CONNECT message a SIP 200 OK (INVITE) is sent to the calling user equipment. Ensure that if a Bearer Capability IE is present in the CONNECT message, a PSTN XML BearerCapability is present in the 200 OK (INVITE) the InformationTransferCabability element is derived from the Information Transfer Capability indicator as indicated in Table 7.2.3.2.2-4.</td></tr>
<tr><td colspan="4">SIP header values
200 OK (INVITE):
<?xml version="1.0" encoding="utf-8"?>
PSTN
  BearerCapability
    BCoctet3
      CodingStandard>00<
      InformationTransferCabability>ITC_value<
    BCoctet4
      TransferMode>00<
      InformationTransferRate>10000<
    BCoctet5
      Layer1Identification>01<
      UserInfoLayer1Protocol>00011<</td></tr>
<tr><td colspan="4">DSS1 Parameter values
CONNECT: Bearer Capability Information Transfer Capability ITC_value</td></tr>
<tr><td colspan="4">Message flow
Test equipment / End device
INVITE → → SETUP
180 Ringing ← ← ALERTING

200 OK (INVITE) ← ← CONNECT
ACK → → CONNECT ACK
Apply post test routine</td></tr>
</table>

Table 7.2.3.2.2-4 – Mapping of Bearer Capability to PSTN XML BearerCapability

ITC_value	XML InformationTransferCabability	BC Information Transfer Capability
ITC_VA_1	'00000'	Speech
ITC_VA_2	'10000'	3,1 kHz audio
ITC_VA_3	'01000'	unrestricted digital information

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_105	Reference subclause 5.1.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1

Test purpose

PSTN XML ProgressIndicator value 7 is sent

Ensure that the PSTN XML ProgressIndicator is present in the 200 OK (INVITE) if no DSS1 Progress Indicator IE was received in the CONNECT message. The ProgressDescription element is set to value 7

SIP header values

200 OK (INVITE):

PSTM XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

    ProgressIndicator

        ProgressOctet3

            CodingStandard>00<

            Location>yyyy<

        ProgressOctet4

            ProgressDescription>**0000111**<

DSS1 Parameter values

Message flow

Test equipment			End device
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→	→	CONNECT ACK

Apply post test routine

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_106	Reference subclause 5.1.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/1

Test purpose

Handling of Fallback connection type at the Coincident S and T reference point, fallback does not occur.

Ensure that on receipt of a Fallback connection type in the initial INVITE a Bearer Capability IE is received in a CONNECT message and the Information Transfer Capability is set to 7 kHz audio. A 200 OK (INVITE) is sent and a PSTN XML BearerCapability element is present the InformationTransferCabability element is set to 7 kHz audio. The first stated codec is the CLEARMODE codec.

SIP header values

INVITE: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>
PSTN
BearerCapability
BCoctet3
CodingStandard>00<
InformationTransferCabability>00000<
or
InformationTransferCabability>10000<
BearerCapability
BCoctet3
CodingStandard>00<
InformationTransferCabability>10001<

SDP: m line contains as the first codec CLEARMODE and as the second codec a G.711 codec

200 OK (INVITE):

<?xml version="1.0" encoding="utf-8"?>
PSTN
BearerCapability
BCoctet3
CodingStandard>00<
InformationTransferCabability>**10001**<

SDP: m line contains as the **first codec CLEARMODE** and as the second codec a G.711 codec

DSS1 Parameter values

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→	→	CONNECT ACK

Apply post test routine

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_107	Reference subclause 5.1.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/2

Test purpose

Handling of Fallback connection type at the Coincident S and T reference point, fallback to speech occurs.

Ensure that on receipt of a Fallback connection type in the initial INVITE a Bearer Capability IE is received in a CONNECT message and the Information Transfer Capability is set to speech. A 200 OK (INVITE) is sent and a PSTN XML BearerCapability element is present the InformationTransferCabability element is set to speech. The first stated codec is not the CLEARMODE codec

SIP header values

INVITE: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

  BearerCapability

    BCoctet3

      CodingStandard>00<

      InformationTransferCabability>00000<

        or

      InformationTransferCabability>10000<

  BearerCapability

    BCoctet3

      CodingStandard>00<

      InformationTransferCabability>10001<

SDP: m line contains as the first codec CLEARMODE and as the second codec a G.711 codec

200 OK (INVITE):

<?xml version="1.0" encoding="utf-8"?>

PSTN

  BearerCapability

    BCoctet3

      CodingStandard>00<

      InformationTransferCabability>**00000**<

DSS1 Parameter values

CONNECT: Bearer Capability Information Transfer Capability =**speech**

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→	→	CONNECT ACK

Apply post test routine

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_108	Reference subclause 5.1.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/2

Test purpose

Handling of Fallback connection type at the Coincident S and T reference point, fallback to audio occurs.

Ensure that on receipt of a Fallback connection type in the initial INVITE a Bearer Capability IE is received in a CONNECT message and the Information Transfer Capability is set to 3.1 kHz audio. A 200 OK (INVITE) is sent and a PSTN XML BearerCapability element is present the InformationTransferCabability element is set to 3.1 kHz audio. The first stated codec is not the CLEARMODE codec.

SIP header values

INVITE: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>
PSTN
BearerCapability
BCoctet3
CodingStandard>00<
InformationTransferCabability>00000<
or
InformationTransferCabability>10000<
BearerCapability
BCoctet3
CodingStandard>00<
InformationTransferCabability>10001<

SDP: m line contains as the first codec CLEARMODE and as the second codec a G.711 codec

200 OK (INVITE):

<?xml version="1.0" encoding="utf-8"?>
PSTN
BearerCapability
BCoctet3
CodingStandard>00<
InformationTransferCabability>**10000**<

DSS1 Parameter values

CONNECT: Bearer Capability Information Transfer Capability =**3.1 kHz audio**

Message flow

Test equipment			**End device**
INVITE	➔	➔	SETUP
180 Ringing	🡐	🡐	ALERTING
200 OK (INVITE)	🡐	🡐	CONNECT
ACK	➔	➔	CONNECT ACK

Apply post test routine

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_109	Reference subclause 5.1.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/2

Test purpose

Handling of Fallback connection type at the Coincident S and T reference point, no BC IE received

Ensure that on receipt of a Fallback connection type in the initial INVITE no Bearer Capability IE is received in a CONNECT message. A 200 OK (INVITE) is sent and a PSTN XML **BearerCapability** element is present the InformationTransferCabability element is set to the value indicated in the INVITE (speech or 2.1 kHz audio). The first stated codec is not the CLEARMODE codec.

SIP header values
INVITE: PSTN XML MIME body
<?xml version="1.0" encoding="utf-8"?>
PSTN
    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>00000<
                or
            InformationTransferCabability>10000<
    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>10001<
SDP: m line contains as the first codec CLEARMODE and as the second codec a G.711 codec
 	
200 OK (INVITE):
<?xml version="1.0" encoding="utf-8"?>
PSTN
    BearerCapability
        BCoctet3
            CodingStandard>00<
            InformationTransferCabability>**10000**<

DSS1 Parameter values
Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→	→	CONNECT ACK
	Apply post test routine		

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_110	Reference subclause 5.1.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/2

Test purpose

Handling of Fallback connection type at the Coincident S and T reference point, HLC IE received

Ensure that on receipt of a Fallback connection type in the initial INVITE and a High Layer Compatibility IE is received in a CONNECT message, a 200 OK (INVITE) is sent and a PSTN XML HighLayerCompatibility is present with the HighLayerCharacteristics element set to the value indicated in Table 7.2.3.2.2-3. The first stated codec is not the CLEARMODE codec.

SIP header values

INVITE: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

  BearerCapability

    BCoctet3

      CodingStandard>00<

      InformationTransferCabability>00000<

        or

      InformationTransferCabability>10000<

  BearerCapability

    BCoctet3

      CodingStandard>00<

      InformationTransferCabability>10001<

SDP: m line contains as the first codec CLEARMODE and as the second codec a G.711 codec

200 OK (INVITE):

<?xml version="1.0" encoding="utf-8"?>

PSTN

  HighLayerCompatibility

    HLOctet3

      CodingStandard>00<

      Interpretation>100<

      PresentationMethod>01<

    HLOctet4

      HighLayerCharacteristics>**HLC_VA**<

DSS1 Parameter values

CONNECT: High Layer Compatibility High Layer Characteristics=HLC_VA

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→	→	CONNECT ACK

Apply post test routine

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_111	Reference subclause 5.1.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/2

Test purpose

Handling of Fallback connection type at the Coincident S and T reference point, HLC IE not received

Ensure that on receipt of a Fallback connection type in the initial INVITE and a High Layer Compatibility IE is not received in a CONNECT message. A 200 OK (INVITE) is sent and a PSTN XML HighLayerCompatibility is present the HighLayerCharacteristics element is set to the value received in the initial INVITE as indicated in Table 7.2.3.2.2-3. The first stated codec is not the CLEARMODE codec

SIP header values

INVITE: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

  BearerCapability

    BCoctet3

      CodingStandard>00<

      InformationTransferCabability>00000<

        or

      InformationTransferCabability>10000<

  BearerCapability

    BCoctet3

      CodingStandard>00<

      InformationTransferCabability>10001<

  HighLayerCompatibility

    HLOctet4

      HighLayerCharacteristics>**HLC_VA**<

SDP: m line contains as the first codec CLEARMODE and as the second codec a G.711 codec

200 OK (INVITE):

<?xml version="1.0" encoding="utf-8"?>

PSTN

  HighLayerCompatibility

..

    HLOctet4

      HighLayerCharacteristics>**HLC_VA**<

DSS1 Parameter values

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→	→	CONNECT ACK

Apply post test routine

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_112	Reference subclause 5.1.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/1

Test purpose

Handling of Fallback connection type at the T reference point, fallback does not occur.

Ensure that on receipt of a Fallback connection type in the initial INVITE a Bearer Capability IE is received in a CONNECT message and the Information Transfer Capability is set to 7 kHz audio. A 200 OK (INVITE) is sent and a PSTN XML BearerCapability element is present the InformationTransferCabability element is set to 7 kHz audio. The first stated codec is the CLEARMODE codec.

SIP header values

INVITE: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

  BearerCapability

    BCoctet3

      CodingStandard>00<

      InformationTransferCabability>00000<

        or

      InformationTransferCabability>10000<

  BearerCapability

    BCoctet3

      CodingStandard>00<

      InformationTransferCabability>10001<

SDP: m line contains as the first codec CLEARMODE and as the second codec a G.711 codec

200 OK (INVITE):

<?xml version="1.0" encoding="utf-8"?>

  BearerCapability

    BCoctet3

      CodingStandard>00<

      InformationTransferCabability>**10001**<

SDP: m line contains as the first codec CLEARMODE and as the second codec a G.711 codec

DSS1 Parameter values

CONNECT: Bearer Capability Information Transfer Capability =7 kHz audio, Progress Indicator Progress description = 5

Message flow

Test equipment			**End device**
INVITE	➔	➔	SETUP
180 Ringing	🡐	🡐	ALERTING
200 OK (INVITE)	🡐	🡐	CONNECT
ACK	➔	➔	CONNECT ACK

Apply post test routine

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_113	Reference subclause 5.1.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/1

Test purpose

Handling of Fallback connection type at the T reference point, fallback to speech occurs.

Ensure that on receipt of a Fallback connection type in the initial INVITE a Bearer Capability IE is received in a CONNECT message and the Information Transfer Capability is set to speech and a ProgressIndicator element is present which is set to value 5. A 200 OK (INVITE) is sent and a PSTN XML BearerCapability element is present the InformationTransferCabability element is set to speech. The first stated codec is not the CLEARMODE codec

SIP header values

INVITE: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

BearerCapability

BCoctet3

CodingStandard>00<

InformationTransferCabability>00000<

or

InformationTransferCabability>10000<

BearerCapability

BCoctet3

CodingStandard>00<

InformationTransferCabability>10001<

SDP: m line contains as the first codec CLEARMODE and as the second codec a G.711 codec

200 OK (INVITE):

<?xml version="1.0" encoding="utf-8"?>

BearerCapability

BCoctet3

CodingStandard>00<

InformationTransferCabability>**00000**<

ProgressIndicator

ProgressOctet4

ProgressDescription>**0000101**<

DSS1 Parameter values

CONNECT: Bearer Capability Information Transfer Capability = speech, Progress Indicator Progress description = 5

Message flow

Test equipment			End device
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→	→	CONNECT ACK

Apply post test routine

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_114	Reference subclause 5.1.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/1

Test purpose

Handling of Fallback connection type at the T reference point, fallback to audio occurs.

Ensure that on receipt of a Fallback connection type in the initial INVITE a Bearer Capability IE is received in a CONNECT message and the Information Transfer Capability is set to 3.1 kHz audio and a ProgressIndicator element is present which is set to value 5. A 200 OK (INVITE) is sent and where a PSTN XML BearerCapability element is present the InformationTransferCabability element is set to 3.1 kHz audio. The first stated codec is not the CLEARMODE codec

SIP header values

INVITE: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

  BearerCapability

    BCoctet3

      CodingStandard>00<

      InformationTransferCabability>00000<

        or

      InformationTransferCabability>10000<

  BearerCapability

    BCoctet3

      CodingStandard>00<

      InformationTransferCabability>10001<

SDP: m line contains as the first codec CLEARMODE and as the second codec a G.711 codec

200 OK (INVITE):

<?xml version="1.0" encoding="utf-8"?>

  BearerCapability

    BCoctet3

      CodingStandard>00<

      InformationTransferCabability>**10000**<

  ProgressIndicator

    ProgressOctet4

      ProgressDescription>**0000101**<

DSS1 Parameter values

CONNECT: Bearer Capability Information Transfer Capability = speech, Progress Indicator Progress description = 5

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→	→	CONNECT ACK

Apply post test routine

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_115	Reference subclause 5.1.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/1

Test purpose

Handling of Fallback connection type at the T reference point, no BC IE received

Ensure that on receipt of a Fallback connection type in the initial INVITE no Bearer Capability IE is received in a CONNECT message. A 200 OK (INVITE) is sent and if a PSTN XML BearerCapability element is present, the InformationTransferCabability element is set to the value indicated in the INVITE (speech or 2.1 kHz audio). The first stated codec is not the CLEARMODE codec

SIP header values

INVITE: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

BearerCapability

BCoctet3

CodingStandard>00<

InformationTransferCabability>00000<

or

InformationTransferCabability>10000<

BearerCapability

BCoctet3

CodingStandard>00<

InformationTransferCabability>10001<

SDP: m line contains as the first codec CLEARMODE and as the second codec a G.711 codec

200 OK (INVITE):

<?xml version="1.0" encoding="utf-8"?>

BearerCapability

BCoctet3

CodingStandard>00<

InformationTransferCabability>**10000**<

ProgressIndicator

ProgressOctet4

ProgressDescription>**0000101**<

DSS1 Parameter values

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→	→	CONNECT ACK

Apply post test routine

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_116	Reference subclause 5.1.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/1

Test purpose

Handling of Fallback connection type at the T reference point, HLC IE received

Ensure that on receipt of a Fallback connection type in the initial INVITE and if a High Layer Compatibility IE is received in a CONNECT message, a 200 OK (INVITE) is sent and a PSTN XML HighLayerCompatibility is present the HighLayerCharacteristics element is set to the value indicated in Table 7.2.3.2.2-3. The first stated codec is not the CLEARMODE codec.

SIP header values

INVITE: PSTN XML MIME body

<?xml version="1.0" encoding="utf-8"?>

PSTN

BearerCapability

BCoctet3

CodingStandard>00<

InformationTransferCabability>00000<

or

InformationTransferCabability>10000<

BearerCapability

BCoctet3

CodingStandard>00<

InformationTransferCabability>10001<

SDP: m line contains as the first codec CLEARMODE and as the second codec a G.711 codec

200 OK (INVITE):

<?xml version="1.0" encoding="utf-8"?>

HighLayerCompatibility

HLOctet3

CodingStandard>00<

Interpretation>100<

PresentationMethod>01<

HLOctet4

HighLayerCharacteristics>**HLC_VA**<

DSS1 Parameter values

CONNECT: High Layer Compatibility, High Layer Characteristics=**HLC_VA**

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→	→	CONNECT ACK

Apply post test routine

TSS Term_Establishment_of_a_confirmed_dialogue	TP_302_117	Reference subclause 5.1.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1 AND 5.1.3/1

Test purpose

Handling of Fallback connection type at the T reference point, HLC IE not received

Ensure that on receipt of a Fallback connection type in the initial INVITE and no High Layer Compatibility IE is received in a CONNECT message. A 200 OK (INVITE) is sent and no PSTN XML HighLayerCompatibility is present. The first stated codec is not the CLEARMODE codec.

SIP header values
INVITE: PSTN XML MIME body
<?xml version="1.0" encoding="utf-8"?>
PSTN
  BearerCapability
    BCoctet3
      CodingStandard>00<
      InformationTransferCabability>00000<
        or
      InformationTransferCabability>10000<
  BearerCapability
    BCoctet3
      CodingStandard>00<
      InformationTransferCabability>10001<
SDP: m line contains as the first codec CLEARMODE and as the second codec a G.711 codec

200 OK (INVITE):

DSS1 Parameter values
CONNECT: High Layer Compatibility, High Layer Characteristics=**HLC_VA**

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→	→	CONNECT ACK

Apply post test routine

7.2.3.3 Call release

7.2.3.3.1 Release initiated by the originating user

TSS Term_Release_initiated_by_the_originating_user	TP_303_001	Reference section 15 of [IETF RFC 3261] and 5.1.2.4 of [ETSI TS 183 036]	Selection expression

Test purpose
BYE received in the confirmed dialogue

Ensure that that the IUT, while a session has been established, on receipt of a BYE request sends a Success (200 OK) response.

SIP header values

Message flow

Test equipment		**End device**
Interworking POTS		
INVITE	→	Ringing
180 Ringing	←	
200 OK INVITE	←	Off hook
ACK	→	
BYE	→	
200 OK BYE	←	

ISDN interworking		
INVITE	→	→ SETUP
180 Ringing	←	← ALERTING
200 OK INVITE	←	← CONNECT
ACK	→	
BYE	→	→ DISCONNECT
200 OK BYE	←	← RELEASE
		→ RELEASE COMPLETE

TSS Term_Release_initiated_ by_the_originating_user	TP_303_002	Reference sections 15 and 12 of [IETF RFC 3261]	Selection expression
Test purpose *BYE received in the early dialogue* Ensure that the IUT, while the dialogue is in an early stage, on receipt of a BYE request sends a response.			
SIP header values			
Message flow			

Test equipment		**End device**
Interworking POTS		
INVITE	→	Ringing
180 Ringing	←	
BYE	→	
200 OK BYE	←	
487 Request Terminated	←	
ACK	→	
ISDN interworking		
INVITE	→	→ SETUP
180 Ringing	←	← ALERTING
BYE	→	→ RELEASE
200 OK BYE	←	← RELEASE COMPLETE
487 Request Terminated	←	
ACK	→	

TSS Term_Release_initiated_ by_the_originating_user	TP_303_003	Reference sections 8.2.2 and 15.1.2/[IETF RFC 3261]	Selection expression
Test purpose *BYE received with unknown header field* Ensure that the IUT, once a dialogue has been established, on receipt of a BYE request including a header that it does not understand sends a Success (200 OK) response.			
SIP header values BYE: Unknown-Header: testing			

Message flow **Test equipment** — **End device** **Interworking POTS** INVITE → Ringing 180 Ringing ← 200 OK INVITE ← ACK → BYE → 200 OK BYE ← **ISDN interworking** INVITE → → SETUP 180 Ringing ← ← ALERTING 200 OK INVITE ← ← CONNECT ACK → BYE → → DISCONNECT 200 OK BYE ← ← RELEASE → RELEASE COMPLETE

TSS Term_Release_initiated_ by_the_originating_user	TP_303_004	Reference sections 8.2.6.2 and 15.1.2 of [IETF RFC 3261]	Selection expression
Test purpose *200 OK BYE contains the same From, Call-Id, CSeq and Via header as in the request* Ensure that the IUT, once a dialogue has been established, on receipt of a BYE request, sends a Success (200 OK) response with From, Call-ID, CSeq and Via headers set to the same value as in the request.			
SIP header values BYE: Via: *any_bye_via_value*;branch= *z9hG4bK any_bye_branch_value* From: *any_bye_from_value*;tag=*any_ bye_from_tag_value* Call-ID: *any_bye_call-id_value* CSeq: *any_bye_cseq* 200 OK BYE Via: *any_bye_via_value*;branch= *z9hG4bK any_bye_branch_value* From: *any_bye_from_value*;tag=*any_ bye_from_tag_value* Call-ID: *any_bye_call-id_value* CSeq: *any_bye_cseq*			
Message flow **Test equipment** — **End device** **Interworking POTS** INVITE → Ringing 180 Ringing ← 200 OK INVITE ← ACK → BYE → 200 OK BYE ←			

ISDN interworking INVITE → → SETUP 180 Ringing ← ← ALERTING 200 OK INVITE ← ← CONNECT ACK → BYE → → DISCONNECT 200 OK BYE ← ← RELEASE → RELEASE COMPLETE

TSS Term_Release_initiated_by_the_originating_user	TP_303_005	Reference section 9.2 of [IETF RFC 3261]	Selection expression
Test purpose *CANCEL received in the early dialogue* Ensure that the IUT when a server INVITE transaction is in the Proceeding state, on receipt of a CANCEL answers to the original INVITE request with a Request Terminated (487 Request Terminated) response.			
SIP header values			
Message flow **Test equipment** **End device** **Interworking POTS** INVITE → Ringing 180 Ringing ← CANCEL → 200 OK CANCEL ← 487 Request Terminated ← ACK → **ISDN interworking** INVITE → → SETUP 180 Ringing ← ← ALERTING CANCEL → → RELEASE/RELEASE COMPLETE 200 OK CANCEL ← 487 Request Terminated ← ACK →			

7.2.3.3.1.2 Test purposes for ISDN

TSS Term_Release_initiated_by_the_originating_user	TP_303_101	Reference subclause 5.1.2.4 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2
Test purpose *BYE received in the confirmed dialogue* Ensure that on receipt of a BYE request in the confirmed dialogue a DISCONNECT message is sent to the called user equipment. The location is coded '1010' network beyond interworking point.			
SIP header values			

DSS1 Parameter values DISCONNECT: location='1010' (network beyond interworking point)
Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→	→	CONNECT ACK
BYE	→	→	DISCONNECT
200 OK BYE	←	←	RELEASE
		→	RELEASE COMPLETE

Apply post test routine

TSS Term_Release_initiated_ by_the_originating_user	TP_303_102	Reference subclause 5.1.2.4 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2

Test purpose

BYE received before an early dialogue is established

Ensure that on receipt of a BYE request, before an early dialogue is established, a RELEASE or RELEASE COMPLETE message is sent to the called user equipment. The location is coded '1010' network beyond interworking point.

SIP header values

RELEASE or RELEASE COMPLETE: location='1010' (network beyond interworking point)

DSS1 Parameter values

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
100 Trying	←		
BYE	→		
200 OK BYE	←		
CASE A		→	**RELEASE**
		←	RELEASE COMPLETE
CASE B			
		→	RELEASE COMPLETE
487 Request Terminated	←		
ACK	→		

TSS Term_Release_initiated_by_the_originating_user	TP_303_103	Reference subclause 5.1.2.4 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2

Test purpose

CANCEL received before an early dialogue is established

Ensure that on receipt of a CANCEL request, before an early dialogue is established, a RELEASE or RELEASE COMPLETE message is sent to the called user equipment. The location is coded '1010' network beyond interworking point.

SIP header values

RELEASE or RELEASE COMPLETE: location='1010' (network beyond interworking point)

DSS1 Parameter values

Message flow

Test equipment			End device
INVITE	→	→	SETUP
100 Trying	←		
CANCEL	→		
200 OK CANCEL	←		
CASE A		→	**RELEASE**
		←	RELEASE COMPLETE
CASE B			
		→	RELEASE COMPLETE
487 Request Terminated	←		
ACK	→		

TSS Term_Release_initiated_by_the_originating_user	TP_303_104	Reference subclause 5.1.2.4 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2

Test purpose

BYE received after an early dialogue is established

Ensure that on receipt of a BYE request, after an early dialogue by means of sending a 183 Session Progress is established, a RELEASE or RELEASE COMPLETE message is sent to the called user equipment. The location is coded '1010' network beyond interworking point.

SIP header values

RELEASE or RELEASE COMPLETE: location='1010' (network beyond interworking point)

DSS1 Parameter values

Message flow

Test equipment			End device
INVITE	→	→	SETUP
183 Session Progress	←	←	CALL PROCEEDING
BYE	→		
200 OK BYE	←		

CASE B		→	**RELEASE**
		←	RELEASE COMPLETE
CASE C			
		→	RELEASE COMPLETE
487 Request Terminated	←		
ACK	→		

TSS Term_Release_initiated_by_the_originating_user	TP_303_105	Reference subclause 5.1.2.4 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2

Test purpose

CANCEL received after an early dialogue is established

Ensure that on receipt of a CANCEL request, after an early dialogue by means of sending a 183 Session Progress is established, a RELEASE or RELEASE COMPLETE message is sent to the called user equipment. The location is coded '1010' network beyond interworking point.

SIP header values

RELEASE or RELEASE COMPLETE: location='1010' (network beyond interworking point)

DSS1 Parameter values

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
183 Session Progress	←	←	CALL PROCEEDING
CANCEL	→		
200 OK CANCEL	←		
CASE A		→	**RELEASE**
		←	RELEASE COMPLETE
CASE B			
		→	RELEASE COMPLETE
487 Request Terminated	←		
ACK	→		

TSS Term_Release_initiated_by_the_originating_user	TP_303_106	Reference subclause 5.1.2.4 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2

Test purpose

BYE received after an early dialogue is established

Ensure that on receipt of a BYE request, after an early dialogue by means of sending a 180 Ringing is established, a RELEASE or RELEASE COMPLETE message is sent to the called user equipment. The location is coded '1010' network beyond interworking point.

SIP header values RELEASE or RELEASE COMPLETE: location='1010' (network beyond interworking point)
DSS1 Parameter values

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
183 Session Progress	←	←	CALL PROCEEDING
BYE	→		
200 OK BYE	←		
CASE B			
		→	**RELEASE**
		←	RELEASE COMPLETE
CASE C			
		→	RELEASE COMPLETE
487 Request Terminated	←		
ACK	→		

TSS Term_Release_initiated_ by_the_originating_user	TP_303_107	Reference subclause 5.1.2.4 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2

Test purpose

CANCEL received after an early dialogue is established

Ensure that on receipt of a CANCEL request, after an early dialogue by means of sending a 180 Ringing is established, a RELEASE or RELEASE COMPLETE message is sent to the called user equipment. The location is coded '1010' network beyond interworking point.

SIP header values

RELEASE or RELEASE COMPLETE: location='1010' (network beyond interworking point)

DSS1 Parameter values

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
183 Session Progress	←	←	CALL PROCEEDING
CANCEL	→		
200 OK CANCEL	←		
CASE A		→	**RELEASE**
		←	RELEASE COMPLETE
CASE B			
		→	RELEASE COMPLETE
487 Request Terminated	←		
ACK	→		

7.2.3.3.2 Release initiated by the terminating user

TSS Term_Release_initiated_ by_the_terminating_user	TP_304_001	Reference sections 17.2.3.1, 13.3.1.4 and Figure 7 of [IETF RFC 3261]	Selection expression
Test purpose *The transaction enters the completed state* Ensure that the IUT when a server INVITE transaction is in the Proceeding state, after sending a unsuccessful final response, enters in the Completed state.			
SIP header values			
Message flow **Test equipment** **End device** **Interworking POTS** INVITE → Ringing 180 Ringing ← 4xx response ← Reject call **Verify completed state** **ISDN interworking** INVITE → → SETUP 180 Ringing ← ← ALERTING 4xx response ← ← RELEASE_COMPLETE **Verify completed state**			

TSS Term_Release_initiated_ by_the_terminating_user	TP_304_002	Reference sections 17.2.3.1, 13.3.1.4 and Figure 7of [IETF RFC 3261]	Selection expression
Test purpose *The transaction enters the confirmed state* Ensure that the IUT when a server INVITE transaction is in the Completed state, on receipt of an ACK request, enters in the Confirmed transaction state			
SIP header values			
Message flow **Test equipment** **End device** **Interworking POTS** INVITE → Ringing 180 Ringing ← 4xx_response ← Reject call ACK → **Verify confirmed state** **ISDN interworking** INVITE → → SETUP 180 Ringing ← ← ALERTING 4xx_response ← ← RELEASE_COMPLETE ACK → **Verify confirmed state**			

TSS Term_Release_initiated_ by_the_terminating_user	TP_304_003	Reference section 12.2.1.1 of [IETF RFC 3261]	Selection expression

Test purpose

To header in the BYE is set to the From header in the previous received request

Ensure that the IUT, once a dialogue has been established to release it, sends a BYE request with a To header set to the same value as in the From header of the previous received request.

SIP header values

INVITE

From: *any_invite_from_value*;tag=*any_invite1_from_tag_value*

BYE:

To: *any_invite_from_value*;tag=*any_invite1_from_tag_value*

Message flow

Test equipment			**End device**
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
200 OK INVITE	←		Off hook
ACK	→		
BYE	←		On hook
200 OK BYE	→		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→		
BYE	←	←	RELEASE
200 OK BYE	→	→	RELEASE COMPLETE

TSS Term_Release_initiated_ by_the_terminating_user	TP_304_004	Reference section 12.2.1.1/ [IETF RFC 3261]	Selection expression

Test purpose

BYE is sent with the From header set to value set in the last sent response

Ensure that the IUT, once a dialogue has been established to release, it sends a BYE request with a From header set to the same value as in the To header of the last sent response.

SIP header values

180:

To: *any_180_to_value*;tag=*any_180_to_tag_value*

BYE:

From: *any_180_to_value*;tag=*any_180_to_tag_value*

Message flow

Test equipment			End device
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
200 OK INVITE	←		Off hook
ACK	→		
BYE	←		On hook
200 OK BYE	→		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→		
BYE	←	←	RELEASE
200 OK BYE	→	→	RELEASE COMPLETE

TSS Term_Release_initiated_ by_the_terminating_user	TP_304_005	Reference section 12.2.1.1 of [IETF RFC 3261]	Selection expression

Test purpose

Request line in the sent BYE is set to the value of the Contact header in the received request

Ensure that the IUT, once a dialogue has been established with an INVITE request including no Record-Route header set to release it, sends a BYE request with the Request-URI set to the Contact URI included in the original INVITE request and with no Route header set.

SIP header values

INVITE:

Contact: <*any_invite_contact_value*>

BYE:

BYE sip: *any_invite_contact_value* SIP/2.0

Message flow

Test equipment			End device
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
200 OK INVITE	←		Off hook
ACK	→		
BYE	←		On hook
200 OK BYE	→		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→		
BYE	←	←	RELEASE
200 OK BYE	→	→	RELEASE COMPLETE

TSS Term_Release_initiated_ by_the_terminating_user	TP_304_006	Reference section 12.2.1.1 of [IETF RFC 3261]	Selection expression

Test purpose

Request line in the sent BYE is set to the value of the Contact header in the received request

Ensure that the IUT, once a dialogue has been established with an INVITE request including a Record-Route header set to a list in which the last element does not contain a lr parameter to release the call, sends a BYE request with the Request-URI set to the Contact URI and a Route header set to the list in a reverse order of the Record-Route included in the original INVITE request.

SIP header values

INVITE:

Contact: <*any_invite_contact_value*>
Record-Route: <sip:*any_invite_value1*>
Record-Route: <sip:*any_invite_value2;lr*>

BYE:

BYE sip: *any_invite_contact_value* SIP/2.0
Route: <sip:*any_invite_value2;lr*>
Route: <sip:*any_invite_value1*>

Message flow

Test equipment			**End device**
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
200 OK INVITE	←		Off hook
ACK	→		
BYE	←		On hook
200 OK BYE	→		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→		
BYE	←	←	RELEASE
200 OK BYE	→	→	RELEASE COMPLETE

TSS Term_Release_initiated_ by_the_terminating_user	TP_304_007	Reference section 12.2.1.1 of [IETF RFC 3261]	Selection expression

Test purpose

Request line in the sent BYE is set to the value of the Record-Route header in the received request

Ensure that the IUT, once a dialogue has been established with an INVITE request including a Record-Route header set to a list in which the last element contains a lr parameter to release the call, sends a BYE request with the Request-URI set to this element and a Route header set to the remainder list in a reverse order of the received Record-Route appended with the received Contact URI.

SIP header values INVITE: Contact: <*any_invite_contact_value*> Record-Route: <sip:*any_invite_value1;lr*> Record-Route: <sip:*any_invite_value2*> BYE: BYE sip: *any_invite_value1* SIP/2.0 Route: <sip:*any_invite_value2*> Route: <sip: *any_invite_contact_value*>

Message flow

Test equipment			End device
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
200 OK INVITE	←		Off hook
ACK	→		
BYE	←		On hook
200 OK BYE	→		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→		
BYE	←	←	RELEASE
200 OK BYE	→	→	RELEASE COMPLETE

7.2.3.3.2.1 Test purposes for ISDN

TSS Term_Release_initiated_by_the_terminating_user	TP_304_101	Reference sections 17.2.3.1, 13.3.1.4 and Figure 7 of [IETF RFC 3261] Subclause 5.1.2.5/ [ETSI TS 183 036]	Selection expression

Test purpose

Session terminated by the called party before an early dialogue is established

Ensure that a final response is sent to the calling user equipment if a DSS1 RELEASE or RELEASE COMPLETE is received from the called user equipment before a Provisional response was sent to the calling user equipment. The Status code to be sent is determined by examining the Cause code value received in the RELEASE or RELEASE COMPLETE message as indicated in Table 7.2.3.3.2.2-1.

SIP header values

Status code : SIP_final_response_VA

DSS1 Parameter values

RELEASE:
Cause - Cause value
or
RELEASE COMPLETE:
Cause - Cause value

Message flow			
Test equipment			**End device**
INVITE	→	→	SETUP
100 Trying	←		
CASE A			
SIP_final_response_VA	←	←	RELEASE
ACK	→	→	RELEASE COMPLETE
CASE B			
SIP_final_response_VA	←	←	RELEASE COMPLETE
ACK	→		

TSS Term_Release_initiated_by_the_terminating_user	TP_304_102	Reference sections 17.2.3.1, 13.3.1.4 and Figure 7 of [IETF RFC 3261] Subclause 5.1.2.5 of [ETSI TS 183 036]	Selection expression

Test purpose

Session terminated by the called party after an early dialogue (183) is established

Ensure that a final response is sent to the calling user equipment if a DSS1 RELEASE or RELEASE COMPLETE is received from the called user equipment after a 183 Session Progress provisional response was sent to the calling user equipment. The Status code to be sent is determined by examining the Cause code value received in the RELEASE or RELEASE COMPLETE message as indicated in Table 7.2.3.3.2.2-2.

SIP header values

Status code

DSS1 Parameter values

RELEASE:
Cause - Cause value
or
RELEASE COMPLETE:
Cause - Cause value

Message flow			
Test equipment			**End device**
INVITE	→	→	SETUP
183 Session Progress	←	←	CALL PROCEEDING / PROGRESS
CASE A			
SIP_final_response_VA	←	←	RELEASE
ACK	→	→	RELEASE COMPLETE
CASE B			
SIP_final_response_VA	←	←	RELEASE COMPLETE
ACK	→		

TSS Term_Release_initiated_by_the_terminating_user	TP_304_103	Reference sections17.2.3.1, 13.3.1.4 and Figure 7 of [IETF RFC 3261] Subclause 5.1.2.5 of [ETSI TS 183 036]	Selection expression

Test purpose

Session terminated by the called party before an early dialogue is established

Ensure that a final response is sent to the calling user equipment if a DSS1 RELEASE or RELEASE COMPLETE is received from the called user equipment before a provisional response was sent to the calling user equipment. The Status code to be sent is determined by examining the Cause code value received in the RELEASE or RELEASE COMPLETE message as indicated in Table 7.2.3.3.2.2-2.

SIP header values

Status code

DSS1 Parameter values

RELEASE:

Cause - Cause value

or

RELEASE COMPLETE:

Cause - Cause value

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
100 Trying	←	←	ALERTING
CASE A			
SIP_final_response_VA	←	←	RELEASE
ACK	→	→	RELEASE COMPLETE
CASE B			
SIP_final_response_VA	←	←	RELEASE COMPLETE
ACK	→		

Table 7.2.3.3.2.2-1 – Interworking of release causes to SIP status codes

	←SIP final response	← DISCONNECT, RELEASE, RELEASE COMPLETE
SIP_final_response_VA	**Status code**	**Cause value**
VA_01	404 Not Found	Cause value No. 1 (unallocated (unassigned) number)
VA_02	486 Busy Here	Cause value No. 17 (user busy)
VA_03	480 Temporarily unavailable	Cause value No 18 (no user responding)
VA_04	480 Temporarily unavailable	Cause value No 19 (no answer from the user)
VA_05	480 Temporarily unavailable	Cause value No. 20 (subscriber absent)
VA_06	603 Decline	Cause value No 21 (call rejected)
VA_07	502 Bad Gateway	Cause value No 27 (destination out of order)
VA_08	484 Address Incomplete	Cause value No. 28 invalid number format (address incomplete)
VA_09	501 (Not Implemented)	Cause value No 29 (facility rejected)
VA_10	480 Temporarily unavailable	Cause value No 31 (normal unspecified) (class default) (Note 1)
VA_11	486 Busy here if CCBS-T-Available invoke component is present) else 503 Service Unavailable	Cause value in the Class 010 (resource unavailable, Cause value No 34)
VA_12	500 Server Internal error	Cause value No 38 (Network out of order)
VA_13	503 Service Unavailable	Cause value No 41 (Temporary failure)
VA_14	500 Server Internal error	Cause value No 43 (Access information discarded)
VA_15	503 Service Unavailable	Cause value No 44 (Requested channel not available)
VA_16	500 Server Internal error	Cause value No 46 (Precedence call blocked)
VA_17	503 Service Unavailable	Cause value No 47 (Resource unavailable, unspecified) (class default)
VA_18	488 Not acceptable here	Cause value No 50 (requested facility no subscribed)
VA_19	603 Decline	Cause value No 57 (bearer capability not authorised)
VA_20	503 Service Unavailable	Cause value No 58 (bearer capability not presently)
VA_21	501 (Not Implemented)	Cause value No 63 (service option not available, unspecified) (class default)
VA_22	500 Server Internal error	Cause value No 65 (Bearer capability not implemented)
VA_23	501 Not Implemented	Cause value No 69 (Requested facility not implemented)
VA_24	501 Not Implemented	Cause value No 70 (Only restricted digital information capability is available)
VA_25	501 Not Implemented	Cause value No 79 (Service or option not implemented,unspecified) (class default)
VA_26	606 Not Acceptable	Cause value No 88 (incompatible destination)
VA_27	513 Message too large	Cause value No 95 (invalid message) (class default)
VA_28	480 Temporarily unavailable	Cause value No. 102 (recovery on timer expiry)
VA_29	501 Not Implemented	Cause value No 110 (Message with unrecognised Parameter, discarded)
VA_30	400 Bad Request	Cause value No. 111 (protocol error, unspecified) (class default)
VA_31	500 Server Internal error	Cause value No. 127 (interworking unspecified) (class default)

Table 7.2.3.3.2.2-2 – Interworking of Release causes to SIP status codes

	←SIP final response	← DISCONNECT, RELEASE, RELEASE COMPLETE
SIP_final_response_VA	**Status code**	**Cause value**
VA_01	486 Busy Here	Cause value No. 17 (user busy)
VA_02	480 Temporarily unavailable	Cause value No 18 (no user responding)
VA_03	603 Decline	Cause value No 21 (call rejected), Location = 000 / user (U)
VA_04	502 Bad Gateway	Cause value No 27 (destination out of order)
VA_05	484 Address Incomplete	Cause value No. 28 invalid number format (address incomplete)
VA_06	480 Temporarily unavailable	Cause value No 31 (normal unspecified) (class default) (Note 1)
VA_07	486 Busy here if CCBS-T-Available invoke component is present) else 503 Service Unavailable	Cause value in the Class 010 (resource unavailable, Cause value No 34)
VA_08	503 Service Unavailable	Cause value No 41 (Temporary failure)
VA_09	500 Server Internal error	Cause value No 43 (Access information discarded)
VA_10	503 Service Unavailable	Cause value No 47 (Resource unavailable, unspecified) (class default)
VA_11	501 (Not Implemented)	Cause value No 63 (service option not available, unspecified) (class default)
VA_12	606 Not Acceptable	Cause value No 88 (incompatible destination)
VA_13	500 Server Internal error	Cause value No. 127 (interworking unspecified) (class default)

7.2.3.4 Timers

TSS Term_Timers	TP_305_001	Reference section 17.2.1 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/1
Test purpose *Timeout timer G 200 OK is repeated* Ensure that if an unreliable transport is used, the IUT, when an INVITE server transaction is in the Completed state repeats its response on the timeout condition of timer G set with a value of T1.			
SIP header values			
Message flow			

End device			**Test equipment**
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
200 OK INVITE	←	Start timer G	On hook
200 OK INVITE	←	Timeout timer G	
ACK	→		

ISDN interworking				
INVITE	→		→	SETUP
180 Ringing	←		←	ALERTING
200 OK INVITE	←	Start timer G	←	CONNECT
200 OK INVITE	←	Timeout timer G		
ACK	→			
		Apply post test routine		

TSS Term_Timers	TP_305_002	Reference section 17.2.1 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/2
Test purpose *Timeout timer G 200 OK is not repeated* Ensure that if a reliable transport (TCP) is used, the IUT, when an INVITE server transaction is in the Completed state does not repeat its response on the timeout condition of timer G set with a value of T1.			
SIP header values			
Message flow			

End device				**Test equipment**
Interworking POTS				
INVITE	→			Ringing
180 Ringing	←			
200 OK INVITE	←	Start timer G		On hook
		Timeout timer G		
ACK	→			
ISDN interworking				
INVITE	→		→	SETUP
180 Ringing	←		←	ALERTING
200 OK INVITE	←	Start timer G	←	CONNECT
		Timeout timer G		
ACK	→			
		Apply post test routine		

TSS Term_Timers	TP_305_003	Reference section 17.2.1 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/1
Test purpose *Timer G is started 2 time* Ensure that if an unreliable transport is used, the IUT, when an INVITE server transaction is in the Completed state and having already sent its response twice, repeats it after timer G set MIN(2*T1,T2) value expires.			
SIP header values			

Message flow				
End device				**Test equipment**
Interworking POTS				
INVITE	→			Ringing
180 Ringing	←			
200 OK INVITE	←	Start timer G		On hook
200 OK INVITE	←	Timeout timer G		
		Start timer G		
200 OK INVITE	←	Timeout timer G		
ACK	→			
ISDN interworking				
INVITE	→		→	SETUP
180 Ringing	←		←	ALERTING
200 OK INVITE	←	Start timer G	←	CONNECT
200 OK INVITE	←	Timeout timer G		
		Start timer G		
200 OK INVITE	←	Timeout timer G		
ACK	→			
		Apply post test routine		

TSS Term_Timers	TP_305_004	Reference section 17.2.1 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/1
Test purpose *Timer G is started 3 time* Ensure that if an unreliable transport is used, the IUT, when an INVITE server transaction is in the Completed state and having already sent its response three times, repeats it after timer G set the MIN(4*T1,T2) value expires.			
SIP header values			

Message flow				
End device				**Test equipment**
Interworking POTS				
INVITE	→			Ringing
180 Ringing	←			
200 OK INVITE	←	Start timer G		On hook
200 OK INVITE	←	Timeout timer G		
		Start timer G		
200 OK INVITE	←	Timeout timer G		
		Start timer G		
200 OK INVITE	←	Timeout timer G		
ACK	→			

ISDN interworking			
INVITE	→		→ SETUP
180 Ringing	←		← ALERTING
200 OK INVITE	←	Start timer G	← CONNECT
200 OK INVITE	←	Timeout timer G	
		Start timer G	
200 OK INVITE	←	Timeout timer G	
		Start timer G	
200 OK INVITE	←	Timeout timer G	
ACK	→		
		Apply post test routine	

TSS Term_Timers	TP_305_005	Reference section 17.2.1 and Annex A of [IETF RFC 3261]	Selection expression

Test purpose

Timeout timer H an ACK is sent

Ensure that the IUT, when an INVITE server transaction is in the Completed state, it enters the Terminated state after the timer H, which is set to value 64*T1, expires.

SIP header values

Message flow

End device			**Test equipment**
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
200 OK INVITE	←	Start timer H	On hook
		Timeout timer H	
ACK	→		
481 Call/Transaction does Not Exist			
ISDN interworking			
INVITE	→		→ SETUP
180 Ringing	←		← ALERTING
200 OK INVITE	←	Start timer H	← CONNECT
		Timeout timer H	
ACK	→		
481 Call/Transaction does Not Exist	←		→ DISCONNECT
			← RELEASE
			→ RELEASE COMPLETE

TSS Term_Timers	TP_305_006	Reference section 17.2.1 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/1

Test purpose

Timeout timer H an ACK is sent

Ensure that the IUT, when an INVITE server transaction is in the Completed state, does not repeat its response after timer H, which is set to 64*T1, expires.

SIP header values

Message flow

End device				Test equipment
Interworking POTS				
INVITE	→			Ringing
180 Ringing	←			
200 OK INVITE	←	Start timer H		On hook
200 OK INVITE	←			
		Timeout timer H		
ACK	→			
481 Call/Transaction does Not Exist	←			
ISDN interworking				
INVITE	→		→	SETUP
180 Ringing	←		←	ALERTING
200 OK INVITE	←	Start timer H	←	CONNECT
200 OK INVITE	←			
		Timeout timer H		
ACK	→		→	DISCONNECT
481 Call/Transaction does Not Exist	←		←	RELEASE
			→	RELEASE COMPLETE

TSS Term_Timers	TP_305_007	Reference section 17.2.1 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/1

Test purpose

Timeout timer I an ACK is sent

Ensure that the IUT, when the IUT, when an INVITE server transaction is in the Confirmed state, enters in the Terminated state after timer I set to T4 value expires

SIP header values

Message flow			
End device			**Test equipment**
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
200 OK INVITE	←		On hook
ACK	→	Start timer I	
		Timeout timer I	
ACK	→		
481 Call/Transaction does Not Exist	←		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→	Start timer I	
		Timeout timer I	
ACK	→		
481 Call/Transaction does Not Exist	←	→	DISCONNECT
		←	RELEASE
		→	RELEASE COMPLETE

TSS Term_Timers	TP_305_008	Reference section 17.2.1 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/2
Test purpose *Timeout timer I an ACK is sent* Ensure that the IUT, when an INVITE server transaction is in the Confirmed state, enters immediately in the Terminated state.			
SIP header values			
Message flow			

End device			**Test equipment**
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
200 OK INVITE	←		On hook
ACK	→	Start timer I	
		Timeout timer I	
ACK	→		
481 Call/Transaction does Not Exist	←		

ISDN interworking

INVITE	→		→	SETUP
180 Ringing	←		←	ALERTING
200 OK INVITE	←		←	CONNECT
ACK	→	Start timer I		
		Timeout timer I		
ACK	→			
481 Call/Transaction does Not Exist	←		→	DISCONNECT
			←	RELEASE
			→	RELEASE COMPLETE

TSS Term_Timers	TP_305_009	Reference section 13.3.1.4 and Annex A of [IETF RFC 3261]	Selection expression

Test purpose

200 OK INVITE received Timer T1 is started

Ensure that the IUT, when it has answered to an INVITE request with 2XX response, repeats it after T1 duration expires without receiving an ACK request.

SIP header values

Message flow

End device				**Test equipment**
Interworking POTS				
INVITE	→			Ringing
180 Ringing	←			
200 OK INVITE	←	Start timer T1		On hook
200 OK INVITE	←	Timeout timer T1		
ACK	→			
		Apply post test routine		

ISDN interworking

INVITE	→		→	SETUP
180 Ringing	←		←	ALERTING
200 OK INVITE	←	Start timer T1	←	CONNECT
200 OK INVITE	←	Timeout timer T1		
ACK	→			
		Apply post test routine		

TSS Term_Timers	TP_305_010	Reference section 13.3.1.4 and Annex A of [IETF RFC 3261]	Selection expression

Test purpose

200 OK INVITE received Timer T1 is started 2 time

Ensure that the IUT, when it has already answered twice to an INVITE request with a 2XX response, repeats it after 2*T1 duration expires without receiving an ACK request.

SIP header values

Message flow

End device				**Test equipment**
Interworking POTS				
INVITE	🡒			Ringing
180 Ringing	🡐			
200 OK INVITE	🡐	Start timer T1		On hook
200 OK INVITE	🡐	Timeout timer T1		
		Start timer 2*T1		
200 OK INVITE	🡐	Timeout timer T1		
ACK	🡒			
		Apply post test routine		
ISDN interworking				
INVITE	🡒		🡒	SETUP
180 Ringing	🡐		🡐	ALERTING
200 OK INVITE	🡐	Start timer T1	🡐	CONNECT
200 OK INVITE	🡐	Timeout timer T1		
		Start timer 2*T1		
200 OK INVITE	🡐	Timeout timer T1		
ACK	🡒			
		Apply post test routine		

TSS Term_Timers	TP_305_011	Reference section 13.3.1.4 and Annex A of [IETF RFC 3261]	Selection expression

Test purpose

Timer T1 was started 64 time

Ensure that the IUT, does not repeat its 2XX response to an INVITE request after 64*T1 duration expires without receiving an ACK request.

SIP header values

Message flow

End device				**Test equipment**
Interworking POTS				
INVITE	→			Ringing
180 Ringing	←			
200 OK INVITE	←	Start timer T1		On hook
200 OK INVITE	←	Timeout timer T1		
		Start timer 64*T1		
		Timeout timer T1		
		Apply post test routine		
ISDN interworking				
INVITE	→		→	SETUP
180 Ringing	←		←	ALERTING
200 OK INVITE	←	Start timer T1	←	CONNECT
200 OK INVITE	←	Timeout timer T1		
		Start timer 64*T1		
		Timeout timer T1		
		Apply post test routine		

TSS Term_Timers	TP_305_012	Reference section 13.3.1.4 and Annex A of [IETF RFC 3261]	Selection expression

Test purpose

*A BYE is sent after expiry 64*T1*

Ensure that the IUT, when it has received no ACK to its 2XX responses during a duration of 64*T1 seconds, sends a BYE request.

SIP header values

Message flow

End device				**Test equipment**
Interworking POTS				
INVITE	→			Ringing
180 Ringing	←			
200 OK INVITE	←	Start timer T1		On hook
200 OK INVITE	←	Timeout timer T1		
		Start timer 64*T1		
		Timeout timer T1		
BYE	←		→	DISCONNECT
200 OK BYE	→		←	RELEASE
			→	RELEASE COMPLETE

ISDN interworking				
INVITE	→		→	SETUP
180 Ringing	←		←	ALERTING
200 OK INVITE	←	Start timer T1	←	CONNECT
200 OK INVITE	←	Timeout timer T1		
		Start timer 64*T1		
		Timeout timer T1		
BYE	←		→	DISCONNECT
200 OK BYE	→		←	RELEASE
			→	RELEASE COMPLETE

TSS Term_Timers	TP_305_013	Reference section 17.2.2 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/1
Test purpose *BYE is retransmitted after exiry of 64*T1* Ensure that the IUT, when a BYE server transaction is in the Completed state, on receipt of the repetitions of the BYE request, retransmits its response until the timer J set to 64*T1 expires.			
SIP header values			

Message flow

End device				**Test equipment**
Interworking POTS				
INVITE	→			Ringing
180 Ringing	←			
200 OK INVITE	←			On hook
ACK	→			
BYE	→	Start timer J 64*T1		
200 OK BYE	←			
		Wait 31.5 sec		
BYE	→			
200 OK BYE	←			
ISDN interworking				
INVITE	→		→	SETUP
180 Ringing	←		←	ALERTING
200 OK INVITE	←		←	CONNECT
ACK	→			
BYE	→	Start timer J 64*T1	→	DISCONNECT
200 OK BYE	←		←	RELEASE
			→	RELEASE COMPLETE
BYE	→	Wait 31.5 sec		
200 OK BYE	←			

TSS Term_Timers	TP_305_014	Reference section 17.2.2 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/1

Test purpose

Timeout timer J

Ensure that the IUT, when a BYE server transaction is in the Completed state, on receipt of the repetitions of the BYE request, retransmits its response until the timer J set to 64*T1 expires. After expiry of timer J the 200 OK response is not retransmitted.

SIP header values

Message flow

End device				Test equipment
Interworking POTS				
INVITE	→			Ringing
180 Ringing	←			
200 OK INVITE	←			On hook
ACK	→			
BYE	→	Start timer J 64*T1		
200 OK BYE	←			
		Timeout timer J		
BYE	→			
Case A				
CASE B				
481 Call/Transaction does Not Exist	←			
ISDN interworking				
INVITE	→		→	SETUP
180 Ringing	←		←	ALERTING
200 OK INVITE	←		←	CONNECT
ACK	→			
BYE	→	Start timer J 64*T1	→	DISCONNECT
200 OK BYE	←		←	RELEASE
			→	RELEASE COMPLETE
		Timeout timer J		
BYE	→			
Case A				
CASE B				
481 Call/Transaction does Not Exist	←			

7.2.3.5 Abnormal situations

<table>
<tr><td>TSS
Term_Abnormal_situations</td><td>TP_306_001</td><td>Reference
section 8.2.2.1 of
[IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
A call setup is rejected due to an unknown IRU scheme

Ensure that the IUT on receipt of an INVITE request with a Request-URI set with a scheme that it does not support, sends an Unsupported URI scheme (416 Unsupported URI scheme) response.</td></tr>
<tr><td colspan="4">SIP header values
INVITE:
INVITE got:<any destination URI> SIP/2.0</td></tr>
<tr><td colspan="4">Message flow
Test equipment End device
Interworking POTS
INVITE →
416 Unsupported URI scheme ←
ACK →

ISDN interworking
INVITE →
416 Unsupported URI scheme ←
ACK →
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Term_Abnormal_situations</td><td>TP_306_002</td><td>Reference
section 8.2.2.1 of
[IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
A call setup is rejected due to an unknown destination user address

Ensure that the IUT on receipt of an INVITE request with a Request-URI set with an address that it does not accept, sends a Not Found (404 Not Found) response.</td></tr>
<tr><td colspan="4">SIP header values
INVITE
INVITE sip:<any unknown destination URI> SIP/2.0</td></tr>
<tr><td colspan="4">Message flow
Test equipment End device
Interworking POTS
INVITE →
404 Not Found ←
ACK →

ISDN interworking
INVITE →
404 Not Found ←
ACK →
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Term_Abnormal_situations</td><td>TP_306_003</td><td>Reference
section 13.3.1 of
[IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
A call setup is rejected due to an Expires header set to 0

Ensure that the IUT on receipt of an INVITE request including an Expires header set to 0, sends a Request Terminated (487 Request Terminated) response.</td></tr>
<tr><td colspan="4">SIP header values
INVITE
Expires: 0</td></tr>
<tr><td colspan="4">Message flow
Test equipment End device
Interworking POTS
INVITE →
487 Request Terminated ←
ACK →

ISDN interworking
INVITE →
487 Request Terminated ←
ACK →
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
Term_Abnormal_situations</td><td>TP_306_004</td><td>Reference
sections 8.2.3, 13.2.1,
13.3.1 and 20.11 of
[IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
A call setup is rejected due to a Content-Disposition header handling parameter empty

Ensure that the IUT on receipt of an INVITE request including a Content-Language header value that it cannot understand, a Content-Disposition header including a handling empty sends an Unsupported Media Type (415 Unsupported Media Type) response with an Accept header that lists the types of all bodies it understands.</td></tr>
<tr><td colspan="4">SIP header values
INVITE
Content-Language: by
Content-Disposition: session

415:
Accept:
Accept-Language:</td></tr>
</table>

Message flow **Test equipment** **End device** **Interworking POTS** INVITE → 415 Unsupported Media Type ← ACK → **ISDN interworking** INVITE → 415 Unsupported Media Type ← ACK → **Apply post test routine**

TSS Term_Abnormal_situations	TP_306_005	Reference sections 8.2.3, 21.4.13, 13.2.1, 13.3.1 and 20.11 of [IETF RFC 3261]	Selection expression
Test purpose *A call setup is rejected due to a Content-Disposition handlingparameter set to required* Ensure that the IUT on receipt of an INVITE request including a Content-Language header value that it cannot understand, a Content-Disposition with a handling set to "required" and a disposition-types set to session, sends an Unsupported Media Type (415 Unsupported Media Type) response with an Accept header that lists the types of all bodies it understands.			
SIP header values INVITE Content-Language: by Content-Disposition: session;handling= required 415: Accept: Accept-Language:			
Message flow **Test equipment** **End device** **Interworking POTS** INVITE → 415 Unsupported Media Type ← ACK → **ISDN interworking** INVITE → 415 Unsupported Media Type ← ACK → **Apply post test routine**			

TSS Term_Abnormal_situations	TP_306_006	Reference sections 8.2.3, 21.4.13, 13.2.1, 13.3.1 and 20.11 of [IETF RFC 3261]	Selection expression
Test purpose *Initial offer not supported, rejected with 415* Ensure that the IUT on receipt of an INVITE request including a Content-Encoding header set to content-coding values that it does not support, sends an Unsupported Media Type (415 Unsupported Media Type) response with an Accept-Encoding header that list the types of coding that it understands.			
SIP header values INVITE Content-Encoding: unsupported 415: Accept-Encoding:			
Message flow			
Test equipment		**End device**	
Interworking POTS			
INVITE	→		
415 Unsupported Media Type	←		
ACK	→		
ISDN interworking			
INVITE	→		
415 Unsupported Media Type	←		
ACK	→		
Apply post test routine			

TSS Term_Abnormal_situations	TP_306_007	Reference section 13.2.1 and 13.3 of [IETF RFC 3261]	Selection expression
Test purpose *Initial offer not set to session, initial offer sent in 200 OK* Ensure that the IUT on receipt of an INVITE request including a message body with a Content-Disposition header not set to session value, includes in its first 2xx response an initial offer session description.			
SIP header values INVITE Content- Disposition: renter;handling=optional 200 OK: SDP			

Message flow
Test equipment — **End device**
Interworking POTS
INVITE ➔ Ringing
200 OK INVITE 🡐 On hook
ACK ➔
Apply post test routine
ISDN interworking
INVITE ➔ ➔ SETUP
200 OK INVITE 🡐 🡐 CONNECT
ACK ➔
Apply post test routine

TSS Term_Abnormal_situations	TP_306_008	Reference sections 13.2.1, 13.3.1 and 20.11 of [IETF RFC 3261]	Selection expression
Test purpose *Content language not understood, initial offer in 200 OK INVITE* Ensure that the IUT on receipt of an INVITE request including a Content-Language header value that it cannot understand and a Content-Disposition header including a handling set to "optional", includes in its first 2xx response an initial offer session description.			
SIP header values INVITE Content-Language: unknown Content- Disposition: session;handling=optional 200 OK: SDP			
Message flow **Test equipment** — **End device** **Interworking POTS** INVITE ➔ Ringing 200 OK INVITE 🡐 On hook ACK ➔ **Apply post test routine** **ISDN interworking** INVITE ➔ ➔ SETUP 200 OK INVITE 🡐 🡐 CONNECT ACK ➔ **Apply post test routine**			

TSS Term_Abnormal_situations	TP_306_009	Reference section 13.3.1.3 and 21.4.13 of [IETF RFC 3261]	Selection expression

Test purpose

SDP not acceptable, rejected with 488

Ensure that the IUT on receipt of an INVITE request including a session description that it can understand but it cannot accept, a Content-Disposition with a handling set to "required" and a disposition-types set to session, sends a Not Acceptable Here (488 Not Acceptable Here) response including a Warning header.

SIP header values

INVITE

Content- Disposition: session;handling=required

SDP not acceptable

488:

Warning:

Message flow

Test equipment		End device
Interworking POTS		
INVITE	➔	
488 Not Acceptable Here	🡐	
ACK	➔	
ISDN interworking		
INVITE	➔	
488 Not Acceptable Here	🡐	
ACK	➔	

TSS Term_Abnormal_situations	TP_306_010	Reference sections 17.2.3, 8.2.2.3 and Figure 7 of [IETF RFC 3261]	Selection expression

Test purpose

Proxy-Require header with an unsupported option-tag in ACK

Ensure that the IUT when a server INVITE transaction is in the Completed state, on receipt of an ACK request including a Proxy-Require header set with an option-tag that it does not support, enters in the Confirmed transaction state.

SIP header values

ACK

Proxy-Require: unsupported

Message flow

Test equipment		End device
Interworking POTS		
INVITE	➔	Ringing
200 OK INVITE	🡐	On hook
ACK	➔	

Check confirmed dialogue

Apply post test routine

ISDN interworking INVITE → → SETUP 200 OK INVITE ← ← CONNECT ACK → **Check confirmed dialogue** **Apply post test routine**

TSS Term_Abnormal_situations	TP_306_011	Reference section 17.2.3, 8.2.2.3 and Figure 7 of [IETF RFC 3261]	Selection expression
Test purpose *Require header with an unsupported option-tag in ACK* Ensure that the IUT when a server INVITE transaction is in the Completed state, on receipt of an ACK request including a Require header set with an option-tag that it does not support, enters in the Confirmed transaction state.			
SIP header values ACK Require: unsupported			
Message flow **Test equipment** **End device** **Interworking POTS** INVITE → Ringing 200 OK INVITE ← On hook ACK → **Check confirmed dialogue** **Apply post test routine** **ISDN interworking** INVITE → → SETUP 200 OK INVITE ← ← CONNECT ACK → **Check confirmed dialogue** **Apply post test routine**			

TSS Term_Abnormal_situations	TP_306_012	Reference section 17.2.1 and 17.2.3 of [IETF RFC 3261]	Selection expression
Test purpose *Additional identical INVITE received, the previous sent final response is repeated* Ensure that the IUT when a server INVITE transaction is in the Completed state, on receipt of an INVITE request, including a Via header set with the same branch parameter and sent-by value in the topmost list value, repeat its last response.			

SIP header values INVITE1: Via: *any_invite1_via_value*;branch= *z9hG4bK any_invite1_branch_value* From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq* INVITE2: Via: *any_invite1_via_value*;branch= z9hG4bK_*any_invite1_branch_value* From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq*

Message flow

Test equipment			
Interworking POTS			
INVITE1	➔		Ringing
3xx – 6xx response	🡐		
INVITE2	➔		
3xx – 6xx response	🡐		
ACK	➔		
ISDN interworking			
INVITE1	➔	➔	SETUP
3xx – 6xx response	🡐	🡐	RELEASE_COMPLETE
INVITE2	➔		
3xx – 6xx response	🡐		
ACK	➔		

(Column headers: **Test equipment** | **End device**)

TSS Term_Abnormal_situations	TP_306_013	Reference section 17.2.1 and 17.2.3 of [IETF RFC 3261]	Selection expression

Test purpose

Additional identical INVITE and Via header without branch parameter received, the previous sent final response is repeated

Ensure that the IUT when a server INVITE transaction is in the Completed state, on receipt of an INVITE request, including a Via header set with no branch parameter but with the Request-URI, To tag, From tag, Call-ID, CSeq and top Via identical as in the first INVITE request, repeat its last response.

SIP header values

INVITE1:

Via: *any_invite1_via_value*;branch= *z9hG4bK any_invite1_branch_value*
From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value*
To: *any_invite1_to_value*
Call-ID: *any_invite1_call-id_value*
CSeq: *any_invite1_cseq*

INVITE2:

Via: *any_invite1_via_value*;branch= z9hG4bK_*any_invite1_branch_value*
Via: *any_invite2_via_value*
From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value*
To: *any_invite1_to_value*
Call-ID: *any_invite1_call-id_value*
CSeq: *any_invite1_cseq*

Message flow

Test equipment			End device
Interworking POTS			
INVITE1	→		Ringing
3xx – 6xx response	←		
INVITE2	→		
3xx – 6xx response	←		
ACK	→		
ISDN interworking			
INVITE2	→	→	SETUP
3xx – 6xx response	←	←	RELEASE_COMPLETE
INVITE2	→		
3xx – 6xx response	←		
ACK	→		

TSS Term_Abnormal_situations	TP_306_014	Reference section 17.2.1 and 17.2.3 of [IETF RFC 3261]	Selection expression

Test purpose

Additional identical INVITE and Via header without magic cookie in the branch parameter received, the previous sent final response is repeated

Ensure that the IUT when a server INVITE transaction is in the Completed state, on receipt of an INVITE request, including a Via header set with a branch parameter without the magic cookie "z9hG4bK" but with the Request-URI, To tag, From tag, Call-ID, CSeq and top Via identical as in the first INVITE request, repeat its last response.

SIP header values Via: *any_invite1_via_value*;branch= *z9hG4bK any_invite1_branch_value* From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq* INVITE2: Via: *any_invite1_via_value*;branch= *z9hG4bK any_invite1_branch_value* Via: *any_invite1_via_value*;branch= *any_invite2_branch_value* From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq*

Message flow

Test equipment				End device
Interworking POTS				
INVITE1	➔			Ringing
3xx – 6xx response	🡐			
INVITE2	➔			
3xx – 6xx response	🡐			
ACK	➔			
ISDN interworking				
INVITE2	➔		➔	SETUP
3xx – 6xx response	🡐		🡐	RELEASE_COMPLETE
INVITE2	➔			
3xx – 6xx response	🡐			
ACK	➔			

TSS Term_Abnormal_situations	TP_306_015	Reference sections 8.2.2.2, 17.2.1 and 17.2.3 of [IETF RFC 3261]	Selection expression

Test purpose

Additional identical INVITE and Via header branch parameter different from the previous request 482 is sent

Ensure that the IUT when a server INVITE transaction is in the Proceeding state, on receipt of an INVITE request, including a Via header set with a different branch parameter starting with the magic cookie "z9hG4bK" but with the Request-URI, To tag, From tag, Call-ID and CSeq identical as in the first INVITE request, sends a Loop Detected (482 Loop Detected) response.

SIP header values
INVITE1: Via: *any_invite1_via_value*;branch= *z9hG4bK any_invite1_branch_value* From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq* INVITE2: Via: *any_invite1_via_value*;branch= z9hG4bK_*any_invite2_branch_value* From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq*

Message flow

Test equipment			End device
Interworking POTS			
INVITE1	→		Ringing
180 Ringing	←		
INVITE2	→		
482 Loop Detected	←		
ACK	→		
	Apply post test routine		
ISDN interworking			
INVITE1	→	→	SETUP
180 Ringing	←	←	ALERTING
INVITE2	→		
482 Loop Detected	←	→	RELEASE
ACK	→	←	RELEASE_COMPLETE

TSS Term_Abnormal_situations	TP_306_016	Reference sections 8.2.2.2, 17.2.1 and 17.2.3 of [IETF RFC 3261]	Selection expression

Test purpose

Additional identical INVITE and Via header sent-by value different from the previous request 482 is sent

Ensure that the IUT when a server INVITE transaction is in the Proceeding state, on receipt of an INVITE request, including a Via header set to an identical branch parameter starting with the magic cookie "z9hG4bK" and a different sent-by value, but with the Request-URI, To tag, From tag, Call-ID and CSeq identical as in the first INVITE request, sends a Loop Detected (482 Loop Detected) response.

SIP header values INVITE1: Via: *any_invite1_via_value*;branch= *z9hG4bK any_invite1_branch_value* From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq* INVITE2: Via: *any_invite2_via_value*;branch= z9hG4bK_*any_invite1_branch_value* From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq*

Message flow

Test equipment			**End device**
Interworking POTS			
INVITE1	→		Ringing
180 Ringing	←		
INVITE2	→		
482 Loop Detected	←		
ACK	→		
ISDN interworking			
INVITE1	→	→	SETUP
180 Ringing	←	←	ALERTING
INVITE2	→		
482 Loop Detected	←	→	RELEASE
ACK	→	←	RELEASE_COMPLETE

TSS Term_Abnormal_situations	TP_306_017	Reference sections 8.2.2.2, 17.2.1 and 17.2.3 of [IETF RFC 3261]	Selection expression

Test purpose

Additional identical INVITE and Via header different from the previous request 482 is sent

Ensure that the IUT when a server INVITE transaction is in the Proceeding state, on receipt of an INVITE request, including a top Via header set to a different value but with the Request-URI, To tag, From tag, Call-ID and CSeq identical as in the first INVITE request, sends a Loop Detected (482 Loop Detected) response.

SIP header values INVITE1: Via: *any_invite1_via_value*;branch= *z9hG4bK any_invite1_branch_value* From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq* INVITE2: Via: *any_invite2_via_value*;branch= z9hG4bK_*any_invite2_branch_value* From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq*
Message flow

Test equipment			**End device**
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
INVITE2	→		
482 Loop Detected	←		
ACK	→		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
INVITE2	→		
482 Loop Detected	←	→	RELEASE
ACK	→	←	RELEASE_COMPLETE

TSS Term_Abnormal_situations	TP_306_018	Reference section 8.2.2.3 of [IETF RFC 3261]	Selection expression
Test purpose *INVITE option value in the Supported header unknown a 420 is sent* Ensure that the IUT, on receipt of an INVITE request with a Require header set to an option value that the IUT does not support, sends a Bad Extension (420 Bad Extension) response including those options in the Unsupported header.			
SIP header values INVITE: Require: unsupported 420: Unsupported: unsupported			

Message flow **Test equipment** **End device** **Interworking POTS** INVITE → 420 Bad Extension ← ACK → **ISDN interworking** INVITE → 420 Bad Extension ← ACK →

TSS Term_Abnormal_situations	TP_306_019	Reference section 8.2.2 and 15.1.2 of [IETF RFC 3261]	Selection expression
Test purpose *BYE option value in the Supported header unknown a 420 is sent* Ensure that the IUT, once a dialogue has been established, on receipt of a BYE request including a Require header set with an option-tag that it does not support, sends a Bad Extension (420 Bad Extension) response including an Unsupported set with this option-tag.			
SIP header values BYE: Require: unsupported 420: Unsupported: unsupported			
Message flow **Test equipment** **End device** **Interworking POTS** INVITE → 180 Ringing ← Ringing 200 OK INVITE ← Off hook ACK → BYE → 420 Bad Extension ← **ISDN interworking** INVITE → → SETUP 180 Ringing ← ← ALERTING 200 OK INVITE ← ← CONNECT ACK → BYE → 420 Bad Extension ←			

TSS Term_Abnormal_situations	TP_306_020	Reference section 15.1.2 of [IETF RFC 3261]	Selection expression

Test purpose

BYE received no dialogue exist

Ensure that the IUT, while no dialogue has been established, on receipt of a BYE request, sends a Call/Transaction does not exist (481 Call/Transaction does not exist).

SIP header values

Message flow

Test equipment				**End device**
BYE	➔			
481 Call/Transaction does not exist	🡐			

TSS Term_Abnormal_situations	TP_306_021	Reference section 15.1.2 of [IETF RFC 3261]	Selection expression

Test purpose

BYE To header tag not received, 481 is sent

Ensure that the IUT, while a dialogue has been established, on receipt of a BYE request without TAG in the To header, sends a Call/Transaction does not exist (481 Call/Transaction does not exist).

SIP header values

INVITE:

From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value*

To: *any_invite1_to_value*

BYE:

From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value*

To: *any_invite1_to_value*

Message flow

Test equipment				**End device**
Interworking POTS				
INVITE	➔			Ringing
180 Ringing	🡐			
200 OK INVITE	🡐			Off hook
ACK	➔			
BYE	➔			
481 Call/Transaction does not exist	🡐			
ISDN interworking				
INVITE	➔		➔	SETUP
180 Ringing	🡐		🡐	ALERTING
200 OK INVITE	🡐		🡐	CONNECT
ACK	➔			
BYE	➔			
481 Call/Transaction does not exist	🡐			

<table>
<tr><td>TSS
Term_Abnormal_situations</td><td>TP_306_022</td><td>Reference
section 12.2.2 of
[IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
BYE CSeq value in BYE higher than in INVITE request

Ensure that the IUT, once a dialogue has been established, on receipt of a BYE request including a CSeq header set with a more than one higher value as in the previous request, sends a Success (200 OK) response with the same CSeq value.</td></tr>
<tr><td colspan="4">SIP header values
INVITE1:
CSeq: any_invite1_cseq

BYE:
CSeq: any_invite1_cseq+x

200 OK BYE
CSeq: any_invite1_cseq+x</td></tr>
<tr><td colspan="4">Message flow
Test equipment — End device
Interworking POTS
INVITE →
180 Ringing ← Ringing
200 OK INVITE ← Off hook
ACK →
BYE →
200 OK BYE ←

ISDN interworking

INVITE → → SETUP
180 Ringing ← ← ALERTING
200 OK INVITE ← ← CONNECT
ACK →
BYE → → DISCONNECT
200 OK BYE ← ← RELEASE
→ RELEASE_COMPLETE</td></tr>
</table>

<table>
<tr><td>TSS
Term_Abnormal_situations</td><td>TP_306_023</td><td>Reference
section 9.2 of
[IETF RFC 3261]</td><td>Selection expression</td></tr>
<tr><td colspan="4">Test purpose
CANCEL with different branch in the Via header received a 481 is sent

Ensure that the IUT when a server INVITE transaction is in the Proceeding state, on receipt of a CANCEL request including a Via header set with a different branch parameter starting with the magic cookie "z9hG4bK" but with the Request-URI, To tag, From tag, Call-ID and CSeq identical as in the original INVITE request, sends a Call Leg/Transaction Does Not Exist (481 Call Leg/Transaction Does Not Exist) response.</td></tr>
</table>

SIP header values INVITE1: Via: *any_invite1_via_value*;branch= *z9hG4bK any_invite1_branch_value* From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq* INVITE2: Via: *any_invite2_via_value*;branch= z9hG4bK_*any_cancel_branch_value* From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq*

Message flow

Test equipment			End device
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
CANCEL	→		
481 Call Leg/Transaction Does Not Exist	←		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
CANCEL	→		
481 Call Leg/Transaction Does Not Exist	←		

TSS Term_Abnormal_situations	TP_306_024	Reference section 9.2 and 17.2.3 of [IETF RFC 3261]	Selection expression

Test purpose

CANCEL with different Via send-by value received a 481 is sent

Ensure that the IUT when a server INVITE transaction is in the Proceeding state, on receipt of a CANCEL request including a Via header set to an identical branch parameter starting with the magic cookie "z9hG4bK" and a different sent-by value, but with the Request-URI, To tag, From tag, Call-ID and CSeq identical as in the original INVITE request, sends a Call Leg/Transaction Does Not Exist (481 Call Leg/Transaction Does Not Exist) response.

SIP header values INVITE1: Via: *any_invite1_via_value*;branch= *z9hG4bK any_invite1_branch_value* From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq* INVITE2: Via: *any_cancel_via_value*;branch= z9hG4bK_*any_ invite1_branch_value* From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq*

Message flow

Test equipment			End device
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
CANCEL	→		
481 Call Leg/Transaction Does Not Exist	←		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
CANCEL	→		
481 Call Leg/Transaction Does Not Exist	←		

TSS Term_Abnormal_situations	TP_306_025	Reference section 9.2 and 17.2.3 of [IETF RFC 3261]	Selection expression
Test purpose *CANCEL received with different Via header value a 481 is sent* Ensure that the IUT when a server INVITE transaction is in the Proceeding state, on receipt of a CANCEL request, including a top Via header set to a different value but with the Request-URI, To tag, From tag, Call-ID and CSeq identical as in the original INVITE request, sends a Call Leg/Transaction Does Not Exist (481 Call Leg/Transaction Does Not Exist) response.			

SIP header values INVITE1: Via: *any_invite1_via_value*;branch= *z9hG4bK any_invite1_branch_value* From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq* INVITE2: Via: *any_cancel_via_value*;branch= z9hG4bK_*any_ cancel_branch_value* From: *any_invite1_from_value*;tag=*any_invite1_from_tag_value* To: *any_invite1_to_value* Call-ID: *any_invite1_call-id_value* CSeq: *any_invite1_cseq*

Message flow

Test equipment			End device
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
CANCEL	→		
481 Call Leg/Transaction Does Not Exist	←		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
CANCEL	→		
481 Call Leg/Transaction Does Not Exist	←		

TSS Term_Abnormal_situations	TP_306_026	Reference sections 17.2.3, 17.2.2, 12.2.1.1 and 15.1.2 of [IETF RFC 3261]	Selection expression
Test purpose *BYE received Via header with the same branch parameter after BYE was answered last response is repeated* Ensure that the IUT, having already answered to a BYE request, on receipt of a BYE request, before timer J fires, including a Via header set with the same branch parameter in the topmost list value, repeat its last response.			
SIP header values BYE1: Via: *any_bye1_via_value*;branch= *z9hG4bK any_bye1_branch_value* BYE2: Via: *any_bye1_via_value*;branch= *z9hG4bK any_bye1_branch_value*			

Message flow			
Test equipment			**End device**
Interworking POTS			
INVITE	→		
180 Ringing	←		Ringing
200 OK INVITE	←		Off hook
ACK	→		
BYE1	→		
200 OK BYE	←		
BYE2	→		
200 OK BYE	←		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→		
BYE1	→	→	DISCONNECT
200 OK BYE	←	←	RELEASE
		→	RELEASE_COMPLETE
BYE2	→		
200 OK BYE	←		

TSS Term_Abnormal_situations	TP_306_027	Reference sections 17.2.3, 17.2.2 and 15.1.2 of [IETF RFC 3261]	Selection expression
Test purpose *BYE received Via header without branch parameter after BYE was answered last response is repeated* Ensure that the IUT, having already answered to a BYE request, on receipt of a BYE request, before timer J fires, including a Via header set with no branch parameter but with the Request-URI, To tag, From tag, Call-ID and CSeq identical as in the first BYE request, repeat its last response.			
SIP header values BYE1: Via: *any_bye1_via_value*;branch= *z9hG4bK any_bye1_branch_value* BYE2: Via: *any_bye1_via_value*			

Message flow			
Test equipment			**End device**
Interworking POTS			
INVITE	→		
180 Ringing	←		Ringing
200 OK INVITE	←		Off hook
ACK	→		
BYE1	→		
200 OK BYE	←		
BYE2	→		
200 OK BYE	←		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→		
BYE1	→	→	DISCONNECT
200 OK BYE	←	←	RELEASE
		→	RELEASE_COMPLETE
BYE2	→		
200 OK BYE	←		

TSS Term_Abnormal_situations	TP_306_028	Reference section 12.2.2 of [IETF RFC 3261]	Selection expression
Test purpose *BYE CSeq value in BYE lower than in INVITE request* Ensure that the IUT on receipt of a BYE request with a CSeq number set to a lower value than in the preceding INVITE request, sends a 500 (Server Internal Error) response.			
SIP header values INVITE: CSeq: *any_invite_cseq* BYE: CSeq: *any_invite1_cseq-1*			

Message flow			
Test equipment			**End device**
Interworking POTS			
INVITE	→		
180 Ringing	←		Ringing
200 OK INVITE	←		Off hook
ACK	→		
BYE	→		
500 Server Internal Error	←		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→		
BYE	→		
500 Server Internal Error	←		

TSS Term_Abnormal_situations	TP_306_029	Reference section 8.2.2.3 of [IETF RFC 3261]	Selection expression
Test purpose *CANCEL received Require header contains not supported value* Ensure that the IUT when a server INVITE transaction is in the Proceeding state, on receipt of a CANCEL request including a Require header set with an option-tag that it does not support, sends a Success (200 OK) response.			
SIP header values Require: 100unsupported			

Message flow			
Test equipment			**End device**
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
CANCEL	→		
200 OK CANCEL	←		
487 Request Terminated	←		
ACK	→		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
CANCEL	→	→	RELEASE/RELEASE COMPLETE
200 OK CANCEL	←		
487 Request Terminated	←		
ACK	→		

TSS Term_Abnormal_situations	TP_306_030	Reference section 8.2.2.3 of [IETF RFC 3261]	Selection expression

Test purpose

CANCEL received Proxy-*Require header contains not supported value*

Ensure that the IUT when a server INVITE transaction is in the Proceeding state, on receipt of a CANCEL request including a Proxy-Require header set with an option-tag that it does not support, sends a Success (200 OK) response.

SIP header values

Proxy-Require: 100unsupported

Message flow

Test equipment			End device
Interworking POTS			
INVITE	→		Ringing
180 Ringing	←		
CANCEL	→		
200 OK CANCEL	←		
487 Request Terminated	←		
ACK	→		
ISDN interworking			
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
CANCEL	→	→	RELEASE/RELEASE COMPLETE
200 OK CANCEL	←		
487 Request Terminated	←		
ACK	→		

TSS Term_Abnormal_situations	TP_306_031	Reference section 9.2 and 15.1.2 of [IETF RFC 3261]	Selection expression

Test purpose

CANCEL received no dialogue exists

Ensure that the IUT while no session has been initiated, on receipt of a CANCEL request, sends a Call/transaction Does Not Exist (481 Call/transaction Does Not Exist) response.

SIP header values

Message flow

Test equipment		End device
CANCEL	→	
481 Call/transaction Does Not Exist	←	

TSS Term_Abnormal_situations	TP_306_032	Reference section 15.1.2 of [IETF RFC 3261]	Selection expression PICS 5.1.2/1

Test purpose

BYE received after session is released

Ensure that the IUT, while a session has been released, on receipt of a BYE request, sends a Call/transaction Does Not Exist (481 Call/transaction Does Not Exist) response.

SIP_CC_TE_CR_I_005

SIP header values

Message flow

Test equipment				**End device**
Interworking POTS				
INVITE	→			
180 Ringing	←			Ringing
200 OK INVITE	←			Off hook
ACK	→			
BYE1	→	Start timer J		
200 OK BYE	←			
		Timeout timer J		
BYE2	→			
481 Call/transaction Does Not Exist	←			
ISDN interworking				
INVITE	→		→	SETUP
180 Ringing	←		←	ALERTING
200 OK INVITE	←		←	CONNECT
ACK	→			
BYE1	→	Start timer J	→	DISCONNECT
200 OK BYE	←		←	RELEASE
		Timeout timer J	→	RELEASE_COMPLETE
BYE2	→			
481 Call/transaction Does Not Exist	←			

7.2.4 Emergency service

TSS Emergency_service	TP_401_001	Reference subclause 5.1.6.8.4 of [ETSI TS 124 229]	Selection expression

Test purpose

Emergency session setup within a non-emergency registration

Ensure that the SUT in case of an emergency call and non-emergency registration sends an INVITE request that contains:

- a service URN in the Request-URI
- a To header field with the same emergency service URN as in the Request-URI
- a From header field that includes the public user identity or the tel URI associated with the public user identity
- if available to the SUT, and if defined for the access type, the SUT shall insert in the P-Access-Network-Info header field
- one P-Preferred-Identity header field that include the public user identity or the tel URI associated with the public user identity
- if the UE has its location information available, or a URI that points to the location information, then the UE shall include a Geolocation header field

SIP header values

INVITE

Request Line: urn:service:{sos/sos.ambulance/sos.animal-control/sos.fire/sos.marine/sos.mountain/sos.physician/sos.police

or

operator specific emergency digits

To: urn:service:{sos/sos.ambulance/sos.animal-control/sos.fire/sos.marine/sos.mountain/sos.physician/sos.police

or

operator specific emergency digits

From: public user identity

P-Access-Network-Info

Message flow

End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
ISDN interworking		
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE

Apply post test routine

7.2.5 Supplementary service control

7.2.5.1 Originating identification presentation and originating identification restriction

7.2.5.1.1 Test purposes for POTS

TSS OIP_OIR	TP_501_101	Reference subclause C.1.2.1 and C5 of [TS183 043]	Selection expression PICS 5.1.1/1

Test purpose

User Identification information received in an INVITE request

User Identification information received in an INVITE request ("P-Asserted-Id" and "From" headers) is used by the SUT. The SUT requests the media gateways to send this message over the analogue line using the Display Data Block parameter of the Display.

SIP header values

INVITE:

From: [A user identification]

P-Asserted-Identification: [A user identification]

Message flow

Test equipment		**End device**
INVITE	→	
180 Ringing	←	Ringing, A user identification

Apply post test routine

TSS OIP_OIR	TP_501_102	Reference subclause C.1.2.1, C5 of [TS183 043]	Selection expression PICS 5.1.1/1

Test purpose

The calling user wishes to override the default privacy setting

A service code command may be received by the SUT in case the calling user wishes to override the default privacy setting for a particular call, in which case the called party number is embedded in the command code as part of the supplementary information. The SUT generates an INVITE and the service code commands and the destination number is present in the Request Line.

SIP header values

INVITE: Request Line START PX SC (SR SI) SX" or "PX SC (SR SI) SX FINISH@pes-scc.operator.com

Note : The command syntax is described in subclause C.1.2.1.1 of [ETSI TS183 043]

Message flow

End device		**Test equipment**
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE

Apply post test routine

TSS OIP_OIR	TP_501_103	Reference subclause C.1.2.1 and C5 of [ETSI TS183 043]	Selection expression PICS 5.1.1/1
Test purpose *User Name information received in an INVITE request* Ensure that when user Name information is received in an INVITE request ("From" header /or "P-Asserted-Identity") it is used by the SUT to generate the appropriate Call Setup message. The SUT requests the media gateways to send this message over the analogue line using the Display Data Block. parameter of the Display.			
SIP header values From: [A user Name] P-Asserted-Identification: [A user Name]			
Message flow			
End device		**Test equipment**	
Off hook			
Dial number	→	INVITE	
	←	407 Proxy Authentication Required	
	→	ACK	
	→	INVITE	
Apply post test routine			

7.2.5.1.2 Test purposes for ISDN

TSS OIP_OIR	TP_501_201	Reference subclause 5.2.3.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2
Test purpose *INVITE received, no P-Asserted-Identity and no Privacy header present* Ensure that on receipt of an INVITE request and the P-Asserted-Identity is not present and no Privacy header is present, a SETUP is sent and a Calling party number Information Element is not included.			
SIP header values			
DSS1 Parameter values			
Message flow			
Test equipment		**End device**	
INVITE	→	→	SETUP
183 Ringing	←	←	ALERTING
Apply post test routine			

TSS OIP_OIR	TP_501_202	Reference subclause 5.2.3.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2

Test purpose

INVITE received, no P-Asserted-Identity present and Privacy header present From header Anonymous

Ensure that on receipt of an INVITE request and the P-Asserted-Identity is not present and a Privacy header is present the priv-value is set to 'id' or 'header' or 'user' the From header is set to an Anonymous value, a SETUP is sent and a Calling party number Information Element is included and coded as follows:

Calling party number

Type of number = Unknown

Numbering plan identification = Unknown

Presentation indicator = **Presentation restricted**

Screening indicator = Network provided

Number digits not present

SIP header values

INVITE: From: <sip:anonymous@anonymous.invalid>; tag=

DSS1 Parameter values

SETUP: Calling party number

Presentation indicator = **Presentation restricted**

Message flow

Test equipment			End device
INVITE	→	→	SETUP
183 Ringing	←	←	ALERTING

Apply post test routine

TSS OIP_OIR	TP_501_203	Reference subclause 5.2.3.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2

Test purpose

INVITE received, no P-Asserted-Identity present and no Privacy header present From header Unavailable

Ensure that on receipt of an INVITE request and the P-Asserted-Identity is not present and a Privacy header is present the priv-value is se to 'id' or 'header' or 'user' the From header is set to an Unavailable value, a SETUP is sent and a Calling party number Information Element is included and coded as follows:

Calling party number

Type of number = Unknown

Numbering plan identification = Unknown

Presentation indicator = **Not available due to interworking**

Screening indicator = Network provided

Number digits not present

SIP header values

INVITE: From: <sip:unavailable@unknown.invalid >; tag=

DSS1 Parameter values

SETUP: Calling party number

Presentation indicator = **Not available due to interworking**

Message flow			
Test equipment			**End device**
INVITE	➔	➔	SETUP
183 Ringing	🡐	🡐	ALERTING
Apply post test routine			

TSS OIP_OIR	TP_501_204	Reference subclause 5.2.3.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2
Test purpose *INVITE received, P-Asserted-Identity present and no Privacy header present From header not in the format of an E.164 address* Ensure that on receipt of an INVITE request and the P-Asserted-Identity is present, the URI in the format of an E.164 address in the local number format and a no Privacy header is present, the URI of the From header is not in the format of an E.164 address, a SETUP is sent and a Calling party number Information Element is included and coded as follows: **Calling party number** Type of number = **National number** Numbering plan identification = ISDN/Telephony numbering plan Presentation indicator = Presentation allowed Screening indicator = User provided, verified and passed Number digits: Number digits are derived from userinfo of the P-Asserted-Identity header. In case for local number format the userinfo is sent as digits			
SIP header values INVITE: P-Asserted-Identity: userinfo in local number format			
DSS1 Parameter values SETUP: Calling party number Type of number = National number Number digits derived from the P-Asserted-Identity			
Message flow **Test equipment** / **End device** INVITE ➔ ➔ SETUP 183 Ringing 🡐 🡐 ALERTING **Apply post test routine**			

TSS OIP_OIR	TP_501_205	Reference subclause 5.2.3.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2
Test purpose *INVITE received, P-Asserted-Identity present and no Privacy header present From header not in the format of an E.164 address* Ensure that on receipt of an INVITE request and the P-Asserted-Identity is present, the URI in the format of an E.164 address in the global number format and a no Privacy header is present, the URI of the From header is not in the format of an E.164 address, a SETUP is sent and a Calling party number Information Element is included and coded as follows: **Calling party number** Type of number = **National number** Numbering plan identification = ISDN/Telephony numbering plan Presentation indicator = Presentation allowed Screening indicator = User provided, verified and passed Number digits: Number digits are derived from userinfo of the P-Asserted-identity. In case for global number and the country code is the same as the SUT or the ISDN line is located, the country code is removed from the userinfo.			
SIP header values INVITE: P-Asserted-Identity: userinfo in local number format			
DSS1 Parameter values SETUP: Calling party number Type of number = National number Number digits derived from the P-Asserted-Identity			
Message flow			
Test equipment			**End device**
INVITE	→	→	SETUP
183 Ringing	←	←	ALERTING
Apply post test routine			

TSS OIP_OIR	TP_501_206	Reference subclause 5.2.3.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2
Test purpose *INVITE received, P-Asserted-Identity present and Privacy header not present From header not in the format of an E.164 address* Ensure that on receipt of an INVITE request and the P-Asserted-Identity is present, the URI in the format of an E.164 address in the global number format and a no Privacy header is present, the URI of the From header is not in the format of an E.164 address, a SETUP is sent and a Calling party number Information Element is included and coded as follows: **Calling party number** Type of number = **International number** Numbering plan identification = ISDN/Telephony numbering plan Presentation indicator = Presentation allowed Screening indicator = User provided, verified and passed Number digits: Number digits are derived from userinfo of the P-Asserted-Identity header. In case for global number and the country code is not the same as the SUT or the ISDN line is located, the userinfo is sent as digits.			

SIP header values INVITE: P-Asserted-Identity: userinfo in global number format
DSS1 Parameter values SETUP: Calling party number Type of number = International number Number digits derived from the P-Asserted-Identity
Message flow

Test equipment			**End device**
INVITE	➔	➔	SETUP
183 Ringing	🡐	🡐	ALERTING
	Apply post test routine		

TSS OIP_OIR	TP_501_207	Reference subclause 5.2.3.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2

Test purpose

INVITE received, P-Asserted-Identity present and Privacy header present From header not in the format of an E.164 address

Ensure that on receipt of an INVITE request and the P-Asserted-Identity is present the URI in the format of an E.164 address and a Privacy header is present the priv-value is set to 'id' or 'header' or 'user' the URI of the From header is not in the format of an E.164 address, a SETUP is sent and a Calling party number Information Element is included and coded as follows:

Calling party number

Type of number = Unknown

Numbering plan identification = Unknown

Presentation indicator = **Presentation restricted**

Screening indicator = Network provided

Number digits not present

SIP header values INVITE: P-Asserted-Identity Privacy: **id**
DSS1 Parameter values SETUP: Calling party number Presentation indicator **Presentation restricted**
Message flow

Test equipment			**End device**
INVITE	➔	➔	SETUP
183 Ringing	🡐	🡐	ALERTING
	Apply post test routine		

TSS OIP_OIR	TP_501_208	Reference subclause 5.2.3.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2

Test purpose

*INVITE received, P-Asserted-Identity **not equal** From header not in the format of an E.164 address Privacy header present*

Ensure that on receipt of an INVITE request and the P-Asserted-Identity is present the URI in the format of an E.164 address and a no Privacy header is present the URI of the From header is in the format of an E.164 address and the userinfo is not equal to the userinfo of the P-Asserted-Identity, a SETUP is sent and two Calling party number Information Elements are included and coded as follows:

1st Calling party number

Numbering plan identification = ISDN/Telephony numbering plan

Presentation indicator = Presentation allowed

Screening indicator = **User provided, not verified**

Number digits:

Number digits are derived from userinfo of the **From header**. In case for global number and the country code is not the same as the SUT or the ISDN line is located, the userinfo is sent as digits

2nd Calling party number

Numbering plan identification = ISDN/Telephony numbering plan

Presentation indicator = Presentation allowed

Screening indicator = **User provided, verified and passed**

Number digits:

Number digits are derived from userinfo of the **P-Asserted-Identity header**. In case for global number and the country code is not the same as the SUT or the ISDN line is located, the userinfo is sent as digits

SIP header values

INVITE: From:
P-Asserted-Identity:

DSS1 Parameter values

SETUP: Calling party number
Screening indicator
User provided, not verified
Calling party number
Screening indicator
User provided, verified and passed

Message flow

Test equipment			End device
INVITE	→	→	SETUP
183 Ringing	←	←	ALERTING

Apply post test routine

TSS OIP_OIR	TP_501_209	Reference subclause 5.2.3.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2

Test purpose

*INVITE received, P-Asserted-Identity **equal** From header present no Privacy header present*

Ensure that on receipt of an INVITE request and if the P-Asserted-Identity is present, the URI is in the format of an E.164 address and a no Privacy header is present, the URI of the From header is in the format of an E.164 address and the userinfo is equal to the userinfo of the P-Asserted-Identity, a SETUP is sent and one Calling party number Information Elements is included and coded as follows:

Calling party number

Numbering plan identification = ISDN/Telephony numbering plan

Presentation indicator = Presentation allowed

Screening indicator = **User provided, verified and passed**

Number digits:

Number digits are derived from userinfo of the **P-Asserted-Identity header**. In a case where the global number and the country code is not located the same as the SUT or the ISDN line , the userinfo is sent as digits.

SIP header values

INVITE: From:

P-Asserted-Identity:

DSS1 Parameter values

SETUP: Calling party number

Screening indicator

User provided, verified and passed

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
183 Ringing	←	←	ALERTING

Apply post test routine

TSS OIP_OIR	TP_501_210	Reference subclause 5.2.3.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2

Test purpose

INVITE received, P-Asserted-Identity present and From header present Privacy header present

Ensure that on receipt of an INVITE request and if the P-Asserted-Identity is present, the URI is in the format of an E.164 address and a Privacy header is present, the URI of the From header is in the format of an E.164 address, a SETUP is sent and a Calling party number Information Elements is included and coded as follows:

Calling party number

Type of number = Unknown

Numbering plan identification = Unknown

Presentation indicator = **Presentation restricted**

Screening indicator = Network provided

Number digits not present

SIP header values

INVITE: From

P-Asserted-Identity

Privacy: **id,header,user**

DSS1 Parameter values SETUP: Calling party number Presentation indicator **Presentation restricted**
Message flow **Test equipment** **End device** INVITE ➔ ➔ SETUP 183 Ringing 🡐 🡐 ALERTING **Apply post test routine**

TSS OIP_OIR	TP_501_211	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2
Test purpose *SETUP received, Calling party number national significant Presentation Restriction Indicator is set to restricted* Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'National number', Presentation Restriction Indicator is set to **Presentation restricted,** an INVITE request is sent. If the Calling party number digits matches with a registered Public identity, a P Preferred Identity header field is present, the userinfo is set to the identified Public user identity. The Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format or set to anonymous@anonymous.invalid. A Privacy header is present set to 'id' and 'header' and 'user'.			
SIP header values INVITE: From: <derived from the Calling party number Address digits or anonymous@anonymous.invalid> P Preferred Identity: <matched Public user identity> Privacy: id, header, user			
DSS1 Parameter values SETUP: Calling party number Type of number = National number Presentation Restriction Presentation restricted Address digits present			
Message flow **End device** **Test equipment** SETUP ➔ INVITE 🡐 407 Proxy Authentication Required ➔ ACK ➔ INVITE SETUP ACKNOWLEDGE 🡐 100 Trying **Apply post test routine**			

TSS OIP_OIR	TP_501_212	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2
Test purpose *SETUP received, Calling party number national significant Presentation Restriction Indicator is set to restricted* Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'National number', Presentation Restriction Indicator is set to **Presentation restricted**, an INVITE request is sent. Ifthe Calling party number digits does not match with a registered Public identity, a P Preferred Identity header field is present, the userinfo is set to the Default Public user identity. The Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format or set to anonymous@anonymous.invalid. A Privacy header is present set to 'id' and 'header' and 'user'.			
SIP header values INVITE: From: <derived from the Calling party number Address digits or anonymous@anonymous.invalid> P Preferred Identity: <Default Public user identity> Privacy: id, header, user			
DSS1 Parameter values SETUP: Calling party number Type of number = National number Presentation Restriction Presentation restricted Address digits present			
Message flow			
End device			**Test equipment**
SETUP		→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
SETUP ACKNOWLEDGE		←	100 Trying
Apply post test routine			

TSS OIP_OIR	TP_501_213	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2
Test purpose *SETUP received, Calling party number international Presentation Restriction Indicator is set to restricted* Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'International number', Presentation Restriction Indicator is set to **Presentation restricted**, an INVITE request is sent. If a P Preferred Identity header field is present, the userinfo is derived from the Address digits. The Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format or set to anonymous@anonymous.invalid. A Privacy header is present set to 'id' and 'header' and 'user'			
SIP header values INVITE: From: <derived from the Calling party number Address digits or anonymous@anonymous.invalid> P Preferred Identity: <derived from the Calling party number Address digits > Privacy: id, header, user			
DSS1 Parameter values SETUP: Calling party number Type of number = International number Presentation Restriction Presentation restricted Address digits present			

Message flow		
End device		**Test equipment**
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying
	Apply post test routine	

TSS OIP_OIR	TP_501_214	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2

Test purpose

SETUP received, Calling party number Subscriber Presentation Restriction Indicator is set to restricted

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'Subscriber', Presentation Restriction Indicator is set to Presentation restricted, an INVITE request is sent. If the Calling party number digits matches with a registered Public identity, a P Preferred Identity header field is present, the userinfo is derived from the Address digits. The Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format or set to anonymous@anonymous.invalid. A Privacy header is present set to 'id' and 'header' and 'user'.

SIP header values

INVITE: From: <derived from the Calling party number Address digits or anonymous@anonymous.invalid>
P Preferred Identity: <matched Public user identity>
Privacy: id, header, user

DSS1 Parameter values

SETUP: Calling party number
Type of number = Subscriber
Presentation Restriction Presentation restricted
Address digits present

Message flow		
End device		**Test equipment**
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying
	Apply post test routine	

TSS OIP_OIR	TP_501_215	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2
Test purpose *SETUP received, Calling party number Subscriber Presentation Restriction Indicator is set to restricted* Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'Subscriber', Presentation Restriction Indicator is set to Presentation restricted, an INVITE request is sent. If the Calling party number digits does not match with a registered Public identity, a P Preferred Identity header field is present, the userinfo is set to the Default Public user identity. The Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format or set to anonymous@anonymous.invalid. A Privacy header is present set to 'id' and 'header' and 'user'			
SIP header values INVITE: From: <derived from the Calling party number Address digits or anonymous@anonymous.invalid> P Preferred Identity: <Default Public user identity> Privacy: id, header, user			
DSS1 Parameter values SETUP: Calling party number Type of number = Subscriber Presentation Restriction Presentation restricted Address digits present			
Message flow			
End device		**Test equipment**	
SETUP	→	INVITE	
	←	407 Proxy Authentication Required	
	→	ACK	
	→	INVITE	
SETUP ACKNOWLEDGE	←	100 Trying	
Apply post test routine			

TSS OIP_OIR	TP_501_216	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2
Test purpose *SETUP received, Calling party number Unknown Presentation Restriction Indicator is set to restricted* Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'Unknown', Presentation Restriction Indicator is set to Presentation restricted, an INVITE request is sent. If the Calling party number digits matches with a registered Public identity a P Preferred Identity header field is present, the userinfo is derived from the Address digits. The Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format or set to anonymous@anonymous.invalid. A Privacy header is present set to 'id' and 'header' and 'user'			
SIP header values INVITE: From: <derived from the Calling party number Address digits or anonymous@anonymous.invalid> P Preferred Identity: <matched Public user identity> Privacy: id, header, user			

DSS1 Parameter values SETUP: Calling party number Type of number = Unknown Presentation Restriction Presentation restricted Address digits present
Message flow **End device** / **Test equipment** SETUP → INVITE ← 407 Proxy Authentication Required → ACK → INVITE SETUP ACKNOWLEDGE ← 100 Trying **Apply post test routine**

TSS OIP_OIR	TP_501_217	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2
Test purpose *SETUP received, Calling party number Unknown Presentation Restriction Indicator is set to restricted* Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'Unknown', Presentation Restriction Indicator is set to Presentation restricted, an INVITE request is sent. If the Calling party number digits does not match with a registered Public identity, a P Preferred Identity header field is present, the userinfo is set to the Default Public user identity. The Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format or set to anonymous@anonymous.invalid. A Privacy header is present set to 'id' and 'header' and 'user'.			
SIP header values INVITE: From: <derived from the Calling party number Address digits or anonymous@anonymous.invalid> P Preferred Identity: <Default Public user identity> Privacy: id, header, user			
DSS1 Parameter values SETUP: Calling party number Type of number = Unknown Presentation Restriction Presentation restricted Address digits present			
Message flow **End device** / **Test equipment** SETUP → INVITE ← 407 Proxy Authentication Required → ACK → INVITE SETUP ACKNOWLEDGE ← 100 Trying **Apply post test routine**			

<table>
<tr><td>TSS
OIP_OIR</td><td>TP_501_218</td><td>Reference
subclause 5.2.3.2 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.1.3/2</td></tr>
<tr><td colspan="4">Test purpose
SETUP received, Calling party number Presentation Restriction Indicator is set to restricted Address digits not present

Ensure that on receipt of a SETUP message containing a Calling party number, if Address digits are not present and the Type of number is set to 'National number', Presentation Restriction Indicator is set to Presentation restricted, an INVITE request is sent. If a P Preferred Identity header field is present, the userinfo is set to the default Public user identity. The Userinfo of the From header is set to unavailable@unknown.invalid,and if a Privacy header is present, set to 'id' and 'header' and 'user'.</td></tr>
<tr><td colspan="4">SIP Header values
INVITE: From: <sip: unavailable@unknown.invalid>
P Preferred Identity: <default Public user identity>
Privacy: id, header, user</td></tr>
<tr><td colspan="4">DSS1 Parameter values
SETUP: Calling party number
Type of number = National number
Presentation Restriction Presentation restricted
Address digits not present</td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><th>End device</th><th></th><th>Test equipment</th></tr>
<tr><td>SETUP</td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td>←</td><td>407 Proxy Authentication Required</td></tr>
<tr><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>SETUP ACKNOWLEDGE</td><td>←</td><td>100 Trying</td></tr>
</table>
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
OIP_OIR</td><td>TP_501_219</td><td>Reference
subclause 5.2.3.2 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.1.3/2</td></tr>
<tr><td colspan="4">Test purpose
SETUP received, Calling party number absent, Privacy header is present

Ensure that on receipt of a SETUP message and if the Calling party number IE is absent, an INVITE request is sent. If a P Preferred Identity header field is present, the userinfo is set to the default Public user identity. The Userinfo of the From header is set to unavailable@unknown.invalid. If a Privacy header is present set to 'id' and 'header' and 'user'.</td></tr>
<tr><td colspan="4">SIP header values
INVITE: From: <sip: unavailable@unknown.invalid>
P Preferred Identity: <default Public user identity>
Privacy: id, header, user</td></tr>
<tr><td colspan="4">DSS1 Parameter values
SETUP: Calling party number not present</td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><th>End device</th><th></th><th>Test equipment</th></tr>
<tr><td>SETUP</td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td>←</td><td>407 Proxy Authentication Required</td></tr>
<tr><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>SETUP ACKNOWLEDGE</td><td>←</td><td>100 Trying</td></tr>
</table>
Apply post test routine</td></tr>
</table>

TSS OIP_OIR	TP_501_220	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2

Test purpose

SETUP received, Calling party number absent, Privacy header is absent

Ensure that on receipt of a SETUP message and if the Calling party number IE is absent, an INVITE request is sent. If a P Preferred Identity header field is present, the userinfo is set to the default Public user identity. The Userinfo of the From header is set. A Privacy header is not present.

SIP header values

INVITE: From: <sip: unavailable@unknown.invalid>
P Preferred Identity: <Default Public user identity>

DSS1 Parameter values

SETUP: Calling party number not present

Message flow

End device		Test equipment
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying

Apply post test routine

TSS OIP_OIR	TP_501_221	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2

Test purpose

SETUP received, Calling party number NOA national significant Presentation Restriction Indicator is set to allowed

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'National number', Presentation Restriction Indicator is set to Presentation allowed and an INVITE request is sent. If the Calling party number digits matches with a registered Public identity and a P Preferred Identity header field is present, the userinfo is set to the identified Public user identity. If the Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format, and if a Privacy header is present, set to 'none'.

SIP header values

INVITE: From: <derived from the Calling party number Address digits>
P Preferred Identity: <identified Public user identity>
Privacy: none

DSS1 Parameter values

SETUP: Calling party number
Type of number = National number
Presentation Restriction Presentation allowed
Address digits present

Message flow

End device		Test equipment
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying

Apply post test routine

TSS OIP_OIR	TP_501_222	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2

Test purpose

SETUP received, Calling party number NOA national significant Presentation Restriction Indicator is set to allowed

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'National number', Presentation Restriction Indicator is set to Presentation allowed and an INVITE request is sent. If the Calling party number digits does not match with a registered Public identity and a P Preferred Identity header field is present, the userinfo is set to the Default Public user identity. If the Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format and a Privacy header is present, set to 'none'.

SIP header values

INVITE: From: <derived from the Calling party number Address digits>
P Preferred Identity: <Default Public user identity>
Privacy: none

DSS1 Parameter values

SETUP: Calling party number
Type of number = National number
Presentation Restriction Presentation allowed
Address digits present

Message flow

End device		Test equipment
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying

Apply post test routine

TSS OIP_OIR	TP_501_223	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2

Test purpose

SETUP received, Calling party number NOA international Presentation Restriction Indicator is set to allowed

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'International number', Presentation Restriction Indicator is set to Presentation allowed and an INVITE request is sent. If the Calling party number digits matches with a registered Public identity, a P Preferred Identity header field is present, the userinfo is set to the identified Public user identity. If the Userinfo of the From header is derived from the address digits of the calling party number IE in the global format and a Privacy header is present, set to 'none'.

SIP header values

INVITE: From: <derived from the Calling party number Address digits>
P Preferred Identity: <matched Public user identity>
Privacy: none

DSS1 Parameter values

SETUP: Calling party number
Type of number = International number
Presentation Restriction Presentation allowed
Address digits present

Message flow		
End device		**Test equipment**
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying
	Apply post test routine	

TSS OIP_OIR	TP_501_224	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2

Test purpose

SETUP received, Calling party number NOA international Presentation Restriction Indicator is set to allowed

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'International number', Presentation Restriction Indicator is set to Presentation allowed and an INVITE request is sent. If the Calling party number digits does not match with a registered Public identity, if a P Preferred Identity header field is present, the userinfo is set to the Default Public user identity. If the Userinfo of the From header is derived from the address digits of the calling party number IE in the global number format and a Privacy header is present, set to 'none'.

SIP header values

INVITE: From: <derived from the Calling party number Address digits>
P Preferred Identity: <Default Public user identity>
Privacy: none

DSS1 Parameter values

SETUP: Calling party number
Type of number = International number'
Presentation Restriction Presentation allowed
Address digits present

Message flow		
End device		**Test equipment**
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying
	Apply post test routine	

TSS OIP_OIR	TP_501_225	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2

Test purpose

SETUP received, Calling party number NOA subscriber Presentation Restriction Indicator is set to allowed

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'Subscriber', Presentation Restriction Indicator is set to Presentation allowed and an INVITE request is sent. If the Calling party number digits matches with a registered Public identity and a P Preferred Identity header field is present, the userinfo is set to the identified Public user identity. If the Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format and a Privacy header is present, set to 'none'.

SIP header values

INVITE: From: <derived from the Calling party number Address digits>
P Preferred Identity: <matched Public user identity>
Privacy: none

DSS1 Parameter values

SETUP: Calling party number
Type of number = Subscriber
Presentation Restriction Presentation allowed
Address digits present

Message flow

End device		Test equipment
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying

Apply post test routine

TSS OIP_OIR	TP_501_226	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2

Test purpose

SETUP received, Calling party number NOA subscriber Presentation Restriction Indicator is set to allowed

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'Subscriber', Presentation Restriction Indicator is set to Presentation allowed and an INVITE request is sent. If the Calling party number digits does not match with a registered Public identity and a P Preferred Identity header field is present, the userinfo is set to the Default Public user identity. If the Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format And a Privacy header is present, set to 'none'.

SIP header values

SETUP: Calling party number
Type of number = Subscriber
Presentation Restriction Presentation allowed
Address digits present

DSS1 Parameter values

INVITE: From: <derived from the Calling party number Address digits>
P Preferred Identity: <Default Public user identity>
Privacy: none

Message flow		
End device		**Test equipment**
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying
	Apply post test routine	

TSS OIP_OIR	TP_501_227	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2

Test purpose

SETUP received, Calling party number NOA unknown Presentation Restriction Indicator is set to allowed

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'Unknown', Presentation Restriction Indicator is set to Presentation allowed and an INVITE request is sent. If the Calling party number digits matches with a registered Public identity and a P Preferred Identity header field is present, the userinfo is set to the identified Public user identity. If the Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format and a Privacy header is present, set to 'none'.

SIP header values

INVITE: From: <derived from the Calling party number Address digits>
P Preferred Identity: <matched Public user identity>
Privacy: none

DSS1 Parameter values

SETUP: Calling party number
Type of number = Unknown
Presentation Restriction Presentation allowed
Address digits present

Message flow		
End device		**Test equipment**
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying
	Apply post test routine	

TSS OIP_OIR	TP_501_228	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2

Test purpose

SETUP received, Calling party number NOA national significant Presentation Restriction Indicator is set to allowed

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'Unknown', Presentation Restriction Indicator is set to Presentation allowed and an INVITE request is sent. If the Calling party number digits does not match with a registered Public identity and a P Preferred Identity header field is present, the userinfo is set to the Default Public user identity. If the Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format and a Privacy header is present, set to 'none'.

SIP header values

INVITE: From: <derived from the Calling party number Address>
P Preferred Identity: <Default Public user identity>
Privacy: none

DSS1 Parameter values

SETUP: Calling party number
Type of number = Unknown
Presentation Restriction Presentation allowed
Address digits present

Message flow

End device		Test equipment
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying

Apply post test routine

TSS OIP_OIR	TP_501_229	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1

Test purpose

Mw interface. SETUP received, Calling party number national significant Presentation Restriction Indicator is set to restricted

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'National number', Presentation Restriction Indicator is set to Presentation restricted, and an INVITE request is sent. If the Calling party number digits matches with a registered Public identity and a P Asserted Identity header field is present, the userinfo is set to the identified Public user identity. If the Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format or set to anonymous@anonymous.invalid and a Privacy header is present, set to 'id' and 'header' and 'user'

SIP header values

INVITE: From: <derived from the Calling party number Address digits or
anonymous@anonymous.invalid>
P Asserted Identity: <matched Public user identity>
Privacy: id, header, user

DSS1 Parameter values SETUP: Calling party number Type of number = **National number** Presentation Restriction **Presentation restricted** Address digits present
Message flow **End device** **Test equipment** SETUP → INVITE SETUP ACKNOWLEDGE ← 100 Trying **Apply post test routine**

TSS OIP_OIR	TP_501_230	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1
Test purpose *Mw interface. SETUP received, Calling party number national significant Presentation Restriction Indicator is set to restricted* Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'National number', Presentation Restriction Indicator is set to Presentation restricted, and an INVITE request is sent. If the Calling party number digits does not matche with a registered Public identity and a P Asserted Identity header field is present, the userinfo is set to the Default Public user identity. If the Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format or set to anonymous@anonymous.invalid, and a Privacy header is present, set to 'id' and 'header' and 'user'.			
SIP header values INVITE:			
DSS1 Parameter values SETUP: Calling party number Type of number = **National number** Presentation Restriction **Presentation restricted** Address digits present			
Message flow **End device** **Test equipment** SETUP → INVITE SETUP ACKNOWLEDGE ← 100 Trying **Apply post test routine**			

TSS OIP_OIR	TP_501_231	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1
Test purpose *Mw interface. SETUP received, Calling party number international Presentation Restriction Indicator is set to restricted* Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'International number', Presentation Restriction Indicator is set to Presentation restricted, and an INVITE request is sent. If the Calling party number digits matches with a registered Public identity and a P Asserted Identity header field is present, the userinfo is set to the identified Public user identity. If the Userinfo of the From header is derived from the address digits of the calling party number IE in the global number format or set to anonymous@anonymous.invalid anda Privacy header is present, set to 'id' and 'header' and 'user'.			

SIP header values INVITE: From: <derived from the Calling party number Address digits or anonymous@anonymous.invalid> P Asserted Identity: <matched Public user identity> Privacy: id, header, user
DSS1 parameter values SETUP: Calling party number Type of number = International number Presentation Restriction Presentation restricted Address digits present
Message flow **End device** / **Test equipment** SETUP → INVITE SETUP ACKNOWLEDGE ← 100 Trying **Apply post test routine**

TSS OIP_OIR	TP_501_232	Reference subclause 5.2.3.2/ [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1
Test purpose *Mw interface. SETUP received, Calling party number international Presentation Restriction Indicator is set to restricted* Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'International number', Presentation Restriction Indicator is set to Presentation restricted, an INVITE request is sent. The Calling party number digits does not matche with a registered Public identity, a P Asserted Identity header field is present, the userinfo is set to the Default Public user identity. If the userinfo of the From header is derived from the address digits of the calling party number IE in the global number format or set to anonymous@anonymous.invalid, a Privacy header is present, set to 'id' and 'header' and 'user'.			
SIP header values INVITE: From: <derived from the Calling party number Address digits or anonymous@anonymous.invalid> P Asserted Identity: <Default Public user identity> Privacy: id, header, user			
DSS1 Parameter values SETUP: Calling party number Type of number = International number Presentation Restriction Presentation restricted Address digits present			
Message flow **End device** / **Test equipment** SETUP → INVITE SETUP ACKNOWLEDGE ← 100 Trying **Apply post test routine**			

<table>
<tr><td>TSS
OIP_OIR</td><td>TP_501_233</td><td>Reference
subclause 5.2.3.2 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.1.3/1</td></tr>
<tr><td colspan="4">Test purpose
Mw interface. SETUP received, Calling party number Subscriber Presentation Restriction Indicator is set to restricted

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'Subscriber', Presentation Restriction Indicator is set to Presentation restricted, an INVITE request is sent. If the Calling party number digits matches with a registered Public identity, a P-Asserted-Identity header field is present, the userinfo is set to the identified Public user identity. If the Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format or set to anonymous@anonymous.invalid, a Privacy header is present, set to 'id' and 'header' and 'user'.</td></tr>
<tr><td colspan="4">SIP header values
INVITE: From: <derived from the Calling party number Address digits or
anonymous@anonymous.invalid>
P Asserted Identity: <matched Public user identity>
Privacy: id, header, user</td></tr>
<tr><td colspan="4">DSS1 Parameter values
SETUP: Calling party number
Type of number = Subscriber
Presentation Restriction Presentation restricted
Address digits present</td></tr>
<tr><td colspan="4">Message flow</td></tr>
<tr><td colspan="2">End device</td><td></td><td>Test equipment</td></tr>
<tr><td colspan="2">SETUP</td><td>→</td><td>INVITE</td></tr>
<tr><td colspan="2">SETUP ACKNOWLEDGE</td><td>←</td><td>100 Trying</td></tr>
<tr><td colspan="4">Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
OIP_OIR</td><td>TP_501_234</td><td>Reference
subclause 5.2.3.2 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.1.3/1</td></tr>
<tr><td colspan="4">Test purpose
Mw interface. SETUP received, Calling party number Subscriber Presentation Restriction Indicator is set to restricted

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'Subscriber', Presentation Restriction Indicator is set to Presentation restricted, an INVITE request is sent. If the Calling party number digits does not matche with a registered Public identity, a P Asserted Identity header field is present, the userinfo is set to the Default Public user identity. If the Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format or set to anonymous@anonymous.invalid, a Privacy header is present, set to 'id' and 'header' and 'user'.</td></tr>
<tr><td colspan="4">SIP header values
INVITE: From: <derived from the Calling party number Address digits or
anonymous@anonymous.invalid>
P Asserted Identity: <Default Public user identity>
Privacy: id, header, user</td></tr>
<tr><td colspan="4">DSS1 Parameter values
SETUP: Calling party number
Type of number = Subscriber
Presentation Restriction Presentation restricted
Address digits present</td></tr>
</table>

Message flow

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying

Apply post test routine

TSS OIP_OIR	TP_501_235	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1

Test purpose

Mw interface. SETUP received, Calling party number Unknown Presentation Restriction Indicator is set to restricted

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'Unknown', Presentation Restriction Indicator is set to Presentation restricted, an INVITE request is sent. If the Calling party number digits matches with a registered Public identity, a P Asserted Identity header field is present, the userinfo is set to the identified Public user identity. If the userinfo of the From header is derived from the address digits of the calling party number IE in the local number format or set to anonymous@anonymous.invalid, a Privacy header is present, set to 'id' and 'header' and 'user'.

SIP header values

INVITE: From: <derived from the Calling party number Address digits or
anonymous@anonymous.invalid>
P Asserted Identity: <matched Public user identity>
Privacy: id, header, user

DSS1 Parameter values

SETUP: Calling party number
Type of number = Unknown
Presentation Restriction Presentation restricted
Address digits present

Message flow

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying

Apply post test routine

TSS OIP_OIR	TP_501_236	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1

Test purpose

Mw interface. SETUP received, Calling party number Unknown Presentation Restriction Indicator is set to restricted

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'Unknown', Presentation Restriction Indicator is set to Presentation restricted, an INVITE request is sent. If the Calling party number digits does not matche with a registered Public identity, a P Asserted Identity header field is present, the userinfo is set to the Default Public user identity. If the Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format or set to anonymous@anonymous.invalid, a Privacy header is present, set to 'id' and 'header' and 'user'

SIP header values

INVITE: From: <derived from the Calling party number Address digits or
anonymous@anonymous.invalid>
P Asserted Identity: <Default Public user identity>
Privacy: id, header, user

DSS1 Parameter values SETUP: Calling party number Type of number = Unknown Presentation Restriction Presentation restricted Address digits present
Message flow **End device** / **Test equipment** SETUP → INVITE SETUP ACKNOWLEDGE ← 100 Trying **Apply post test routine**

TSS OIP_OIR	TP_501_237	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1
Test purpose *Mw interface. SETUP received, Calling party number Address digits not present Presentation Restriction Indicator is set to restricted* Ensure that on receipt of a SETUP message containing a Calling party number Address digits not present, Presentation Restriction Indicator is set to Presentation restricted, an INVITE request is sent. If a P Asserted Identity header field is present, the userinfo is set to the Default Public user identity. The Userinfo of the From header is set to 'unavailable@unknown.invalid'. If a Privacy header is present set to 'id' and 'header' and 'user'			
SIP header values INVITE: From: <unavailable@unknown.invalid> P Asserted Identity: <Default Public user identity> Privacy: id, header, user			
DSS1 Parameter values SETUP: Calling party number Presentation Restriction Presentation restricted Address digits not present			
Message flow **End device** / **Test equipment** SETUP → INVITE SETUP ACKNOWLEDGE ← 100 Trying **Apply post test routine**			

TSS OIP_OIR	TP_501_238	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1
Test purpose *Mw interface. SETUP received, Calling party number is not present* Ensure that on receipt of a SETUP message containing a Calling party number Information Element not present Presentation, an INVITE request is sent. If a P Asserted Identity header field is present, the userinfo is set to the Default Public user identity. The Userinfo of the From header is set to 'unavailable@unknown.invalid'. If a Privacy header is present set to 'id' and 'header' and 'user'.			
SIP header values INVITE: From: <unavailable@unknown.invalid> P Asserted Identity: <Default Public user identity> Privacy: id, header, user			

DSS1 Parameter values

SETUP: Calling party number not present

Message flow

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying

Apply post test routine

TSS OIP_OIR	TP_501_239	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1

Test purpose

Mw interface. SETUP received, Calling party number is not present

Ensure that on receipt of a SETUP message containing a Calling party number Information Element not present Presentation, an INVITE request is sent. If a P Asserted Identity header field is present, the userinfo is set to the Default Public user identity. The Userinfo of the From header is set to 'unavailable@unknown.invalid'. A Privacy header is not present.

SIP header values

INVITE: From: <unavailable@unknown.invalid>
P Asserted Identity: <Default Public user identity>

DSS1 Parameter values

SETUP: Calling party number not present

Message flow

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying

Apply post test routine

TSS OIP_OIR	TP_501_240	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1

Test purpose

SETUP received, Calling party number NOA national significant Presentation Restriction Indicator is set to allowed

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'National number', Presentation Restriction Indicator is set to Presentation allowed, an INVITE request is sent. If the Calling party number digits matches with a registered Public identity, a P Asserted Identity header field is present, the userinfo is set to the identified Public user identity. If the Userinfo of the From header is derived from the address digits of the calling party number IE in the local number formata Privacy header is present set to 'none'.

SIP header values

INVITE: From: <derived from the Calling party number Address digits>
P Asserted Identity: <identified Public user identity>
Privacy: none

DSS1 Parameter values

SETUP: Calling party number
Type of number = National number
Presentation Restriction Presentation allowed
Address digits present

Message flow		
End device		**Test equipment**
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying
Apply post test routine		

TSS OIP_OIR	TP_501_241	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1

Test purpose

SETUP received, Calling party number NOA national significant Presentation Restriction Indicator is set to allowed

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'National number', Presentation Restriction Indicator is set to Presentation allowed and an INVITE request is sent. If the Calling party number digits does not match with a registered Public identity, a P Asserted Identity header field is present, the userinfo is set to the Default Public user identity. If the Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format,a Privacy header is present set to 'none'.

SIP header values

INVITE: From: <derived from the Calling party number Address digits>
P Asserted Identity: <Default Public user identity>
Privacy: none

DSS1 Parameter values

SETUP: Calling party number
Type of number = National number
Presentation Restriction Presentation allowed
Address digits present

Message flow		
End device		**Test equipment**
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying
Apply post test routine		

TSS OIP_OIR	TP_501_242	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1

Test purpose

SETUP received, Calling party number NOA international Presentation Restriction Indicator is set to allowed

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'International number', Presentation Restriction Indicator is set to Presentation allowed and an INVITE request is sent. If the Calling party number digits matches with a registered Public identity, a P Asserted Identity header field is present, the userinfo is set to the identified Public user identity. If the Userinfo of the From header is derived from the address digits of the calling party number IE in the global format, a Privacy header is present set to 'none'.

SIP header values

INVITE: From: <derived from the Calling party number Address digits>
P Asserted Identity: <matched Public user identity>
Privacy: none

DSS1 Parameter values

SETUP: Calling party number

Type of number = International number

Presentation Restriction Presentation allowed

Address digits present

Message flow

End device		**Test equipment**
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying

Apply post test routine

TSS OIP_OIR	TP_501_243	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1

Test purpose

SETUP received, Calling party number NOA international Presentation Restriction Indicator is set to allowed

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'International number', Presentation Restriction Indicator is set to Presentation allowed and an INVITE request is sent. If the Calling party number digits does not match with a registered Public identity, a P Asserted Identity header field is present, the userinfo is set to the Default Public user identity. If the Userinfo of the From header is derived from the address digits of the calling party number IE in the global number format, a Privacy header is present set to 'none'.

SIP header values

INVITE: From: <derived from the Calling party number Address digits>

P Asserted Identity: <Default Public user identity>

Privacy: none

DSS1 Parameter values

SETUP: Calling party number

Type of number = International number'

Presentation Restriction Presentation allowed

Address digits present

Message flow

End device		**Test equipment**
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying

Apply post test routine

TSS OIP_OIR	TP_501_244	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1

Test purpose

SETUP received, Calling party number NOA subscriber Presentation Restriction Indicator is set to allowed

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'Subscriber', Presentation Restriction Indicator is set to Presentation allowed and an INVITE request is sent. If the Calling party number digits matches with a registered Public identity, a P Asserted Identity header field is present, the userinfo is set to the identified Public user identity. If the Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format,a Privacy header is present set to 'none'.

SIP header values INVITE: From: <derived from the Calling party number Address digits> P Asserted Identity: <matched Public user identity> Privacy: none
DSS1 Parameter values SETUP: Calling party number Type of number = Subscriber Presentation Restriction Presentation allowed Address digits present
Message flow **End device** / **Test equipment** SETUP ➔ INVITE SETUP ACKNOWLEDGE ← 100 Trying **Apply post test routine**

TSS OIP_OIR	TP_501_245	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1
Test purpose *SETUP received, Calling party number NOA subscriber Presentation Restriction Indicator is set to allowed* Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'Subscriber', Presentation Restriction Indicator is set to Presentation allowed aand n INVITE request is sent. If the Calling party number digits does not match with a registered Public identity, a P Asserted Identity header field is present, the userinfo is set to the Default Public user identity. The Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format. A Privacy header is present set to 'none'.			
SIP header values INVITE: From: <derived from the Calling party number Address digits> P Asserted Identity: <Default Public user identity> Privacy: none			
DSS1 Parameter values SETUP: Calling party number Type of number = Subscriber Presentation Restriction Presentation allowed Address digits present			
Message flow **End device** / **Test equipment** SETUP ➔ INVITE SETUP ACKNOWLEDGE ← 100 Trying **Apply post test routine**			

TSS OIP_OIR	TP_501_246	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1

Test purpose

SETUP received, Calling party number NOA unknown Presentation Restriction Indicator is set to allowed

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'Unknown', Presentation Restriction Indicator is set to Presentation allowed and an INVITE request is sent. If the Calling party number digits matches with a registered Public identity, a P Asserted Identity header field is present, the userinfo is set to the identified Public user identity. The Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format. A Privacy header is present set to 'none'.

SIP header values

INVITE: From: <derived from the Calling party number Address digits>
P Asserted Identity: <matched Public user identity>
Privacy: none

DSS1 Parameter values

SETUP: Calling party number
Type of number = Unknown
Presentation Restriction Presentation allowed
Address digits present

Message flow

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying

Apply post test routine

TSS OIP_OIR	TP_501_247	Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1

Test purpose

SETUP received, Calling party number NOA national significant Presentation Restriction Indicator is set to allowed

Ensure that on receipt of a SETUP message containing a Calling party number Address digits present and the Type of number is set to 'Unknown', Presentation Restriction Indicator is set to Presentation allowed and an INVITE request is sent. If the Calling party number digits does not match with a registered Public identity, a P Asserted Identity header field is present, the userinfo is set to the Default Public user identity. The Userinfo of the From header is derived from the address digits of the calling party number IE in the local number format. A Privacy header is present set to 'none'.

SIP header values

INVITE: From: <derived from the Calling party number Address>
P Asserted Identity: <Default Public user identity>
Privacy: none

DSS1 Parameter values

SETUP: Calling party number
Type of number = Unknown
Presentation Restriction Presentation allowed
Address digits present

Message flow

End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying

Apply post test routine

7.2.5.2 Terminating identification presentation and terminating identification restriction)

7.2.5.2.1 Test purposes for ISDN

<table>
<tr><td>TSS
TIP_TIR</td><td>TP_502_101</td><td>Reference
subclause 5.2.2.1.1 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.1.3/1</td></tr>
<tr><td colspan="4">Test purpose
CONNECT no Connected number or no valid Connected number received, 200 OK (INVITE) is sent

Ensure that a 200 OK (INVITE) is sent when a CONNECT message was received. The P-Asserted-Identity contains the value saved from the P-Called-Party-ID header that was received in the INVITE request if no Connected number or no valid Connected number is present in the received CONNECT. No "from-change" tag in the supported header.</td></tr>
<tr><td colspan="4">SIP header values
INVITE:
P-Called-Party-ID = <called number>

200 OK (INVITE):
P-Asserted-Identity = <called number></td></tr>
<tr><td colspan="4">DSS1 Parameter values</td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><th colspan="2">Test equipment</th><th colspan="2">End device</th></tr>
<tr><td>INVITE</td><td>→</td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td>←</td><td>ALERTING</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td>←</td><td>CONNECT</td></tr>
<tr><td>ACK</td><td>→</td><td>→</td><td>CONNECT ACK</td></tr>
</table>
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
TIP_TIR</td><td>TP_502_102</td><td>Reference
subclause 5.2.2.1.1 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.1.3/1</td></tr>
<tr><td colspan="4">Test purpose
CONNECT unknown or Subscriber number received, 200 OK (INVITE) is sent

Ensure that a 200 OK (INVITE) is sent when a CONNECT message was received. The P-Asserted-Identity is present with that value, including the display name if previously stored during registration representing the terminating user indicated in the connected number type of number: unknown or Subscriber number. No "from-change" tag in the supported header</td></tr>
<tr><td colspan="4">SIP header values
200 OK (INVITE):
P-Asserted-Identity = <registered number></td></tr>
<tr><td colspan="4">DSS1 Parameter values
CONNECT: Connected number
Nature of address = unknown or Subscriber number
Number Digits = <connected number></td></tr>
</table>

Message flow **Test equipment** **End device** INVITE → → SETUP 180 Ringing ← ← ALERTING 200 OK INVITE ← ← CONNECT ACK → → CONNECT ACK **Apply post test routine**

TSS TIP_TIR	TP_502_103	Reference subclause 5.2.2.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1
Test purpose *CONNECT National number received, 200 OK (INVITE) is sent* When an INVITE is received and no 'from-change' tag is contained in the Supported header,ensure that a 200 OK (INVITE) is sent when a CONNECT message was received. The P-Asserted-Identity is present with that value, including the display name if previously stored during registration representing the terminating user indicated in the connected number type of number: **National number**. No "from-change" tag in the supported header.			
SIP header values 200 OK (INVITE): P-Asserted-Identity = <registered number>			
DSS1 Parameter values CONNECT: Connected number Nature of address = National number Number Digits = <connected number>			
Message flow **Test equipment** **End device** INVITE → → SETUP 180 Ringing ← ← ALERTING 200 OK INVITE ← ← CONNECT ACK → → CONNECT ACK **Apply post test routine**			

TSS TIP_TIR	TP_502_104	Reference subclause 5.2.2.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1
Test purpose *CONNECT international number received, 200 OK (INVITE) is sent* When an INVITE is received and no 'from-change' tag contained in the Supported header,ensure that a 200 OK (INVITE) is sent when a CONNECT message was received. The P-Asserted-Identity is present with that value, including the display name if previously stored during registration representing the terminating user indicated in the connected number type of number: **international number**. No "from-change" tag in the supported header			
SIP header values 200 OK (INVITE): P-Asserted-Identity = <registered number>			

DSS1 Parameter values CONNECT: Connected number Nature of address = international number Number Digits = <connected number>
Message flow **Test equipment** / **End device** INVITE → → SETUP 180 Ringing ← ← ALERTING 200 OK INVITE ← ← CONNECT ACK → → CONNECT ACK **Apply post test routine**

TSS TIP_TIR	TP_502_105	Reference subclause 5.2.2.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1 AND NOT 5.4/2
Test purpose *CONNECT National number received, 200 OK (INVITE) is sent* When an INVITE is received and a 'from-change' tag contained in the Supported header,ensure that a 200 OK (INVITE) is sent when a CONNECT message was received. The P-Asserted-Identity is present with that value, including the display name if previously stored during registration representing the terminating user indicated in the connected number type of number: **National number**. No "from-change" tag in the supported header.			
SIP header values 200 OK (INVITE): P-Asserted-Identity = <registered number>			
DSS1 Parameter values CONNECT: Connected number Nature of address = National number Number Digits = <connected number>			
Message flow **Test equipment** / **End device** INVITE → → SETUP 180 Ringing ← ← ALERTING 200 OK INVITE ← ← CONNECT ACK → → CONNECT ACK **Apply post test routine**			

TSS TIP_TIR	TP_502_106	Reference subclause 5.2.2.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1 AND NOT 5.4/2
Test purpose *CONNECT international number received, 200 OK (INVITE) is sent* If an INVITE is received and a 'from-change' tag contained in the Supported header,ensure that a 200 OK (INVITE) is sent when a CONNECT message was received. If the P-Asserted-Identity is present with that value, including the display name if previously stored during registration, representing the terminating user indicated in the connected number type of number: **international number,** no "from-change" tag in the supported header.			

SIP header values

200 OK (INVITE):

P-Asserted-Identity = <registered number>

DSS1 Parameter values

CONNECT: Connected number

Nature of address = international number

Number Digits = <connected number>

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→	→	CONNECT ACK

Apply post test routine

TSS TIP_TIR	TP_502_107	Reference subclause 5.2.2.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1 AND 5.4/2

Test purpose

CONNECT National number received, 200 OK (INVITE) is sent

If an INVITE is received and a 'from-change' tag contained in the Supported header,ensure that a 200 OK (INVITE) is sent when a CONNECT message was received. If the P-Asserted-Identity is present with that value, including the display name if previously stored during registration, representing the terminating user indicated in the connected number type of number: National number,a "from-change" tag is present in the supported header. An UPDATE is sent and the From header contains the digits received in the Connected Number.

SIP header values

INVITE:

Supported: from-change

200 OK (INVITE):

P-Asserted-Identity = <registered number>

UPDATE:

From: <connected number>

DSS1 Parameter values

CONNECT: Connected number

Nature of address = National number

Number Digits = <connected number>

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→	→	CONNECT ACK
UPDATE	←		
200 OK UPDATE	→		

Apply post test routine

<table>
<tr><td>TSS
TIP_TIR</td><td>TP_502_108</td><td>Reference
subclause 5.2.2.1.1 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.1.3/1 AND 5.4/2</td></tr>
<tr><td colspan="4">Test purpose
CONNECT international number received, 200 OK (INVITE) is sent

If an INVITE is received and a 'from-change' tag contained in the Supported header,ensure that a 200 OK (INVITE) is sent when a CONNECT message was received. If the P-Asserted-Identity is present with that value, including the display name if previously stored during registration, representing the terminating user indicated in the connected number type of number: international number,a "from-change" tag is present in the supported header, an UPDATE is sent and the From header contains the digits received in the Connected Number.</td></tr>
<tr><td colspan="4">SIP header values
INVITE:
Supported: from-change

200 OK (INVITE):
P-Asserted-Identity = <registered number>

UPDATE:
From: <connected number></td></tr>
<tr><td colspan="4">DSS1 Parameter values
CONNECT: Connected number
Nature of address = international number
Number Digits = <connected number></td></tr>
<tr><td colspan="4">Message flow
Test equipment | End device
INVITE ➔ ➔ SETUP
180 Ringing ← ← ALERTING

200 OK INVITE ← ← CONNECT
ACK ➔ ➔ CONNECT ACK

UPDATE ←
200 OK UPDATE ➔
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
TIP_TIR</td><td>TP_502_109</td><td>Reference
subclause 5.2.2.1.1 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.1.3/1 AND 5.4/2</td></tr>
<tr><td colspan="4">Test purpose
No 'from-change' tag in the INVITE received, 200 OK (INVITE) is sent and no 'from-change' tag is present

If an INVITE is received and no 'from-change' tag contained in the Supported header,ensure that a 200 OK (INVITE) is sent when a CONNECT message was received. a "from-change" tag is not present in the supported header nor UPDATE is sent</td></tr>
<tr><td colspan="4">SIP header values
200 OK (INVITE):
P-Asserted-Identity = <registered number></td></tr>
</table>

DSS1 Parameter values

CONNECT: Connected number

Nature of address = national number

Number Digits = <connected number>

Message flow

Test equipment			End device
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→	→	CONNECT ACK

Apply post test routine

TSS TIP_TIR	TP_502_110	Reference subclause 5.2.2.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2

Test purpose

No 'from-change' tag in the INVITE received, CONNECT received, 200 OK (INVITE) is sent no 'from-change' tag present

Ensure that no 'from-change' tag in a Supported header in the sent 200 OK INVITE is present if no 'from-change' tag was received in the supported header in the received INVITE. No P-Asserted-Identity or P-Preferred-Identity header is present in the 200 OK INVITE. No UPDATE is sent.

SIP header values

DSS1 Parameter values

Message flow

Test equipment			End device
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→	→	CONNECT ACK

Apply post test routine

TSS TIP_TIR	TP_502_111	Reference subclause 5.2.2.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2 AND 5.4/2

Test purpose

'from-change' tag in the INVITE received, CONNECT received, 200 OK (INVITE) is sent no 'from-change' tag present

Ensure that no 'from-change' tag in a Supported header in the sent 200 OK INVITE is present if a 'from-change' tag was received in the supported header in the received INVITE and the nature of address of the received CONNECT is set to Nature_of_address as indicated in Table 7.2.5.2.1-1

No P-Asserted-Identity or P-Preferred-Identity header is present in the 200 OK INVITE. No UPDATE is sent.

SIP header values

DSS1 Parameter values

Message flow **Test equipment** / **End device** INVITE ➔ ➔ SETUP 180 Ringing ← ← ALERTING 200 OK INVITE ← ← CONNECT ACK ➔ ➔ CONNECT ACK **Apply post test routine**

TSS TIP_TIR	TP_502_112	Reference subclause 5.2.2.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2 AND 5.4/2
Test purpose *'from-change' tag in the INVITE received, CONNECT received, 200 OK (INVITE) is sent no 'from-change' tag present* Ensure that no 'from-change' tag in a Supported header in the sent 200 OK INVITE is present if a 'from-change' tag was received in the supported header in the received INVITE and no or invalid connected number information element was present in the received CONNECT. No P-Asserted-Identity or P-Preferred-Identity header is present in the 200 OK INVITE. No UPDATE is sent.			
SIP header values **INVITE**: 'from-change' 200 OK INVITE: no 'from-change' tag			
DSS1 Parameter values **CONNECT**: no or invalid connected number information element			
Message flow **Test equipment** / **End device** INVITE ➔ ➔ SETUP 180 Ringing ← ← ALERTING 200 OK INVITE ← ← CONNECT ACK ➔ ➔ CONNECT ACK **Apply post test routine**			

Table 7.2.5.2.1-1 – Handling of Nature of Address indicator

Nature_of_address	Nature of address value
VA_1	Unknown
VA_2	Subscriber number

TSS TIP_TIR	TP_502_113	Reference subclause 5.2.2.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2 AND 5.4/2

Test purpose

'from-change' tag in the INVITE received, CONNECT received, 200 OK (INVITE) is sent 'from-change' tag present

Ensure that the 'from-change' tag in a Supported header in the sent 200 OK INVITE is present if a 'from-change' tag was received in the supported header in the received INVITE and the nature of address of the received CONNECT is set to Nature_of_address as indicated in Table 7.2.5.2.1-2

No P-Asserted-Identity or P-Preferred-Identity header is present in the 200 OK INVITE. An UPDATE is sent and the From header contains the digits received in the Connected Number.

SIP header values

INVITE: Supported: from-change

200 OK (INVITE):

Supported: from-change

UPDATE: From: <connected number>

DSS1 Parameter values

CONNECT: Connected number

Nature of address = National number or international number

Number Digits = <connected number>

Message flow

Test equipment			**End device**
INVITE	➔	➔	SETUP
180 Ringing	🡐	🡐	ALERTING
200 OK INVITE	🡐	🡐	CONNECT
ACK	➔	➔	CONNECT ACK
UPDATE	🡐		
200 OK UPDATE	➔		

Apply post test routine

TSS TIP_TIR	TP_502_114	Reference subclause 5.2.2.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2 AND 5.4/2

Test purpose

Connected Sub-address is transported in the From header of the UPDATE

Ensure that on receipt of a Connected Sub-Address in the received CONNECT message, an as isub parameter is sent in the From header of the sent UPDATE.

SIP header values

INVITE: Supported: from-change

200 OK (INVITE):

Supported: from-change

UPDATE: From: URI, isub = <sub-address>

DSS1 Parameter values

CONNECT: Connected Sub-Address <sub-address>

Message flow			
Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→	→	CONNECT ACK
UPDATE	←		
200 OK UPDATE	→		
Apply post test routine			

Table 7.2.5.2.1-2 – Mapping of Nature of Address indicator into From header in the UPDATE

Nature_of_address	Nature of address value	Connected number digits	Form header URI
VA_1	National number	NDC+SN	'+' CC+NDC+SN
VA_2	international number	CC+NDC+SN	'+' CC+NDC+SN

TSS TIP_TIR	TP_502_115	Reference subclause 5.2.2.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2 AND 5.4/3
Test purpose *CONNECT Connected number Presentation restricted received, 200 OK (INVITE) is sent* Ensure that on receipt of a CONNECT message containing a Connected number Information Element and the Presentation restriction indicator is set to 'Presentation restricted', a 200 OK INVITE is sent, a Privacy header contains the values "id" and "header" and "user".			
SIP header values **200 OK:** Privacy: id, header, user			
DSS1 Parameter values CONNECT: Connected number Presentation restriction = Presentation restricted			

Message flow			
Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→	→	CONNECT ACK
Apply post test routine			

TSS TIP_TIR	TP_502_116	Reference subclause 5.2.2.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/2

Test purpose

A 'from-change' tag is contained in the initial INVITE request if a SETUP message was received

Ensure that on receipt of a SETUP message an INVITE message is sent to the terminating user and a 'from-change' tag is contained in the Supported header.

SIP header values

INVITE: Supported: from-change

DSS1 Parameter values

Message flow

End device			**Test equipment**
SETUP	→	→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
CONNECT ACK	→	→	ACK

Apply post test routine

TSS TIP_TIR	TP_502_117	Reference subclause 5.2.2.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1

Test purpose

The Userinfo of the P-Asserted-Identity is sent in a connected number. No Privacy header present or Privacy value 'none' 'from-change' tag not present in the 200 OK (INVITE).

Ensure that on receipt of a 200 OK INVITE where the 'from-change' tag is not present in the Supported header and no Privacy header is present, the Userinfo of the P-Asserted-Identity is in the form of a tel URI containing a E.164 number, a CONNECT message is sent to the calling user and a Connected number is present coded as described below.

Connected number

Type of number

National number

sip: local-number-digits; phone-context=nat@hostportion; user=phone

International number

sip: global-number-digits@hostportion; user=phone

Numbering plan identification = ISDN/Telephony numbering plan

Presentation indicator = **Presentation allowed**

Screening indicator = Network provided

Number digits derived from the userinfo of the P-Asserted-Identity.

In case where the global number and the country code is the same as the SUT or line is located, the country code is removed from the number of the Type of number is set to "national number.

SIP header values

200 OK: P-Asserted-Identity

DSS1 Parameter values

CONNECT: Connected number

Message flow			
End device			**Test equipment**
SETUP	→	→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
CONNECT ACK	→	→	ACK
Apply post test routine			

TSS	TP_502_118	Reference	Selection expression
TIP_TIR		subclause 5.2.2.1.2 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/3

Test purpose

The Userinfo of the P-Asserted-Identity is sent in a connected number. No Privacy header present or Privacy value 'none' 'from-change' tag present in the 200 OK (INVITE).

Ensure that on receipt of a 200 OK INVITE where the 'from-change' tag is present in the Supported header and no Privacy header is present, the sending of the CONNECT message is held until the UPDATE is received. If the Userinfo in the From header in the UPDATE is in the form of a tel URI containing a E.164 number a CONNECT message is sent to the calling user and a Connected number is present coded as described below.

Connected number

Type of number

- National number
  - sip: local-number-digits; phone-context=nat@hostportion; user=phone
- International number
  - sip: global-number-digits@hostportion; user=phone

Numbering plan identification = ISDN/Telephony numbering plan

Presentation indicator = **Presentation allowed**

Screening indicator = User provided, not verified

Number digits derived from the userinfo of the From header.

- In case where the global number and the country code is the same as the SUT or line is located, the country code is removed from the number of the Type of number is set to "national number.

SIP header values

UPDATE:From

DSS1 Parameter values

CONNECT: Connected number

Message flow			
End device			**Test equipment**
SETUP	→	→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
CONNECT ACK	→	→	ACK
UPDATE	←		
200 OK UPDATE	→		
Apply post test routine			

TSS TIP_TIR	TP_502_119	Reference subclause 5.2.2.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/3

Test purpose

The Userinfo of the From header is sent in a connected number. Privacy header value 'id' or 'header' 'from-change' tag not present in the 200 OK (INVITE).

Ensure that on receipt of a 200 OK INVITE where the 'from-change' tag is not present in the Supported header and a Privacy header is present, the priv-value set to 'id' or 'header' the Userinfo of the P-Asserted-Identity is in the form of a tel URI containing a E.164 number and a CONNECT message is sent to the calling user and a Connected number is present coded as described below.

Connected number

Type of number = Unknown

Numbering plan identification = Unknown

Presentation indicator = **Presentation restricted**

Screening indicator = Network provided

Number digits derived from the userinfo of the P-Asserted-Identity.

Number digits not present

SIP header values

200 OK:

P-Asserted-Identity

Privacy: id,header,user

DSS1 Parameter values

CONNECT: Connected number

Message flow

End device			Test equipment
SETUP	→	→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
CONNECT ACK	→	→	ACK

Apply post test routine

TSS TIP_TIR	TP_502_120	Reference subclause 5.2.2.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/3

Test purpose

The Userinfo of the From header is sent in a connected number. Privacy header value 'id' or 'header 'from-change' tag present in the 200 OK (INVITE).

Ensure that on receipt of a 200 OK INVITE and the 'from-change' tag is present in the Supported header and a Privacy header is present the priv-value set to 'id' or 'header', the sending of the CONNECT message is held until the UPDATE is received. If the Userinfo in the From header in the UPDATE is in the form of a tel URI containing a E.164 number a CONNECT message is sent to the calling user and a Connected number is present coded as described below.

Connected number

Type of number = Unknown

Numbering plan identification = Unknown

Presentation indicator = **Presentation restricted**

Screening indicator = Network provided

Number digits not present

SIP header values UPDATE:From: anonymous Privacy: id,header
DSS1 Parameter values CONNECT: Connected number
Message flow

End device			Test equipment
SETUP	→	→	INVITE
ALERTING	←	←	180 Ringing
		←	200 OK INVITE
		→	ACK
CONNECT	←	←	UPDATE
CONNECT ACK	→	→	200 OK UPDATE

Apply post test routine

TSS TIP_TIR	TP_502_121	Reference subclause 5.2.2.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/3

Test purpose

The Userinfo of the From header is sent in a connected number. Privacy header value 'id' or 'header 'from-change' tag present in the 200 OK (INVITE).

Ensure that on receipt of a 200 OK INVITE where the 'from-change' tag is present in the Supported header and a Privacy header is present, the priv-value set to 'id' or 'header', the sending of the CONNECT message is held until the UPDATE is received. If the Userinfo in the From header in the UPDATE is in the form of a tel URI containing a E.164 number, a CONNECT message is sent to the calling user and a Connected number is present coded as described below.

Connected number

Type of number = Unknown

Numbering plan identification = Unknown

Presentation indicator = **Presentation restricted**

Screening indicator = Network provided

Number digits not present

SIP header values UPDATE:From: anonymous Privacy: id,header
DSS1 Parameter values CONNECT: Connected number
Message flow

End device			Test equipment
SETUP	→	→	INVITE
ALERTING	←	←	180 Ringing
		←	200 OK INVITE
		→	ACK
CONNECT	←	←	UPDATE
CONNECT ACK	→	→	200 OK UPDATE

Apply post test routine

TSS TIP_TIR	TP_502_122	Reference subclause 5.2.2.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/3
Test purpose *No P-Asserted-Identity header and no Privacy id or header received, no Connected number is sent* Ensure that on receipt of a 200 OK INVITE where no P-Asserted-Identity and no Privacy value 'id' or 'header' is present, a CONNECT message is sent to the calling user and no Connected number is present. NOTE – This is the indication, the calling user has not subscribed to the COLP service.			
SIP header values			
DSS1 Parameter values			
Message flow			

End device			**Test equipment**
SETUP	→	→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
CONNECT ACK	→	→	ACK
Apply post test routine			

TSS TIP_TIR	TP_502_123	Reference subclause 5.2.2.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/3
Test purpose *No P-Asserted-Identity header and no Privacy id or header received, no Connected number is sent* Ensure that on receipt of a 200 OK INVITE and no P-Asserted-Identity and no Privacy value 'id' or 'header' is present, a CONNECT message is sent to the calling user and no Connected number is present. NOTE: This is the indication, the calling user has not subscribed to the COLP service.			
SIP header values			
DSS1 Parameter values			
Message flow			

End device			**Test equipment**
SETUP	→	→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
CONNECT ACK	→	→	ACK
Apply post test routine			

<table>
<tr><td>TSS
TIP_TIR</td><td>TP_502_124</td><td>Reference
subclause 5.2.2.1.2 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.4/3</td></tr>
<tr><td colspan="4">Test purpose
'from-change' tag in 200 OK (INVITE) received, expiry of timer T_{TIR1}

Ensure that on receipt of a 200 OK INVITE and the 'from-change' tag is present in the Supported header and no Privacy header is present, timer TTIR1 is started and the CONNECT message is held. After expiry of TTIR1, the CONNECT message is sent, the Userinfo of the P-Asserted-Identity is mapped into the Connected number and coded as described below:
Connected number
Type of number
National number
sip: local-number-digits; phone-context=nat@hostportion; user=phone
International number
sip: global-number-digits@hostportion; user=phone
Numbering plan identification = ISDN/Telephony numbering plan
Presentation indicator = Presentation allowed
Screening indicator = User provided, not verified
Number digits derived from the userinfo of the From header.
In case where the global number and the country code is the same as the SUT or line is located, the country code is removed from the number of the Type of number is set to "national number.</td></tr>
<tr><td colspan="4">SIP header values</td></tr>
<tr><td colspan="4">DSS1 Parameter values
CONNECT: Connected number</td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><td>End device</td><td></td><td></td><td>Test equipment</td></tr>
<tr><td>SETUP</td><td>→</td><td>→</td><td>INVITE</td></tr>
<tr><td>ALERTING</td><td>←</td><td>←</td><td>180 Ringing</td></tr>
<tr><td>CONNECT</td><td>←</td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td>CONNECT ACK</td><td>→</td><td>→</td><td>ACK</td></tr>
</table>Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
TIP_TIR</td><td>TP_502_125</td><td>Reference
subclause 5.2.2.1.2 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.4/3</td></tr>
<tr><td colspan="4">Test purpose
Several identities received in different responses.

Ensure that on receipt of several P-Asserted-Identity headers in different responses, the Connected number sent to the calling party is derived from the latest received P-Asserted header.</td></tr>
<tr><td colspan="4">SIP header values
180: P-Asserted-Identity identity 1
200 OK: P-Asserted-Identity identity 2</td></tr>
<tr><td colspan="4">DSS1 Parameter values
CONNECT: Connected number identity 2</td></tr>
</table>

Message flow			
End device			**Test equipment**
SETUP	→	→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
CONNECT ACK	→	→	ACK
Apply post test routine			

TSS TIP_TIR	TP_502_126	Reference subclause 5.2.2.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/3
Test purpose *Mapping of isub parameter in the UPDATE From header into Connected subaddress* Ensure that on receipt of several P-Asserted-Identity headers in different responses, the Connected number sent to the calling party is derived from the latest received P-Asserted header.			
SIP header values 180: P-Asserted-Identity identity 1 200 OK: P-Asserted-Identity identity 2			
DSS1 Parameter values CONNECT: Connected number identity 2			
Message flow			
End device			**Test equipment**
SETUP	→	→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
CONNECT ACK	→	→	ACK
Apply post test routine			

7.2.5.3 Communication HOLD

7.2.5.3.1 Test purposes for POTS

TSS HOLD	TP_503_101	Reference subclause B.4.2.2 and C.1.2.1 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1
Test purpose *Originating user sets the remote party on hold* Ensure that when a ReINVITE or UPDATE request is received, the SDP description for the active media stream is set to a=sendonly, a 200 OK is sent the a line in the SDP is set to “recvonly”.			
SIP header values INVITE/UPDATE SDP a=sendonly 200 OK SDP a=recvonly			

Message flow **Test equipment** **End device** **A confirmed dialogue exists** **CASE A** INVITE → 200 OK INVITE ← ACK → **CASE B** UPDATE → 200 OK UPDATE ← **Apply post test routine**

TSS HOLD	TP_503_102	Reference subclause B.4.2.2 and C.1.2.1 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1
Test purpose *User resumes the remote party* Ensure that when the user is set on hold, upon receipt of a ReINVITE or UPDATE request, the SDP description for the held media stream is set to a=sendrecv, a 200 OK is sent and the a line in the SDP is set to “sendrecv”. NOTE – sendrecvis the default value.			
SIP header values INVITE SDP a=sendrecv (or absent) 200 OK SDP a=sendrecv (or absent)			
Message flow **Test equipment** **End device** **A confirmed dialogue exists** **CASE A** INVITE1 → 200 OK INVITE1 ← ACK → INVITE2 → 200 OK INVITE2 ← ACK → **CASE B** UPDATE1 → 200 OK UPDATE1 ← UPDATE2 → 200 OK UPDATE3 ← **Apply post test routine**			

TSS HOLD	TP_503_103	Reference subclause 5.3.1.4 of [TS183 043]	Selection expression PICS 5.1.1/1 AND 5.3/6

Test purpose

User sets the remote party on hold

Ensure that when flash-hook notification from the MGC component is received a ReINVITE request is sent. The SDP description for the active media stream is set to a=sendonly.

SIP header values

INVITE
SDP
a=sendonly

200 OK
SDP
a=recvonly

Message flow

End device		Test equipment
	A confirmed dialogue exists	
flash-hook	→	INVITE
	←	200 OK INVITE
	→	ACK
	Apply post test routine	

TSS HOLD	TP_503_104	Reference subclause 5.3.1.4 of [TS183 043]	Selection expression PICS 5.1.1/1 AND 5.3/6

Test purpose

User resumes the remote held party

Ensure that if the remote party is on hold when flash-hook notification from the MGC component is received a ReINVITE request is sent. The SDP description for the active media stream is set to a=sendrecv

SIP header values

INVITE2
SDP
a=sendrecv

200 OK2
SDP
a=sendrecv

Message flow

End device		Test equipment
	A confirmed dialogue exists	
flash-hook	→	INVITE1
	←	200 OK INVITE1
	→	ACK
flash-hook	→	INVITE2
	←	200 OK INVITE2
	→	ACK
	Apply post test routine	

7.2.5.3.2 Test purposes for ISDN

TSS HOLD	TP_503_201	Reference subclause 5.2.1.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4
Test purpose *HOLD requested by the calling party in an reINVITE request (a=sendonly)* Ensure that the calling party is able to set the communication on HOLD. The received reINVITE contains an a attribute in the SDP set to 'sendonly'. The SUT sends a 200 OK (INVITE) containing an a attribute in the SDP set to 'recvonly'. A DSS1 NOTIFY message is sent to the terminating user equipment and the Notification indicator information element is set to 'Remote hold'.			
SIP header values INVITE: SDP a=sendonly 200 OK (INVITE): SDP a=recvonly			
DSS1 Parameter values NOTIFY: Notification indicator = Remote hold			
Message flow			

Test equipment			**End device**
INVITE	→	→	SETUP
183 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→		
INVITE (sendonly)	→	→	NOTIFY (Remote hold)
200 OK (INVITE)	←		
ACK	→		
Apply post test routine			

TSS HOLD	TP_503_202	Reference subclause 5.2.1.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4
Test purpose *HOLD requested by the calling party in an UPDATE request (a=sendonly)* Ensure that the calling party is able to set the communication on HOLD. The received UPDATE contains an a attribute in the SDP set to 'sendonly'. The SUT sends a 200 OK (UPDATE) containing an a attribute in the SDP set to 'recvonly'. A DSS1 NOTIFY message is sent to the terminating user equipment and the Notification indicator information element is set to 'Remote hold'.			
SIP header values UPDATE: SDP a=sendonly 200 OK (UPDATE): SDP a=recvonly			
DSS1 Parameter values NOTIFY: Notification indicator = Remote hold			

Message flow			
Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→		
UPDATE (sendonly)	→	→	NOTIFY (Remote hold)
200 OK (UPDATE)	←		
Apply post test routine			

TSS HOLD	TP_503_203	Reference subclause 5.2.1.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4
Test purpose *HOLD requested by the calling party in an reINVITE request (a=inactive)* Ensure that the calling party is able to set the communication on HOLD. The received reINVITE contains an a attribute in the SDP set to 'inactive'. The SUT sends a 200 OK (INVITE) containing an a attribute in the SDP set to 'inactive'. A DSS1 NOTIFY message is sent to the terminating user equipment and the Notification indicator information element is set to 'Remote hold'. The communication was previously set on HOLD by the called party.			
SIP header values INVITE: SDP a=inactive 200 OK (INVITE): SDP a=inactive			
DSS1 Parameter values NOTIFY: Notification indicator = Remote hold			

Message flow			
Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→		
		←	NOTIFY (Remote hold)
CASE A			
INVITE (sendonly)	←		
200 OK INVITE	→		
ACK	←		
CASE B			
UPDATE (sendonly)	←		
200 OK (UPDATE)	→		
INVITE (inactive)	→	→	NOTIFY (Remote hold)
200 OK INVITE	←		
ACK	→		
Apply post test routine			

TSS HOLD	TP_503_204	Reference subclause 5.2.1.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4

Test purpose

HOLD requested by the calling party in an UPDATE request (a=inactive)

Ensure that the calling party is able to set the communication on HOLD. The received UPDATE containes an a attribute in the SDP set to 'inactive'. The SUT sends a 200 OK (UPDATE) containing an a attribute in the SDP set to 'inactive'. A DSS1 NOTIFY message is sent to the terminating user equipment and the Notification indicator information element is set to 'Remote hold'. The communication was previously set on HOLD by the called party.

SIP header values

UPDATE: SDP a=inactive

200 OK (UPDATE): SDP a=inactive

DSS1 Parameter values

NOTIFY: Notification indicator = Remote hold

Message flow

Test equipment			End device
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→		
		←	NOTIFY (Remote hold)
CASE A			
INVITE (sendonly)	←		
200 OK INVITE	→		
ACK	←		
CASE B			
UPDATE (sendonly)	←		
200 OK (UPDATE)	→		
UPDATE (inactive)	→	→	NOTIFY (Remote hold)
200 OK (UPDATE)	←		

Apply post test routine

TSS HOLD	TP_503_205	Reference subclause 5.2.1.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4

Test purpose

RETRIEVE requested by the calling party in an INVITE request

Ensure that the calling party is able to retrieve an earlier communication previously set on HOLD. The received INVITE contains an attribute in the SDP set to 'sendrecv'. The SUT sends a 200 OK (UPDATE) containing an a attribute in the SDP set to 'sendrecv'. A DSS1 NOTIFY message is sent to the terminating user equipment and the Notification indicator information element is set to 'Remote retrieval'.

SIP header values

INVITE: SDP a=sendrecv

200 OK (INVITE): SDP a=sendrecv

DSS1 Parameter values NOTIFY: Notification indicator = Remote retrieval
Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→		
INVITE (sendonly)	→	→	NOTIFY (Remote hold)
200 OK INVITE	←		
ACK	→		
INVITE (sendrecv)	→	→	NOTIFY (Remote retrieval)
200 OK INVITE	←		
ACK	→		
Apply post test routine			

TSS HOLD	TP_503_206	Reference subclause 5.2.1.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4
Test purpose *RETRIEVE requested by the calling party in an UPDATE request* Ensure that the calling party is able to retrieve an earlier communication previously set on HOLD. The received UPDATE contains an attribute in the SDP set to 'sendrecv'. The SUT sends a 200 OK (INVITE) containing an a attribute in the SDP set to 'sendrecv'. A DSS1 NOTIFY message is sent to the terminating user equipment and the Notification indicator information element is set to 'Remote retrieval'.			
SIP header values UPDATE: SDP a=sendrecv 200 OK (UPDATE): SDP a=sendrecv			
DSS1 Parameter values NOTIFY: Notification indicator = Remote retrieval			
Message flow			

Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→		
UPDATE (sendonly)	→	→	NOTIFY (Remote hold)
200 OK (UPDATE)	←		
UPDATE (sendrecv)	→	→	NOTIFY (Remote retrieval)
200 OK (UPDATE)	←		
Apply post test routine			

TSS HOLD	TP_503_207	Reference subclause 5.2.1.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4

Test purpose

HOLD requested by the called party a reINVITE or UPDATE is sent

Ensure that the called party is able to set the communication on HOLD. A HOLD message is received from the called party. The SUT sends a reINVITE request or an UPDATE request and the a attribute in the SDP is set to 'sendonly'. A 200 OK (INVITE) or a 200 OK (UPDATE) is received and the a attribute in the SDP is set to 'recvonly'.

SIP header values

INVITE/UPDATE: SDP a=sendonly

200 OK (INVITE/UPDATE): SDP a=recvonly

DSS1 Parameter values

HOLD

Message flow

Test equipment			End device
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→		
		←	HOLD
CASE A			
INVITE (sendonly)	←		
200 OK INVITE	→		
ACK	←		
CASE B			
UPDATE (sendonly)	←		
200 OK (UPDATE)	→		

Apply post test routine

TSS HOLD	TP_503_208	Reference subclause 5.2.1.1.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4 AND 5.1.3/1

Test purpose

HOLD requested by the called party in a private network

Ensure that the called party is able to set the communication on HOLD. A HOLD message is received from the called party. The SUT sends an reINVITE request or an UPDATE request and the a attribute in the SDP is set to 'sendonly'. A 200 OK (INVITE) or a 200 OK (UPDATE) is received and the a attribute in the SDP is set to 'recvonly'.

SIP header values

INVITE/UPDATE: SDP a=sendonly

200 OK (INVITE/UPDATE): SDP a=recvonly

DSS1 Parameter values

HOLD

Message flow			
Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→		
		←	NOTIFY (Remote hold)
CASE A			
INVITE (sendonly)	←		
200 OK INVITE	→		
ACK	←		
CASE B			
UPDATE (sendonly)	←		
200 OK (UPDATE)	→		
Apply post test routine			

TSS HOLD	TP_503_209	Reference subclause 5.2.1.1.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4 AND 5.1.3/1

Test purpose

HOLD requested by the called party in a private network

Ensure that the called party in a private network is able to set a communication on HOLD. A NOTIFY message is received and the Notification indicator information element is set to 'Remote hold'. A reINVITE request or an UPDATE request is sent and the a attribute in the SDP is set to 'inactive', A 200 OK (INVITE) or 200 OK (UPDATE) is received and the a attribute in the SDP is set to 'inactive')if the session was previously set on HOLD by the calling user.

SIP header values

INVITE/UPDATE: SDP a= sendrecv

200 OK (INVITE/UPDATE): SDP a= sendrecv

DSS1 Parameter values

NOTIFY: Notification indicator = Remote retrieval

Message flow			
Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK INVITE	←	←	CONNECT
ACK	→		
CASE A		←	NOTIFY (Remote hold)
INVITE (sendonly)	←		
200 OK INVITE	→		
ACK	←		

CASE B		
UPDATE (sendonly)	🡐	
200 OK (UPDATE)	🡒	
CASE A	🡐	NOTIFY (remote retrieval)
INVITE (sendrecv)	🡐	
200 OK INVITE	🡒	
ACK	🡐	
CASE B		
UPDATE (sendrecv)	🡐	
200 OK (UPDATE)	🡒	
Apply post test routine		

TSS HOLD	TP_503_210	Reference subclause 5.2.1.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4

Test purpose

HOLD requested by the called party. INVITE was received

Ensure that the SUT is able to send a DSS1 NOTIFY message and the Notification indicator information element is set to 'Remote hold' if an INVITE request was received and the a attribute in the SDP is set to 'sendonly'. A 200 OK (INVITE) is send and the a attribute in the SDP is set to 'recvonly'.

SIP header values

INVITE: SDP a=sendonly

200 OK (INVITE) SDP a=recvonly

DSS1 Parameter values

NOTIFY: Remote hold

Message flow

Test equipment			**Test equipment**
SETUP	🡒	🡒	INVITE
		🡐	407 Proxy Authentication Required
		🡒	ACK
		🡒	INVITE
ALERTING	🡐	🡐	180 Ringing
CONNECT	🡐	🡐	200 OK INVITE
		🡒	ACK
NOTIFY (Remote hold)	🡐	🡐	INVITE(sendonly)
		🡒	200 OK INVITE
		🡐	ACK

Apply post test routine

TSS HOLD	TP_503_210	Reference subclause 5.2.1.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4

Test purpose

HOLD requested by the called party. UPDATE was received

Ensure that the SUT is able to send a DSS1 NOTIFY message and the Notification indicator information element is set to 'Remote hold' if an UPDATE request was received and the a attribute in the SDP is set to 'sendonly'. A 200 OK (UPDATE) is send and the a attribute in the SDP is set to 'recvonly'.

SIP header values

INVITE: SDP a=sendonly

200 OK (INVITE) SDP a=recvonly

DSS1 Parameter values

NOTIFY: Remote hold

Message flow

Test equipment			**Test equipment**
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
		→	ACK
NOTIFY (Remote hold)	←	←	UPDATE (sendonly)
		→	200 OK (UPDATE)

Apply post test routine

TSS HOLD	TP_503_211	Reference subclause 5.2.1.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4

Test purpose

HOLD requested by the called party. INVITE was received

Ensure that the SUT is able to send a DSS1 NOTIFY message and the Notification indicator information element is set to 'Remote hold' if an INVITE request was received and the a attribute in the SDP is set to 'inactive'. A 200 OK (INVITE) is send and the a attribute in the SDP is set to 'inactive' if the session was previously set on HOLD by the called user.

SIP header values

INVITE: SDP a=inactive

200 OK (INVITE) SDP a=inactive

DSS1 Parameter values

NOTIFY: Remote hold

Message flow			
Test equipment			**Test equipment**
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
		→	ACK
HOLD	→		
CASE A			
		→	INVITE(sendonly)
		←	200 OK INVITE
		→	ACK
CASE B			
		→	UPDATE (sendonly)
		←	200 OK (UPDATE)
NOTIFY (Remote hold)	←	←	INVITE(inactive)
		→	200 OK INVITE
		←	ACK
Apply post test routine			

TSS HOLD	TP_503_212	Reference subclause 5.2.1.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4
Test purpose *HOLD requested by the called party. UPDATE was received* Ensure that the SUT is able to send a DSS1 NOTIFY message and the Notification indicator information element is set to 'Remote hold' if an UPDATE request was received and the a attribute in the SDP is set to 'inactive'. A 200 OK (UPDATE) is send and the a attribute in the SDP is set to 'inactive' if the session was previously set on HOLD by the called user.			
SIP header values UPDATE: SDP a=inactive 200 OK (UPDATE) SDP a=inactive			
DSS1 Parameter values NOTIFY: Remote hold			

Message flow			
Test equipment			**Test equipment**
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
		→	ACK
HOLD	→		
CASE A			
		→	INVITE(sendonly)
		←	200 OK INVITE
		→	ACK
CASE B			
		→	UPDATE (sendonly)
		←	200 OK (UPDATE)
NOTIFY (Remote hold)	←	←	UPDATE (inactive)
		→	200 OK (UPDATE)
Apply post test routine			

TSS HOLD	TP_503_213	Reference subclause 5.2.1.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4
Test purpose *Retrieve requested by the called party. INVITE was received* Ensure that the SUT is able to send a DSS1 NOTIFY message and the Notification indicator information element is set to 'Remote retrieval' if an INVITE request was received and the a attribute in the SDP is set to 'sendrecv'. A 200 OK (INVITE) is send and the a attribute in the SDP is set to 'sendrecv.			
SIP header values INVITE: SDP a=sendrecv 200 OK (INVITE) SDP a=sendrecv			
DSS1 Parameter values NOTIFY: Remote retrieval			

Message flow

Test equipment			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
		→	ACK
NOTIFY(Remote hold)	←	←	INVITE(sendonly)
		→	200 OK INVITE
		←	ACK
NOTIFY (Remote retrieval)	←	←	INVITE(sendonly)
		→	200 OK INVITE
		←	ACK

Apply post test routine

TSS HOLD	TP_503_214	Reference subclause 5.2.1.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4
Test purpose *Retrieve requested by the called party. UPDATE was received* Ensure that the SUT is able to send a DSS1 NOTIFY message and the Notification indicator information element is set to 'Remote retrieval' if an UPDATE request was received and the a attribute in the SDP is set to 'sendrecv'. A 200 OK (UPDATE) is send and the a attribute in the SDP is set to 'sendrecv.			
SIP header values UPDATE: SDP a=sendrecv 200 OK (UPDATE) SDP a=sendrecv			
DSS1 Parameter values NOTIFY: Remote retrieval			

Message flow

Test equipment			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
		→	ACK
NOTIFY(Remote hold)	←	←	UPDATE (sendonly)
		→	200 OK (UPDATE)
NOTIFY (Remote retrieval)	←	←	UPDATE (sendrecv)
		→	200 OK (UPDATE)

Apply post test routine

TSS HOLD	TP_503_215	Reference subclause 5.2.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4
Test purpose *HOLD requested by the calling party* Ensure that the SUT is able to send a reINVITE request or UPDATE request and the a attribute in the SDP is set to 'sendonly' if a DSS1 HOLD message was received.			
SIP header values INVITE/UPDATE: SDP a=sendonly 200 OK (INVITE/UPDATE) SDP a=recvonly			
DSS1 Parameter values			
Message flow			

Test equipment			**Test equipment**
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
		→	ACK
HOLD	→		
CASE A		→	INVITE(sendonly)
		←	200 OK INVITE
		→	ACK
CASE B			
		→	UPDATE (sendonly)
		←	200 OK (UPDATE)
Apply post test routine			

TSS HOLD	TP_503_216	Reference subclause 5.2.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4
Test purpose *Retrieve requested by the calling party* Ensure that the SUT is able to send a reINVITE request or UPDATE request and the a attribute in the SDP is set to 'sendrecv' if a DSS1 RETRIVE message was received.			
SIP header values INVITE/UPDATE: SDP a=sendrecv 200 OK (INVITE/UPDATE) SDP a=sendrecv			
DSS1 Parameter values			

Message flow			
Test equipment			**Test equipment**
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
		→	ACK
HOLD	→		
CASE A		→	INVITE(sendonly)
		←	200 OK INVITE
		→	ACK
CASE B			
		→	UPDATE (sendonly)
		←	200 OK (UPDATE)
RETRIVE	→		
CASE A		→	INVITE(sendrecv)
		←	200 OK INVITE
		→	ACK
CASE B			
		→	UPDATE (sendrecv)
		←	200 OK (UPDATE)
Apply post test routine			

TSS HOLD	TP_503_217	Reference subclause 5.2.1.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4 AND 5.1.3/1
Test purpose *HOLD requested by the calling party* Ensure that the SUT is able to send a reINVITE request or UPDATE request and the a attribute in the SDP is set to 'sendonly' if a DSS1 NOTIFY message was received from an user in a private network and the Notification description is set to 'Remote Hold'.			
SIP header values INVITE/UPDATE: SDP a=sendonly 200 OK (INVITE/UPDATE) SDP a=recvonly			
DSS1 Parameter values NOTIFY: Remote Hold			

Message flow		
Test equipment		**Test equipment**
SETUP	→	→ INVITE
		← 407 Proxy Authentication Required
		→ ACK
		→ INVITE
ALERTING	←	← 180 Ringing
CONNECT	←	← 200 OK INVITE
		→ ACK
NOTIFY (Remote Hold)	→	
CASE A		→ INVITE(sendonly)
		← 200 OK INVITE
		→ ACK
CASE B		
		→ UPDATE (sendonly)
		← 200 OK (UPDATE)
Apply post test routine		

TSS HOLD	TP_503_218	Reference subclause 5.2.1.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4 AND 5.1.3/1
Test purpose *HOLD requested by the calling party* Ensure that the SUT is able to send a reINVITE request or UPDATE request and the a attribute in the SDP is set to 'inactive' if a DSS1 NOTIFY message was received from an user in a private network and the Notification description is set to 'Remote Hold'.			
SIP header values INVITE/UPDATE: SDP a=sendonly 200 OK (INVITE/UPDATE) SDP a=recvonly			
DSS1 Parameter values NOTIFY: Remote Hold			

Message flow		
Test equipment		**Test equipment**
SETUP	→	→ INVITE
		← 407 Proxy Authentication Required
		→ ACK
		→ INVITE
ALERTING	←	← 180 Ringing
CONNECT	←	← 200 OK INVITE
		→ ACK
NOTIFY (Remote Hold)	→	

CASE A	→	INVITE(sendonly)
	←	200 OK INVITE
	→	ACK
CASE B		
	→	UPDATE (sendonly)
	←	200 OK (UPDATE)
Apply post test routine		

TSS HOLD	TP_503_219	Reference subclause 5.2.1.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/4 AND 5.1.3/1

Test purpose

Retrieve requested by the calling party.

Ensure that the SUT is able to retrive the held session if a DSS1 NOTIFY message is received and the Notification description is set to 'Remote retrieval'. A SIP reINVITE or UPDATE request is sent to the called user and the a attribute of the SDP is set to 'sendrecv'.

SIP header values

INVITE/UPDATE: SDP a= sendrecv

200 OK (INVITE/UPDATE) SDP a= sendrecv

DSS1 Parameter values

NOTIFY: Remote retrieval

Message flow

Test equipment				Test equipment
SETUP	→		→	INVITE
			←	407 Proxy Authentication Required
			→	ACK
			→	INVITE
ALERTING	←		←	180 Ringing
CONNECT	←		←	200 OK INVITE
			→	ACK
NOTIFY (Remote Hold)	→			
CASE A			→	INVITE(sendonly)
			←	200 OK INVITE
			→	ACK
CASE B				
			→	UPDATE (sendonly)
			←	200 OK (UPDATE)
NOTIFY (remote retrieval)	→			
CASE A			→	INVITE(sendrecv)
			←	200 OK INVITE
			→	ACK

<table>
<tr><td colspan="3">CASE B</td></tr>
<tr><td></td><td>→</td><td>UPDATE (sendrecv)</td></tr>
<tr><td></td><td>←</td><td>200 OK (UPDATE)</td></tr>
<tr><td colspan="3" align="center">Apply post test routine</td></tr>
</table>

7.2.5.4 Communication Diversion (CDIV)

7.2.5.4.1 Test purposes for POTS

<table>
<tr><td>TSS
CDIV</td><td>TP_504_101</td><td>Reference
clause C.7 and D.2 of [ETSI TS 183 043]</td><td>Selection expression
PICS 5.1.1/1 AND 5.3/7</td></tr>
<tr><td colspan="4">Test purpose
First call line identity

Ensure that Call forwarding information received in an INVITE request ("history-info" header) is used by the SUT to generate the appropriate Call Setup message to be delivered to the called terminal, using the ITU-T H.248 andisp package.</td></tr>
<tr><td colspan="4">SIP header values
INVITE:
History-Info: [<first call forwarding URI];user=phone>;index=1, [<last call forwarding URI];user=phone>;index=1.1</td></tr>
<tr><td colspan="4">Message flow</td></tr>
<tr><td colspan="2" align="center">Test equipment</td><td colspan="2" align="center">End device</td></tr>
<tr><td>INVITE</td><td>→</td><td colspan="2"></td></tr>
<tr><td>180 Ringing</td><td>←</td><td colspan="2">Ringing, display call forwarding information</td></tr>
<tr><td colspan="4" align="center">Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
CDIV</td><td>TP_504_102</td><td>Reference
clause C.7 of [ETSI TS183 043]</td><td>Selection expression
PICS 5.1.1/1</td></tr>
<tr><td colspan="4">Test purpose
Activate, deactivate or interrogate the call forwarding service

Ensure that the SUT is able to send an INVITE request to activate, deactivate or interrogate the call forwarding service.</td></tr>
<tr><td colspan="4">SIP header values
INVITE
Request-Line: PX SC (SR SI) SX@pes-scc.operator.com</td></tr>
<tr><td colspan="4">Message flow</td></tr>
<tr><td colspan="2" align="center">End device</td><td colspan="2" align="center">Test equipment</td></tr>
<tr><td>Off hook</td><td></td><td colspan="2"></td></tr>
<tr><td>Dial service code command</td><td>→</td><td colspan="2">INVITE</td></tr>
<tr><td></td><td>←</td><td colspan="2">407 Proxy Authentication Required</td></tr>
<tr><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td></td><td>→</td><td colspan="2">INVITE</td></tr>
<tr><td colspan="4" align="center">Apply post test routine</td></tr>
</table>

7.2.5.4.2 Test purposes for ISDN

TSS CDIV	TP_504_201	Reference subclause 5.2.5.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5
Test purpose *181 received, a NOTIFY is sent* Ensure that on receipt of a 181 Call Being Forwarded provisional response, a NOTIFY message is sent to the DSS1 User equipment. The Notification indicator is set to 'Call is diverting'.			
SIP header values			
DSS1 Parameter values NOTIFY: Notification indicator Call is diverting			
Message flow			
Test equipment			**Test equipment**
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
NOTIFY	←	←	181 (Call Being Forwarded)
Apply post test routine			

TSS CDIV	TP_504_202	Reference subclause 5.2.5.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5
Test purpose *181 received, a PROGRESS is sent* Ensure that on receipt of a 181 Call Being Forwarded provisional response where a 183 (Session Progress) containing a P-Early-Media header authorizing early media was previously received, a PROGRESS message is sent to the DSS1 User equipment. The Notification indicator is set to 'Call is diverting'.			
SIP header values			
DSS1 Parameter values PROGRESS: Notification indicator Call is diverting			
Message flow			
Test equipment			**Test equipment**
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
CALL PROCEEDING	←	←	183 (Session Progress)
PROGRESS	←	←	181 (Call Being Forwarded)
Apply post test routine			

TSS CDIV	TP_504_203	Reference subclause 5.2.5.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5

Test purpose

181 received, a PROGRESS is sent

Ensure that on receipt of a 181 Call Being Forwarded provisional response where a 180 (Ringing) was previously received, a PROGRESS message is sent to the DSS1 User equipment. The Notification indicator is set to 'Call is diverting'.

SIP header values

DSS1 Parameter values

PROGRESS: Notification indicator
Call is diverting

Message flow

Test equipment			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 (Ringing)
PROGRESS	←	←	181 (Call Being Forwarded)

Apply post test routine

TSS CDIV	TP_504_204	Reference subclause 5.2.5.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5

Test purpose

Subsequent diversion cause 487 and 408, a notification is sent

Ensure that on receipt of a subsequent 181 provisional response indicating a subsequent diversion, only the cause parameter value 487 (Deflection during alerting) or 408 (No reply) in the History-Info header, the notification 'Call is diverting' is sent to the DSS1 User equipment.

SIP header values

181: History-Info: <appropriate value>; index=1,
<appropriate value; cause=487>; index=1.1
or
<appropriate value; cause=408>; index=1.1

DSS1 Parameter values

PROGRESS: Notification indicator
Call is diverting

Message flow

Test equipment			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 (Ringing)
PROGRESS	←	←	181 (Call Being Forwarded)

Apply post test routine

<table>
<tr><td>TSS
CDIV</td><td>TP_504_205</td><td>Reference
subclause 5.2.5.1 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.4/5</td></tr>
<tr><td colspan="4">Test purpose
Subsequent diversion cause not equal to 487 and 408, no notification is sent

Ensure that on receipt of a subsequent 181 provisional response indicating a subsequent diversion no notification 'Call is diverting' is sent to the DSS1 User equipment if the cause parameter value other than 487 (Deflection during alerting) or 408 (No reply) in the History-Info header as indicated in Table 7.2.5.4.2-1.</td></tr>
<tr><td colspan="4">SIP header values
181 2: History-Info:
<appropriate value>; index=1, <appropriate value; cause=Reason>; index=1.1</td></tr>
<tr><td colspan="4">DSS1 Parameter values
NOTIFY: Notification indicator
Call is diverting</td></tr>
<tr><td colspan="4">Message flow</td></tr>
<tr><td>Test equipment</td><td></td><td></td><td>Test equipment</td></tr>
<tr><td>SETUP</td><td>→</td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td>←</td><td>407 Proxy Authentication Required</td></tr>
<tr><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>ALERTING</td><td>←</td><td>←</td><td>180 (Ringing)</td></tr>
<tr><td>NOTIFY</td><td>←</td><td>←</td><td>181 (Call Being Forwarded) 1</td></tr>
<tr><td></td><td></td><td>←</td><td>181 (Call Being Forwarded) 2</td></tr>
<tr><td colspan="4">Apply post test routine</td></tr>
</table>

Table 7.2.5.4.2-1 – Cause values not sent to the DSS1 UE

Reason	Value
Reason_01	486
Reason_02	302
Reason_03	480
Reason_04	404
Reason_05	503

TSS CDIV	TP_504_206	Reference subclause 5.2.5.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5

Test purpose

181 received. Mapping of latest History-Info entry into the Redirection number if no Privacy value is present in the entry

Ensure that on receipt of a 181 (Call Being Forwarded) provisional response where a History-Info header is present and no Privacy header is present in the latest hist-entry, a NOTIFY is sent to the DSS1 User equipment. The NOTIFY contains a Redirection number Information Element coded as described below:

Type of number

National number if Userinfo of latest entry is in the local number format or if in global number format and the country code of the URI is equal to the country where the SUT is located.

International number if Userinfo of latest entry is in the global number format and the country code of the URI is not equal to the country where the SUT is located

Numbering plan identification = ISDN (telephony) numbering plan

Presentation indicator = **Presentation allowed**

Number digits

User portion received in the URI of the latest entry; if the country code of the URI: In case where global number and the country code is the same as the AGCF/VGW or line is located, the country code is removed from the number.

SIP header values

181: History-Info:

<URI>; index=1, <diverted-to URI; cause=any appropriate value>; index=1.1

DSS1 Parameter values

NOTIFY: Redirection number

Presentation indicator = Presentation allowed

Message flow

Test equipment			**Test equipment**
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
NOTIFY	←	←	181 (Call Being Forwarded)

Apply post test routine

TSS CDIV	TP_504_207	Reference subclause 5.2.5.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5

Test purpose

180 received. Mapping of latest History-Info entry into the Redirection number if no Privacy value is present in the entry

Ensure that on receipt of a 180 (Ringing) provisional response where a History-Info header is present and no Privacy header is present in the latest hist-entry, an ALERTING is sent to the DSS1 User equipment. The ALERTING contains a Redirection number Information Element coded as described below:

Type of number

National number if Userinfo of latest entry is in the local number format or if in global number format and the country code of the URI is equal to the country where the SUT is located.

International number if Userinfo of latest entry is in the global number format and the country code of the URI is not equal to the country where the SUT is located

Numbering plan identification = ISDN (telephony) numbering plan

Presentation indicator = **Presentation allowed**

Number digits

User portion received in the URI of the latest entry; if the country code of the URI: In case for global number and the country code is the same as the AGCF/VGW or line is located, the country code is removed from the number.

SIP header values

180: History-Info:

<URI>; index=1, <diverted-to URI; cause=any appropriate value>; index=1.1

DSS1 Parameter values

ALERTING: Redirection number

Presentation indicator = Presentation allowed

Message flow

Test equipment			**Test equipment**
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 (Ringing)

Apply post test routine

TSS CDIV	TP_504_208	Reference subclause 5.2.5.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5

Test purpose

200 OK received. Mapping of latest History-Info entry into the Redirection number if no Privacy value is present in the entry

Ensure that on receipt of a 200 OK (INVITE) provisional response where a History-Info header is present and no Privacy header is present in the latest hist-entry, a CONNECT is sent to the DSS1 User equipment. The CONNECT contains a Redirection number Information Element coded as described below:

Type of number

National number if Userinfo of latest entry is in the local number format or if in global number format an the country code of the URI is equal to the country where the SUT is located.

International number if Userinfo of latest entry is in the global number format and the country code of the URI is not equal to the country where the SUT is located

Numbering plan identification = ISDN (telephony) numbering plan

Presentation indicator = **Presentation allowed**

Number digits

User portion received in the URI of the latest entry; if the country code of the URI: In case for global number and the country code is the same as the AGCF/VGW or line is located, the country code is removed from the number.

SIP header values

200: History-Info:

<URI>; index=1, <diverted-to URI; cause=any appropriate value>; index=1.1

DSS1 Parameter values

CONNECT: Redirection number

Presentation indicator = Presentation allowed

Message flow

Test equipment			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 (Ringing)
CONNECT	←	←	200 OK INVITE
		→	ACK

Apply post test routine

TSS CDIV	TP_504_209	Reference subclause 5.2.5.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1

Test purpose

181 received. Mapping of latest History-Info entry into the Redirection number if a Privacy value history is present in the entry

Ensure that on receipt of a 181 (Call Being Forwarded) provisional response where a History-Info header is present and a Privacy header is present in the latest hist-entry value set to 'history', a NOTIFY is sent to the DSS1 User equipment. The NOTIFY contains a Redirection number Information Element coded as described below:

Type of number = Unknown

Numbering plan identification = Unknown

Presentation indicator = Presentation restricted

Number digits not present

SIP header values

181: History-Info:

<URI>; index=1, <diverted-to URI; cause=any appropriate value>?Privacy=history; index=1.1

DSS1 Parameter values

NOTIFY: Redirection number

Presentation indicator = Presentation restricted

Message flow

Test equipment			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
NOTIFY	←	←	181 (Call Being Forwarded)

Apply post test routine

TSS CDIV	TP_504_210	Reference subclause 5.2.5.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1

Test purpose

180 received. Mapping of latest History-Info entry into the Redirection number if a Privacy value history is present in the entry

Ensure that on receipt of a 180 (Ringing) provisional response where a History-Info header is present and a Privacy header is present in the latest hist-entry value set to 'history', an ALERTING is sent to the DSS1 User equipment. The ALERTING contains a Redirection number Information Element coded as described below:

Type of number = Unknown

Numbering plan identification = Unknown

Presentation indicator = Presentation restricted

Number digits not present

SIP header values

180: History-Info:

<URI>; index=1, <diverted-to URI; cause=any appropriate value>?Privacy=history; index=1.1

DSS1 Parameter values

ALERTING: Redirection number

Presentation indicator = Presentation restricted

Message flow
Test equipment → → **Test equipment**
SETUP → → INVITE
← 407 Proxy Authentication Required
→ ACK
→ INVITE
ALERTING ← ← 180 (Ringing)
Apply post test routine

TSS CDIV	TP_504_211	Reference subclause 5.2.5.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1
Test purpose *200 received. Mapping of latest History-Info entry into the Redirection number if a Privacy value history is present in the entry* Ensure that on receipt of a 200 OK (INVITE) provisional response where a History-Info header is present and a Privacy header is present in the latest hist-entry value set to 'history', a CONNECT is sent to the DSS1 User equipment. The CONNECT contains a Redirection number Information Element coded as described below: Type of number = Unknown Numbering plan identification = Unknown Presentation indicator = **Presentation restricted** Number digits not present			
SIP header values 200: History-Info: <URI>; index=1, <diverted-to URI; cause=any appropriate value>?Privacy=history; index=1.1			
DSS1 Parameter values CONNECT: Redirection number Presentation indicator = Presentation restricted			
Message flow **Test equipment** **Test equipment** SETUP → → INVITE ← 407 Proxy Authentication Required → ACK → INVITE ALERTING ← ← 180 (Ringing) CONNECT ← ← 200 OK INVITE → ACK **Apply post test routine**			

TSS CDIV	TP_504_212	Reference subclause 5.2.5.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2

Test purpose

181 received. Sending of Redirection number 'not available' if no History-Info header is present.

A 180 Ringing was already received. Ensure that on receipt of a 181 (Call Being Forwarded) provisional response and a History-Info header is not present, a NOTIFY is sent to the DSS1 User equipment. The PROGRESS contains a Redirection number Information Element coded as described below:

Type of number = Unknown

Numbering plan identification = Unknown

Presentation indicator = **Number not available due to interworking**

Number digits not present

SIP header values

DSS1 Parameter values

PROGRESS: Redirection number
Presentation indicator = Presentation restricted

Message flow

Test equipment			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 (Ringing)
PROGRESS	←	←	181 (Call Being Forwarded)

Apply post test routine

TSS CDIV	TP_504_213	Reference subclause 5.2.5.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1

Test purpose

Mapping of the DivertingLegInformation2 invoke component into the History-Info header first diversion

Ensure that on receipt of a SETUP message and a Facility Information Element DivertingLegInformation2 invoke component is present, the diversionCounter is set to '1', an INVITE request is sent the History-Info header contains two history entries.

- The first history entry is derived from the 'divertingNr' Parameter - index=1
- The second entry is derived from Request URI - index=1.1
- The cause parameter in the second entry is derived from the 'diversionReason' parameter Reason_VA as described in Table 7.2.5.4.2-2

SIP header values

INVITE: Request URI
History-Info:
< divertingNr >; index=1,
<Request URI>; cause= Reason_VA>; index=1.1

DSS1 Parameter values SETUP: Facility diversionCounter diversionReason divertingNr originalCalledNr Called party number
Message flow **Test equipment** **Test equipment** SETUP ➔ ➔ INVITE 🡐 407 Proxy Authentication Required ➔ ACK ➔ INVITE **Apply post test routine**

TSS CDIV	TP_504_214	Reference subclause 5.2.5.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1
Test purpose *Mapping of the DivertingLegInformation2 invoke component into the History-Info header second diversion* Ensure that on receipt of a SETUP message and a Facility Information Element DivertingLegInformation2 invoke component is present, the diversionCounter is set to '2', an INVITE request is sent the History-Info header contains two history entries. • The first history entry is derived from the 'originalCalledNr' Parameter - index=1 • The second entry is derived from 'divertingNr' parameter - index=1.1 • The cause parameter in the second entry is set to '404' • The third history entry is derived from Request URI - index=1.1.1 • The cause parameter in the third entry is derived from the 'diversionReason' parameter Reason_VA as described in Table 7.2.5.4.2-2			
SIP header values INVITE: Request URI History-Info: < divertingNr >; index=1, < divertingNr; cause=404>; index=1.1, < Request URI>; cause= Reason_VA>; index=1.1.1			
DSS1 Parameter values SETUP: Facility diversionCounter diversionReason divertingNr originalCalledNr Called party number			
Message flow **Test equipment** **Test equipment** SETUP ➔ ➔ INVITE 🡐 407 Proxy Authentication Required ➔ ACK ➔ INVITE **Apply post test routine**			

Table 7.2.5.4.2-2 – Mapping of diversionReason value into cause parameter value in the second history entry

Reason_VA	diversionReason	cause
Reason_VA_01	unknown	"404"
Reason_VA_02	cfu	"302"
Reason_VA_03	cfb	"486"
Reason_VA_04	cfnr	"408"
Reason_VA_05	cdAlerting	"487"
Reason_VA_06	cdImmediate	"480"

TSS CDIV	TP_504_215	Reference subclause 5.2.5.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1
Test purpose *181 received. Mapping of the value of the Privacy parameter in latest History Entry into the DivertingLegInformation3 invoke component* Ensure that on receipt of a 181 (Call Being Forwarded) provisional response, the privacy requirements in the latest history entry are mapped into Facility Information Element DivertingLegInformation3 presentationAllowedIndicator parameter in an FACILITY as described in Table 7.2.5.4.2-3.			
SIP header values 181: History-Info: < divertingNr >; index=1, <Request URI>; cause= privacy_VA>; index=1.1			
DSS1 Parameter values FACILITY: Facility DivertingLegInformation3 presentationAllowedIndicator = privacy_VA			
Message flow			
Test equipment			**Test equipment**
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
FACILITY	←	←	181 (Call Being Forwarded)
Apply post test routine			

TSS CDIV	TP_504_216	Reference subclause 5.2.5.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1
Test purpose *180 received. Mapping of the value of the Privacy parameter in latest History Entry into the DivertingLegInformation3 invoke component* Ensure that on receipt of a 180 (Ringing) provisional response the privacy requirements in the latest history entry are mapped into Facility Information Element DivertingLegInformation3 presentationAllowedIndicator parameter in an ALERTING as described in Table 7.2.5.4.2-3.			

SIP header values

180: History-Info:

< divertingNr >; index=1,

<Request URI>; cause= privacy_VA>; index=1.1

DSS1 Parameter values

ALERTING: Facility

DivertingLegInformation3

presentationAllowedIndicator = privacy_VA

Message flow

Test equipment			**Test equipment**
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	181 (Call Being Forwarded)

Apply post test routine

TSS CDIV	TP_504_217	Reference subclause 5.2.5.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1

Test purpose

200 received. Mapping of the value of the Privacy parameter in latest History Entry into the DivertingLegInformation3 invoke component

Ensure that on receipt of a 200 OK (INVITE) final response the privacy requirements in the latest history entry are mapped into Facility Information Element DivertingLegInformation3 presentationAllowedIndicator parameter in a CONNECT as described in Table 7.2.5.4.2-3.

SIP header values

200: History-Info:

< divertingNr >; index=1,

<Request URI>; cause= privacy_VA>; index=1.1

DSS1 Parameter values

CONNECT: Facility

DivertingLegInformation3

presentationAllowedIndicator = privacy_VA

Message flow

Test equipment			**Test equipment**
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 (Ringing)
CONNECT	←		200 OK (INVITE)
			ACK

Apply post test routine

Table 7.2.5.4.2-3 – Mapping of the value of the Privacy parameter in latest History Entry into the DivertingLegInformation3 invoke component

privacy_VA	Privacy header in SIP response	presentationAllowedIndicator
privacy_VA_01	No Privacy parameter	true
privacy_VA_02	history	false

TSS CDIV	TP_503_218	Reference subclause 5.2.5.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1
Test purpose *ALERTING received. Mapping of DivertingLegInformation1 into History-Info header No History-Info header is sent.* Ensure that on receipt of an ALERTING and a Facility Information Element DivertingLegInformation1 is present, the diversionReason as indicated in Table 5.3.6.2-1, the subscriptionOption is set to 'noNotification', a 180 (Ringing) provisional response is sent and no History-Info header is present.			
SIP header values			
DSS1 Parameter values ALERTING: Facility DivertingLegInformation1 diversionReason = reason_VA subscriptionOption = noNotification divertedToNumber = diverted to number (PIXIT)			
Message flow			
Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
Apply post test routine			

TSS CDIV	TP_503_219	Reference subclause 5.2.5.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1
Test purpose *FACILITY received. Mapping of DivertingLegInformation1 into History-Info header No History-Info header is sent.* Ensure that on receipt of an FACILITY and a Facility Information Element DivertingLegInformation1, the diversionReason as indicated in Table 7.2.5.4.2-2 is present, the subscriptionOption is set to 'noNotification', no SIP response is sent.			
SIP header values			
DSS1 Parameter values ALERTING: Facility DivertingLegInformation1 diversionReason = reason_VA subscriptionOption = noNotification divertedToNumber = diverted to number (PIXIT)			

Message flow **Test equipment** / **End device** INVITE → → SETUP 180 Ringing ← ← FACILITY **Apply post test routine**

TSS CDIV	TP_503_220	Reference subclause 5.2.5.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1
Test purpose *PROGRESS received. Mapping of DivertingLegInformation1 into History-Info header No History-Info header is sent.* Ensure that on receipt of an PROGRESS and a Facility Information Element DivertingLegInformation1, the diversionReason as indicated in Table 7.2.5.4.2-2 is present, the subscriptionOption is set to 'noNotification', a 183 (Session Progress) provisional response is sent and no History-Info header is present.			
SIP header values			
DSS1 Parameter values PROGRESS: Facility DivertingLegInformation1 diversionReason = reason_VA subscriptionOption = noNotification divertedToNumber = diverted to number (PIXIT)			
Message flow **Test equipment** / **End device** INVITE → → SETUP ← FACILITY **Apply post test routine**			

TSS CDIV	TP_503_221	Reference subclause 5.2.5.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1
Test purpose *ALERTING received. Mapping of DivertingLegInformation1 into History-Info header a restricted History-Info header is sent.* Ensure that on receipt of an ALERTING and a Facility Information Element DivertingLegInformation1, the diversionReason as indicated in Table 7.2.5.4.2-2 is present, the subscriptionOption is set to 'notificationWithoutDivertedToNr', a 180 (Ringing) provisional response is sent and a History-Info header is present a Privacy header is escaped in the last entry coded as follows: First entry: URI not significant; index=1 Second entry URI derived from the 'divertedToNumber'; Privacy= history; cause=reason_VA; index=1.1.			
SIP header values 180: History-Info: <URI non significant value; index=1, < divertedToNumber?Privacy=history; cause=reason_VA; index=1.1			

DSS1 Parameter values ALERTING: Facility DivertingLegInformation1 diversionReason = reason_VA subscriptionOption = notificationWithoutDivertedToNr divertedToNumber = diverted to number (PIXIT)
Message flow **Test equipment** **End device** INVITE → → SETUP 180 Ringing ← ← ALERTING **Apply post test routine**

TSS CDIV	TP_503_222	Reference subclause 5.2.5.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1
Test purpose *PROGRESS received. Mapping of DivertingLegInformation1 into History-Info header a restricted History-Info header is sent.* Ensure that on receipt of an PROGRESS and a Facility Information Element DivertingLegInformation1, the diversionReason as indicated in Table 7.2.5.4.2-2 is present, the subscriptionOption is set to 'notificationWithoutDivertedToNr', a 183 (Session Progress) provisional response is sent and a History-Info header is present a Privacy header is escaped in the last entry coded as follows: First entry: URI not significant; index=1 Second entry URI derived from the 'divertedToNumber'; Privacy= history; cause=reason_VA; index=1.1.			
SIP header values 183: History-Info: <URI non significant value; index=1, < divertedToNumber?Privacy=history; cause=reason_VA; index=1.1			
DSS1 Parameter values PROGRESS: Facility DivertingLegInformation1 diversionReason = reason_VA subscriptionOption = notificationWithoutDivertedToNr divertedToNumber = diverted to number (PIXIT)			
Message flow **Test equipment** **End device** INVITE → → SETUP 183 (Session Progress) ← ← PROGRESS **Apply post test routine**			

TSS CDIV	TP_503_223	Reference subclause 5.2.5.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1

Test purpose

ALERTING received. Mapping of DivertingLegInformation1 into History-Info header a restricted History-Info header is sent.

Ensure that on receipt of an ALERTING and a Facility Information Element DivertingLegInformation1, the diversionReason as indicated in Table 7.2.5.4.2-2 is present, the subscriptionOption is set to 'notificationWithDivertedToNr', a 180 (Ringing) provisional response is sent and a History-Info header is present no Privacy header is present in the last entry coded as follows:

First entry: URI not significant; index=1

Second entry URI derived from the 'divertedToNumber'; cause=reason_VA; index=1.1.

SIP header values

180: History-Info:

<URI non significant value; index=1,

< divertedToNumber?Privacy=history; cause=reason_VA; index=1.1

DSS1 Parameter values

ALERTING: Facility

DivertingLegInformation1

diversionReason = reason_VA

subscriptionOption = notificationWithDivertedToNr

divertedToNumber = diverted to number (PIXIT)

Message flow

Test equipment			**End device**
INVITE	➔	➔	SETUP
180 (Ringing)	←	←	ALERTING

Apply post test routine

TSS CDIV	TP_503_224	Reference subclause 5.2.5.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1

Test purpose

FACILITY received. Mapping of DivertingLegInformation1 into History-Info header a restricted History-Info header is sent.

Ensure that on receipt of an FACILITY and a Facility Information Element DivertingLegInformation1, the diversionReason as indicated in Table 7.2.5.4.2-2 is present, the subscriptionOption is set to 'notificationWithDivertedToNr', a 181 (Being forwarded) provisional response is sent and a History-Info header is present no Privacy header is present in the last entry coded as follows:

First entry: URI not significant; index=1

Second entry URI derived from the 'divertedToNumber'; cause=reason_VA; index=1.1.

SIP header values

181: History-Info:

<URI non significant value; index=1,

< divertedToNumber?Privacy=history; cause=reason_VA; index=1.1

DSS1 Parameter values

FACILITY: Facility

DivertingLegInformation1

diversionReason = reason_VA

subscriptionOption = notificationWithDivertedToNr

divertedToNumber = diverted to number (PIXIT)

Message flow			
Test equipment			**End device**
INVITE	→	→	SETUP
181 (Being forwarded)	←	←	FACILITY
Apply post test routine			

TSS CDIV	TP_503_225	Reference subclause 5.2.5.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1
Test purpose *PROGRESS received. Mapping of DivertingLegInformation1 into History-Info header a restricted History-Info header is sent.* Ensure that on receipt of an PROGRESS and a Facility Information Element DivertingLegInformation1, the diversionReason as indicated in Table 7.2.5.4.2-2 is present, the subscriptionOption is set to 'notificationWithDivertedToNr', a 183 (Session Progress) provisional response is sent and a History-Info header is present no Privacy header is present in the last entry coded as follows: First entry: URI not significant; index=1 Second entry URI derived from the 'divertedToNumber'; cause=reason_VA; index=1.1.			
SIP header values 183: History-Info: <URI non significant value; index=1, < divertedToNumber?Privacy=history; cause=reason_VA; index=1.1			
DSS1 Parameter values PROGRESS: Facility DivertingLegInformation1 diversionReason = reason_VA subscriptionOption = notificationWithDivertedToNr divertedToNumber = diverted to number (PIXIT)			
Message flow **Test equipment** — **End device** INVITE → → SETUP 183 (Session Progress) ← ← PROGRESS **Apply post test routine**			

TSS CDIV	TP_503_226	Reference subclause 5.2.5.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1
Test purpose *DivertingLegInformation3 received in an ALERTING and History-Info header is send in the 180* Ensure that on receipt of a Facility Information Element containing a DivertingLegInformation3 invoke component and the presentationAllowedIndicator is set to 'true', a 180 (Ringing) is sent and a History-Info header is present. The latest entry in the History-Info header does not contain a Privacy header field.			

SIP header values

INVITE: History-Info:

<any URI>index=1

<URI set to the value of the Request URI; cause=any>; index=1.1

180: History-Info:

<any URI>index=1

<URI set to the value of the Request URI; cause=any>; index=1.1

DSS1 Parameter values

ALERTING: Facility

DivertingLegInformation3

presentationAllowedIndicator=true

Message flow

Test equipment			End device
INVITE	→	→	SETUP
180 (Ringing)	←	←	ALERTING

Apply post test routine

TSS CDIV	TP_503_227	Reference subclause 5.2.5.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1

Test purpose

DivertingLegInformation3 received in an ALERTING and History-Info header is send in the 180

Ensure that on receipt of a Facility Information Element containing a DivertingLegInformation3 invoke component and the presentationAllowedIndicator is set to privacy_VA in an ALERTING message, a 180 (Ringing) is sent and a History-Info header is present. A Privacy header field in the latest entry in the History-Info header is processed as described in Table 7.2.5.4.2-4.

SIP header values

INVITE: History-Info:

<any URI>index=1

<URI set to the value of the Request URI; cause=any>; index=1.1

180: History-Info:

<any URI>index=1

<URI set to the value of the Request URI[privacy_VA]; cause=any>; index=1.1

DSS1 Parameter values

ALERTING: Facility

DivertingLegInformation3

presentationAllowedIndicator= privacy_VA

Message flow

Test equipment			End device
INVITE	→	→	SETUP
180 (Ringing)	←	←	ALERTING

Apply post test routine

TSS CDIV	TP_503_228	Reference subclause 5.2.5.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1
Test purpose *DivertingLegInformation3 received in an FACILITY and History-Info header is send in the 181* Ensure that on receipt of a Facility Information Element containing a DivertingLegInformation3 invoke component and the presentationAllowedIndicator is set to privacy_VA in a FACILITY message, a 181 (Being forwarded) is sent and a History-Info header is present. A Privacy header field in the latest entry in the History-Info header is processed as described in Table 7.2.5.4.2-4.			
SIP header values INVITE: History-Info: <any URI>index=1 <URI set to the value of the Request URI; cause=any>; index=1.1 181: History-Info: <any URI>index=1 <URI set to the value of the Request URI[privacy_VA]; cause=any>; index=1.1			
DSS1 Parameter values FACILITY: Facility DivertingLegInformation3 presentationAllowedIndicator= privacy_VA			
Message flow			
Test equipment			**End device**
INVITE	→	→	SETUP
181 (Being forwarded)	←	←	FACILITY
Apply post test routine			

TSS CDIV	TP_503_229	Reference subclause 5.2.5.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1
Test purpose *DivertingLegInformation3 received in an CONNECT and History-Info header is send in the 180* Ensure that on receipt of a Facility Information Element containing a DivertingLegInformation3 invoke component and the presentationAllowedIndicator is set to privacy_VA in a CONNECT message, a 180 (Ringing) is sent and a History-Info header is present. A Privacy header field in the latest entry in the History-Info header is processed as described in Table 7.2.5.4.2-4.			
SIP header values INVITE: History-Info: <any URI>index=1 <URI set to the value of the Request URI; cause=any>; index=1.1 200 OK: History-Info: <any URI>index=1 <URI set to the value of the Request URI[privacy_VA]; cause=any>; index=1.1			
DSS1 Parameter values CONNECT: Facility DivertingLegInformation3 presentationAllowedIndicator= privacy_VA			

Message flow			
Test equipment			**End device**
INVITE	→	→	SETUP
180 (Ringing)	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→		
Apply post test routine			

Table 7.2.5.4.2-4 – Mapping of the value of the Privacy parameter in latest History Entry into the DivertingLegInformation3 invoke component

privacy_VA	presentationAllowedIndicator	Privacy header in SIP response
privacy_VA_01	true	No Privacy parameter
privacy_VA_02	false	Privacy=history

TSS CDIV	TP_503_230	Reference subclause 5.2.5.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5

Test purpose

INVITE received History-Info header contains two entries not restricted, Redirecting number sent in SETUP

Ensure that on receipt of an INVITE request and a History-Info header is present containing two entries and the first entry is not escaped with an Privacy header, a SETUP is sent and the Redirecting number address signals is derived from the URI of the first entry of the History-Info header and the Reason of diversion is derived from the cause parameter of the last entry coded as described below and in Table 7.2.5.4.2-5:

Redirecting number

Type of number

National if the URI is coded as follows sip: local-number@hostportion

international if the URI is coded as follows sip: global –number@hostportion and CC is not the same as the country where the user or line is located

Numbering plan identification: ISDN numbering plan

Presentation indicator: presentation allowed

Reason of diversion: reason_VA

Number digits:

Userinfo as received in the first URI; global-number-digits: if the country code of the URI is the same as the country where the user or line is located, the country code is removed from the Userinfo

SIP header values

INVITE: History-Info:

<any URI 1 (PIXIT)>index=1

<URI set to the value of the Request URI; cause=reason_VA >; index=1.1

DSS1 Parameter values

SETUP: Redirecting number

Presentation indicator: presentation allowed

Reason of diversion: reason_VA

Message flow			
Test equipment			**End device**
INVITE	→	→	SETUP
Apply post test routine			

TSS CDIV	TP_503_231	Reference subclause 5.2.5.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5

Test purpose

IINVITE received History-Info header contains two entries restricted, Redirecting number sent in SETUP

Ensure that on receipt of an INVITE request and a History-Info header is present containing two entries and the first entry is escaped with an Privacy header value 'history', a SETUP is sent and the Redirecting number address signals are absent and coded as described below and in Table 7.2.5.4.2-5:

Redirecting number

Type of number; unknown

Numbering plan identification: unknown

Presentation indicator: **presentation restricted**

Reason of diversion: **reason_VA**

Number digits: not present

SIP header values

INVITE: History-Info:

<any URI**?Privacy=history** >index=1

<URI set to the value of the Request URI; cause=reason_VA >; index=1.1

DSS1 Parameter values

SETUP: Redirecting number

Presentation indicator: **presentation restricted**

Reason of diversion: **reason_VA**

Message flow

Test equipment			**End device**
INVITE	➔	➔	SETUP

Apply post test routine

TSS CDIV	TP_503_232	Reference subclause 5.2.5.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5

Test purpose

INVITE received History-Info header contains one entry, Redirecting number sent in SETUP

Ensure that on receipt of an INVITE request and a History-Info header is present containing only the entry index=1.1, a SETUP is sent and the Redirecting number address signals are absent and coded as described below and in Table 7.2.5.4.2-5:

Redirecting number

Type of number; unknown

Numbering plan identification: unknown

Presentation indicator: **number not available due to interworking**

Reason of diversion: **reason_VA**

Number digits: not present

SIP header values

INVITE: History-Info:

<URI set to the value of the Request URI; cause=**reason_VA** >; index=1.1

DSS1 Parameter values

SETUP: Redirecting number

Presentation indicator: **number not available due to interworking**

Reason of diversion: **reason_VA**

Message flow			
	Test equipment		**End device**
INVITE	→	→	SETUP
	Apply post test routine		

TSS CDIV	TP_503_233	Reference subclause 5.2.5.2.2/ [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5

Test purpose

INVITE received History-Info header contains more than two entries not restricted, Redirecting number sent in SETUP

Ensure that on receipt of an INVITE request and a History-Info header is present containing more than two entries and the second last entry is not escaped with an Privacy header, a SETUP is sent and the first Redirecting number address signals are derived from the URI of the second last entry of the History-Info header and the Reason of diversion is derived from the cause parameter of the last entry, the second Redirecting number address signals are derived from the URI of the first entry of the History-Info header and the Reason of diversion is derived from the cause parameter of the second entry coded as described below and in Table 7.2.5.4.2-5:

First Redirecting number

Type of number

National if the URI is coded as follows sip: local-number@hostportion

international if the URI is coded as follows sip: global –number@hostportion and CC is not the same as the country where the user or line is located

Numbering plan identification: ISDN numbering plan

Presentation indicator: **presentation allowed**

Reason of diversion: **reason_VA**

Number digits:

Userinfo as received in the second last URI; global-number-digits: if the country code of the URI is the same as the country where the user or line is located, the country code is removed from the Userinfo

Second Redirecting number

Type of number

National if the URI is coded as follows sip: local-number@hostportion

international if the URI is coded as follows sip: global –number@hostportion and CC is not the same as the country where the user or line is located

Numbering plan identification: ISDN numbering plan

Presentation indicator: presentation allowed

Number digits:

Userinfo as received in the first URI; global-number-digits: if the country code of the URI is the same as the country where the user or line is located, the country code is removed from the Userinfo

SIP header values

INVITE: History-Info:

<any URI 1 (PIXIT)>index=1

<any URI 2 (PIXIT); cause=any appropriate value>; index=1.1

<URI set to the value of the Request URI; cause=**reason_VA** >; index=1.1.1

DSS1 Parameter values

SETUP: **Redirecting number**

Presentation indicator: presentation allowed

Reason of diversion: reason_VA

Redirecting number

Presentation indicator: presentation allowed

Message flow			
	Test equipment		**End device**
INVITE	➔	➔	SETUP
	Apply post test routine		

TSS CDIV	TP_503_234	Reference subclause 5.2.5.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5
Test purpose *INVITE received History-Info header contains more than two entries restricted, Redirecting number sent in SETUP* Ensure that on receipt of an INVITE request and a History-Info header is present containing two entries and the second last entry is escaped with an Privacy header value 'history', a SETUP is sent and the first Redirecting number address signals are not present and the Reason of diversion is derived from the cause parameter of the last entry coded as described below and in Table 7.2.5.4.2-5: **1st Redirecting number** Type of number; unknown Numbering plan identification: unknown Presentation indicator: **presentation restricted** Reason of diversion: reason_VA Number digits: not present **2nd Redirecting number**			
SIP header values INVITE: History-Info: <any URI 1 (PIXIT)>index=1 <any URI 2 (PIXIT)?Privacy=history; cause=any appropriate value>;index=1.1 <URI set to the value of the Request URI; cause=reason_VA>; index=1.1.1			
DSS1 Parameter values SETUP: **Redirecting number** Presentation indicator: presentation restricted Reason of diversion: reason_VA Redirecting number			
Message flow **Test equipment** — **End device** INVITE ➔ ➔ SETUP **Apply post test routine**			

Table 7.2.5.4.2-5 – Mapping of cause parameter in the history-entry into reason of diversion

reason_VA	cause parameter	Reason of diversion
reason_VA_01	"404"	Unknown
reason_VA_02	"302 "	Call forwarding unconditional
reason_VA_03	"486"	Call forwarding busy
reason_VA_04	"408"	Call forwarding no reply
reason_VA_05	"487"	Deflection
reason_VA_06	"480"	Deflection
reason_VA_07	"503"	Unknown

TSS CDIV	TP_503_235	Reference subclause 5.2.5.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1

Test purpose

INVITE received a History-Info header present with more than two entries, a SETUP is sent at the T Reference point

Ensure that on receipt of an INVITE request and a History-Info header is present, a SETUP message is sent. The SETUP contains a Facility Information Element with a DivertingLegInformation2 invoke component. The diversionCounter is derived from the number of dots in the index parameter of the last entry of the History-Info header. The diversionReason is derived from the cause parameter of the last entry as described in Table 7.2.5.4.2-6, the divertingNr is derived from the URI of second last entry and the originalCalledNr is derived from the URI of the first entry in the History-Info header.

SIP header values

INVITE: History-Info:

<any URI 1 (PIXIT)>index=1

<any URI 2 (PIXIT); cause=any appropriate value>; index=1.1

<URI set to the value of the Request URI; cause=**reason_VA** >; index=1.1.1

DSS1 Parameter values

SETUP: Facility

DivertingLegInformation2

diversionCounter: 1

diversionReason: **reason_VA**

divertingNr: derived from the first entry

Message flow

Test equipment			**End device**
INVITE	➔	➔	SETUP

Apply post test routine

Table 7.2.5.4.2-6 – Mapping of cause parameter in the history-entry into reason of diversion in the DivertingLegInformation2 component

reason_VA	cause	diversionReason
reason_VA_01	"404"	unknown
reason_VA_02	"302"	cfu
reason_VA_03	"486"	cfb
reason_VA_04	"408"	cfnr
reason_VA_05	"487"	cdAlerting
reason_VA_06	"480"	cdImmediate

TSS CDIV	TP_503_236	Reference subclause 5.2.5.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1

Test purpose

No further diversion occurs a 180 is sent

When an INVITE is received and a History-Info header is present,ensure that on receipt of an ALERTING message a 180 Ringing provisional response is sent. The 180 Ringing contains a History-Info header and the value is the same as received in the initial INVITE, a Privacy header is escaped in the last entry and the value is set to 'history'.

SIP header values

INVITE: History-Info:

<any URI (PIXIT)>index=1

<URI set to the value of the Request URI; cause=any appropriate value>;
index=1.1

180: History-Info:

<any URI (PIXIT)>index=1

<URI set to the value of the Request URI of the INVITE?**Privacy=history**;
cause=any appropriate value>; index=1.1

DSS1 Parameter values

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING

Apply post test routine

TSS CDIV	TP_503_237	Reference subclause 5.2.5.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1

Test purpose

No further diversion occurs a 183 is sent

When an INVITE is received and a History-Info header is present, ensure that on receipt of a PROGRESS message a 183 Session Progress provisional response is sent. The 183 Session Progress contains a History-Info header and the value is the same as received in the initial INVITE, a Privacy header is escaped in the last entry and the value is set to 'history'.

SIP header values

INVITE: History-Info:

<any URI (PIXIT)>index=1

<URI set to the value of the Request URI; cause=any appropriate value>;
index=1.1

180: History-Info:

<any URI (PIXIT)>index=1

<URI set to the value of the Request URI of the INVITE?**Privacy=history**;
cause=any appropriate value>; index=1.1

DSS1 Parameter values

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
183 Session Progress	←	←	PROGRESS

Apply post test routine

TSS CDIV	TP_503_238	Reference subclause 5.2.5.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1

Test purpose

No further diversion occurs a 200 OK is sent

When an INVITE is received and a History-Info header is present,ensure that on receipt of a CONNECT message a 200 OK INVITE final response is sent. The 200 OK INVITE contains a History-Info header and the value is the same as received in the initial INVITE, a Privacy header is escaped in the last entry and the value is set to 'history'.

SIP header values

INVITE: History-Info:
<any URI (PIXIT)>index=1
<URI set to the value of the Request URI; cause=any appropriate value>;
index=1.1

200 OK: History-Info:
<any URI (PIXIT)>index=1
<URI set to the value of the Request URI of the INVITE?**Privacy=history**;
cause=any appropriate value>; index=1.1

DSS1 Parameter values

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→		

Apply post test routine

TSS CDIV	TP_503_239	Reference subclause 5.2.5.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1

Test purpose

A call diversion occurs in the private network a 180 is sent

When an INVITE is received and a History-Info header is present, ensure that on receipt of an ALERTING message containing a Facility Information Element with a DivertingLegInformation1 invoke component, a 180 (Ringing) provisional response is sent. The 180 (Ringing) contains a History-Info header the value is the same as received in the initial INVITE. A further entry is added and the URI of the added entry is derived from the divertedToNumber element, the cause parameter is derived from the diversionReason element as described in Table 7.2.5.4.2-7, the Privacy header in the entry is derived from the subscriptionOption element as described in Table 7.2.5.4.2-8. If the subscriptionOption is set to 'noNotification' no History-Info header is present in the 180 (Ringing).

SIP header values

INVITE: History-Info:
<any URI (PIXIT)>index=1
<URI set to the value of the Request URI; cause=any appropriate value>;
index=1.1

180: History-Info:
<any URI (PIXIT)>index=1
<URI set to the value of the Request URI in the INVITE;
cause=any appropriate value>; index=1.1
<URI derived from the divertedToNumber [**Privacy_VA**]; cause=**reason_VA**>;index=1.1.1

DSS1 Parameter values ALERTING: Facility DivertingLegInformation1 diversionReason subscriptionOption divertedToNumber
Message flow **Test equipment** **End device** INVITE → → SETUP 180 Ringing ← ← ALERTING **Apply post test routine**

TSS CDIV	TP_503_240	Reference subclause 5.2.5.2.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1
Test purpose *A call diversion occurs in the private network a 183 is sent* When an INVITE is received and a History-Info header is present,ensure that on receipt of a PROGRESS message containing a Facility Information Element with a DivertingLegInformation1 invoke component, a 183 (Session Progress) provisional response is sent. If the 183 (Session Progress) contains a History-Info header the value is the same as received in the initial INVITE. If a further entry is added and the URI of the added entry is derived from the divertedToNumber element, the cause parameter is derived from the diversionReason element as described in Table 7.2.5.4.2-7, the Privacy header in the entry is derived from the subscriptionOption element as described in Table 7.2.5.4.2-8. If the subscriptionOption is set to 'noNotification'no History-Info header is present in the 183 (Session Progress).			
SIP header values INVITE: History-Info: <any URI (PIXIT)>index=1 <URI set to the value of the Request URI; cause=any appropriate value>; index=1.1 183: History-Info: <any URI (PIXIT)>index=1 <URI set to the value of the Request URI in the INVITE; cause=any appropriate value>; index=1.1 <URI derived from the divertedToNumber [**Privacy_VA**]; cause=**reason_VA**>;index=1.1.1			
DSS1 Parameter values PROGRESS: Facility **DivertingLegInformation1** diversionReason subscriptionOption divertedToNumber			
Message flow **Test equipment** **End device** INVITE → → SETUP 180 Ringing ← ← ALERTING 183 Session Progress ← ← PROGRESS **Apply post test routine**			

<table>
<tr><td>TSS
CDIV</td><td>TP_503_241</td><td>Reference
subclause 5.2.5.2.3 of [ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1</td></tr>
<tr><td colspan="4">Test purpose
A call diversion occurs in the private network a 181 is sent

When an INVITE is received and a History-Info header is present,ensure that on receipt of a FACILITY message containing a Facility Information Element with a DivertingLegInformation1 invoke component, a 181 (Being Forwarded) provisional response is sent. The 181 (Being Forwarded) contains a History-Info header the value is the same as received in the initial INVITE. A further entry is added and the URI of the added entry is derived from the divertedToNumber element, the cause parameter is derived from the diversionReason element as described in Table 7.2.5.4.2-7, the Privacy header in the entry is derived from the subscriptionOption element as described in Table 7.2.5.4.2-8. If the subscriptionOption is set to 'noNotification' no provisional response is sent.</td></tr>
<tr><td colspan="4">SIP header values
INVITE: History-Info:
<any URI (PIXIT)>index=1
<URI set to the value of the Request URI; cause=any appropriate value>;
index=1.1

181: History-Info:
<any URI (PIXIT)>index=1
<URI set to the value of the Request URI in the INVITE;
cause=any appropriate value>; index=1.1
<URI derived from the divertedToNumber [Privacy_VA]; cause=reason_VA>;index=1.1.1</td></tr>
<tr><td colspan="4">DSS1 Parameter values
FACILITY: Facility
DivertingLegInformation1
diversionReason
subscriptionOption
divertedToNumber</td></tr>
<tr><td colspan="4">Message flow
Test equipment End device
INVITE → → SETUP
181 (Being Forwarded) ← ← FACILITY
Apply post test routine</td></tr>
</table>

Table 7.2.5.4.2-7 – Mapping of the DivertingLegInformation1 invoke component into added history entry

reason_VA	cause	diversionReason
reason_VA_01	"404"	unknown
reason_VA_02	"302"	cfu
reason_VA_03	"486"	cfb
reason_VA_04	"408"	cfnr
reason_VA_05	"487"	cdAlerting
reason_VA_06	"480"	cdImmediate

Table 7.2.5.4.2-8 – Mapping of subscriptionOption into the Privacy header in the latest history entry

Privacy_VA	subscriptionOptio	Privacy header
Privacy_VA_01	*noNotification*	No History-Info header present or in case of FACILITY not interworked
Privacy_VA_02	notificationWithoutDivertedToNr	Privacy=history"
Privacy_VA_03	notificationWithDivertedToNr	No Privacy parameter present

TSS CDIV	TP_503_242	Reference subclause 5.2.5.2.4 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1

Test purpose

A further call diversion beyond the private network

When an INVITE is received and a History-Info header is present, ensure that on receipt of an ALERTING message and a Facility Information Element is present containing a DivertingLegInformation3 invoke component with a presentationAllowedIndicator set to privacy_VA as described in Table 7.2.5.4.2-9, a 180 (Ringing) is sent. The 180 (Ringing) contains the History-Info header as received in the initial INVITE, according the privacy requirements described in Table 7.2.5.4.2-9 a Privacy header is escaped in the last entry and set to 'history' or not present.

SIP header values

INVITE: History-Info:

<any URI (PIXIT)>index=1

<URI set to the value of the Request URI; cause=any appropriate value>; index=1.1

180: History-Info:

<any URI (PIXIT)>index=1

<URI derived from the divertedToNumber [**Privacy_VA**];
cause= any appropriate value >; index=1.1

DSS1 Parameter values

ALERTING: Facility

DivertingLegInformation3

presentationAllowedIndicator: **privacy_VA**

Message flow

Test equipment			End device
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING

Apply post test routine

Table 7.2.5.4.2-9 – Mapping of the value of the DivertingLegInformation3 invoke component into the Privacy parameter in latest History Entry

privacy_VA	presentationAllowedIndicator	Privacy header in SIP response
privacy_VA_01	true	No Privacy parameter
privacy_VA_02	false	Privacy=history

TSS CDIV	TP_503_243	Reference subclause 5.2.5.2.4 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1

Test purpose

'CallRerouteing invoke' received, rerouteingCounter is set to 1 302 is sent

Ensure that on receipt of FACILITY message containing a Facility Information Element CallRerouteing invoke component, a 302 (Moved Temporarily) final response is sent. The URI of the Contact header is derived from the calledAddress element, the PSTN XML BearerCapability is derived from the q931InfoElemen Bearer Capability, the PSTN XML Low layer compatibility is derived from the q931InfoElemen High layer compatibility, the User-to-User header is derived from the q931InfoElemen User-user information.

SIP header values

302: Contact: <derived from the calledAddress>
  PSTN XML MIME body
    <?xml version="1.0" encoding="utf-8"?>
    PSTN
      BearerCapability derived from the q931InfoElemen Bearer Capability
      LowLayerCompatibility derived from the q931InfoElemen Low layer compatibility
      HighLayerCompatibility derived from the q931InfoElemen High layer compatibility
  User-to-User derived from the q931InfoElemen User-user information
  History-Info:
      <derived from lastRerouteingNr>index=1
      <derived from calledAddress; cause=**reason_VA**>; index=1.1

DSS1 Parameter values

FACILITY: Facility
    CallRerouteing invoke
      rerouteingReason: reason_VA
      calledAddress
      rerouteingCounter
      q931InfoElement
        Bearer Capability
        Low layer compatibility
        High layer compatibility
        User-user information
      lastRerouteingNr
      subscriptionOption: any appropriate value

Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
302 (Moved Temporarily)	←	←	FACILITY
ACK	→		

Apply post test routine

Table 7.2.5.4.2-10 – Mapping of the CallRerouteing invoke component into added history entry

reason_VA	rerouteingReason	cause
reason_VA_01	unknown	"404"
reason_VA_02	cfu	"302"
reason_VA_03	cfb	"486"
reason_VA_04	cfnr	"408"
reason_VA_05	cdAlerting	"487"
reason_VA_06	cdImmediate	"480"

7.2.5.5 Three Party Service (3PTY)

7.2.5.5.1 Test purposes for POTS

<table>
<tr><td>TSS
3PTY</td><td>TP_505_101</td><td>Reference
subclause C.14.1.1 of
[ETSI TS 183 043]</td><td>Selection expression
PICS 5.1.1/1 AND 5.3/6</td></tr>
<tr><td colspan="4">Test purpose
The user establishes a three party conversation

Ensure that when the initial party is on hold and On receipt of a flash-hook event, the SUT performs the following actions:
• Request the media gateway to play a dial tone and collect digits
• Send a re-INVITE request to place the current call on hold.
On receipt of the dialled digits, the AGCF opens a new dialogue by sending an INVITE request with the following elements:
• The dialled digits used as a Request-URI.
• An SDP Offer for a voice call.
On receipt of 180 (Ringing) without P Early Media header or with a P Early Media header set to a value different from "sendonly" or from "sendreceive", the AGCF performs the following actions:
• Request the media gateway to play a ringback tone
On receipt of 180 (Ringing) or 183 (Session Progress) with a P Early Media header set to "sendonly" or "sendreceive", the AGCF performs the following actions:
• Request the media gateway to modify the configuration of the ephemeral termination so as to ensure that the end user will perceive early media.
On receipt of an SDP Answer in a 200 (OK) or in one of the above provisional responses, the AGCF performs the following actions:
• Request the media gateway to modify the Remote Descriptor of the ephemeral termination associated with the physical termination representing the analogue line.</td></tr>
</table>

SIP header values INVITE1 (initial party) SDP a=sendonly 200 OK 1 SDP a=recvonly INVITE3 (initial party) SDP a=sendrecv 200 OK 3 SDP a=sendrecv
Message flow

Test equipment		**End device**
Terminating user is connected with the initial party		
		Flash hook
		Dial tone
		Dial switching order command
INVITE1 (initial party)	←	
200 OK INVITE1	→	
ACK	←	
INVITE2 (additional party)	←	
180 Ringing2/183 Session Progress2	→	
200 OK INVITE2	→	
ACK	←	
INVITE3 (initial party)	←	Three party communication
200 OK INVITE4	→	
ACK	←	
Apply post test routine		

TSS 3PTY	TP_505_102	Reference subclause C.14.2 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/6

Test purpose

The served user wishes to reject one of the parties

Ensure that when the three-party call is established with the waiting party, processing of the flash-hook event is similar to the call waiting service. If the switching order command indicates that the served user wishes to reject one of the parties, the SUT performs the following actions:

- Send a BYE request towards the held party
- Request the media gateway to:
  - Remove the corresponding ephemeral termination

SIP header values

INVITE1 (additional party)
SDP
a=sendonly

200 OK 1
SDP
a=recvonly

INVITE2 (initial party)
SDP
a=sendrecv

200 OK 2
SDP
a=sendrecv

Message flow

Test equipment		**End device**
	The initial party is set on hold	
	A three party conversation is established	
		Flash hook
INVITE1 (initial party)	←	Dial tone
200 OK INVITE1	→	
ACK	←	
		Dial switching order command
BYE (additional party)	←	
200 OK BYE	→	
INVITE2 (initial party)	←	Two party ommunication
200 OK INVITE2	→	
ACK	←	
	Apply post test routine	

TSS 3PTY	TP_505_103	Reference subclause C.14.2 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/6 AND 5.3/9 AND 5.3/11

Test purpose

The user establishes a three party conversation (Loose coupling, INVITE method)

Ensure that when the initial party is on hold and a confirmed session with the waiting party exists, a flash hook event applies followed by a switching order command. If the switching order command indicates that the user wishes to establish a three-party conference with the held parties.

- Request the media gateway to:
  - Add a termination to the current context based on SDP information associated with the held party
- Send a re INVITE request towards the first held party (i.e., the party that was already held when the flash hook event was detected). The re INVITE request is built as follows:
  - The Request URI is set to the held party's identity.
  - The SDP description is set to a=sendrecv. The address and port are set according to the contents of the local descriptor of the new termination.
- Send a re INVITE request towards the second held party (i.e., the party that has been held for the purpose of collecting the switch order command). The re INVITE request is built as follows:
  - The Request URI is set to the held party's identity.
  - The SDP description is set to a=sendrecv.

SIP header values

INVITE2 (additional party)
SDP
a=sendrecv

200 OK 2
SDP
a=sendrecv

INVITE3 (initial party)
SDP
a=sendrecv

200 OK 3
SDP
a=sendrecv

Message flow

Test equipment		**End device**
The initial party is set on hold		
Terminating user is connected with the active party (hold or call waiting)		
		Flash hook
INVITE1 (initial party, sendonly)	←	Dial tone
200 OK INVITE1	→	
ACK	←	
		Dial switching order command
INVITE2 (additional party)	←	
200 OK INVITE2	→	
ACK	←	
INVITE3 (initial party)	←	Three party communication
200 OK INVITE3	→	
ACK	←	
Apply post test routine		

TSS 3PTY	TP_505_104	Reference subclause C.14.2 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/6 AND 5.3/9 AND 5.3/11

Test purpose

The served user wishes to reject one of the parties(Loose coupling, INVITE method)

Ensure that when the three-party call is established with the waiting party, processing of the flash-hook event is similar to the call waiting service. If the switching order command indicates that the served user wishes to reject one of the parties, the SUT performs the following actions:

- Send a BYE request towards the held party
- Request the media gateway to:
  - Remove the corresponding ephemeral termination

SIP header values

INVITE1 (additional party)
SDP
a=sendonly

200 OK 1
SDP
a=recvonly

INVITE2 (initial party)
SDP
a=sendrecv

200 OK 2
SDP
a=sendrecv

Message flow

Test equipment		**End device**
The initial party is set on hold		
A three party conversation is established		
		Flash hook
INVITE1 (initial party)	←	Dial tone
200 OK INVITE1	→	
ACK	←	
		Dial switching order command
BYE (additional party)	←	
200 OK BYE	→	
INVITE2 (initial party)	←	Two party ommunication
200 OK INVITE2	→	
ACK	←	
Apply post test routine		

TSS 3PTY	TP_505_105	Reference subclause C.14.2A of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/6 AND 5.3/9 AND 5.3/12

Test purpose

The user establishes a three party conversation (Loose coupling, REFER method)

Ensure that when the initial party is on hold and a confirmed session with the waiting party exists, a flash hook event applies followed by a switching order command. If the switching order command indicates that the user wishes to establish a three-party conference with the held parties.

- Request the media gateway to:
  - Add a termination to the current context based on SDP information associated with the held party
- Send an initial INVITE with the To and the Request URI containing the Conference bridge URI provisioned in the SUT.
- Send a REFER request within the existing dialogue with user B with the Refer To header containing the Conference bridge contact URI and the dialogue associated with the B party. The ReferredBy header field is set to the served user's identity.
- Send a REFER request within the existing dialogue with user C with the Refer To header containing the Conference bridge contact URI and the dialogue associated with the C party. The ReferredBy header field is set to the served user's identity.

When the SIP end point receives 202 ACCEPTED response to each sent REFER request, it:

- performs conference establishment notification;
- considers only the call established with the conference bridge as active.

The SIP end point will receive a BYE message in both calls established.

SIP header values

INVITE1 (additional party)
SDP
a=sendonly

200 OK 2
SDP
a=recvonly

INVITE2 (Conference bridge URI)

Message flow

Test equipment		End device
The initial party is set on hold		
Terminating user is connected with the active party (hold or call waiting)		
		Flash hook
INVITE1 (initial party, sendonly)	←	Dial tone
200 OK INVITE1	→	
ACK	←	
		Dial switching order command
INVITE2 (Conference bridge)	←	
200 OK INVITE1	→	
ACK	←	
REFER1 (additional party)	←	
202 ACCEPTED	→	
REFER2 (initial party)	←	

202 ACCEPTED	→	
BYE (initial party)	→	
200 OK BYE	←	
BYE (additional party)	→	
200 OK BYE	←	Three party communication

Apply post test routine

TSS 3PTY	TP_505_106	Reference subclause C.14.2A of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/6 AND 5.3/9 AND 5.3/12

Test purpose

The served user wishes to reject one of the parties (Loose coupling, REFER method)

Ensure that when the three-party call is established with the waiting party, processing of the flash-hook event is similar to the call waiting service. If the switching order command indicates that the served user wishes to reject one of the parties, the SUT performs the following actions:

- Send a BYE request towards the held party
- Request the media gateway to:
  - Remove the corresponding ephemeral termination

SIP header values

INVITE1 (additional party)
SDP
a=sendonly

200 OK 1
SDP
a=recvonly

INVITE2 (initial party)
SDP
a=sendrecv

200 OK 2
SDP
a=sendrecv

Message flow

Test equipment		End device

The initial party is set on hold

A three party conversation is established

		Flash hook
INVITE1 (initial party)	←	Dial tone
200 OK INVITE1	→	
ACK	←	
		Dial switching order command
BYE (additional party)	←	
200 OK BYE	→	
INVITE2 (initial party)	←	Two party ommunication
200 OK INVITE2	→	
ACK	←	

Apply post test routine

TSS 3PTY	TP_505_107	Reference subclause C.14.2B of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/6 AND 5.3/9 AND 5.3/13

Test purpose

The user establishes a three party conversation (Loose coupling INVITE request with URI list)

Ensure that when the initial party is on hold and a confirmed session with the waiting party exists, a flash hook event applies followed by a switching order commandif the switching order command indicates that the user wishes to establish a three-party conference with the held parties.

SUT creates a conference and invites user B and user C to the conference by sending an INVITE to the Conference Factory URI including URI list in the INVITE request. SUT indicates the particular dialogues which can be re used for this conference in the uri list by ? mechanism.

SIP header values

INVITE1 (additional party)
SDP
a=sendonly

200 OK 2
SDP
a=recvonly

INVITE2 (Conference bridge URI)
<resource lists xmlns="urn:ietf:params:xml:ns:resource lists" xmlns:cp="urn:ietf:params:xml:ns:copyControl">
<list>
<entry uri="B?Call ID=1a&From=A%3Btag%3Da&To=B%3Btag%3Db" cp:copyControl="to"/>
</list>
</resource lists>

Message flow

Test equipment		End device
The initial party is set on hold		
Terminating user is connected with the active party (hold or call waiting)		
		Flash hook
INVITE1 (initial party, sendonly)	←	Dial tone
200 OK INVITE1	→	
ACK	←	
		Dial switching order command
INVITE2 (Conference bridge)	←	
200 OK INVITE1	→	
ACK	←	
		Three party communication
Apply post test routine		

<table>
<tr><td>TSS
3PTY</td><td>TP_505_108</td><td>Reference
subclause C.14.2B of
[ETSI TS 183 043]</td><td>Selection expression
PICS 5.1.1/1 AND
5.3/6 AND 5.3/9 AND
5.3/13</td></tr>
<tr><td colspan="4">Test purpose
The served user wishes to reject one of the parties (Loose coupling INVITE request with URI list)

Ensure that when the three-party call is established with the waiting party, processing of the flash-hook event is similar to the call waiting service. If the switching order command indicates that the served user wishes to reject one of the parties, the SUT performs the following actions:
• Send a BYE request towards the held party
• Request the media gateway to:
– Remove the corresponding ephemeral termination</td></tr>
<tr><td colspan="4">SIP header values
INVITE1 (additional party)
SDP
a=sendonly

200 OK 1
SDP
a=recvonly

INVITE2 (initial party)
SDP
a=sendrecv

200 OK 2
SDP
a=sendrecv</td></tr>
<tr><td colspan="4">Message flow</td></tr>
<tr><td>Test equipment</td><td></td><td colspan="2">End device</td></tr>
<tr><td colspan="4">The initial party is set on hold</td></tr>
<tr><td colspan="4">A three party conversation is established</td></tr>
<tr><td></td><td></td><td colspan="2">Flash hook</td></tr>
<tr><td>INVITE1 (initial party)</td><td>←</td><td colspan="2">Dial tone</td></tr>
<tr><td>200 OK INVITE1</td><td>→</td><td colspan="2"></td></tr>
<tr><td>ACK</td><td>←</td><td colspan="2"></td></tr>
<tr><td></td><td></td><td colspan="2">Dial switching order command</td></tr>
<tr><td>BYE (additional party)</td><td>←</td><td colspan="2"></td></tr>
<tr><td>200 OK BYE</td><td>→</td><td colspan="2"></td></tr>
<tr><td>INVITE2 (initial party)</td><td>←</td><td colspan="2">Two party ommunication</td></tr>
<tr><td>200 OK INVITE2</td><td>→</td><td colspan="2"></td></tr>
<tr><td>ACK</td><td>←</td><td colspan="2"></td></tr>
<tr><td colspan="4">Apply post test routine</td></tr>
</table>

TSS 3PTY	TP_505_109	Reference clause C.14.3 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/6 AND 5.3/9

Test purpose

The user establishes a three party conversation (Tight coupling)

Ensure that when the initial party is on hold and a confirmed session with the waiting party exists, a flash hook event applies. On receipt of a notification of Register RECALL from the SUT, the SUT opens a new dialogue (D3) and sends an INVITE (flash) to an originating AS. This INVITE includes the following:

- The Request URI is structured as follows:
  - A user part containing "flash".
  - A domain name that together with the user part provides sufficient information for the AS Network to forward the request to the appropriate AS, based on Initial Filter Criteria stored in the user profile, e.g."flash@pes.operator.com"
- A From header containing the public identity of the line on which the RECALL occurred.
- An SDP offer for a speech call.

The AGCF now awaits receipt of a 484 Address Incomplete from the originating AS, and when received the AGCF takes the following actions:

- Requests the A MGW to play Dial Tone and collect one digit.
- Sends an INVITE (D3) containing this single digit (as this is a Recall sequence with more than one active dialogue) and await receipt of 200 OK (Invite) or a failure response code. This INVITE is built in the same way as the previous INVITE except that the dialled digit replaces "flash".

The AGCF then awaits a re INVITE (D2) with the SDP of a Media Server in the AS Network (acting as a 3 party bridge) and when received it takes the following actions:

- Sends an instruction to the A MGW to change the address to which RTP packets are sent and from which they are received (e.g. it modifies the ITU-T H.248 Remote Descriptor).

Sends a 200 OK (Invite) to the AS, awaits receipt of a BYE to end dialogue D3, and when this is received it sends a 200 OK (Bye).

SIP header values

INVITE D2-1

Request-Line: flash@pes.operator.com

INVITE D2-2

Request-Line: flash@pes.operator.com

INVITE3 (additional party)

SDP

a=sendrecv

200 OK 3

SDP

a=sendrecv

INVITE4 (initial party)

SDP

a=sendrecv

200 OK 4

SDP

a=sendrecv

Message flow		
Test equipment		**End device**
The initial party is set on hold		
Terminating user is connected with the active party (hold or call waiting)		
		Flash hook
INVITE1 (initial party, sendonly)	←	
200 OK INVITE1	→	
ACK	←	
INVITE D2-1	←	
484 Address Incomplete	→	
ACK	←	
		Dial tone
INVITE D2-2	←	Dial switching order command
180 Ringing	→	
200 OK INVITE2	→	
ACK	←	
		Flash hook
INVITE1 (D2-2)	←	
200 OK INVITE1	→	
ACK	←	
INVITE (D3-1)	←	
484 Address Incomplete	→	
ACK	←	
INVITE3 (additional party)	←	
200 OK INVITE3	→	
ACK	←	
INVITE4 (initial party)	←	Three party communication
200 OK INVITE4	→	
ACK	←	
Apply post test routine		

TSS 3PTY	TP_505_110	Reference C.14.2B of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/6 AND 5.3/9 AND 5.3/13

Test purpose

The served user wishes to reject one of the parties (Loose coupling INVITE request with URI list)

Ensure that when the three-party call is established with the waiting party, processing of the flash-hook event is similar to the call waiting service. If the switching order command indicates that the served user wishes to reject one of the parties, the SUT performs the following actions:

- Send a BYE request towards the held party
- Request the media gateway to:
  - Remove the corresponding ephemeral termination

SIP header values

INVITE1 (additional party)
SDP
a=sendonly

200 OK 1
SDP
a=recvonly

INVITE2 (initial party)
SDP
a=sendrecv

200 OK 2
SDP
a=sendrecv

Message flow

Test equipment		**End device**
The initial party is set on hold		
A three party conversation is established		
		Flash hook
INVITE1 (initial party)	←	Dial tone
200 OK INVITE1	→	
ACK	←	
		Dial switching order command
BYE (additional party)	←	
200 OK BYE	→	
INVITE2 (initial party)	←	Two party ommunication
200 OK INVITE2	→	
ACK	←	
Apply post test routine		

7.2.5.5.2 Test purposes for ISDN

<table>
<tr><td>TSS
3PTY</td><td>TP_505_201</td><td>Reference
subclause 5.2.13 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.4/6
AND 5.6/7</td></tr>
<tr><td colspan="4">Test purpose
Conference creation by three-way session creation. REFER request to the Focus, Conference notification service is subscribed

The conference creator is participating in two SIP sessions (S1 and S2) which are put on hold and wants to join together two of these active sessions to a so-called three-way session. The conference notification service is subscribed. The conference creator shall perform the following steps:

• Create a conference at the conference factory by sending an INVITE request with the conference factory URI. Receive and store the conference URI in the 200 OK response.
• For each of the active sessions, that are requested to be joined to a three-way session, send two REFER requests with the Request URI indicating the previously received conference URI and the Refer-To header indicating the SIP URI or tel URL of the respective remote user.
• The conference creator releases the sessions 1 and 2 after the receipt of NOTIFY requests indicating that the remote users have successfully joined the three-way session.</td></tr>
<tr><td colspan="4">SIP header values
INVITE1
Request Line user inactive idle stae
SDP a=sendonly

INVITE2
Request Line = conference factory URI
Contact: …; isfocus

SUBSCRIBE
Request URI contained the conference URI
Event: conference"

NOTIFY 1 Event contains conference; Subscription-State contains active; expires=[any value]

REFER1
Request URI indicating the conference URI
Refer-to header contains URI of remote user 1

REFER2
Request URI indicating the conference URI
Refer-to header contains URI of remote user 2

NOTIFY 2 Event contains conference; Subscription-State contains active
message/sipfrag contains SIP/2.0 100 Trying
NOTIFY 3 Event contains conference; Subscription-State contains active
message/sipfrag contains SIP/2.0 200 OK
application/conference-info+xml contains (S1) connected, dialled-in
NOTIFY 4 Event contains conference; Subscription-State contains active
message/sipfrag contains SIP/2.0 100 Trying
NOTIFY 5 Event contains conference; Subscription-State contains active</td></tr>
</table>

DSS1 Parameter values FACILITY Begin3PTY-Inv Call reference of call A-B

Message flow

Test equipment			Test equipment
	Session 1 is in active hold state		
	Session 2 is in active idle state		
FACILITY	→	→	INVITE1
		←	200 OK
		→	ACK
		→	SUBSCRIBE
		←	200 OK
		←	NOTIFY 1
		→	200 OK NOTIFY
		→	INVITE2
		←	200 OK
		→	ACK
		→	REFER1
		←	200 OK REFER
		←	NOTIFY 2 (S1, 100)
		→	200 OK NOTIFY
		←	NOTIFY 3 (S1, 200)
		→	200 OK NOTIFY
DISCONNECT 1	←	←	BYE S1
RELEASE COMPLETE	→	→	200 OK (BYE)
		→	REFER2
		←	200 OK REFER
		←	NOTIFY 4 (S2, 100)
		→	200 OK NOTIFY
		←	NOTIFY 5 (S1, 200)
		→	200 OK NOTIFY
DISCONNECT 2	←	←	BYE S2
RELEASE COMPLETE	→	→	200 OK (BYE)
	Apply post test routine		

<table>
<tr><td>TSS
3PTY</td><td>TP_505_202</td><td>Reference
5.2.13 of [ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.4/6 AND 5.6/7</td></tr>
<tr><td colspan="4">Test purpose
Conference creation by three-way session creation. REFER request to the user, Conference notification service is subscribed.

The conference creator is participating in two SIP sessions (S1 and S2) which are put on hold and wants to join two of these active sessions to a so-called three-way session. The conference notification service is subscribed. The conference creator shall perform the following steps:

• Create a conference at the conference factory by sending an INVITE request with the conference factory URI. Receive and store the conference URI in the 200 OK response.
• For each of the active sessions, that are requested to be joined to a three-way session, send two REFER requests with the Request URI indicating SIP URI or tel URL of the respective remote user and the Refer-To header indicating the previously received conference URI.
• The conference creator releases the sessions 1 and 2 after the receipt of NOTIFY requests indicating that the remote users have successfully joined the three-way session.</td></tr>
<tr><td colspan="4">SIP header values
INVITE1
Request Line user inactive idle stae
SDP a=sendonly

INVITE2
Request Line = conference factory URI
Contact: …; isfocus

SUBSCRIBE
Request URI contained the conference URI
Event: conference"

NOTIFY 1 Event contains conference; Subscription-State contains active; expires=[any value]

REFER1
Request URI indicating the conference URI
Refer-to header contains URI of remote user 1

REFER2
Request URI indicating the conference URI
Refer-to header contains URI of remote user 2

NOTIFY 2 Event contains conference; Subscription-State contains active
message/sipfrag contains SIP/2.0 100 Trying
NOTIFY 3 Event contains conference; Subscription-State contains active
message/sipfrag contains SIP/2.0 200 OK
application/conference-info+xml contains (S1) connected, dialled-in
NOTIFY 4 Event contains conference; Subscription-State contains active
message/sipfrag contains SIP/2.0 100 Trying
NOTIFY 5 Event contains conference; Subscription-State contains active</td></tr>
<tr><td colspan="4">DSS1 Parameter values
FACILITY
Begin3PTY-Inv
Call reference of call A-B</td></tr>
</table>

Message flow			
Test equipment			**Test equipment**
	Session 1 is in active hold state		
	Session 2 is in active idle state		
FACILITY	→	→	INVITE1
		←	200 OK
		→	ACK
		→	SUBSCRIBE
		←	200 OK
		←	NOTIFY 1
		→	200 OK NOTIFY
		→	INVITE2
		←	200 OK
		→	ACK
		→	REFER1
		←	200 OK REFER
		←	NOTIFY 2 (S1, 100)
		→	200 OK NOTIFY
		←	NOTIFY 3 (S1, 200)
		→	200 OK NOTIFY
RELEASE COMPLETE1	→	→	BYE S1
		←	200 OK (BYE)
		→	REFER2
		←	200 OK REFER
		←	NOTIFY 4 (S2, 100)
		→	200 OK NOTIFY
		←	NOTIFY 5 (S1, 200)
		→	200 OK NOTIFY
RELEASE COMPLETE2	→	→	BYE S2
		←	200 OK (BYE)
	Apply post test routine		

TSS 3PTY	TP_505_203	Reference subclause 5.2.13 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND NOT 5.4/6 AND 5.6/7

Test purpose

Conference creation by three-way session creation. REFER request to the Focus, Conference notification service not subscribed

The conference creator is participating in two SIP sessions (S1 and S2) which are put on hold and wants to join two of these active sessions to a so-called three-way session. The conference notification service is not subscribed. The conference creator shall perform the following steps:

- Create a conference at the conference factory by sending an INVITE request with the conference factory URI. Receive and store the conference URI in the 200 OK response.
- For each of the active sessions, that are requested to be joined to a three-way session, send two REFER requests with the Request URI indicating the previously received conference URI and the Refer-To header indicating the SIP URI or tel URI of the respective remote user.

The remote instance disconnects the previously held sessions.

SIP header values

INVITE1
 SDP a=sendonly

INVITE2
 Request Line = conference factory URI
 Contact: …; isfocus

REFER1
 Request URI indicating the conference URI
 Refer-to header contains URI of remote user 1

REFER2
 Request URI indicating the conference URI
 Refer-to header contains URI of remote user 2

DSS1 Parameter values

FACILITY
 Begin3PTY-Inv
 Call reference of call A-B

Message flow

Test equipment			Test equipment
	Session 1 is in active hold state		
	Session 2 is in active idle state		
FACILITY	→	→	INVITE1
		←	200 OK
		→	ACK
		→	INVITE2
		←	200 OK
		→	ACK
		→	REFER1
		←	200 OK REFER

DISCONNECT 2	←	←	BYE S1
RELEASE COMPLETE	→	→	200 OK (BYE)
		→	REFER2
		←	200 OK REFER
DISCONNECT 2	←	←	BYE S2
RELEASE COMPLETE	→	→	200 OK (BYE)
Apply post test routine			

TSS 3PTY	TP_505_204	Reference subclause 5.2.13 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/6 AND 5.6/7

Test purpose

Conference creation by three-way session creation. REFER request to the Focus, Conference notification service not subscribed

The conference creator is participating in two SIP sessions (S1 and S2) which are put on hold and wants to join two of these active sessions to a so-called three-way session. The conference notification service is not subscribed. The conference creator shall perform the following steps:

- Create a conference at the conference factory by sending an INVITE request with the conference factory URI. Receive and store the conference URI in the 200 OK response.
- For each of the active sessions, that are requested to be joined to a three-way session, send two REFER requests with the Request URI indicating the previously received conference URI and the Refer-To header indicating the SIP URI or tel URI of the respective remote user.

The SUT disconnects the previously held sessions.

SIP header values

INVITE1
SDP a=sendonly

INVITE2
Request Line = conference factory URI
Contact: …; isfocus

REFER1
Request URI indicating the conference URI
Refer-to header contains URI of remote user 1

REFER2
Request URI indicating the conference URI
Refer-to header contains URI of remote user 2

DSS1 Parameter values

FACILITY
Begin3PTY-Inv
Call reference of call A-B

Message flow

Test equipment			Test equipment
	Session 1 is in active hold state Session 2 is in active idle state		
FACILITY	→	→	INVITE1
		←	200 OK
		→	ACK
		→	INVITE2
		←	200 OK
		→	ACK
		→	REFER1
		←	200 OK REFER
RELEASE COMPLETE1	→	→	BYE S1
		←	200 OK (BYE)
		→	REFER2
		←	200 OK REFER
RELEASE COMPLETE2	→	→	BYE S2
		←	200 OK (BYE)

Apply post test routine

TSS 3PTY	TP_505_205	Reference subclause 5.2.13 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.6/7AND 5.4/8

Test purpose

The user equipment has the capability join a conference.

Ensure that the user equipment on receipt of a REFER request that contains a Refer-To header indicating a conference URI including the “method" parameter set to INVITE and contains a Referred-By header, sends an INVITE request to the conference URI including the received Referred-By header.
The SUT disconnects the previously held session.

SIP header values

REFER: Refer-To=conference URI; method=INVITE
Referred-By=Remote user equipment URI
INVITE S2: Request URI indicating the received conference URI
Referred-By=Remote user equipment URI

DSS1 Parameter values

Message flow

Test equipment			End device
	A session is in active hold state		
REFER	→		
200 OK REFER	←		
INVITE S2	←		
200 OK	→		
ACK	←		
BYE S1	←	←	RELEASE COMPLETE
200 OK (BYE)	→		

Apply post test routine

TSS	TP_505_206	Reference subclause 5.2.13 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.6/7AND 5.4/8

Test purpose

The user equipment has the capability join a conference.

Ensure that the user equipment on receipt of a REFER request that contains a Refer-To header indicating a conference URI including the “method" parameter set to INVITE and contains a Referred-By header, sends an INVITE request to the conference URI including the received Referred-By header.

The remote instance disconnects the previously held session.

SIP header values

REFER: Refer-To=conference URI; method=INVITE
Referred-By=Remote User Equipment URI

INVITE S2: Request URI indicating the received conference URI
Referred-By=Remote User Equipment URI

DSS1 Parameter values

Message flow

Test equipment			End device
	A session is in active hold state		
REFER	→		
200 OK REFER	←		
INVITE S2	←		
200 OK	→		
ACK	←		
BYE S1	→	→	DISCONNECT
200 OK (BYE)	←	←	RELEASE COMPLETE
	Apply post test routine		

TSS 3PTY	TP_505_207	Reference subclause 5.2.13 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/6 AND 5.6/7 AND 5.3/13

Test purpose

The user equipment has the capability to invite a participant to the conference. Resource list is used

Ensure that the SUT is able to send a resource list to the conference AS to invite participant(s) to a conference. The remote instance disconnects the previously held sessions.

SIP header values INVITE1 SDP a=sendonly INVITE2 Request Line = conference factory URI Contact: …; isfocus Content-Type: application/resource-lists+xml Content-Disposition: recipient-list <?xml version="1.0" encoding="UTF-8"?> <resource-lists xmlns="urn:ietf:params:xml:ns:resource-lists" xmlns:cp="urn:ietf:params:xml:ns:copyControl"> <list> <entry uri="S1 URI" cp:copyControl="to"/> </list> </resource-lists>
DSS1 Parameter values FACILITY Begin3PTY-Inv Call reference of call A-B
Message flow

Test equipment				**Test equipment**
	Session 1 is in active hold state			
	Session 2 is in active idle state			
FACILITY	→		→	INVITE1
			←	200 OK
			→	ACK
			→	INVITE2
			←	200 OK
			→	ACK
DISCONNECT 2	←		←	BYE S1
RELEASE COMPLETE	→		→	200 OK (BYE)
DISCONNECT 2	←		←	BYE S2
RELEASE COMPLETE	→		→	200 OK (BYE)
	Apply post test routine			

TSS 3PTY	TP_505_208	Reference subclause 5.2.13 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/6 AND 5.6/7 AND 5.3/13
Test purpose *The user equipment has the capability to invite a participant to the conference. Resource list is used* Ensure that the SUT is able to send a resource list to the conference AS to invite participant(s) to a conference. The SUT disconnects the previous held sessions.			

SIP header values

INVITE1

SDP a=sendonly

INVITE2

Request Line = conference factory URI

Contact: …; isfocus

Content-Type: application/resource-lists+xml

Content-Disposition: recipient-list

<?xml version="1.0" encoding="UTF-8"?>
<resource-lists xmlns="urn:ietf:params:xml:ns:resource-lists" xmlns:cp="urn:ietf:params:xml:ns:copyControl">
<list>
<entry uri="S1 URI" cp:copyControl="to"/>
</list>
</resource-lists>

DSS1 Parameter values

FACILITY

Begin3PTY-Inv

Call reference of call A-B

Message flow

Test equipment			Test equipment
	Session 1 is in active hold state		
	Session 2 is in active idle state		
FACILITY	→	→	INVITE1
		←	200 OK
		→	ACK
		→	INVITE2
		←	200 OK
		→	ACK
RELEASE COMPLETE1	→	→	BYE S1
		←	200 OK (BYE)
RELEASE COMPLETE2	→	→	BYE S2
		←	200 OK (BYE)
	Apply post test routine		

7.2.5.6 Closed User Group (CUG)

TSS CUG	TP_506_001	Reference subclause 5.2.9.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/9

Test purpose

Setup for a CUG call with outgoing access request CUG index present

Ensure that on receipt of an DSS1 SETUP message containing a Facility Information Element with CUGCallOperation invoke component, an SIP INVITE request is sent. The INVITE request contains an 'cug' XML MIME body. The cugCallOperation element contains the outgoingAccessRequest element is mapped from the DSS1 outgoingAccessRequest parameter set to 'TRUE' and the cugIndex element is mapped from the DSS1 cUGIndex parameter.

SIP header values

INVITE:

cug

cugCallOperation

outgoingAccessRequest>true<

cugIndex>[mapped from cUGIndex]<

DSS1 Parameter values

SETUP:

Facility

CUGCallOperation invoke

outgoingAccessRequest TRUE

cUGIndex configured index value

Message flow

End device		Test equipment
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying

Apply post test routine

TSS CUG	TP_505_002	Reference subclause 5.2.9.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/9

Test purpose

Setup for a CUG call with outgoing access request CUG index ***not*** *present*

Ensure that on receipt of a DSS1 SETUP message containing a Facility Information Element with CUGCallOperation invoke component, a SIP INVITE request is sent. The INVITE request contains an 'cug' XML MIME body. The cugCallOperation element that contains the outgoingAccessRequest element is mapped from the DSS1 outgoingAccessRequest parameter set to 'TRUE' and the DSS1 cUGIndex parameter is not present.

SIP header values

INVITE:

cug

cugCallOperation

outgoingAccessRequest>true<

DSS1 Parameter values

SETUP:

Facility

CUGCallOperation invoke

outgoingAccessRequest TRUE

Message flow

End device		Test equipment
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying

Apply post test routine

<table>
<tr><td>TSS
CUG</td><td>TP_505_003</td><td>Reference
subclause 5.2.9.1 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.4/9</td></tr>
<tr><td colspan="4">Test purpose
Setup for a CUG call without outgoing access request CUG index present

Ensure that on receipt of a DSS1 SETUP message containing a Facility Information Element with CUGCallOperation invoke component, a SIP INVITE request is sent. The INVITE request contains a 'cug' XML MIME body. The cugCallOperation element that contains the outgoingAccessRequest element is mapped from the DSS1 outgoingAccessRequest parameter set to 'FALSE' and the cugIndex element is mapped from the DSS1 cUGIndex parameter.</td></tr>
<tr><td colspan="4">SIP header values
INVITE:
1) cug
cugCallOperation
outgoingAccessRequest>false<
cugIndex>[mapped from cUGIndex]<</td></tr>
<tr><td colspan="4">DSS1 Parameter values
SETUP:
Facility
CUGCallOperation invoke
outgoingAccessRequest FALSE
cUGIndex configured index value</td></tr>
<tr><td colspan="4">Message flow</td></tr>
<tr><td colspan="2">End device</td><td></td><td>Test equipment</td></tr>
<tr><td colspan="2">SETUP</td><td>→</td><td>INVITE</td></tr>
<tr><td colspan="2"></td><td>←</td><td>407 Proxy Authentication Required</td></tr>
<tr><td colspan="2"></td><td>→</td><td>ACK</td></tr>
<tr><td colspan="2"></td><td>→</td><td>INVITE</td></tr>
<tr><td colspan="2">SETUP ACKNOWLEDGE</td><td>←</td><td>100 Trying</td></tr>
<tr><td colspan="4">Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
CUG</td><td>TP_505_004</td><td>Reference
clause 5.2.9.1 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.4/9</td></tr>
<tr><td colspan="4">Test purpose
Setup for a CUG call without outgoing access request CUG index not present

Ensure that on receipt of a DSS1 SETUP message containing a Facility Information Element with CUGCallOperation invoke component, a SIP INVITE request is sent. The INVITE request contains a 'cug' XML MIME body. The cugCallOperation element that contains the outgoingAccessRequest element is mapped from the DSS1 outgoingAccessRequest parameter set to 'FALSE' and the DSS1 cUGIndex parameter is not present.</td></tr>
<tr><td colspan="4">SIP header values
INVITE:
1) cug
cugCallOperation
outgoingAccessRequest>false<</td></tr>
</table>

DSS1 Parameter values

SETUP:

Facility

CUGCallOperation invoke

outgoingAccessRequest **FALSE**

Message flow

End device		Test equipment
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
SETUP ACKNOWLEDGE	←	100 Trying

Apply post test routine

TSS CUG	TP_505_005	Reference subclause 5.2.9.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/9

Test purpose

Setup for a CUG call with outgoing access request is rejected

Ensure that on receipt of a DSS1 SETUP message containing a Facility Information Element with CUGCallOperation invoke component, the DSS1 outgoingAccessRequest element is set to 'TRUE' and the DSS1 cUGIndex parameter is not present. If the AGCF/VGW receives a 403 Forbidden final response a DSS1 RELEASE COMPLETE message is sent and a Facility Information Element is present. The CUGCallOperation Return Error component is set to 'inconsistencyInDesignatedFacilityAndSubscriberClass'.

SIP header values

INVITE:

Cug

cugCallOperation

outgoingAccessRequest>**true**<

DSS1 Parameter values

SETUP:

Facility

CUGCallOperation invoke

outgoingAccessRequest **TRUE**

RELEASE COMPLETE

CUGCallOperation Return Error

inconsistencyInDesignatedFacilityAndSubscriberClass

Message flow

End device		Test equipment
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
	←	403 Forbidden
CASE A		
RELEASE COMPLETE	←	

CASE B		
DISCONNECT	←	
RELEASE COMPLETE	→	
	→	ACK

<table>
<tr><td>TSS
CUG</td><td>TP_505_006</td><td>Reference
subclause 5.2.9.1 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.4/9</td></tr>
<tr><td colspan="4">Test purpose
Setup for a CUG call without outgoing access request is rejected

Ensure that on receipt of a DSS1 SETUP message containing a Facility Information Element with CUGCallOperation invoke component, the DSS1 outgoingAccessRequest element is set to 'FALSE' and the DSS1 cUGIndex parameter is not present. If the AGCF/VGW receives a 403 Forbidden final response a DSS1 RELEASE COMPLETE message is sent and a Facility Information Element is present. The CUGCallOperation Return Error component is set to 'inconsistencyInDesignatedFacilityAndSubscriberClass'</td></tr>
<tr><td colspan="4">SIP header values
INVITE:
Cug
cugCallOperation
outgoingAccessRequest>false<</td></tr>
<tr><td colspan="4">DSS1 Parameter values
SETUP:
Facility
CUGCallOperation invoke
outgoingAccessRequest FALSE

RELEASE COMPLETE
CUGCallOperation Return Error
inconsistencyInDesignatedFacilityAndSubscriberClass</td></tr>
<tr><td colspan="4">Message flow</td></tr>
<tr><td colspan="2">End device</td><td></td><td>Test equipment</td></tr>
<tr><td colspan="2">SETUP</td><td>→</td><td>INVITE</td></tr>
<tr><td colspan="2"></td><td>←</td><td>407 Proxy Authentication Required</td></tr>
<tr><td colspan="2"></td><td>→</td><td>ACK</td></tr>
<tr><td colspan="2"></td><td>→</td><td>INVITE</td></tr>
<tr><td colspan="2"></td><td>←</td><td>403 Forbidden</td></tr>
<tr><td colspan="2">CASE A</td><td></td><td></td></tr>
<tr><td colspan="2">RELEASE COMPLETE</td><td>←</td><td></td></tr>
<tr><td colspan="2">CASE B</td><td></td><td></td></tr>
<tr><td colspan="2">DISCONNECT</td><td>←</td><td></td></tr>
<tr><td colspan="2">RELEASE COMPLETE</td><td>→</td><td></td></tr>
<tr><td colspan="2"></td><td>→</td><td>ACK</td></tr>
</table>

7.2.5.7 Communication waiting

7.2.5.7.1 Test purposes for POTS

TSS CW	TP_507_101	Reference subclause 6.3.2.4 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/14
Test purpose *The terminating user receives indication "a call is waiting"* Ensure that when a terminating user in a confirmed session receives an INVITE request to another dialogue, the SUT sends a 180 Ringing. The user is informed that a call is waiting.			
SIP header values INVITE:			
Message flow			

Test equipment			End device
INVITE1	→		Ringing
180 Ringing	←		
200 OK INVITE	←		Off hook
ACK	→		
INVITE2	→		
180 Ringing	←		Call waiting indication
	Apply post test routine		

TSS CW	TP_507_102	Reference clause C.9 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/14
Test purpose *The terminating user receives indication "a call is waiting" ignored* Ensure that when a terminating user in a confirmed session receives an INVITE request to another dialogue, the SUT sends a 180 Ringing. The user ignores the waiting call, when the answer timer expires the SUT sends a 486 (Busy Here) response.			
SIP header values INVITE:			
Message flow			

Test equipment			End device
INVITE1	→		Ringing
180 Ringing	←		
200 OK INVITE	←		Off hook
ACK	→		
INVITE2	→		
180 Ringing	←	Start $T_{no\ answer}$	Call waiting indication
		Expiry $T_{no\ answer}$	
486 Busy Here2	←		
ACK	→		
	Apply post test routine		

TSS CW	TP_507_103	Reference clause C.9 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/14

Test purpose

The terminating user receives indication "a call is waiting". Terminal based communication waiting (CW) is supported

Ensure that when a terminating user in a confirmed session receives an INVITE request to another dialogue, the SUT sends a 180 Ringing. The user is informed that a call is waiting.

SIP header values

180

Alert-Info: <urn:alert:service:call-waiting>

Message flow

Test equipment		End device
INVITE	→	Ringing
180 Ringing	←	
200 OK INVITE	←	Off hook
ACK	→	
INVITE	→	
180 Ringing	←	Call waiting indication

Apply post test routine

TSS CW	TP_507_104	Reference clause C.9 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/14 AND 5.3/9

Test purpose

The terminating user confirms the waiting call

Ensure that when a terminating user in a confirmed session receives an INVITE request to another dialogue, the SUT sends a 180 Ringing. The SUT evaluates the switch order command based on a provisioned mapping table.

The SOC indicates that the served user wishes the be connected to the waiting party, the AGCF performs the following actions:

- Send a 200 OK response to the INVITE request received from the waiting party
- Request the media gateway to:
  - Modify the Remote Descriptor of the ephemeral termination according to the SDP information received from the waiting party.

The AGCF also sends a re-INVITE request on the initial dialogue to hold the associated media stream.

SIP header values

INVITE3 (initial party)

SDP

a=sendonly

200 OK 3

SDP

a=recvonly

Message flow		
Test equipment		**End device**
INVITE1	→	Ringing
180 Ringing	←	
200 OK INVITE1	←	Off hook
ACK	→	
INVITE2	→	
180 Ringing2	←	Call waiting indication
		Flash hook
		Dial tone
		Dial switching order command
INVITE3	←	
200 OKINVITE3	→	
ACK	←	
200 OK INVITE2	←	
ACK	→	
Apply post test routine		

TSS CW	TP_507_105	Reference clause C.9 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/14 AND 5.3/9

Test purpose

The terminating user rejects the waiting call

Ensure that when a terminating user in a confirmed session receives an INVITE request to another dialogue, the SUT sends a 180 Ringing. The SUT evaluates the switch order command based on a provisioned mapping Table.

SOC indicates that the served user wishes to reject the waiting call, the AGCF performs the following actions:

- Send a provisioned error response code (e.g. 603) to the INVITE request received from the waiting party.
- Request the media gateway to:
  - Set the stream mode to send-receive.
  - Monitor the flash-hook event.
- Send a re-INVITE request towards the held party (i.e., the party that has been held for the purpose of collecting the switching order command). The re-INVITE request is built as follows:
  - The Request-URI is set to the held party's identity.
  - The SDP description for the active media stream is set to a=sendrecv.

SIP header values INVITE3 (initial party) SDP a=sendonly 200 OK 3 SDP a=recvonly INVITE4 (initial party) SDP a=sendrecv 200 OK 4 SDP a=sendrecv

Message flow

Test equipment		End device
INVITE	→	Ringing
180 Ringing	←	
200 OK INVITE	←	Off hook
ACK	→	
INVITE	→	
180 Ringing	←	Call waiting indication
		Flash hook
		Dial tone
		Dial switching order command
INVITE3	←	
200 OKINVITE3	→	
ACK	←	
603 Decline	←	
ACK	→	
INVITE4	←	
200 OKINVITE4	→	
ACK	←	

Apply post test routine

TSS CW	TP_507_106	Reference clause C.9 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/14 AND 5.3/9

Test purpose

The terminating user decide to switch back to the initial party

Ensure that when the initial party is on hold and a confirmed session with the waiting party exists, a flash hook event applies followed by a switching order command. If the value of the switching order command indicates that the initial party is to be switched back, the AGCF performs the following actions:

- Send a re-INVITE request towards the held party. The re-INVITE request is built as follows:
  - The Request-URI is set to the held party's identity.
  - The SDP description for the active media stream is set to a=sendrecv.
- Send a re-INVITE request towards the active party. The re-INVITE request is built as follows:
  - The Request URI is set to the active party's identity.
  - The SDP description for the active media stream is set to a=sendonly.
- Request the media gateway to:
  - Modify the Remote Descriptor of the ephemeral termination according to the SDP information associated with the held party.

SIP header values

INVITE1 (initial party)
SDP
a=sendrecv

200 OK 3
SDP
a=sendrecv

INVITE4 (waiting party)
SDP
a=sendonly

200 OK 4
SDP
a=recvonly

Message flow

Test equipment		End device
Terminating user is connected with the waiting party		
		Flash hook
		Dial tone
		Dial switching order command
INVITE1 (initial party)	←	
200 OK INVITE1	→	
ACK	←	
INVITE2 (waiting party)	←	
200 OK INVITE2	→	
ACK	←	
Apply post test routine		

TSS CW	TP_507_107	Reference clause C.9 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/14 AND 5.3/10

Test purpose

The terminating user confirms the waiting call

Ensure that when a terminating user in a confirmed session receives an INVITE request to another dialogue, the SUT sends a 180 Ringing. The SUT evaluates the switch order command (SOC) based on a provisioned mapping table.

The SOC indicates that the served user wishes to be connected to the waiting party, the AGCF performs the following actions:

- Send a 200 OK response to the INVITE request received from the waiting party
- Request the media gateway to:
  - Modify the Remote Descriptor of the ephemeral termination according to the SDP information received from the waiting party.

The AGCF also sends a re-INVITE request on the initial dialogue to hold the associated media stream.

- The Request URI is structured as follows:
  - A user part containing a provisioned prefix followed by the switching order command without the start and finish fields.
  - A domain name that provides sufficient information to the S-CSCF to forward the INVITE request to the appropriate AS, based on Initial Filter Criteria stored in the user profile, e.g. SOC- "SO (SR SI)"@pes.operator.com
  - A P-Asserted-Identity containing the public identity of the subscriber issuing the switching control command.
  - An SDP offer for a voice call.

SIP header values

INVITE3
Request-Line: SO (SR SI)@pes.operator.com
SDP
a=sendonly

200 OK 3
SDP
a=recvonly

Message flow

Test equipment		**End device**
INVITE1	→	Ringing
180 Ringing	←	
200 OK INVITE1	←	Off hook
ACK	→	
INVITE2	→	
180 Ringing2	←	Call waiting indication
		Flash hook
		Dial tone
		Dial switching order command
INVITE3	←	
200 OKINVITE3	→	
ACK	←	
200 OK INVITE2	←	
ACK	→	

Apply post test routine

TSS CW	TP_507_108	Reference clause C.9 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/14 AND 5.3/10
Test purpose *The terminating user rejects the waiting call* Ensure that when a terminating user in a confirmed session receives an INVITE request to another dialogue, the SUT sends a 180 Ringing. The SUT evaluates the switch order command based on a provisioned mapping table SOC indicates that the served user wishes to reject the waiting call, the AGCF performs the following actions: • Send a provisioned error response code (e.g. 603) to the INVITE request received from the waiting party. • Request the media gateway to: – Set the stream mode to send-receive. – Monitor the flash-hook event. • Send a re-INVITE request towards the held party (i.e. the party that has been held for the purpose of collecting the switching order command). The re-INVITE request is built as follows: – The Request-URI is set to the held party's identity. – The SDP description for the active media stream is set to a=sendrecv. An SDP offer for a voice call. The AGCF also sends a re-INVITE request on the initial dialogue to hold the associated media stream. – A user part containing a provisioned prefix followed by the switching order command without the start and finish fields. – A domain name that provides sufficient information to the S-CSCF to forward the INVITE request to the appropriate AS, based on Initial Filter Criteria stored in the user profile, e.g. SOC- "SO (SR SI)"@pes.operator.com – A P-Asserted-Identity containing the public identity of the subscriber issuing the switching control command.			
SIP header values INVITE3 (initial party) Request-Line: SO (SR SI)@pes.operator.com SDP a=sendonly 200 OK 3 SDP a=recvonly INVITE4 (initial party) SDP a=sendrecv 200 OK 4 SDP a=sendrecv			

Message flow		
Test equipment		**End device**
INVITE	→	Ringing
180 Ringing	←	
200 OK INVITE	←	Off hook
ACK	→	
INVITE	→	
180 Ringing	←	Call waiting indication
		Flash hook
		Dial tone
		Dial switching order command
INVITE3	←	
200 OKINVITE3	→	
ACK	←	
603 Decline	←	
ACK	→	
INVITE4	←	
200 OKINVITE4	→	
ACK	←	
Apply post test routine		

TSS CW	TP_507_109	Reference clause C.9 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/14 AND 5.3/10

Test purpose

The terminating user decide to switch back to the initial party

Ensure that when the initial party is on hold and a confirmed session with the waiting party exists a flash hook event applies followed by a switching order command. If the value of the switching order command indicates that the initial party is to be switched back, the AGCF performs the following actions:

- Send a re-INVITE request towards the held party. The re-INVITE request is built as follows:
  - The Request-URI is set to the held party's identity.
  - The SDP description for the active media stream is set to a=sendrecv.
- Send a re-INVITE request towards the active party. The re-INVITE request is built as follows:
  - The Request URI is set to the active party's identity.
  - The SDP description for the active media stream is set to a=sendonly.
- Request the media gateway to:
  - Modify the Remote Descriptor of the ephemeral termination according to the SDP information associated with the held party.

The AGCF also sends a re-INVITE request on the initial dialogue to hold the associated media stream.

- A user part containing a provisioned prefix followed by the switching order command without the start and finish fields.
- A domain name that provides sufficient information to the S-CSCF to forward the INVITE request to the appropriate AS, based on Initial Filter Criteria stored in the user profile, e.g. SOC- "SO (SR SI)"@pes.operator.com

A P-Asserted-Identity containing the public identity of the subscriber issuing the switching control command.

SIP header values INVITE1 (initial party) Request-Line: SO (SR SI)@pes.operator.com SDP a=sendrecv 200 OK 3 SDP a=sendrecv INVITE4 (waiting party) SDP a=sendonly 200 OK 4 SDP a=recvonly
Message flow

Test equipment		**End device**
Terminating user is connected with the waiting party		
		Flash hook
		Dial tone
		Dial switching order command
INVITE1 (initial party)	←	
200 OK INVITE1	→	
ACK	←	
INVITE2 (waiting party)	←	
200 OK INVITE2	→	
ACK	←	
Apply post test routine		

7.2.5.7.2 Test purposes for ISDN

TSS CW	TP_507_201	Reference subclause 5.2.11.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/10
Test purpose *Call waiting indication received in a 183 Session Progress* Ensure that on receipt of a 183 Session Progress containing an Alert-Info header set to the urn '<urn:alert:service:call-waiting>' and no provisional response was received before, a PROGRESS message is sent and a Notification indicator Information Element is present set to 'Call is a waiting call'.			
SIP header values 183 Alert-Info: <urn:alert:service:call-waiting>			
DSS1 Parameter values PROGRESS Notification indicator Call is a waiting call			

Message flow		
End device		**Test equipment**
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
PROGRESS	←	183 (Session Progress)
	Apply post test routine	

TSS CW	TP_507_202	Reference subclause 5.2.11.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/10
Test purpose *Call waiting indication received in a 180 Ringing* Ensure that on receipt of a 180 Ringing containing an Alert-Info header set to the urn '<urn:alert:service:call-waiting>', an ALERTING message is sent and a Notification indicator Information Element is present set to 'Call is a waiting call'.			
SIP header values 180 Alert-Info: <urn:alert:service:call-waiting>			
DSS1 Parameter values ALERTING Notification indicator Call is a waiting call			
Message flow **End device** / **Test equipment** SETUP → INVITE ← 407 Proxy Authentication Required → ACK → INVITE ALERTING ← 180 (Ringing) **Apply post test routine**			

TSS CW	TP_507_203	Reference subclause 5.2.11.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/10
Test purpose *Call waiting indication received in a 180 Ringing a 183 session Progress was received before* Ensure that on receipt of a 180 Ringing containing an Alert-Info header set to the urn '<urn:alert:service:call-waiting>' and a provisional response was received before, an ALERTING message is sent and a Notification indicator Information Element is present set to 'Call is a waiting call'.			
SIP header values 180 Alert-Info: <urn:alert:service:call-waiting>			

DSS1 Parameter values

ALERTING

  Notification indicator

    Call is a waiting call

Message flow

End device		Test equipment
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
PROGRESS	←	183 (Session Progress)
ALERTING	←	180 (Ringing)

Apply post test routine

TSS CW	TP_507_204	Reference subclause 5.2.11.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/10 AND 5.1.3/2

Test purpose

CW indication received in the INVITE busy condition met a CW notification is sent

Ensure that on receipt of a CW indication in an INVITE request due to the 'ims-cw' XML element where the busy condition is met (two communication are already established), a SETUP is sent. The Channel identification is set to 'no channel'.

Ensure that on receipt of the ALERTING message a 180 Ringing is sent containing an Alert-Info header set to 'urn:alert:service:call-waiting'.

SIP header values

INVITE

  <?xml version="1.0"?>

  <ims-cw xmlns="urn:3gpp:ns:cw:1.0">

    <communication-waiting-indication/>

  </ims-cw>

180

  Alert-Info: <urn:alert:service:call-waiting>

DSS1 Parameter values

SETUP

  Channel identification

    no channel

Message flow

Test equipment			End device
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING

Apply post test routine

TSS CW	TP_507_205	Reference subclause 5.2.11.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/10 AND 5.1.3/2

Test purpose

CW indication received in the INVITE busy condition not met a CW notification is not sent

Ensure that on receipt of a CW indication in an INVITE request due to the 'ims-cw' XML element where the busy condition is not met, a SETUP is sent. The Channel identification is not set to 'no channel'.

Ensure that on receipt of the ALERTING message a 180 Ringing is sent containing and the Alert-Info header set to 'urn:alert:service:call-waiting' is not present.

SIP header values

INVITE

<?xml version="1.0"?>
<ims-cw xmlns="urn:3gpp:ns:cw:1.0">
<communication-waiting-indication/>
</ims-cw>

DSS1 Parameter values

Message flow

Test equipment			End device
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING

Apply post test routine

TSS CW	TP_507_206	Reference subclause 5.2.11.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/10 AND 5.1.3/2

Test purpose

CW indication received in the INVITE busy condition met a CW notification is sent Call Waiting condition terminated after expiry of $T_{UE\text{-}CW}$

Ensure that on receipt of a CW indication in an INVITE request due to the 'ims-cw' XML element and an Expires header where the busy condition is met (two communication are already established), a SETUP is sent. The Channel identification is set to 'no channel'.

The TUE-CW timer is started.

Ensure that on receipt of the ALERTING message a 180 Ringing is sent containing an Alert-Info header set to 'urn:alert:service:call-waiting'.

Ensure that on expiry of TUE-CW a 480 Temporarily Unavailable is sent containing a Reason header set to 'Q.850' and 'cause=19'

SIP header values

INVITE

Expires: <any value>
<?xml version="1.0"?>
<ims-cw xmlns="urn:3gpp:ns:cw:1.0">
<communication-waiting-indication/>
</ims-cw>

180

Alert-Info: <urn:alert:service:call-waiting>

480

Reason: Q.850; cause=19

DSS1 Parameter values

SETUP

Channel identification

no channel

Message flow

Test equipment			End device
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
	Start T_{UE-CW}		
	Expiry T_{UE-CW}		
480 (Temporarily Unavailable)	←	→	RELEASE COMPLETE
ACK	→		

Apply post test routine

TSS CW	TP_507_207	Reference subclause 5.2.11.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/10 AND 5.1.3/2

Test purpose

CW indication not received in the INVITE busy condition met a CW notification is sent

Ensure that on receipt of an INVITE request where the CW indication is not present and the busy condition is met (two communication are already established), a SETUP is sent. The Channel identification is set to 'no channel'.

Ensure that on receipt of the ALERTING message a 180 Ringing is sent containing an Alert-Info header set to 'urn:alert:service:call-waiting'.

SIP header values

180

Alert-Info: <urn:alert:service:call-waiting>

DSS1 Parameter values

SETUP

Channel identification

no channel

Message flow

Test equipment			End device
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING

Apply post test routine

TSS CW	TP_507_208	Reference subclause 5.2.11.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/10 AND 5.1.3/2

Test purpose

CW notification received from the private network

Ensure that on receipt of an ALERTING message or a PROGRESS message or a NOTIFY message containing a Notification indicator Information Element, the Notification description is set to 'Call is a waiting call' a 180 Ringing is sent containing an Alert-Info header set to 'urn:alert:service:call-waiting'.

SIP header values 180 Alert-Info: <urn:alert:service:call-waiting> 183 Alert-Info: <urn:alert:service:call-waiting>
DSS1 Parameter values ALERTING Notification indicator Notification description Call is a waiting call PROGRESS Notification indicator Notification description Call is a waiting call NOTIFY Notification indicator Notification description Call is a waiting call
Message flow

Test equipment			**End device**
INVITE	→	→	SETUP
CASE A			
180 Ringing	←	←	ALERTING
CASE B			
183 (Session Progress)	←	←	PROGRESS
CASE C			
183 (Session Progress)	←	←	NOTIFY
Apply post test routine			

7.2.5.8 Terminal Portability (TP)

TSS TP	TP_508_001	Reference subclause 5.2.12.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/11
Test purpose *Suspend initiated* Ensure that on receipt of a SUSPEND message in the confirmed state an INVITE request or UPDATE request is sent. A 'a' line in the SDP is set to 'sendonly' or 'inactive'			
SIP header values INVITE SDP a= sendonly or a=inactive			

DSS1 Parameter values SUSPEND   Call identity

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE1
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
		→	ACK
SUSPEND	→		
SUSPEND ACKNOWLEDGE	←		
CASE A			
		→	INVITE
		←	200 OK INVITE
		→	ACK
CASE B			
		→	UPDATE
		←	200 OK UPDATE
Apply post test routine			

TSS TP	TP_508_002	Reference subclause 5.2.12.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/11 AND 5.1.3/2
Test purpose *Resumed initiated* Ensure that on receipt of a RESUME message in the confirmed state and the ISDN connection is in the suspended state an INVITE request or UPDATE request is sent. A 'a' line in the SDP is set to 'sendrecv'			

SIP header values
INVITE1/UPDATE 1
    SDP
        a=sendonly
 	
INVITE2/UPDATE 2
    SDP
        a=sendrecv			

DSS1 Parameter values
SUSPEND
        Call identity
 	
RESUME
        Call identity			

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
		→	ACK
CASE A			
SUSPEND	→	→	INVITE1
SUSPEND ACKNOWLEDGE	←	←	200 OK INVITE
		→	ACK
		→	INVITE2
RESUME	→	←	200 OK INVITE
SUSPEND ACKNOWLEDGE	←	→	ACK
CASE B			
SUSPEND	→	→	UPDATE1
SUSPEND ACKNOWLEDGE	←	←	200 OK UPDATE
RESUME	→	→	UPDATE2
SUSPEND ACKNOWLEDGE	←	←	200 OK UPDATE
Apply post test routine			

TSS TP	TP_508_003	Reference subclause 5.2.12.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/11 AND 5.1.3/2
Test purpose *Suspend initiated T307 expiry* Ensure that on receipt of a SUSPEND message timer T307 is started. After the expiry of timer T307 the SIP communication is terminated. A CANCEL or BYE is sent. A Reason header is present set to 'Q.850' and the cause parameter set to '102'.			
SIP header values INVITE SDP a= sendonly or a=inactive CANCEL/BYE Reason: Q.850; cause=102			
DSS1 Parameter values SUSPEND Call identity Call identity			

End device				Test equipment
SETUP	→		→	INVITE
			←	407 Proxy Authentication Required
			→	ACK
			→	INVITE
ALERTING	←		←	180 Ringing
CONNECT	←		←	200 OK INVITE
			→	ACK
CASE A				
SUSPEND	→	**Start T307**	→	INVITE1
SUSPEND ACKNOWLEDGE	←		←	200 OK INVITE
			→	ACK
		T307 expiry	→	CANCEL/BYE
			←	200 OK CANCEL/BYE
			←	487 Request Terminated
			→	ACK
CASE B				
SUSPEND	→	**Start T307**	→	UPDATE1
SUSPEND ACKNOWLEDGE	←		←	200 OK UPDATE
		T307 expiry	→	CANCEL/BYE
			←	200 OK CANCEL/BYE
			←	487 Request Terminated
			→	ACK
		Apply post test routine		

TSS TP	TP_508_004	Reference subclause 5.2.12.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/11 AND 5.1.3/1
Test purpose *Suspend initiated by the private network* Ensure that on receipt of a NOTIFY message the Notification indicator Information Element is set to 'User suspended' in the confirmed state and an INVITE request or UPDATE request is sent. A 'a' line in the SDP is set to 'sendonly' or 'inactive'			
SIP header values INVITE SDP a= sendonly			
DSS1 Parameter values NOTIFY Notification indicator Notification description User suspended			

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
		→	ACK
NOTIFY	→		
CASE A		→	INVITE1
		←	200 OK INVITE
		→	ACK
CASE B		→	UPDATE1
		←	200 OK UPDATE
Apply post test routine			

TSS TP	TP_508_005	Reference subclause 5.2.12.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/11 AND 5.1.3/1
Test purpose *Resumed initiated from the private network* Ensure that on receipt of a NOTIFY message, the Notification indicator Information Element is set to 'User resumed' in the confirmed state, the ISDN connection is in the suspended state and an INVITE request or UPDATE request is sent. A 'a' line in the SDP is set to 'sendrecv'			
SIP header values INVITE1/UPDATE 1 SDP a=sendonly INVITE2/UPDATE 2 SDP a=sendrecv			
DSS1 Parameter values NOTIFY 1 Notification indicator Notification description User suspended NOTIFY 2 Notification indicator Notification description User resumed			

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
		→	ACK
CASE A			
NOTIFY 1	→	→	INVITE1
		←	200 OK INVITE
		→	ACK
NOTIFY 2	→	→	INVITE2
		←	200 OK INVITE
		→	ACK
CASE B			
NOTIFY 1	→	→	UPDATE1
		←	200 OK UPDATE
NOTIFY 2	→	→	UPDATE2
		←	200 OK UPDATE
Apply post test routine			

TSS TP	TP_508_006	Reference subclause 5.2.12.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/11 AND 5.1.3/2
Test purpose *Suspend initiated* Ensure that on receipt of an INVITE or UPDATE message in the confirmed state the 'a' line in the SDP is set to 'sendonly' and a SUSPEND message is sent to the end device.			
SIP header values INVITE2/UPDATE SDP a= sendonly			
DSS1 Parameter values SUSPEND Call identity			

Message flow			
Test equipment			**End device**
INVITE1	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→		
CASE A			
INVITE2	→	→	SUSPEND
200 OK (INVITE)	←	←	SUSPEND ACKNOWLEDGE
ACK	→		
CASE B			
UPDATE	→	→	SUSPEND
200 OK UPDATE	←	←	SUSPEND ACKNOWLEDGE
Apply post test routine			

TSS TP	TP_508_007	Reference subclause 5.2.12.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/11 AND 5.1.3/2
Test purpose *Resumed initiated* Ensure that on receipt of an INVITE or UPDATE message in the confirmed state the 'a' line in the SDP is set to 'sendrecv' and a SUSPEND message is sent to the end device.			
SIP header values INVITE3/UPDATE2 SDP a= sendonly			
DSS1 Parameter values RESUME Call identity			

Message flow			
Test equipment			**End device**
INVITE1	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→		
CASE A			
INVITE2	→	→	SUSPEND
200 OK (INVITE)	←	←	SUSPEND ACKNOWLEDGE
ACK	→		
INVITE3	→	→	RESUME
200 OK (INVITE)	←	←	SUSPEND ACKNOWLEDGE
ACK	→		

CASE B			
UPDATE1	→	→	SUSPEND
200 OK UPDATE	←	←	SUSPEND ACKNOWLEDGE
CASE B			
UPDATE2	→	→	RESUME
200 OK UPDATE	←	←	SUSPEND ACKNOWLEDGE
Apply post test routine			

TSS TP	TP_508_008	Reference subclause 5.2.12.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/11 AND 5.1.3/1

Test purpose

Suspend initiated from the private network

Ensure that on receipt of an INVITE or UPDATE message in the confirmed state the 'a' line in the SDP is set to 'sendonly'. A NOTIFY message the Notification indicator Information Element is set to 'User suspended' is sent to the end device.

SIP header values

INVITE2/UPDATE
  SDP
    a= sendonly

DSS1 Parameter values

NOTIFY
  Notification indicator
    Notification description
      User suspended

Message flow

Test equipment			End device
INVITE1	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→		
CASE A			
INVITE2	→	→	NOTIFY
200 OK (INVITE)	←		
ACK	→		
CASE B			
UPDATE	→	→	NOTIFY
200 OK UPDATE	←		
Apply post test routine			

TSS TP	TP_508_009	Reference 5.2.12.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/11 AND 5.1.3/1

Test purpose

Resumed initiated from the private network

Ensure that on receipt of an INVITE or UPDATE message in the confirmed state, the 'a' line in the SDP is set to 'sendrecv'. A NOTIFY message the Notification indicator Information Element is set to 'User resumed' is sent to the end device.

SIP header values

INVITE2/UPDATE
    SDP
        a= sendrecv

DSS1 Parameter values

NOTIFY 2
    Notification indicator
        Notification description
            User resumed

Message flow

Test equipment			End device
INVITE1	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→		
CASE A			
INVITE2	→	→	NOTIFY1
200 OK (INVITE)	←		
ACK	→		
INVITE2	→	→	NOTIFY2
200 OK (INVITE)	←		
ACK	→		
CASE B			
UPDATE	→	→	NOTIFY1
200 OK UPDATE	←		
UPDATE	→	→	NOTIFY2
200 OK UPDATE	←		

Apply post test routine

7.2.5.9 Explicit Communication Transfer (ECT)

TSS ECT	TP_509_001	Reference subclause 5.2.7.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/12

Test purpose

Activation of ECT using the EctExecute invoke component in active state

A call to a user in active state is on hold and a call to an additional user is in active state.

Ensure that on receipt of a FACILITY message containing a 'EctExecute' invoke component regarding the Call Reference to the user in held state a REFER is sent.

- The request line is set to the address of the held user.
- The Refer-To header is set to the address of the active user, the method parameter is set to 'invite', the Replaces header is set to the session identification values of the active connection.
- The Referred-By header is set to the Public user identity of the served user

After an NOTIFY containing the message/sipfrag 'SIP/2.0 200 OK' a DISCONNECT is sent for all connection to the ISDN equipment, the DISCONNECT regarding the Call Reference of the received FACILIT message contains the EctExecute return result component.

SIP header values

REFER: [Connection CR 1]
Refer-To: <sip:[Connection CR 2]>; method=invite?Replaces=[Call CR 2]
Referred-By: <sip:[Public user identity]>

NOTIFY 1:
Content-Type: message/sipfrag
SIP/2.0 100 Trying

NOTIFY 2:
Content-Type: message/sipfrag
SIP/2.0 200 OK

DSS1 Parameter values

FACILITY:
CR 1
EctExecute invoke

DISCONNECT 1
EctExecute return result

End device			Test equipment
	A call is established in held state CR 1		
	A call is established in active state CR2		
FACILITY	→	→	INVITE (CR2, sendonly)
		←	200 OK INVITE
		→	ACK
		→	REFER
		←	202 Accepted
		←	NOTIFY 1
		→	200 OK NOTIFY
		←	NOTIFY 2

		→	200 OK NOTIFY
		←	BYE (CR 2)
		→	200 OK BYE
		→	BYE (CR 1)
DISCONNECT 1	←	←	200 OK BYE
RELEASE	→		
RELEASE COMPLETE	←		
DISCONNECT 2	←		
RELEASE	→		
RELEASE COMPLETE	←	←	200 OK UPDATE
	Apply post test routine		

TSS ECT	TP_509_002	Reference subclause 5.2.7.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/12

Test purpose

Activation of ECT using the EctExecute invoke component in alerting state

A call to a user in active state is on hold and a call to a additional user is in alerting state.

Ensure that on receipt of a FACILITY message containing a 'EctExecute' invoke component regarding the Call Reference to the user in held state a REFER is sent.

- The request line is set to the address of the held user.
- The Refer-To header is set to the address of the active user, the method parameter is set to 'invite', the Replaces header is set to the session identification values of the active connection.
- The Referred-By header is set to the Public user identity of the served user

After a NOTIFY containing the message/sipfrag 'SIP/2.0 200 OK' a DISCONNECT is sent for all connection to the ISDN equipment, the DISCONNECT regarding the Call Reference of the received FACILIT message contains the EctExecute return result component.

SIP header values

REFER: [Connection CR 1]
Refer-To: <sip:[Connection CR 2]>; method=invite?Replaces=[Call CR 2]
Referred-By: <sip:[Public user identity]>

NOTIFY 1:
Content-Type: message/sipfrag
SIP/2.0 100 Trying

NOTIFY 2:
Content-Type: message/sipfrag
SIP/2.0 200 OK

DSS1 Parameter values

FACILITY:

CR 1

EctExecute invoke

DISCONNECT 1

EctExecute return result

End device			**Test equipment**
	A call is established in held state CR 1		
	A call is alerting state CR2		
FACILITY	→	→	INVITE (CR2, sendonly)
		←	200 OK INVITE
		→	ACK
		→	REFER
		←	202 Accepted
		←	NOTIFY 1
		→	200 OK NOTIFY
		←	NOTIFY 2
		→	200 OK NOTIFY
		←	BYE (CR 2)
		→	200 OK BYE
		→	BYE (CR 1)
DISCONNECT 1	←	←	200 OK BYE
RELEASE	→		
RELEASE COMPLETE	←		
DISCONNECT 2	←		
RELEASE	→		
RELEASE COMPLETE	←	←	200 OK UPDATE
	Apply post test routine		

TSS ECT	TP_509_003	Reference subclause 5.2.7.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/12 AND 5.1.3/1

Test purpose

Activation of ECT performed in a private network using the EctInform invoke component set to alerting, no mapping

One or two sessions are established. Ensure that on receipt of a DSS1 FACILITY message for a Call Reference containing a EctInform invoke component set to Alerting, no SIP message is sent and no SIP action occurs.

SIP header values

DSS1 Parameter values FACILITY: EctInform invoke Alerting
End device **Test equipment** **A call is established CR 1** **A call is established CR2** FACILITY ➔ **Apply post test routine**

TSS ECT	TP_509_004	Reference subclause 5.2.7.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/12 AND 5.1.3/1
Test purpose *Activation of ECT performed in a private network using the EctInform invoke component set to active, no mapping* One or two sessions are established. Ensure that on receipt of a DSS1 FACILITY message for a Call Reference containing a EctInform invoke component set to Active, no SIP message is sent and no SIP action occurs.			
SIP header values			
DSS1 Parameter values FACILITY: EctInform invoke Active redirectionNumber			
End device **Test equipment** **A call is established CR 1** **A call is established CR2** FACILITY ➔ **Apply post test routine**			

TSS ECT	TP_509_005	Reference subclause 5.2.7.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/12 AND 5.1.3/1
Test purpose *Request of Subaddress using the SubaddressTransfer invoke component set to active, no mapping* One or two sessions are established. Ensure that on receipt of a DSS1 FACILITY message for a Call Reference containing a RequestSubaddress invoke component, no SIP message is sent and no SIP action occurs.			
SIP header values			
DSS1 Parameter values FACILITY: RequestSubaddress invoke			

End device		**Test equipment**
	A call is established CR 1	
	A call is established CR2	
FACILITY	→	
	Apply post test routine	

TSS	TP_509_006	Reference subclause 5.2.7.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/12
Test purpose *Receipt of an INVITE request to a call in held state* A communication is in the held state. Ensure that on receipt of an INVITE request, the line in the SDP is set to 'sendrecv', a DSS1 NOTIFY message is sent to the user equipment indicating the user is resumed.			
SIP header values INVITE: SDP a=sendrecv			
DSS1 Parameter values NOTIFY: Notification description User resumed			
Message flow			

Test equipment			**End device**
	A call is established in held state		
INVITE	→	→	NOTIFY
200 OK INVITE	←		
ACK	→		
	Apply post test routine		

TSS ECT	TP_509_007	Reference subclause 5.2.7.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/12
Test purpose *Receipt of an UPDATE request to a call in held state* A communication is in the held state. Ensure that on receipt of an UPDATE request and the a line in the SDP is set 'sendrecv' a DSS1 NOTIFY message is sent to the user equipment indicating the user is resumed.			
SIP header values UPDATE: SDP a=sendrecv			
DSS1 Parameter values NOTIFY: Notification description User resumed			

Message flow **Test equipment** **End device** **A call is established in held state** UPDATE → → NOTIFY 200 OK UPDATE ← **Apply post test routine**

TSS ECT	TP_509_008	Reference subclause 5.2.7.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/12
Test purpose *Receipt of an INVITE request to a call in active state* A communication is in the active state. Ensure that on receipt of an INVITE request the SDP is not different from the previously received SDP containing a: • P-Asserted-Identity • Privacy absent or not "id" • isub parameter in the P-Asserted-Identity No Message is sent to the DSS1 user equipment.			
SIP header values INVITE: P-Asserted-Identity Isub			
DSS1 Parameter values			
Message flow **Test equipment** **End device** **A call is established in held state** INVITE → 200 OK INVITE ← ACK → **Apply post test routine**			

TSS ECT	TP_509_009	Reference subclause 5.2.7.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/12
Test purpose *Receipt of an UPDATE request to a call in active state* A communication is in the active state. Ensure that on receipt of an UPDATE request, the SDP is not different from the previously received SDP containing a: • P-Asserted-Identity • Privacy absent or not "id" • isub parameter in the P-Asserted-Identity No Message is sent to the DSS1 user equipment.			

SIP header values UPDATE: P-Asserted-Identity Isub
DSS1 Parameter values
Message flow **Test equipment** **End device** **A call is established in held state** UPDATE → 200 OK UPDATE ← **Apply post test routine**

7.2.5.10 User User Service (UUS)

TSS UUS	TP_510_001	Reference subclause 5.2.10.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/13
Test purpose *Call setup contains a DSS1 User-user service Information Element implicit request* Ensure that on receipt of a DSS1 SETUP an INVITE request is sent. The User information and the Protocol discriminator in the User-user Information Element is mapped into the uuidata parameter in the User-to-User header present in the sent INVITE request.			
SIP header values INVITE User-to-User uuidata			
DSS1 Parameter values SETUP User-user Protocol discriminator User information			
End device **Test equipment** SETUP → → INVITE ← 407 Proxy Authentication Required → ACK → INVITE1 **Apply post test routine**			

TSS UUS	TP_510_002	Reference subclause 5.2.10.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/13
Test purpose *Call setup contains a DSS1 User-user Information Element implicit request, response in 180 Ringing* Ensure that on receipt of a 180 Ringing an ALERTING is sent. The uuidata parameter in the User-to-User header is mapped into the User information and the Protocol discriminator in the User-user Information Element is present in the sent ALERTING.			

SIP header values
INVITE
    User-to-User
      Uuidata
 	
180
    User-to-User
      uuidata

DSS1 Parameter values
SETUP
      User-user
        Protocol discriminator
        User information
 	
ALERTING
      User-user
        Protocol discriminator
        User information

End device			**Test equipment**
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE1
ALERTING	←	←	180 Ringing
	Apply post test routine		

TSS UUS	TP_510_003	Reference 5.2.10.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/13

Test purpose *Call setup contains a DSS1 User-user Information Element implicit request, response in 183 Session Progress* Ensure that on receipt of a 183 Session Progress a PROGRESS is sent. The uuidata parameter in the User-to-User header is mapped into the User information and the Protocol discriminator in the User-user Information Element is present in the sent PROGRESS.

SIP header values
INVITE
    User-to-User
      Uuidata
 	
180
    User-to-User
      uuidata

DSS1 Parameter values SETUP User-user Protocol discriminator User information **PROGRESS** User-user Protocol discriminator User information

End device			**Test equipment**
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE1
ALERTING	←	←	180 Ringing
PROGRESS	←	←	183 (Session Progress)
Apply post test routine			

TSS UUS	TP_510_004	Reference subclause 5.2.10.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/13
Test purpose *Call setup contains a DSS1 User-user Information Element implicit request, response in 200 OK* Ensure that on receipt of a 200 OK INVITE a PROGRESS is sent. The uuidata parameter in the User-to-User header is mapped into the User information and the Protocol discriminator in the User-user Information Element is present in the sent CONNECT.			
SIP header values INVITE User-to-User Uuidata 180 User-to-User uuidata			
DSS1 Parameter values SETUP User-user Protocol discriminator User information CONNECT User-user Protocol discriminator User information			

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE1
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
		→	ACK
Apply post test routine			

TSS UUS	TP_510_005	Reference subclause 5.2.10.1.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/13

Test purpose

Call setup contains a DSS1 User-user Information Element implicit request, response in BYE

Ensure that on receipt of a BYE request a PROGRESS is sent. The uuidata parameter in the User-to-User header is mapped into the User information and the Protocol discriminator in the User-user Information Element the present in the sent DISCONNECT in backward direction.

SIP header values

INVITE
- User-to-User
  - uuidata

BYE
- User-to-User
  - uuidata

DSS1 Parameter values

SETUP
- User-user
  - Protocol discriminator
  - User information

DISCONNECT
- User-user
  - Protocol discriminator
  - User information

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE1
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
		→	ACK
DISCONNECT	←	←	BYE
RELEASE	→	→	200 OK BYE
RELEASE COMPLETE	←		
Apply post test routine			

TSS UUS	TP_510_006	Reference subclause 5.2.10.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/13

Test purpose

Call setup with User-to-user request service 1 explicit request preferred rejected in CALL PROCEEDING

Ensure that on receipt of a SETUP message containing a User-user Information Element and a Facility Information Element UserUserService request invoke Service 1 preferred an INVITE request is sent. Upon receipt of a 183 (Session Progress) provisional response a CALL PROCEEDING message is sent. To reject the requested User to User service a Facility Information element is present and is set to UserUserService return error rejectedByNetwork

DSS1 Parameter values

SETUP
- User-user
  - Protocol discriminator
  - User information
- Facility
  - UserUserService request invoke
    - Service 1 preferred

CALL PROCEEDING
- Facility
  - UserUserService return error
    - rejectedByNetwork

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE1
CALL PROCEEDING	←	←	183 (Session Progress)

Apply post test routine

TSS UUS	TP_510_007	Reference 5.2.10.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/13

Test purpose

Call setup with User-to-user request service 1 explicit request preferred rejected in ALERTING

Ensure that on receipt of a SETUP message containing a user-user Information Element and a Facility Information Element UserUserService request invoke Service 1 preferred an INVITE request is sent. Upon receipt of a 180 (Ringing) provisional response an ALERTING message is sent. To reject the requested User to User service a Facility Information element is present set to UserUserService return error rejectedByNetwork

DSS1 Parameter values

SETUP
User-user
Protocol discriminator
User information
Facility
UserUserService request invoke
Service 1 preferred

ALERTING
Facility
UserUserService return error
rejectedByNetwork

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 Ringing

Apply post test routine

TSS UUS	TP_510_008	Reference subclause 5.2.10.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/13

Test purpose

Call setup with user-to-user request service 1 explicit request preferred rejected in CONNECT

Ensure that on receipt of a SETUP message containing a user-user Information Element and a Facility Information Element UserUserService request invoke Service 1 preferred an INVITE request is sent. Upon receipt of a 200 OK INVITE final response a CONNECT message is sent. To reject the requested User to User service a Facility Information element is present set to UserUserService return error rejectedByNetwork

DSS1 Parameter values

SETUP
User-user
Protocol discriminator
User information
Facility
UserUserService request invoke
Service 1 preferred

CONNECT
Facility
UserUserService return error
rejectedByNetwork

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK INVITE
		→	ACK
Apply post test routine			

TSS UUS	TP_510_009	Reference 5.2.10.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/13
Test purpose *Call setup with user-to-user request service 1 explicit request required rejected in DISCONNECT* Ensure that on receipt of a SETUP message containing an User-user Information Element and a Facility Information Element UserUserService request invoke Service 1 required a DISCONNECT message is sent. To reject the requested User to User service a Facility Information element is present set to UserUserService return error rejectedByNetwork and the Cause Information Element set 69 (requested facility not implemented)			
DSS1 Parameter values SETUP User-user Protocol discriminator User information Facility UserUserService request invoke Service 1 required DISCONNECT Facility UserUserService return error rejectedByNetwork Cause Value 69 (requested facility not implemented)			

End device			Test equipment
SETUP	→		
DISCONNECT	←		
RELEASE	→		
RELEASE COMPLETE	←		

TSS UUS	TP_510_010	Reference subclause 5.2.10.1.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/14

Test purpose

Call setup with user-to-user request service 2 explicit request preferred rejected in CALL PROCEEDING

Ensure that on receipt of a SETUP message containing a Facility Information Element UserUserService request invoke Service 2 preferred an INVITE request is sent. Upon receipt of a 183 (Session Progress) provisional response a CALL PROCEEDING message is sent. To reject the requested User to User service a Facility Information element is present set to UserUserService return error rejectedByNetwork

DSS1 Parameter values

SETUP
Facility
UserUserService request invoke
Service 2 preferred

CALL PROCEEDING
Facility
UserUserService return error
rejectedByNetwork

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
CALL PROCEEDING	←	←	183 (Session Progress)

Apply post test routine

TSS UUS	TP_510_011	Reference subclause 5.2.10.1.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/14

Test purpose

Call setup with user-to-user request service 2 explicit request preferred rejected in ALERTING

Ensure that on receipt of a SETUP message containing a Facility Information Element UserUserService request invoke Service 2 preferred an INVITE request is sent. Upon receipt of a 180 (Ringing) provisional response an ALERTING message is sent. To reject the requested User to User service a Facility Information element is present set to UserUserService return error rejectedByNetwork

DSS1 Parameter values

SETUP
Facility
UserUserService request invoke
Service 2 preferred

ALERTING
Facility
UserUserService return error
rejectedByNetwork

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 Ringing
Apply post test routine			

TSS UUS	TP_510_012	Reference subclause 5.2.10.1.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/14

Test purpose

Call setup with user-to-user request service 2 explicit request preferred rejected in CONNECT

Ensure that on receipt of a SETUP message containing a Facility Information Element UserUserService request invoke Service 2 preferred an INVITE request is sent. Upon receipt of a 200 OK INVITE final response a CONNECT message is sent. To reject the requested User to User service a Facility Information element is present set to UserUserService return error rejectedByNetwork.

DSS1 Parameter values

SETUP
  Facility
    UserUserService request invoke
      Service 2 preferred

CONNECT
  Facility
    UserUserService return error
      rejectedByNetwork

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
CONNECT	←	←	200 OK INVITE
		→	ACK
Apply post test routine			

TSS UUS	TP_510_013	Reference subclause 5.2.10.1.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/14

Test purpose

Call setup with user-to-user request service 2 explicit request required rejected in DISCONNECT

Ensure that on receipt of a SETUP message containing a user-user Information Element and a Facility Information Element UserUserService request invoke Service 1 required, a DISCONNECT message is sent. To reject the requested User to User service a Facility Information element is present and is set to UserUserService return error rejectedByNetwork and the Cause Information Element set 69 (requested facility not implemented).

DSS1 Parameter values
SETUP
        Facility
           UserUserService request invoke
              Service 2 required
 	
DISCONNECT
        Facility
           UserUserService return error
              rejectedByNetwork
        Cause
           Value 69 (requested facility not implemented)

End device			Test equipment
SETUP	→		
DISCONNECT	←		
RELEASE	→		
RELEASE COMPLETE	←		

TSS UUS	TP_510_014	Reference subclause 5.2.10.1.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/15

Test purpose *Call setup with user-to-user request service 3 explicit request preferred rejected in CALL PROCEEDING* Ensure that on receipt of a SETUP message containing a Facility Information Element UserUserService request invoke Service 3 preferred an INVITE request is sent. Upon receipt of a 183 (Session Progress) provisional response a CALL PROCEEDING message is sent. To reject the requested User to User service a Facility Information element is present set to UserUserService return error rejectedByNetwork

DSS1 Parameter values
SETUP
        Facility
           UserUserService request invoke
              Service 3 preferred
 	
CALL PROCEEDING
        Facility
           UserUserService return error
              rejectedByNetwork

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
CALL PROCEEDING	←	←	183 (Session Progress)
	Apply post test routine		

TSS UUS	TP_510_015	Reference 5.2.10.1.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/15

Test purpose

Call setup with user-to-user request service 3 explicit request preferred rejected in ALERTING

Ensure that on receipt of a SETUP message containing a Facility Information Element UserUserService request invoke Service 3 preferred an INVITE request is sent. Upon receipt of a 180 (Ringing) provisional response an ALERTING message is sent. To reject the requested User to User service a Facility Information element is present and set to UserUserService return error rejectedByNetwork

DSS1 Parameter values

SETUP
- Facility
  - UserUserService request invoke
    - Service 3 preferred

ALERTING
- Facility
  - UserUserService return error
    - rejectedByNetwork

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
ALERTING	←	←	180 Ringing

Apply post test routine

TSS UUS	TP_510_016	Reference 5.2.10.1.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/15

Test purpose

Call setup with user-to-user request service 3 explicit request preferred rejected in CONNECT

Ensure that on receipt of a SETUP message containing a Facility Information Element UserUserService request invoke Service 3 preferred an INVITE request is sent. Upon receipt of a 200 OK INVITE final response a CONNECT message is sent. To reject the requested user to user service a facility information element is present and set to UserUserService return error rejectedByNetwork

DSS1 Parameter values

SETUP
- Facility
  - UserUserService request invoke
    - Service 3 preferred

CONNECT
- Facility
  - UserUserService return error
    - rejectedByNetwork

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
CONNECT	←	←	200 OK INVITE
		→	ACK
Apply post test routine			

TSS UUS	TP_510_017	Reference subclause 5.2.10.1.3 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/15

Test purpose

Call setup with user-to-user request service 3 explicit request required rejected in DISCONNECT

Ensure that on receipt of a SETUP message containing a Facility Information Element UserUserService request invoke Service 3 required a DISCONNECT message is sent. To reject the requested user to user service a facility information element is present, it is set to UserUserService return error rejectedByNetwork and the Cause Information Element set to 69 (requested facility not implemented).

DSS1 Parameter values

SETUP
    Facility
        UserUserService request invoke
            Service 3 required

DISCONNECT
    Facility
        UserUserService return error
            rejectedByNetwork
    Cause
        Value 69 (requested facility not implemented)

End device			Test equipment
SETUP	→		
DISCONNECT	←		
RELEASE	→		
RELEASE COMPLETE	←		

TSS UUS	TP_510_018	Reference subclause 5.2.10.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/13

Test purpose

Call setup INVITE contains a user-to-user header response in ALERTING

An INVITE request containing a user-to-user header was received. Ensure that on receipt of an ALERTING message where a user-user information element is present, the Protocol discriminator and user information is mapped into the uuidata parameter of the user-to-user header in the sent 180 Ringing.

SIP header values
INVITE
    User-to-User
        uuidata
 	
180
    User-to-User
        uuidata

DSS1 Parameter values
SETUP
        User-user
            Protocol discriminator
            User information
 	
ALERTING
        User-user
            Protocol discriminator
            User information

Message flow

Test equipment			End device
INVITE1	→	→	SETUP
180 Ringing	←	←	ALERTING

Apply post test routine

TSS UUS	TP_510_019	Reference subclause 5.2.10.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/13

Test purpose *Call setup INVITE contains a user-to-user header response in CONNECT* An INVITE request containing a user-to-user header was received. Ensure that on receipt of a CONNECT message where a user-user information element is present, the Protocol discriminator and user information is mapped into the uuidata parameter of the user-to-user header in the sent 200 OK INVITE.

SIP header values
INVITE
    User-to-User
        uuidata
 	
200
    User-to-User
        uuidata

DSS1 Parameter values
SETUP
        User-user
            Protocol discriminator
            User information
 	
CONNECT
        User-user
            Protocol discriminator
            User information

Message flow			
Test equipment			**End device**
INVITE1	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→		
Apply post test routine			

TSS UUS	TP_510_020	Reference 5.2.10.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/13
Test purpose *Call setup INVITE contains a User-to-User header response in RELEASE in confirmed dialogue* An INVITE request containing a user-to-user header was received. Ensure that on receipt of a RELEASE message in confirmed dialogue where a user-user information element is present, the Protocol discriminator and user information is mapped into the uuidata parameter of the user-to-user header in the sent BYE request.			
SIP header values INVITE User-to-User uuidata BYE User-to-User uuidata			
DSS1 Parameter values SETUP User-user Protocol discriminator User information RELEASE User-user Protocol discriminator User information			

Message flow			
Test equipment			**End device**
INVITE1	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→		
BYE	←	←	RELEASE
200 OK BYE	→	→	RELEASE COMPLETE

TSS UUS	TP_510_021	Reference subclause 5.2.10.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/13
Test purpose *Call setup INVITE contains a user-to-user header response in RELEASE COMPLETE in confirmed dialogue* An INVITE request containing a user-to-user header was received. Ensure that on receipt of a RELEASE COMPLETE message in confirmed dialogue where a user-user information element is present, the Protocol discriminator and user information is mapped into the uuidata parameter of the user-to-user header in the sent BYE request.			
SIP header values INVITE User-to-User uuidata BYE User-to-User uuidata			
DSS1 Parameter values SETUP User-user Protocol discriminator User information RELEASE User-user Protocol discriminator User information			
Message flow			
Test equipment			**End device**
INVITE1	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→		
BYE	←	←	RELEASE COMPLETE
200 OK BYE	→		

TSS UUS	TP_510_022	Reference subclause 5.2.10.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/13
Test purpose *Call setup INVITE contains a user-to-user header response in RELEASE COMPLETE* An INVITE request containing a user-to-user header was received. Ensure that on receipt of a RELEASE COMPLETE message in confirmed dialogue where a user-user information element is present, the Protocol discriminator and user information is mapped into the uuidata parameter of the user-to-user header in the sent BYE request.			

SIP header values INVITE User-to-User uuidata 4xx User-to-User uuidata
DSS1 Parameter values SETUP User-user Protocol discriminator User information RELEASE COMPLETE User-user Protocol discriminator User information

Message flow

Test equipment			End device
INVITE1	→	→	SETUP
4xx	←	←	RELEASE COMPLETE
ACK	→		

7.2.5.11 Subaddressing (SUB)

TSS SUB	TP_511_001	Reference 5.2.8.1: Table: 5.2.8.1-1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/16
Test purpose *Mapping of Called party sub-address into INVITE* Ensure that on receipt of a SETUP message and if a Called party sub-address is present, an INVITE request is sent and the To header contains an 'isub' parameter set to the 'Subaddress information' parameter as received in the SETUP.			
SIP header values INVITE: To: <any URI>; isub= Subaddress information; isub-encoding=nsap-ia5			
DSS1 Parameter values SETUP: Called party sub-address Type of subaddress = NSAP Subaddress information			

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE1

Apply post test routine

TSS SUB	TP_511_002	Reference subclause 5.2.8.1 and Table 5.2.8.1-1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/16 AND 5.1.3/2

Test purpose

Mapping of Calling party sub-address into INVITE

Ensure that on receipt of a SETUP message where a Calling party sub-address is present, an INVITE request is sent and the From header contains an 'isub' parameter set to the 'Subaddress information' parameter as received in the SETUP.

SIP header values

INVITE:

From: <any URI>; isub= Subaddress information; isub-encoding=nsap-ia5

or

P-Preferred-Identity: <any URI>; isub= Subaddress information; isub-encoding=nsap-ia5

DSS1 Parameter values

SETUP:

Calling party subaddress

Type of subaddress = NSAP

Subaddress information

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE1

Apply post test routine

TSS SUB	TP_511_003	Reference 5.2.8.1: Table: 5.2.8.1-1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/16 AND 5.1.3/1

Test purpose

Mapping of Calling party sub-address into INVITE

Ensure that on receipt of a SETUP message and a Calling party sub-address is present, an INVITE request is sent and the From header contains an 'isub' parameter set to the 'Subaddress information' parameter as received in the SETUP.

SIP header values

INVITE:

P-Asserted-Identity: <any URI>; isub= Subaddress information; isub-encoding=nsap-ia5

DSS1 Parameter values

SETUP:

Calling party subaddress

Type of subaddress = NSAP

Subaddress information

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE1

Apply post test routine

TSS SUB	TP_511_004	Reference subclause 5.2.8.1 and Table 5.2.8.1-2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/16

Test purpose

Mapping of Connected party sub-address from 200 OK

Ensure that on receipt of a 200 OK INVITE response containing an isub parameter in the P-Asserted-Identiy header, a CONNECT message is sent to the DSS1 user equipment and a Connected subaddress. The Type of subaddress is set according to variable ToSUB_VA in Table 7.2.5.11-1 and the subaddress information is set to isub value received in the 200 OK INVITE.

SIP header values

UPDATE:

P-Asserted-Identity: <any URI>; isub= Subaddress information;isub-encoding=**ToSUB_VA**

DSS1 Parameter values

CONNECT:

Connected subaddress

Type of subaddress = NSAP

Subaddress information

End device			Test equipment
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE1
ALERTING	←	←	180 Ringing
CONNECT	←	←	200 OK (INVITE)
		→	ACK

Apply post test routine

Table 7.2.5.11-1 – Mapping of isub-encoding value into Type of Subaddress

ToSUB_VA	isub-encoding	Type of Subaddress
ToSUB_VA_01	isub-encoding not present	"NSAP" (000)
ToSUB_VA_02	"isub-encoding=nsap-ia5"	"NSAP" (000)
ToSUB_VA_03	"isub-encoding=nsap-bcd"	"NSAP" (000)
ToSUB_VA_04	"isub-encoding=nsap"	"NSAP" (000)

TSS SUB	TP_511_005	Reference 5.2.8.2: Table 5.2.8.2-1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/16

Test purpose

Mapping of Called party subaddress into SETUP

Ensure that on receipt of an INVITE request containing a isup parameter in the To header, a SETUP message is sent and a Called party subaddress is present. The type of subaddress is set according to variable ToSUB_VA indicated in Table 7.2.5.11-2 and the subaddress information is set to isub value.

SIP header values

INVITE:

To: <any URI>; isub= isub value; **isub-encoding**= ToSUB_VA

DSS1 Parameter values SETUP: Called party sub-address **Type of subaddress** = ToSUB_VA Subaddress information = isub value
Message flow **Test equipment** **End device** INVITE1 ➔ ➔ SETUP **Apply post test routine**

TSS SUB	TP_511_006	Reference subclause 5.2.8.2 and Table 5.2.8.2-1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/16
Test purpose *Mapping of calling party subaddress into SETUP* Ensure that on receipt of an INVITE request containing a isup parameter in the To header, a SETUP message is sent and a Called party subaddress is present.The Type of subaddress is set according to variable ToSUB_VA indicated in Table 7.2.5.11-2 and the subaddress information is set to isub value.			
SIP header values INVITE: P-Asserted-Identiy: <any URI>; isub= isub value; **isub-encoding**= ToSUB_VA			
DSS1 Parameter values SETUP: Calling party subaddress **Type of subaddress** = ToSUB_VA Subaddress information			
Message flow **Test equipment** **End device** INVITE1 ➔ ➔ SETUP **Apply post test routine**			

Table 7.2.5.11-2 – Mapping of isub-encoding value into Type of Subaddress

ToSUB_VA	isub-encoding	Type of Subaddress
ToSUB_VA_01	isub-encoding not present	"NSAP" (000)
ToSUB_VA_02	"isub-encoding=nsap-ia5"	"NSAP" (000)
ToSUB_VA_03	"isub-encoding=nsap-bcd"	"NSAP" (000)
ToSUB_VA_04	"isub-encoding=nsap"	"NSAP" (000)

<table>
<tr><td>TSS
SUB</td><td>TP_511_007</td><td>Reference
subclause 5.2.8.2 and
Table 5.2.8.2-1 of
[ETSI TS 183 036]</td><td>Selection expression
PICS 5.1.1/2 AND 5.4/16</td></tr>
<tr><td colspan="4">Test purpose
Mapping of Connected subaddress from CONNECT

Ensure that on receipt of a CONNECT message containing a Connected subaddress and the Type of subaddress is set to NSAP, a 200 OK INVITE is sent. An isub parameter is present in the P-Asserted-Identity header set to the value of the Subaddress information of the received Connected subaddress. The isub-encoding parameter is set to nsap-ia5</td></tr>
<tr><td colspan="4">SIP header values
200 OK INVITE:
P-Asserted-Identity: <any URI>; isub= isub value;isub-encoding=nsap-ia5</td></tr>
<tr><td colspan="4">DSS1 Parameter values
CONNECT:
Connected subaddress
Type of subaddress = NSAP
Subaddress information</td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><td>Test equipment</td><td></td><td></td><td>End device</td></tr>
<tr><td>INVITE1</td><td>→</td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td>←</td><td>ALERTING</td></tr>
<tr><td>200 OK (INVITE)</td><td>←</td><td>←</td><td>CONNECT</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td></tr>
</table>
Apply post test routine</td></tr>
</table>

7.2.5.12 Malicious communication identification

7.2.5.12.1 Test purposes for POTS

<table>
<tr><td>TSS
MCID</td><td>TP_512_101</td><td>Reference
subclause C.11.4 of
[ETSI TS 183 043]</td><td>Selection expression
PICS 5.1.1/1 AND 5.3/15</td></tr>
<tr><td colspan="4">Test purpose
The user requires the register of details of the last incoming call in a special record

Ensure that the SUT is able to send an INVITE request after a session was terminated and a flash hook event was detected. The Request-Line of the INVITE request contains the special service code command dialled by the user.</td></tr>
<tr><td colspan="4">SIP header values
INVITE2:
Request-Line: Public user identity of SUT@pes.operator.com</td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><td>Test equipment</td><td></td><td>End device</td></tr>
<tr><td>INVITE1</td><td>→</td><td>Ringing</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td>Off hook</td></tr>
<tr><td>ACK</td><td>→</td><td></td></tr>
<tr><td>CASE A</td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td></tr>
</table></td></tr>
</table>

CASE B		
BYE	←	On hook
200 OK BYE	→	
		Off hook
		Play a dial tone
		Dial special service code command
INVITE	→	
407 Proxy Authentication Required	←	
ACK	→	
INVITE2	←	
200 OK INVITE	→	
ACK	←	
	Apply post test routine	

TSS MCID	TP_512_102	Reference subclause B.4.2.2.2.4 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/15
Test purpose *The user requires the register of details of the last incoming call in a special record* Ensure that the SUT is able to send an INVITE while a session is active and a flash hook event was detected. If the malicious communication identification (MCID) service is provisioned and requires initial flash-hook detection then the Feature Manager on receipt of flash-hook notification from the MGC component shall follow one of the following options: • assume direct invocation of MCID The SUT send a re INVITE request towards the AS. The re INVITE request is built as follows: • the Request URI is set to the served user's identity, and • include no Body in the re-INVITE			
SIP header values INVITE2 (initial party) Request-Line: Public user identity of remote user SDP a=sendonly 200 OK 2 SDP a=recvonly INVITE3: Request-Line: Public user identity of SUT			

Message flow		
Test equipment		**End device**
INVITE1	→	Ringing
180 Ringing	←	
200 OK INVITE	←	Off hook
ACK	→	
		flash hook
INVITE2	←	
200 OK INVITE	→	
ACK	←	
INVITE3	←	
200 OK INVITE	→	
ACK	←	
Apply post test routine		

TSS MCID	TP_512_103	Reference subclause B.4.2.2.2 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/15
Test purpose *The user requires the register of details of the last incoming call in a special record* Ensure that the SUT is able to send an INVITE while a session is active and a flash hook event was detected. If the collected feature code indicates MCID, the Feature Manager shall request the SIP UA to send a re INVITE request towards the AS. The re INVITE request is built as follows: • the Request URI is set to the served user's identity, and • include no Body in the re-INVITE			
SIP header values INVITE2 (initial party) Request-Line: Public user identity of remote user SDP a=sendonly 200 OK 2 SDP a=recvonly INVITE3: Request-Line: Public user identity of SUT			

Message flow		
Test equipment		**End device**
INVITE1	→	Ringing
180 Ringing	←	
200 OK INVITE	←	Off hook
ACK	→	
		flash hook
		Play a dial tone
		Dial special service code command
INVITE2	←	
200 OK INVITE	→	
ACK	←	
INVITE3	←	
200 OK INVITE	→	
ACK	←	
Apply post test routine		

TSS MCID	TP_512_104	Reference B.4.2.2.2.4 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/15 AND 5.3/17
Test purpose *The user requires the register of details of the last incoming call in a special record, McidRequestIndicator is used* Ensure that the SUT is able to send an INVITE while a session is active and a flash hook event was detected. If the MCID service is provisioned and requires initial flash-hook detection then the Feature Manager on receipt of flash-hook notification from the MGC component shall follow one of the following options: • assume direct invocation of MCID The SUT send a re INVITE request towards the AS. The re INVITE request is built as follows: • the Request URI is set to the served user's identity, and • include no Body in the re-INVITE • the re-INVITE including a XML-MIME with XML mcid body with MCID XML Request schema containing a McidRequestIndicator set to 1.			
SIP header values INVITE2 (initial party) Request-Line: Public user identity of remote user SDP a=sendonly 200 OK 2 SDP a=recvonly INVITE3: Request-Line: Public user identity of SUT mcid request McidRequestIndicator = 1 HoldingIndicator=<any value>			

Message flow

Test equipment		End device
INVITE1	→	Ringing
180 Ringing	←	
200 OK INVITE	←	Off hook
ACK	→	
		flash hook
INVITE2	←	
200 OK INVITE	→	
ACK	←	
INVITE3	←	
200 OK INVITE	→	
ACK	←	

Apply post test routine

TSS MCID	TP_512_105	Reference subclause B.4.2.2.2 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/15 AND 5.3/17

Test purpose

The user requires the register of details of the last incoming call in a special record McidRequestIndicator is used

Ensure that the SUT is able to send an INVITE while a session is active and a flash hook event was detected. If the collected feature code indicates MCID, the Feature Manager shall request the SIP UA to send a re INVITE request towards the AS. The re INVITE request is built as follows:

- the Request URI is set to the served user's identity, and
- include no Body in the re-INVITE
- the re-INVITE including a XML-MIME with XML mcid body with MCID XML Request schema containing a McidRequestIndicator set to 1.

SIP header values

INVITE2 (initial party)
Request-Line: Public user identity of remote user
SDP
a=sendonly

200 OK 2
SDP
a=recvonly

INVITE3:
Request-Line: Public user identity of SUT
mcid
    request
        McidRequestIndicator = 1
        HoldingIndicator=<any value>

Message flow

Test equipment		End device
INVITE1	→	Ringing
180 Ringing	←	
200 OK INVITE	←	Off hook
ACK	→	
		flash hook
		Play a dial tone
		Dial special service code command
INVITE2	←	
200 OK INVITE	→	
ACK	←	
INVITE3	←	
200 OK INVITE	→	
ACK	←	

Apply post test routine

7.2.512.2 Test purposes for ISDN

TSS MCID	TP_512_201	Reference 5.2.6.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/17 AND 5.4/18

Test purpose

MCID Invocation during the active state

Ensure that on receipt of a FACILITY message containing a Facility Information Element with a McidRequest invoke component, an INVITE request is sent. A MCID XML MIME body is present and the McidRequestIndicator element is set to 1

SIP header values

INVITE2:

MCID XML MIME body

<?xml version="1.0" encoding="utf-8"?>

mcid

request>

McidRequestIndicator>1</

HoldingIndicator>0</

DSS1 Parameter values

FACILITY: Facility

McidRequest invoke

Message flow

Test equipment			End device
INVITE1	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK			
INVITE2	←	←	FACILITY
200 OK (INVITE)	→	→	FACILITY
ACK	←		

Apply post test routine

TSS MCID	TP_512_202	Reference subclause 5.2.6.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/17 AND NOT 5.4/18
Test purpose *MCID Invocation during the active state* Ensure that on receipt of a FACILITY message containing a Facility Information Element with a McidRequest invoke component, an INVITE request is sent. A MCID XML mMIME body is present and the McidRequestIndicator element is set to 1			
SIP header values INVITE2:			
DSS1 Parameter values FACILITY: Facility McidRequest invoke			
Message flow			

Test equipment			End device
INVITE1	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK			
INVITE2	←	←	FACILITY
200 OK (INVITE)	→	→	FACILITY
ACK	←		

Apply post test routine

7.2.5.13 Message waiting indication (MWI)

7.2.5.13.1 Test purposes for POTS

TSS MWI	TP_513_101	Reference clause C.12.1 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/19
Test purpose *Update of message waiting information* Ensure that after an update of the information of waiting messages the user is indicated. On receipt of a NOTIFY request reporting the "message-summary" event, the SUT requests the following actions from the media gateway: • Modify the default dial tone. • Send a Message Waiting Indicator message using the ITU-T H.248 andisp/data signal.			
SIP header values NOTIFY Event: message-summary Subscription-State: active Content-Type: application/simple-message-summary MIME body: Messages-Waiting: yes Message-Account: sip:served_user@Server Voice-Message: 4/1 (2/0) Video-Message: 3/1 (1/0) Fax-Message: 2/1 (0/1)			

Message flow		
Test equipment		**End device**
NOTIFY	→	
200 OK (NOTIFY)	←	Display of waiting messages
Apply post test routine		

7.2.5.13.2 Test purposes for ISDN

TSS MWI	TP_513_201	Reference clause 5.2.17 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/19

Test purpose

Update of message waiting information

Ensure that after an update of the information of waiting messages the user is indicated. On receipt of a NOTIFY request reporting the "message-summary" event, the SUT requests the following actions from the media gateway:

- Modify the default dial tone.
- Send a facility information element: MWIIndicate invoke

SIP header values

NOTIFY

Event: message-summary

Subscription-State: active

Content-Type: application/simple-message-summary

MIME body:

Messages-Waiting: yes

Message-Account: sip:served_user@Server

Voice-Message: 4/1 (2/0)

Video-Message: 3/1 (1/0)

Fax-Message: 2/1 (0/1)

DSS1 Parameter values

FACILITY: Facility
MWIIndicate invoke
controllingUserProvidedNr= served_user
basicService
speech
telefaxGroup2-3
videotelephony

Message flow			
Test equipment			**End device**
The user has subscribed the MWI Service			
NOTIFY	→	→	FACILITY
200 OK (NOTIFY)	←		
Apply post test routine			

7.2.5.14 Completion of communications to busy subscriber (CCBS) and completion of communications by no reply (CCNR)

7.2.5.14.1 Test purposes for POTS

<table>
<tr><td>TSS
CCBS_CCNR</td><td>TP_514_101</td><td>Reference
clauses C.18.5 and C.18.6 of [ETSI TS 183 043]</td><td>Selection expression
PICS 5.1.1/1</td></tr>
<tr><td colspan="4">Test purpose
CC recall after INVITE was received

Ensure that the SUT is able to send an INVITE request and the Request line contained a 'm' parameter set to 'BS' or 'NR'</td></tr>
<tr><td colspan="4">SIP header values
INVITE
Request-Line URI <served user>; m=BS
or
URI <served user>; m=NR</td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><th>Test equipment</th><th></th><th>End device</th></tr>
<tr><td>INVITE1</td><td>→</td><td>Ringing</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td>Off hook</td></tr>
<tr><td>ACK</td><td>→</td><td></td></tr>
</table>
Apply post test routine</td></tr>
</table>

<table>
<tr><td>TSS
CCBS_CCNR</td><td>TP_514_102</td><td>Reference
subclause C.18.1 of [ETSI TS 183 043]</td><td>Selection expression
PICS 5.1.1/1</td></tr>
<tr><td colspan="4">Test purpose
CCBS invocation

Ensure that the SUT receives, in case a called user is busy and CCBS is possible (in the network), a 183 Session Progress and in succession an announcement with the prompt to activate the call completion procedure. After the activation procedure is completed, the call is terminated by receiving a 486 Busy Here.</td></tr>
<tr><td colspan="4">SIP header values</td></tr>
<tr><td colspan="4">Message flow
<table>
<tr><th>End device</th><th></th><th>Test equipment</th></tr>
<tr><td>Off hook</td><td></td><td></td></tr>
<tr><td>Dial number</td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td>←</td><td>407 Proxy Authentication Required</td></tr>
<tr><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td>←</td><td>183 Session Progress</td></tr>
<tr><td colspan="3">CCBS activation in-band</td></tr>
<tr><td></td><td>←</td><td>486 Busy Here</td></tr>
<tr><td></td><td>→</td><td>ACK</td></tr>
</table></td></tr>
</table>

TSS CCBS_CCNR	TP_514_103	Reference subclause C.18.1 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1

Test purpose

CCNR invocation

Ensure that the SUT receives in case a called user is not responding and CCNR is possible (in the network) a 183 Session Progress is received and in succession an announcement with the prompt to activate the call completion procedure. After the activation procedure is completed, the call is terminated by receiving a 199 Early Dialogue Terminated.

SIP header values

Message flow

End device		Test equipment
Off hook		
Dial number	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE
	←	183 Session Progress
	CCBS activation in-band	
	←	199 Early Dialogue Terminated

7.2.5.14.2 Test purposes for ISDN

TSS CCBS_CCNR	TP_514_201	Reference subclause 6.3.2.4 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/2

Test purpose

CC recall after INVITE was received

Ensure that SUT sends a SETUP to the ISDN user equipment after an INVITE request was received and the Request line contained a 'm' parameter set to 'BS' or 'NR'

SIP header values

INVITE

Request-Line URI <served user>; m=BS

or

URI <served user>; m=NR

DSS1 Parameter values

Message flow

Test equipment			End device
INVITE	→	→	SETUP
180 Ringing	←	←	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→		

Apply post test routine

TSS CCBS_CCNR	TP_514_202	Reference subclause 5.3.1.5.3 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/2
Test purpose *CCBS invocation* Ensure that the SUT receives, in case a called user is busy and CCBS is possible (in the network), a 183 Session Progress and in succession an announcement with the prompt to activate the call completion procedure. After the activation procedure is completed, the call is terminated by receiving a 486 Busy Here.			
SIP header values			
DSS1 Parameter values			
Message flow			

Test equipment			**Test equipment**
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
PROGRESS	←	←	183 Session Progress
	CCBS activation in-band		
RELEASE	←	←	486 Busy Here
RELEASE COMPLETE	→	→	ACK

TSS CCBS_CCNR	TP_514_203	Reference subclause 5.3.1.5.3 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/2
Test purpose *CCNR invocation* Ensure that the SUT receives, in case a called user is not responding and CCNR is possible (in the network), a 183 Session Progress and in succession an announcement with the prompt to activate the call completion procedure. After the activation procedure is completed, the call is terminated by receiving a 199 Early Dialogue Terminated.			
SIP header values			
DSS1 Parameter values			
Message flow			

Test equipment			**Test equipment**
SETUP	→	→	INVITE
		←	407 Proxy Authentication Required
		→	ACK
		→	INVITE
PROGRESS	←	←	183 Session Progress
	CCNR activation in-band		
RELEASE	←	←	199 Early Dialogue Terminated
RELEASE COMPLETE	→		

SERIES OF ITU-T RECOMMENDATIONS

Series A	Organization of the work of ITU-T
Series D	Tariff and accounting principles and international telecommunication/ICT economic and policy issues
Series E	Overall network operation, telephone service, service operation and human factors
Series F	Non-telephone telecommunication services
Series G	Transmission systems and media, digital systems and networks
Series H	Audiovisual and multimedia systems
Series I	Integrated services digital network
Series J	Cable networks and transmission of television, sound programme and other multimedia signals
Series K	Protection against interference
Series L	Environment and ICTs, climate change, e-waste, energy efficiency; construction, installation and protection of cables and other elements of outside plant
Series M	Telecommunication management, including TMN and network maintenance
Series N	Maintenance: international sound programme and television transmission circuits
Series O	Specifications of measuring equipment
Series P	Telephone transmission quality, telephone installations, local line networks
Series Q	**Switching and signalling, and associated measurements and tests**
Series R	Telegraph transmission
Series S	Telegraph services terminal equipment
Series T	Terminals for telematic services
Series U	Telegraph switching
Series V	Data communication over the telephone network
Series X	Data networks, open system communications and security
Series Y	Global information infrastructure, Internet protocol aspects, next-generation networks, Internet of Things and smart cities
Series Z	Languages and general software aspects for telecommunication systems

Printed in Switzerland
Geneva, 2019