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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



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

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^{*} 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, <u>http://handle.itu.int/11.1002/1000/</u> <u>11830-en</u>.

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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), <i>Gateway control protocol: Version 3</i> .
[ITU-T Q.4014.1]	Recommendation ITU-T Q.4014.1 (2019), <i>PSTN/ISDN terminal</i> 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), <i>Telecommunications and</i> <i>Internet converged Services and Protocols for Advanced Networking</i> (TISPAN); ISDN/SIP interworking; Protocol specification.
[ETSI TS 183 043]	Recommendation ETSI TS 183 043 (2011), <i>Telecommunications and</i> <i>Internet converged Services and Protocols for Advanced Networking</i> (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

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
	ЗРТҮ	TP_505_xxx
	CUG	TP_506_xxx
	CW	TP_507_xxx
	ТР	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

Table 6-1 – Test suite structure

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	TP 101 001	Reference	Selection expression
Initial_registration		clause 8.1.3.5 and 22.2 of	1
		[IETF RFC 3261]	
Test purpose			
REGISTER request have	to be authorized receipt o	f 401	
		eceives a 401 Unauthorized and the V	
	1	The Authorization header is present	including 'username', 'realm',
Ţ.	esponse' HTTP parameters		
SIP header values			
401			
WWW-Authenticate:			
REGISTER 2	r 1)1r		
	me[proper value];realm=[proper value];nonce=[proper value];	aigest-uri=[proper value]
Message flow			
En	d device	Test eq	uipment
Make end de	vice available		
		\rightarrow REGISTER 1	
		← 401 Unauthorized	
		\rightarrow REGISTER 2	
		← 200 OK (REGISTER)	
	Арр	ly post test routine	

TSS Initial_registration	TP_101_002	Reference section 8.1.3.5 and 22.2 o	Selection expression
Initial_registration		[IETF RFC 3261]	1
Test purpose			·
REGISTER request have t	o be authorized, receipt	of 407	
Proxy-Authenticate heade	r is present. The SUT rej	receives a 407 Proxy Authentication peats the REGISTER request. The P st-uri' and 'response' HTTP paramet	roxy-Authorization header is
SIP header values			
407			
407			
407 Proxy-Authenticate:			
Proxy-Authenticate: REGISTER 2	username=[proper valu	e];realm=[proper value];nonce=[pr	oper value];digest-uri=[proper
Proxy-Authenticate: REGISTER 2 Proxy-Authorization:	username=[proper valu	e];realm=[proper value];nonce=[pr	oper value];digest-uri=[proper
Proxy-Authenticate: REGISTER 2 Proxy-Authorization: value] Message flow	username=[proper valu		oper value];digest-uri=[proper quipment
Proxy-Authenticate: REGISTER 2 Proxy-Authorization: value] Message flow	device		
Proxy-Authenticate: REGISTER 2 Proxy-Authorization: value] Message flow End	device		
Proxy-Authenticate: REGISTER 2 Proxy-Authorization: value] Message flow End	device	Test e	quipment
Proxy-Authenticate: REGISTER 2 Proxy-Authorization: value] Message flow End	device	→ REGISTER 1	quipment
Proxy-Authenticate: REGISTER 2 Proxy-Authorization: value] Message flow End	device	 → REGISTER 1 ← 407 Proxy Authenticat 	quipment

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

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)
Ар	ply post tes	t routine

TSS Initial_registration	TP_101_004		ce 10.2 and 22 of IFC 3261]	Selection expression		
Test purpose	Fest purpose					
Request line in the REGIS	TER request					
Ensure that the SUT sends Request-line.	s a REGISTER request to	its registrar and	there is no 'userpar	t' in the SIP-URI of the		
SIP header values						
REGISTER						
Request-line: sip:[host	part=domain of location s	ervice]				
Message flow						
End	l device		Test eo	quipment		
Make end dev	ice available					
		→ REC	SISTER 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	o the REGISTER request			
	res a 200 OK Final response to a ameter in the Contact header.	the REGISTER request sent to	its registrar and the Contact	
SIP header values	SIP header values			
200 OK (REGISTER)				
Contact: sip:[any regist	ered URI];expires=[any value]			
Message flow				
End	device	Test equ	iipment	
Make end devi	ce 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.				

SIP header values	
REGISTER	
Request-line: sip:[hostpart=preconfigured registrar ad	dress]
Message flow	
End device	Test equipment
Make end device available	
÷	REGISTER 1
•	• 401 Unauthorized
÷	REGISTER 2
•	• 200 OK (REGISTER)
Apply pos	t test routine

TSS Initial_registration	TP_101_007	Reference section 10.2.6 of [IETF RFC 3261]	Selection expression PICS 5.2/2		
Test purpose	·				
Successful registration usin	ng the 'multicast' mechanism				
Ensure that the SUT sends 'userpart'.	a REGISTER request to the "al	l SIP servers" multicast addres	ss "sip.mcast.net" without a		
SIP header values					
REGISTER					
Request-line: sip: <sip.< td=""><td>mcast.net></td><td></td><td></td></sip.<>	mcast.net>				
OR					
Request-line: sip: <224	.0.1.75>				
Message flow					
End	device	Test equ	ipment		
Make end device available					
	+	REGISTER			
	← 401 Unauthorized				
	→ REGISTER				
← 200 OK (REGISTER)					
Apply post test routine					

TSS Initial_registration	TP_101_008	Reference section 10.2 of [IETF RFC 3261]	Selection expression	
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'				
SIP header values				
REGISTER				
To: 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_009	Reference section 10.2 of	Selection expression		
		[IETF RFC 3261]			
Test purpose					
Successful registration Fro	om header				
Ensure that the SUT sends To header.	a REGISTER request to the reg	sistrar and the 'From' header is	set to the same value as the		
SIP header values					
REGISTER					
To: sip:[address of regi	strant]				
From: sip:[address of re	egistrant]				
Message flow	Message flow				
End	device	Test equ	upment		
Make end device availabl	e.				
	→	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 te	est 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				
	a REGISTER request to the r the Via header field when TC	egistrar and the 'Via' header co CP is used.	ontains a "rport" header field		
SIP header values	SIP header values				
REGISTER					
Via: SIP/2.0/UDP [IP A	Address]:[Port number];branch	n=z9hG4bK[branch];rport			
Message flow					
End	device	Test ec	quipment		
Make end device availabl	e.				
	-	→ 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 opt	ion-tag 'path' in the Supported	header			
Ensure that the SUT sends "path".	a REGISTER request to the re	gistrar and the 'Supported' head	der contains the option-tag		
SIP header values					
REGISTER					
Supported: path					
Message flow					
End	device	Test equ	uipment		
Make end device available					
		REGISTER			
	← 401 Unauthorized				
	-	REGISTER			
← 200 OK (REGISTER)					
	Apply pos	t 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 opt	ion-tag 'gruu' in the Supported I	header			
Ensure that the SUT sends "gruu" if the SUT supports	a REGISTER request to the reg GRUU.	sistrar and the 'Supported' head	ler contains the option-tag		
SIP header values					
REGISTER					
Supported: gruu					
Message flow					
End	device	Test equ	iipment		
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 opt	ion-tag 'gin' in the Require he	pader			
	a REGISTER request to the r the functions of an external at	egistrar and the 'Require' header tached network.	r contains the option-tag		
SIP header values					
REGISTER					
Require: gin					
Message flow					
End	device	Test eq	uipment		
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		I			
Successful registration. No	new registration before succes	sful final response			
	not send a new REGISTER request or before expiration of the re		ccessful final response to its		
SIP header values					
REGISTER					
Message flow					
End	device	Test equ	iipment		
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 alread	ly registered
→	REGISTER 1
÷	401 Unauthorized
→	REGISTER 2
(200 OK (REGISTER)
Apply post t	test routine

TSS	TP_101_018	Reference	Selection expression
Initial_registration		section 10.2.4 of [IETF	-
		RFC 3261]	
Test purpose			
Successful registration. Re	freshing of registration		
	ER final response. The C	In the expires parameter of the Cont Call-ID is the same as in the previou	
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 e	quipment
	SUT i	is already registered	
		→ REGISTER	
		← 401 Unauthorized	
		\rightarrow REGISTER 2	
		← 200 OK (REGISTER)	
	Арр	ly post test routine	

TSS	TP_101_019	Reference	Selection expression
Initial_registration		section 10.2.2 of [IETF RFC 3261]	PICS 5.2/6
Test purpose			
Successful registration ren	noving a Binding		
Ensure that the SUT is abl parameter for that Binding		r Binding by sending a REGISTER s header is set to '0'.	R request and the expires
SIP header values			
REGISTER			
Contact: sip:[any regis	tered URI];expires=0		
OR			
Contact: *			
Expires: 0			
Message flow			
End	device	Test e	quipment
	SUT i	s already registered	
		→ REGISTER	
		← 401 Unauthorized	
		→ REGISTER	
		← 200 OK (REGISTER)	
	App	ly post test routine	

TSS Initial_registration	TP_101_020	Reference section 10.2.1 of [IETF RFC 3261]	Selection expression PICS 5.2/7	
Test purpose				
The SUT gets it registered	contacts			
Ensure that the SUT, in ord Contact header.	der to get its registered contacts,	sends a REGISTER request to	o its registrar without	
SIP header values				
REGISTER				
[without Contact header]				
Message flow				
End	device	Test equ	ipment	
Make end device availabl	le			
→ REGISTER				
← 200 OK (REGISTER)				
Apply post test routine				

TSS	TP_101_021	Reference	Selection expression
Initial_registration		Annex A and section 17.1.2.2 of [IETF RFC 3261]	PICS 5.2/1
Test purpose			
REGISTER request is re	epeated if no response is re	eceived	
Ensure that the SUT or expires, if an unreliable		GISTER request, repeats its request	after timer E set to T1 value
SIP header values			
Message flow			
Ε	nd device		equipment
		→ REGISTER	
	Start timer E		
То	omeout timer E		
		→ REGISTER	
	Apj	ply post test routine	
TSS	TP_101_022	Reference	Selection expression
Initial_registration		Annex A and section 17.1.3 of	PICS 5.2/1
		[IETF RFC 3261]	
Test purpose			
	epeated if no response is re	eceived	
	aving sent a REGISTER rec spires, if an unreliable trans	quest twice, repeats its request after	timer E set to the
SIP header values	.p		
Message flow			
-	nd device	Test 6	equipment
		→ REGISTER	quipment
St	art timer E=T1		
Te	omeout timer E		
		→ REGISTER	
Sta	rt timer E=2*T1		
	omeout timer E		
Te			
Т		→ REGISTER	

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	
А	pply post te	st 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 repe	ated 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 te	st routine

TSS Initial_registration TP_101_025	Reference Annex A and section 17.1.3 of [IETF RFC 3261]	Selection expression
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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 tes	est routine

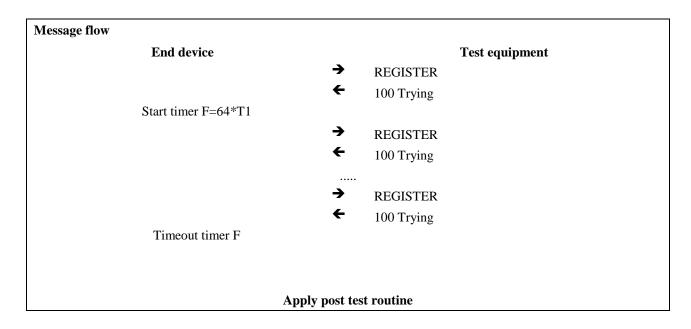
TSS Initial_registration	TP_101_026	A	Reference Annex A and ection 17.1.3 of [IETF RFC 3261]	Selection expression
Tost numasa		I	(FC 3201)	
Test purpose <i>RECISTER request in t</i>	he Proceeding state is rep	oated if no	rosponso is received	
KEOISTEK request in t	ne i roceeung siule is rep	euleu ij no i	response is received	
Ensure that the SUT, w	hen a REGISTER client tr	ansaction is	s in the Proceeding state	e, repeats its REGISTER request
after timer E set to T1 y	alue expires.			
SIP header values				
Message flow				
E	and device		Test	t equipment
		→	REGISTER	
		←	100 Trying	
St	art timer E=T1			
Τ	omeout timer E			
		→	REGISTER	
		←	100 Trying	
St	art timer E=T2			
Т	omeout timer E			
		→	REGISTER	
		ply post te		

TSS Initial_registration	TP_101_027	Reference Annex A and section 17.1.3 of [IETF RFC 3261]	Selection expression PICS 5.2/12
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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



TSS Initial_registration	TP_101_028	Reference subclause 5.1.1.2.1 of [ETSI TS124 229]	Selection expression
Test purpose <i>Receipt of a 305 (Use Prox</i>	y) response to the unprotected	REGISTER request	
	ceipt of a 305 (Use Proxy) resp address list, which is different f ration.		

SIP	header	values	

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	Selection expression
		[ETSI TS124 229]	

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	Test equipment	
	→ REGISTER	
	← 423 Interval Too Brief	
	→ REGISTER2	
	Apply post test routine	

TSS	TP_101_030	Reference	Selection expression
Initial_registration		subclause 5.1.1.2.1 of	-
		[ETSI TS124 229]	
Test purpose			
Receipt of a 408 (Requ	est Timeout) response to the	unprotected REGISTER request	
	, x		
Ensure that the SUT, or	n receipt of a 408 (Request T	imeout) response to the unprotect	ed REGISTER request attempt
to perform initial regist			
SIP header values			
Message flow			
F	End device	Test	equipment
		→ REGISTER	
		← 408 Request Timeout	
		→ REGISTER	
		REGISTER	

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					
attempt to perform initial re	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
A	pply post test routine

TSS	TP_101_033	Reference	Selection expression
Initial_registration		Annex A and	1
		section 17.1.3 of	
		[IETF RFC 3261]	
Test purpose			
Receipt of a 600 (Busy	Everywhere) response to th	e unprotected REGISTER reques	st
to perform initial regist		erywhere) response to the unprot	tected REGISTER request attempt
Message flow			
]	End device	Tes	st equipment
		→ REGISTER	
		← 600 Busy Everywh	ere
		→ REGISTER	
	Ap	ply post test routine	

TSS	TP_101_034	Reference	Selection expression
		Annex A and	_
		section 17.1.3 of [IETF	
		RFC 3261]	
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 te	est routine

TSS Initial_registration	TP_101_035	S	Reference ubclause 5.1.1.3 of ETSI TS 124 229]	Selection expression PICS VGW
Test purpose		Ľ		
	ration-state event package			
2				
	receipt of a 200 OK REGIST			' event package. The SUT
-	lest to the resource to which	the UE	wants to be subscribed to.	
SIPheader values				
SUBSCRIBE	high is the default public use	or idanti	ty used for subscription	
-	hich is the default public use r identity used for subscripti		ty used for subscription	
	lentity used for subscription			
Event: reg	v 1	-		
Expires: 600000				
200 OK SUBSCRIBE	00001			
Expires: [any value <=60	0000]			
NOTIFY				
Event: reg				
Subscription-State: active	; expires= <any value=""></any>			
Message flow				
En	d device	•	Test equ	ipment
		→ ∠	REGISTER	
		+	401 Unauthorized	
		→ ∠	REGISTER	
		÷	200 OK REGISTER	
		→	SUBSCRIBE	
CASE A		÷		
		÷	2xx OK SUBSCRIBE NOTIFY	
		` →		
		-	200 OK NOTIFY	
CASE B		→	SUBSCRIBE	
		←	407 Proxy Authentication	n Required
		→	SUBSCRIBE	
		←	2xx OK SUBSCRIBE	
		←	NOTIFY	
		→	200 OK NOTIFY	
	Annly	post te	st routine	

TSS Initial_registration	TP_101_036	Reference subclause 5.1.1.3 of [ETSI TS124 229]	Selection expression
Test purpose	·		·
Re-subscription to the re-	egistration-state event package		
	ter the registration state event is s The SUT sends a new SUBSCRI		
SIPheader values			
SUBSCRIBE			
-	which is the default public user id	entity used for subscription	
	ser identity used for subscription]		
Event: reg	identity used for subscription]		
Expires: 600000			
2			
200 OK SUBSCRIBE			
Expires: [any value <=6	500000]		
NOTIFY			
Event: reg			
Subscription-State : acti	ve; expires= <any value=""></any>		
Message flow			
8			
0	nd device	Test o	equipment
0		Test o ady registered	equipment
E		ady registered	equipment
-	SUT is alre	ady registered SUBSCRIBE	
E	SUT is alre	ady registered SUBSCRIBE 2xx OK SUBSCRIBE	
E	SUT is alre - •	ady registered SUBSCRIBE 2xx OK SUBSCRIBE NOTIFY	
E CASE A	SUT is alre - • •	ady registered SUBSCRIBE 2xx OK SUBSCRIBE NOTIFY 200 OK NOTIFY	
E	SUT is alre	ady registered SUBSCRIBE 2xx OK SUBSCRIBE NOTIFY 200 OK NOTIFY SUBSCRIBE	
E CASE A	SUT is alre - - - - -	ady registered SUBSCRIBE 2xx OK SUBSCRIBE NOTIFY 200 OK NOTIFY SUBSCRIBE 407 Proxy Authentica	
E CASE A	SUT is alre - - - - - - - - - - - - - - - - - - -	ady registered SUBSCRIBE 2xx OK SUBSCRIBE NOTIFY 200 OK NOTIFY SUBSCRIBE 407 Proxy Authentica SUBSCRIBE	tion Required
E CASE A	SUT is alre	ady registered SUBSCRIBE 2xx OK SUBSCRIBE NOTIFY 200 OK NOTIFY SUBSCRIBE 407 Proxy Authentica SUBSCRIBE 2xx OK SUBSCRIBE	tion Required
E CASE A	SUT is alre	ady registered SUBSCRIBE 2xx OK SUBSCRIBE NOTIFY 200 OK NOTIFY SUBSCRIBE 407 Proxy Authentica SUBSCRIBE 2xx OK SUBSCRIBE NOTIFY	tion Required

TSS Initial_registration	TP_101_037	Reference subclause 5.1.1.5A and 5.1.1.4.1of [ETSI TS124 229]	Selection expression	
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.				

SIPheader values	
NOTIFY	
Event: reg	
Subscription-State : active	
Contact: <sip:[registered address]="" contact="">;event</sip:[registered>	t= shortened;expires=[any value]
REGISTER	
Request-line: sip:[hostpart=domain of location	on 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
SUI	T is already registered
	← NOTIFY
	→ 200 OK NOTIFY
	→ REGISTER
	← 401 Unauthorized
	→ REGISTER
	← 200 OK (REGISTER)
۸r	pply post test routine

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

	6	e			
TSS User-initiated_re-registration	TP_102_001	Reference subclause 5.1.1.4.1 of [ETSI TS124 229]	Selection expression		
Test purpose					
User initiated reregistration					
Ensure that the SUT, after the S user identity. SIP: Header values	U1 is registered, starts i	ne reregistration procedure of	a previously registered public		
REGISTER 1					
Request-line: sip:[hostpart=	domain of location servi	ce]			
Call-ID [any value]					
Cseq x REGISTER					
From: sip:[address of registrant]					
To: sip:[address of registran	To: sip:[address of registrant]				
Expires: 600000					

Message flow	
End device	Test equipment
SUT is already	registered
The SUT has determined that a conti	nued registration is not required
→	REGISTER
+	401 Unauthorized
→	REGISTER
←	200 OK REGISTER
Apply post tes	st 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 – C	GRUU supported			
Ensure that the SUT, after the S user identity.	UT is registered, starts the	reregistration procedure of a	previously registered public	
SIPheader values				
REGISTER 1				
Supported: path,gruu				
Message flow				
End devi	ce	Test eq	uipment	
	SUT is alrea	ndy registered		
The SU	T has determined that a co	ontinued registration is not rec	quired	
	→	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 contin	nued registration is not required
→	REGISTER
(401 Unauthorized
→	REGISTER
÷	200 OK REGISTER
Apply post tes	t 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 - fu	unctions of an external atta	iched network is supported			
Ensure that the SUT, after the S user identity.	UT is registered, starts the	reregistration procedure of a p	previously registered public		
SIPheader values					
REGISTER 1					
Require: gin					
Message flow					
End devi	End device Test equipment				
SUT is already registered					
The SU	T has determined that a co	ontinued registration is not req	uired		
	→	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

М

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		
Ар	ply post tes	t 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 user identity when GRUU is su		leregistration procedure of a p	reviously registered public
SIPheader values			
REGISTER 1			
Supported: path,gruu			
Expires: 0			
Message flow			
End dev	vice	Test equ	upment
	SUT is alrea	dy registered	
T	he SUT has determined tha	at a deregistration is require	d
	→	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 sup	pported		
Ensure that the SUT, after the user identity when multiple re		leregistration procedure of a p	reviously registered public	
SIPheader values				
REGISTER 1				
Supported: path,outbound				
Expires: 0				
Message flow				
End dev	vice	Test equ	iipment	
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 atte	ached network is supported		
Ensure that the SUT, after the user identity when external att			reviously registered public	
SIPheader values				
REGISTER 1				
Require: gin				
Expires: 0				
Message flow				
End dev	vice	Test equ	upment	
	SUT is alrea	dy 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

TSS Network-initiated_deregistration	TP_104_001	Reference subclause 5.1.1.7 of	Selection expression
		[ETSI TS124 229]	
Test purpose			
Network-initiated deregistration			
Ensure that the SUT, upon receipt procedures.	of a NOTIFY to a su	bscribed registration state eve	nt, starts the deregistration
SIPheader values			
NOTIFY			
Event: reg			
Subscription-State: terminated			
Contact: <sip:[registered a<="" contact="" td=""><td>ddress]>;event= shor</td><td>tened;expires=[any value]</td><td></td></sip:[registered>	ddress]>;event= shor	tened;expires=[any value]	
REGISTER 1		1	
Request-line: sip:[hostpart=do:	main of location servi	cej	
Call-ID [any value] Cseq x REGISTER			
From: sip:[address of registran	t]		
To: sip:[address of registrant]	t]		
Expires: 0			
Message flow			
End device		Test	equipment
	SUT is all	ready registered	equipment
	501 15 an	← NOTIFY	
		$\Rightarrow 200 \text{ OK NOTIFY}$	
		200 OK NOTIF I	
		➡ REGISTER	
		→ REGISTER	
		✔ 200 OK (REGISTER)
		ost test routine	

7.2.1.5 Subscription and notification

TSS Subscription_and_notification	TP_105_001	Reference subclause 5.3.1of [ETSI TS183 043]	Selection expression PICS 5.3/1	
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.				

SIPheader values			
SUBCRIBE			
Event: ua-profile;profile-type=user ; vendor="[any reasonal value]";version="[any reasonable value]"	ble value]";model="[any reasonable		
NOTIFY			
Event: ua-profile;profile-type=user ; vendor="[any reasonal value]";version="[any reasonable value]"	ble value]";model="[any reasonable		
<dial-tone-management> <dial-tone-pattern>standard-dial-tone<dial-tone-pattern>special-condition-tonedial-tone-pattern>message-waiting-tone</dial-tone-pattern></dial-tone-pattern></dial-tone-management>	one-pattern> (optional)		
Message flow			
End device	Test equipment		
SUT is already registered			
+	SUBCRIBE		
← 200 OK (SUBCSRIBE)			
+	NOTIFY		
→	200 OK (NOTIFY)		
Apply post t	est routine		

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

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 alrea	ndy registered
÷	SUBCRIBE
←	200 OK (SUBCSRIBE)
÷	NOTIFY
÷	200 OK (NOTIFY)
Apply post	test routine

TSS	TP_105_003	Reference	Selection expression				
Subscription_and_notification		subclause 5.3.1 of	PICS 5.3/1				
		[ETSI TS183 043]					
Test purpose							
Subscription to the Message wa	iting indication rejected	d with 503					
	navailable) response to	a SUBSCRIBE request cont	we Waiting indication service. The taining a Retry-After header. The the Retry-After header contents				
SIPheader values							
SUBCRIBE							
Accept: application/simple-mes	sage-summary						
Expires: 							
Event: message-summary							
503							
Retry-After: 20							
Message flow							
End devi	ce	Test	t equipment				
	SUT is al	ready registered					
		→ SUBCRIBE					
		← 200 OK (SUBCSRII	BE)				
		← 503 Service Unavail	able				
Wait for the time indicated	in the Retry-After hea	nder					
		→ SUBCRIBE					
	← 200 OK (SUBCSRIBE)						
	Apply n	oost test routine	<i>`</i>				

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		
То		
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 tes	st routine

TSS	TP_201_002	Reference	Selection expression	
Orig_Establishment_of_an		section 8.1.1. of		
_early_dialogue		[IETF RFC 3261]		
Test purpose				
The Request line is set to the	same URI value of the To hea	ader		
	o establish a session and the R	Request line URI is set to the s	ame value as the To header	
URI				
SIP header values				
INVITE sip: [Request URI]	SIP/2.0			
To: < sip: [Request URI]	>			
Message flow				
End d	evice	Test equ	iipment	
Interworking POTS				
Off hook				
Dial number	→	INVITE		
	+	100 Trying		
ISDN interworking				
SETUP	+	INVITE		
SETUR	· +	INVIIL		
	τ.	100 Trying		
Apply post test routine				

TSS Orig_Establishment_of_an _early_dialogue	TP_201_003	Reference section 8.1.1.2 of [IETF RFC 3261]	Selection expression		
Test purpose					
The To header does not cont	tain a 'tag' parameter				
Ensure that the IUT is able to parameter	o establish a session and the 'I	To' header in the INVITE reque	est does not contain the 'tag'		
SIP header values					
INVITE					
To: <to header="" uri=""></to>					
Message flow					
End d	evice	Test equ	ipment		
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_004	Referen section 3261]	ce 8.1.1.3 of [IETF RFC	Selection expression		
Test purpose				·		
The From header contains a 't	ag' parameter					
Ensure that the IUT is able to a parameter	establish a session ar	nd the 'From	m' header in the INVITE re	equest contains the 'tag'		
SIP header values						
INVITE						
From: <to header="" uri="">;ta</to>	g=[any value]					
Message flow						
End dev	vice		Test equ	upment		
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_005	Reference section 8.1.1.5 of [IETF RFC 3261]		Selection expression					
Test purpose	Test purpose								
The CSeq header contains th	ne method INVITE								
Ensure that the IUT is able to is set to 'INVITE'.	o establish a session and	d the INV	ITE includes a CSeq heade	r and the method parameter					
SIP header values									
INVITE									
CSeq: <any number=""> IN</any>	VITE								
Message flow									
End d	evice		Test equ	ipment					
Interworking POTS									
Off hook									
Dial number		→	INVITE						
		←	100 Trying						
ISDN interworking									
SETUP		→	INVITE						
		←	100 Trying						
Apply post test routine									

TSS	TP_201_006	Reference		Selection expression
Orig_Establishment_of_an		section 8.1.		
_early_dialogue		[IETF RFC	3261]	
Test purpose				
Max-Forwards initial value				
Ensure that the IUT is able value '70'	to establish a session a	nd the INVIT	TE contains a Max-Forv	vards header set to the initial
SIP header values				
INVITE				
Max-Forwards: 70				
Message flow				
End d	evice		Test equ	ipment
Interworking POTS				
Off hook				
Dial number		→ I	NVITE	
		← 1	00 Trying	
ISDN interworking				
SETUP		→ I	NVITE	
		← 1	00 Trying	
	App	ly post test ro	outine	

TSS Orig_Establishment_of_an _early_dialogue	TP_201_007		ce 3.1.1.7 of FC 3261]	Selection expression					
Test purpose									
Coding of Via header in the	INVITE								
Ensure that the IUT is able to									
SIP, a protocol version set to	2.0 and a branch parar	neter set t	o a value beginning with "z	z9hG4bK".					
SIP header values									
INVITE									
Via:SIP 2.0 <transport></transport>	<ip address="">;branch=</ip>	z9hG4bK							
Message flow									
End d	evice		Test equ	ipment					
Interworking POTS									
Off hook									
Dial number		→	INVITE						
		←	100 Trying						
ISDN interworking									
SETUP		→	INVITE						
		←	100 Trying						
	App	ly post tes	st routine						

TSS Orig_Establishment_of_an _early_dialogue	TP_201_008		ce 13.2.1 of FC 3261]	Selection expression			
Test purpose		·					
Allow and Supported header	in the INVITE						
Ensure that the IUT is able to	o establish a session a	nd the INV	ITE contains the Allow a	and Supported header			
SIP header values							
INVITE							
Allow: <any extensions=""></any>							
Supported: <any method<="" td=""><td>S></td><td></td><td></td><td></td></any>	S>						
Message flow							
End d	evice		Test e	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 enter									
Ensure that the IUT is able to of a Trying (100 Trying) resp				the Calling state, on receipt					
SIP header values									
Message flow									
End d	evice		Test equ	iipment					
Interworking POTS									
Off hook									
Dial number		→	INVITE						
	← 100 Trying								
ISDN interworking									
SETUP		→	INVITE						
SETUP ACKNOWLEDGE + 100 Trying									
	Apply post test routine								

TSS	TP_201_010	Reference	ce	Selection expression
Orig_Establishment_of_an		sections	8, 8.1.3.2, 13.2.2.1	
_early_dialogue		and Figure 5 of [IETF RFC 3261]		
Test purpose				
183 received, the client enter	rs the Proceeding state			
	0			
Ensure that the IUT is able to	o establish a session and	d the INV	TE client transaction is in	the Calling state, on receipt
of a Session Progress (183 S				
SIP header values				
Message flow				
End device Test equipment				
Interworking POTS				
Off hook				
Dial number		→	INVITE	
		←	183 Session Progress	
			C	
ISDN interworking				
SETUP		→	INVITE	
CALL PROCEEDING		←	183 Session Progress	
Apply post test routine				

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	Test purpose						
198 received, the client enter	rs 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 d	evice		Test equ	ipment			
Interworking POTS							
Off hook							
Dial number		→	INVITE				
		←	198 Unknown				
ISDN interworking							
SETUP		→	INVITE				
		←					
SETUP ACKNOWLEDGE		T	198 Unknown				

TSS Orig_Establishment_of_an _early_dialogue	TP_201_013	Reference sections 8, 8.1.3.2, 13.2.2.1 and Figure 5 [IETF RFC 3261]	Selection expression		
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 Orig_Establishment_of_an _early_dialogue	TP_201_014	ReferenceSelectionsection 20.14 and 13.2.1 of[IETF RFC 3261]		Selection expression
Test purpose				· ·
Content-Length header and	SDP in the initial IN	VITE		
Ensure that the IUT is able to the size of the body in the m				he Content-Length header set to
SIP header values				
INVITE	_			
Content-Length: <any td="" va<=""><td>lue></td><td></td><td></td><td></td></any>	lue>			
SDP				
Message flow				
End d	evice		Tes	t 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 SL	<i>OP in the initial INVII</i>	E			
Ensure that the IUT is able to message that contains the set		nd the INVITE request contains the	he Content-Type header in the		
SIP header values					
INVITE	n/adn				
Content-Type: applicatio	n/sap				
SDP					
Message flow					
End d	evice	Tes	t equipment		
Interworking POTS					
Off hook					
Dial number		→ INVITE			
		← 100 Trying			
ISDN interworking					
SETUP		→ INVITE			
		← 100 Trying			
Apply post test routine					

TSS	TP_201_016	Reference	Selection expression					
Orig_Establishment_of_an _early_dialogue		section 20.14 and 13.2.1 of [IETF RFC 3261] PICS 5.1.2/2						
Test purpose								
Content-Length header and	SDP in the initial INVI	TE and TCP						
			ns the Content-Length header set to liable transport (TCP) is used.					
SIP header values								
INVITE								
Content-Type: applicatio	n/sdp							
(DD)								
SDP								
Message flow								
End d	evice	,	Fest 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_017	Reference section 8.1.3.5 and 22.2 of [IETF RFC 3261]Selection expression on			
Test purpose					
The CSeq in the repeated IN	VITE is incremented				
Ensure that the IUT is able to Unauthorized) response inclu header and with an increment	uding a WWW-Auther				
SIP header values					
INVITE1:					
CSeq: <value> IN</value>	IVITE				
INVITE2:					
Authorization:					
CSeq: <value +1=""></value>	INVITE				
Message flow					
End d	End device Test equipment				
Interworking POTS					
Off hook					
Dial number		→	INVITE 1		
		←	401 Unauthorized		
		→	ACK		
		→	INVITE 2		
ISDN interworking					
SETUP		→	INVITE 1		
		←	401 Unauthorized		
		→	ACK		
		→	INVITE 2		
Apply post test routine					

TSS Orig_Establishment_of_an _early_dialogue	TP_201_018	Reference section 8.1.3.5 and 22.2 of [IETF RFC 3261]	Selection expression		
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.					
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>

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 exp		Selection expression
Test purpose				
The CSeq in the repeated IN	VITE is incremented			
Ensure that the IUT is able to Required response including				
header and with an increment		fieader, re	pears its in vir E request v	viui ali Floxy-Autiorization
SIP header values	*			
INVITE1:				
CSeq: <value> I</value>	IVITE			
INVITE2:				
Proxy-Authentica				
CSeq: <value +1=""></value>	· INVITE			
Message flow				
End d	End deviceTest equipment			
Interworking POTS				
Off hook				
Dial number		→	INVITE	
		÷	407 Proxy Authentication	on Required
		→	ACK	
		→	INVITE2	
ISDN interworking				
SETUP		→	INVITE	
		←	407 Proxy Authentication	on Required
		→	ACK	
		→	INVITE2	
	Арр	ly post te	st routine	

TSS Orig_Establishment_of_an _early_dialogue	TP_201_020		ce 8.1.3.5 and 22.2 of RFC 3261]	Selection expression
Test purpose				
Proxy-Authorization header	present in the repeated	l INVITE		
Ensure that the IUT is able to Required response including header including proper value	a Proxy-Authenticate	header, re	peats its INVITE request v	
SIP header values				
INVITE2:				
			lue>", realm="= <any td="" valu<=""><td></td></any>	
value>",uri="= <a value>",algorithm</a 		cnonce="	= <any value="">",response="</any>	= <any< td=""></any<>
	= <ally value=""></ally>			
Message flow				
	End deviceTest equipment			uipment
Interworking POTS				
Off hook				
Dial number		→	INVITE	
		÷	407 Proxy Authentication	on Required
		→	ACK	
		→	INVITE2	
ISDN interworking				
SETUP		→	INVITE	
		←	407 Proxy Authentication	on Required
		→	ACK	
		→	INVITE2	
Apply post test routine				

TSS Orig_Establishment_of_an _early_dialogue	TP_201_021	Reference subclause 5.1.3.1 of [ETSI TS124 229]	Selection expression		
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".					
SIP header values INVITE2: Accept: < application/sdp >					

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

TSS Orig_Establishment_of_an _early_dialogue	TP_201_022		ce se 5.1.3.1 of [S124 229]	Selection expression PICS 5.2/14
Test purpose				
Precondition support in the	initial INVITE			
Ensure that the IUT is able to the SUT shall include the Su				on of support of resource reservation,
SIP header values				
INVITE2:				
Supported: precondition,	100rel			
SDP				
a=curr:qos local none a=curr:qos remote none a=des:qos mandatory/opt a=des:qos mandatory/opt		7		
Message flow	ional remote sendreev	/		
-			7	
End device		1	Sest equipment	
Interworking POTS Off hook				
Dial number		→	INVITE	
Diai number		÷		ntication Required
		÷	ACK	nucuuon required
		→	INVITE2	
ISDN interworking SETUP		→	INVITE	
SETUR		→ ←		ntication Required
		← →	ACK	
		→	INVITE2	
	Ani	ply post te		

TSS Orig_Establishmen _early_dialogue	it_of_an	TP_201_023	Reference subclause 5.1.3.1 of [ETSI TS124 229]	Selection expression PICS 5.2/14		
Test purpose						
Successful resource	e reservati	on procedure				
				reliable) 183 contains in the SDP the		
			he SUT side. The SUT indicates the			
_		note side confirms	the resource reservation in the 20	0 OK UPDATE.		
SIP header values	5					
INVITE2: Supported: pred	ondition 1	00rol				
Supported. pred	condition, i	00101				
SDP a=curr:qos l	ocal none					
a=curr:qos i		e				
a=des:qos n	nandatory/o	optional local send	lrecv			
a=des:qos n	nandatory/o	optional remote se	endrecv			
102						
183: Require: presendit	ion 100rol					
Require: precondit SDP a=curr:qos l						
a=curr:qos i		e				
-		optional local send	lrecv			
-	•	optional remote se				
a=conf:qos	•	-				
_						
UPDATE						
SDP a=curr:qos l						
a=curr:qos i						
_		optional local send				
a=des:qos n	handatory/o	optional remote se	endrecv			
200 OK UPDATE	200 OK LIPDATE					
SDP a=curr:qos l	ocal sendro	ecv				
a=curr:qos i						
	a=des:qos mandatory/optional local sendrecv					
a=des:qos n	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 reservat	tion procedure		
1	U	1 / 1	ial INVITE request in which the

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
A	Apply post te	st 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 t	est 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 pr	ovisional response _l	procedure				
present in the INVITE reque Upon a 18x provisional resp PRACK request containing t	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	-	<u>^</u>				
INVITE2:						
Supported: 100rel						
CSeq: 2 INVITE	CSeq: 2 INVITE					
183:						
Require: 100rel	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 te	st routine

TSS Orig_Establishment_of_an _early_dialogue	TP_201_027	Reference subclause 5.1.3.1 of [ETSI TS124 229]	Selection expression PICS5.2/16
		Section 6 and 8 of [IETF RFC 5009]	

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
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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
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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 te	st 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
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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	→	WICUIA
Media	-	
CASE A sendrecv		
Media	+	Media
Media	→	Media
Media	-	Media
ISDN interworking		
SETUP	→	INVITE
	←	407 Proxy Authentication Required
	→	ACK
	→	INVITE2
	+	183 Session Progress
CASE A sendonly		105 5055001 11051055
Media	←	Media
	÷	ти
Media	-	
CASE A sendrecv		
Media	←	Media
Media	→	Media
	Apply post te	

TSS Orig_Establishment_of_an _early_dialogue	TP_201_031	[ETSI T	ce se 5.1.3.1 of 'S124 229] 6 and 8 [IETF RFC 5009]	Selection expression PICS5.2/16
Test numeros		section		
Test purpose SUT does not present early r	nedia to the user 183 i	eceived r	no P-Farly-Media header n	resent
501 does not present early h	neuta to the user. 105 f	<i>cccivca</i> , <i>i</i>	to I Larry mean nearer p	resent.
Ensure that the IUT upon rec header field, the SUT does n media is not gated.				
SIP header values				
INVITE2:				
P-Early-Media: supported				
SDP offer				
102				
183: SDP answer yes/no				
Message flow			m	
End device Test equipment				
Interworking POTS				
Off hook		→		
Dial number		÷	INVITE	
			407 Proxy Authenticatio	n Required
		→	ACK	
			INVITE2	
		÷	183 Session Progress	
ISDN interworking				
SETUP		→	INVITE	
		←	407 Proxy Authenticatio	n Required
		→	ACK	
		→	INVITE2	
		←	183 Session Progress	
	Ann		st routine	
	App	iy posi te	si i vuille	

Orig_Establishment_of_an _early_dialogue [ET sect	rence Selection expression ause 5.1.3.1 of PICS5.2/16 I TS124 229] on 6 and 8 of F RFC 5009]
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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
	\rightarrow INVITE2
	← 183 Session Progress
ISDN interworking	
SETUP	→ INVITE
	 407 Proxy Authentication Required
	→ ACK
	 ACK INVITE2
	105 565500 11651655
A	pply 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
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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 te	st routine

TSS	TP_201_034	Referen	<u></u>	Selection expression		
Orig_Establishment_of_an	11_201_034		se 5.1.3.1 of	PICS5.2/16		
_early_dialogue		[ETSI TS124 229]		11055.2/10		
		section 6 and 8 of				
		[IETF R	FC 5009]			
Test purpose						
SUT local ring back tone is p	presented to the user. 1	83 receive	ed, P-Early-Media header r	not present.		
Ensure that the IUT upon rec	ceiving a 183 Session P	rogress pi	rovisional response a P-Ear	lv-Media header field is not		
received, the SUT play ringing						
not gated.	-	-	-	•		
SIP header values						
INVITE2:						
P-Early-Media: supported						
SDP offer						
102						
183:						
SDP answer yes/no						
Message flow						
End de	evice		Test equ	lipment		
Interworking POTS						
Off hook		_				
Dial number		→	INVITE			
		+	407 Proxy Authenticatio	n Required		
		→	ACK			
		→	INVITE2			
		←	183 Session Progress			
ISDN interworking						
SETUP		→	INVITE			
		÷	407 Proxy Authenticatio	n Required		
		→	ACK	1		
		→	INVITE2			
		÷	183 Session Progress			
	Apply post test routine					

TSS	TP_201_035	Referen	<u></u>	Selection expression			
Orig_Establishment_of_an	11_201_033		se 5.1.3.1 of	PICS5.2/16			
_early_dialogue			[S124 229]	11055.2/10			
8			6 and 8 of				
			RFC 5009]				
Test purpose							
SUT local ring back tone is	presented to the user	r. 183 rece	ived, P-Early-Media heade	r value sendonly or sendrecv			
present.							
Ensure that the IUT upon re	ceiving a 183 Session	Progress p	rovisional response a P-Ear	ly-Media header field is			
received value sendonly or s							
alerted if no media is received	ed from the network. I	Forward ea	rly 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				iipment			
Interworking POTS							
Off hook							
Dial number		→	INVITE				
		+	407 Proxy Authenticatio	n Required			
		→	ACK				
		→	INVITE2				
		←	183 Session Progress				
Local Ring back Tone		÷					
ISDN interworking							
SETUP		→	INVITE				
		÷	407 Proxy Authenticatio	n Required			
		→	ACK				
		→	INVITE2				
		÷	183 Session Progress				
Local Ring back Tone		÷	100 50051011 1051000				
Local Ming Dack Tolle	Apply post test routine						
	Ap	pry post te	st i vutine				

TSS Orig_Establishment_of_an _early_dialogue	TP_201_036	Reference subclause 5.1.3.1 of [ETSI TS124 229] section 6 and 8 of [IETF RFC 5009]	Selection expression PICS5.2/16
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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 recovnly or inactive, the SUT plays 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:			
P-Early-Media: [recvonly]/[inactive]			
SDP answer yes/no			
Message flow			
End device		Test equipment	
Interworking POTS		Test equipment	
Off hook			
Dial number	÷	INVITE	
	÷	407 Proxy Authentication Required	
	` →	ACK	
	→	INVITE2	
	+ +		
	+ +	183 Session Progress	
Local Ring back Tone	۲		
ISDN interworking			
SETUP	→	INVITE	
	←	407 Proxy Authentication Required	
	→	ACK	
	→	INVITE2	
	←	183 Session Progress	
Local Ring back Tone	÷	-	
~	Apply post te	st routine	

TSS Orig_Establishment_of_an _early_dialogue	TP_201_037		ce se 6.1.2 of 'S124 229]	Selection expression
Test purpose				
Handling of SDP offer				
Ensure that the IUT when se SDP offer shall reflect the ca				at least one media description. This s for the session.
SIP header values				
INVITE2:				
SDP offer				
Message flow				
End d	evice]	lest equipment
Interworking POTS				
Off hook				
Dial number		→	INVITE	
		÷	407 Proxy Authe	ntication 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		ce se 6.1.2 of [\$124 229]	Selection expression
Test purpose Handling of SDP answer				
Ensure that the IUT Upon ge Acceptable Here) response, from the SDP message bodic establishment attempt (i.e. a the SUT shall order the code the 488 (Not Acceptable Her	the SDP offer contains as of all 488 (Not Acce set of INVITE request cs in the SDP offer acc	a subset o ptable Her s used for	f the allowed media types, re) responses so far received the same session establish	codecs and other parameters d for the same session nent). For each media line,
SIP header values				
INVITE2:				
SDP offer1				
488				
488 SDP answer1				
INVITE3:				
SDP offer2 equal SDP answ	er1			
Message flow				
End device			Test equ	ipment
Interworking POTS				
Off hook				
Dial number		→	INVITE	
		←	407 Proxy Authenticatio	n Required
		→	ACK	
		→	INVITE2	
		←	488 (Not Acceptable He	re
		→	ACK	
		→	INVITE3	
ISDN interworking				
SETUP		→	INVITE	
52101		+ +	407 Proxy Authenticatio	n Required
		` →	ACK	n required
		→	INVITE2	
		+	488 Not Acceptable Her	e
		→	ACK	
		→	INVITE3	
	Ann		st routine	

TSS Orig_Establishment_of_an _early_dialogue	TP_201_039	Referen	nce	Selection expression
Test purpose				·
SUT terminates the forked n	ot confirmed dialogue	with a BY	E request.	
Ensure that the IUT when a sup the SIP session. An ACK remaining early dialogues.				
SIP header values				
INVITE2:				
P-Early-Media: supported				
SDP offer				
180 1:				
P-Early-Media: [recvonly]/[i	inactive]			
SDP answer yes/no	,			
Message flow				
End d	evice		Test equ	ipment
Interworking POTS				
Off hook				
Dial number		→	INVITE	
		←	407 Proxy Authenticatio	n Required
		→	ACK	
		→	INVITE2	
		←	180 Ringing1	
		←	180 Ringing2	
		←	200 OK (INVITE)2	
		→	ACK	
		→	BYE1	
		←	200 OK (BYE)	
ISDN interworking				
SETUP		→	INVITE	
		÷	407 Proxy Authenticatio	n Required
		→	ACK	
		→	INVITE2	
		÷	180 Ringing1	
		+	180 Ringing2	
		←	200 OK (INVITE)2	
		→	ACK	
		→	BYE1	
		←	200 OK (BYE)	
	App	ly post te	st routine	

TSS Orig_Establishment_of_an _early_dialogue	TP_201_040	s	Reference ection 8.1.1 of IETF RFC 3261]	Selection expression
Test purpose				
Retry of INVITE request after	er 503 with Retry-After	header re	eceived	
				al INVITE request containing a lindicated by the Retry-After
SIP header values				
Message flow				
End d	levice		Test	equipment
Interworking POTS				
Off hook				
Dial number		→	INVITE	
		÷	407 Proxy Authentic	cation Required
		→	ACK	
		→	INVITE	
		÷	503 Service Unavailable	
		→	ACK	
Wait for the time indicat	ed in the Retry-After	header		
		→	INVITE	
ISDN interworking				
SETUP		→	INVITE	
		←	407 Proxy Authentic	cation Required
		→	ACK	
		→	INVITE	
		←	503 Service Unavail	able
		→	ACK	
Wait for the time indicat	ed in the Retry-After	header		
		→	INVITE	
	Appl	ly post te	st 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-Id	entity or From header j	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	TP_201_042	Reference	Selection expression
Orig_Establishment_of_an	1P_201_042	subclause 6.3.2.4 of	Selection expression
_early_dialogue		[ETSI TS 183 043]	
Test purpose			
Modifying SDP in early dial	ogue		
		TE request to modify the SD	P 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 d	evice		Test equipment
Off hook			
Dial number		→ INVITE	
		← 407 Proxy Au	thentication Required
		→ ACK	
		→ INVITE	
		← 183/180	
		← UPDATE	
		→ 200 OK UPD	ATE
	Δ	pply 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-extended>		
<aoc-s></aoc-s>		
<charged-items></charged-items>		
<basic></basic>		
<price-time></price-time>		
<currency-amount>1<td></td><td></td></currency-amount>		
<charging-type>normal-ch</charging-type>	narging <td>g-type></td>	g-type>
<granularity></granularity>	•	
<time-unit>1<td></td><td></td></time-unit>		
<scale>ten-seconds<td>cale></td><td></td></scale>	cale>	
Message flow		
End device		Test equipment
Off hook		
Dial number	→	INVITE
	÷	407 Proxy Authentication Required
	→	ACK
	→	INVITE
	÷	18x
	→	PRACK
	÷	200 OK (PRACK)
	Apply post te	
	Trppiy post te	st i vuint

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
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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-extended>		
<aoc-d></aoc-d>		
<charging-info>subtotal</charging-info>		
< recorded-charges>		
<recorded-currency-units></recorded-currency-units>		
<currency-id>UNIT</currency-id>		
<currency-amount>2<td>nt></td><td></td></currency-amount>	nt>	
recorded-charges		
 d>illing-id> normal-charging		
aoc-d		
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)
	7	ACK
Update of charging info	←	INFO
	→	200 OK (INFO)
Apply	post te	st 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					

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-extended>			
< aoc-e >			
< recorded-charges>			
<recorded-currency-units></recorded-currency-units>			
<currency-id>UNIT</currency-id>			
<currency-amount>2<td>nt></td><td></td><td></td></currency-amount>	nt>		
recorded-charges			
<billing-id> normal-charging </billing-id>			
aoc-e			
Message flow			
End device		Test equipment	
A communica	tion is a	lready 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	TP_201_101	Reference	Selection expression
Orig_Establishment_of_an _early_dialogue		subclause 4.9.2.1 of [ETSI TS 124 229]	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 dig	it2
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	non
	INVITE2
+	404/484
*	
	ACK
Dial digit2	
	INVITE3
-	180
Apply post t	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		T	
		Test equipment	
Off hook		l est equipment	
		l est equipment	
Off hook	→	I est equipment	
Off hook	→ ←		
Off hook		INVITE1	
Off hook	←	INVITE1 407 Proxy Authentication Required	
Off hook	← →	INVITE1 407 Proxy Authentication Required ACK	
Off hook	← → →	INVITE1 407 Proxy Authentication Required ACK INVITE1 404/484	
Off hook Dial number1	← → → ←	INVITE1 407 Proxy Authentication Required ACK INVITE1	
Off hook	← → → ←	INVITE1 407 Proxy Authentication Required ACK INVITE1 404/484 ACK	
Off hook Dial number1	←	INVITE1 407 Proxy Authentication Required ACK INVITE1 404/484 ACK INVITE2	
Off hook Dial number1	$\begin{array}{c} \leftarrow \\ \rightarrow \\ \rightarrow \\ \leftarrow \\ \rightarrow \end{array}$	INVITE1 407 Proxy Authentication Required ACK INVITE1 404/484 ACK	
Off hook Dial number1 Dial digit1	←	INVITE1 407 Proxy Authentication Required ACK INVITE1 404/484 ACK INVITE2	
Off hook Dial number1	$\begin{array}{ccc} \bullet & \bullet & \bullet \\ \bullet & \bullet & \bullet & \bullet \\ \bullet & \bullet & \bullet &$	INVITE1 407 Proxy Authentication Required ACK INVITE1 404/484 ACK INVITE2 183 Session Progress	
Off hook Dial number1 Dial digit1	 + → + → + → + → 	INVITE1 407 Proxy Authentication Required ACK INVITE1 404/484 ACK INVITE2 183 Session Progress INFO	
Off hook Dial number1 Dial digit1	$\begin{array}{ccc} \bullet & \bullet & \bullet \\ \bullet & \bullet & \bullet & \bullet \\ \bullet & \bullet & \bullet &$	INVITE1 407 Proxy Authentication Required ACK INVITE1 404/484 ACK INVITE2 183 Session Progress	
Off hook Dial number1 Dial digit1 Dial digit2	$\begin{array}{cccc} \bullet & \bullet \\ \bullet & $	INVITE1 407 Proxy Authentication Required ACK INVITE1 404/484 ACK INVITE2 183 Session Progress INFO 200 OK INFO	
Off hook Dial number1 Dial digit1 Dial digit2 Ringing tone	+ + + + + + + + + + + + + + + + + + +	INVITE1 407 Proxy Authentication Required ACK INVITE1 404/484 ACK INVITE2 183 Session Progress INFO	

7.2.2.1.3 Test purposes for ISDN

TSS Orig_Establishment_of_an _early_dialogue	TP_201_201		ee e 5.3.1.5.3 of S 183 043]	Selection expression PICS 5.1.1/2 and 5.4/1		
Test purpose						
Indication of support of the I	PSTN XML schema					
Ensure that the SUT support application/vnd.etsi.pstn+xm			tial INVITE request contain	ns the		
SIP header values						
INVITE:						
Accept: application/vnd.etsi.	pstn+xml					
Message flow						
End d	evice	_	Test equ	ipment		
SETUP		→	INVITE			
		÷	407 Proxy Authentication	n Required		
		→	ACK			
	→ INVITE					
SETUP ACKNOWLEDGE	SETUP ACKNOWLEDGE ← 100 Trying					
	Appl	ly post tes	t routine			
TSS	TP_201_202	Reference	e	Selection expression		
Orig_Establishment_of_an			e D.1.1.2 of	PICS 5.1.1/2		
_early_dialogue		[ETSI T	S 183 036]			
Test purpose						
SETUP without called party	number IE received					
Ensure that on receipt of a Si 'speech' or 'audio 3 kBit/s' ar The AGCF/VGW sends a Si description is set to 8.	nd no called party numb	er Inform	ation Element is present, no	D INVITE request is sent.		
SIP header values						
Message flow						
End d	evice		Test equ	ipment		
SETUP		→	INVITE			
SETUP ACKNOWLEDGE		←	407 Proxy Authentication	n Required		
		→	ACK			
		→	INVITE			
SETUP ACKNOWLEDGE		←	100 Trying			
	Appl	ly post tes	t routine			

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

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

End device	Test equipment	
SETUP		
SETUP ACKNOWLEDGE		
INFO1		
INFO2		
INFO3	\rightarrow INVITE	
	← 407 Proxy Authentication Required	
	→ ACK	
	\rightarrow INVITE1	
	← 404 Not Found	
	→ ACK	
INFO4	→ INVITE2	
	← 404 Not Found	
	→ ACK	
INF05	→ INVITE3	
ALERTING	← 180 Ringing	
	Apply post test routine	

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

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

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

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 now	—
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

TSS	TP_201_206	Reference	Selection expression
Orig_Establishment_of_an		subclause 5.1.1.1 of	PICS 5.1.1/2 and 5.4/1
_early_dialogue		[ETSI TS 183 036]	
Test purpose			
INVITE is sent after receipt o	f "#" character in t	he called party number, XML ser	ndingCompleteIndication is contained
		call establishment procedures, d MS domain, the originating SUT	
The end of address is detemi complete indication.	ned by receipt of a	"#" character Sending complete	Information Element as a sending
 The INVITE contains the 	e XML sendingCom	pleteIndication element	
SIP header values			
INVITE:			
xml version="1.0" encodi</td <td>ng="utf-8"?></td> <td></td> <td></td>	ng="utf-8"?>		
<sendingcompleteindication< td=""><td>n /></td><td></td><td></td></sendingcompleteindication<>	n />		
DSS1 parameter values			
SETUP: Called Party numbe	er contained the "#"	character	
or			
Sending complete			
Message flow			
End d	evice		Test equipment
SETUP		→ INVITE	
SETUP ACKNOWLEDGE		← 407 Proxy Authe	entication Required
		→ ACK	1
		→ INVITE	
	Δ	apply post test routine	
TSS	TP_201_207	Reference	Selection expression
Orig_Establishment_of_an		subclause 5.1.1.1 of	PICS 5.1.1/2
_early_dialogue		[ETSI TS 183 036]	
Test purpose			
	complete called par	rty number is contained in the ca	ulled party number
~	-		
Ensure that if the SETUP me	essage contains the	complete called number informa	tion 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.

is being processed and that no further address information	r is expected.
SIP header values	
INVITE: Request URI Called party number digits	
To: Called party number digits; isub= Called paery subad	dress; isub-encoding=nsap-ia5
DSS1 Parameter values	
SETUP: Called party number: Complete number informa	tion
Called party subaddress	
Message flow	
End device	Test equipment
SETUP	INVITE
SETUP ACKNOWLEDGE	407 Proxy Authentication Required
-	ACK
-	INVITE
Apply pos	t test routine

TSS Orig_Establishment_of_an _early_dialogue	TP_201_208		ce e 5.1.1.1.2 of S 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1	
Test purpose					
One bearer capability inform	nation element is rec	eived. Mapp	ed into the XML Bea	rerCapability element	
	oility body in SIP. A	bearer capal		TUP message, shall be mapped to ment is received. The Information	
v		.2.1.4-1.			
SIP header values INVITE:					
<pre>////////////////////////////////////</pre>	ag_"utf 9"2				
PSTN	ig= uii-8 ?>				
BearerCapability					
BCoctet3					
CodingStandard>)0<				
•	ferCabability>ITC_v	value<			
< BCoctet4					
TransferMode>00	<				
InformationTrans	ferRate>10000<				
BCoctet5					
Layer1Identificati	Layer1Identification>01<				
UserInfoLayer1Protocol>00011<					
DSS1 parameter values					
SETUP:					
bearer capability Information	n transfer capability =	= ITC_value			
Message flow					
End d	evice		T	est equipment	
SETUP		→	INVITE		
SETUP ACKNOWLEDGE		+	407 Proxy Authen	tication Required	
		→	ACK	······································	
		→	INVITE		
Apply post test routine					
	A	ppry post tes	a routine		

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

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 capab	ility = Speech or 3.1 kHz		
audio			
Second Bearer Capability Information transfer cap	ability = 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		ce se 5.1.1.1.3 of (S 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1	
Test purpose					
Mapping of Progress Indica	tor				
				N XML ProgressIndicator value as	
	2. An additionally P	rogressIndica	tor element is is cor	ntained and the ProgressDescription	
value is set to '0000110'					
SIP header values					
INVITE:					
xml version="1.0" encodi</td <th>ng="utf-8"?></th> <td></td> <td></td> <td></td>	ng="utf-8"?>				
PSTN					
ProgressIndicator ProgressOctet	2				
	Standard>00<				
	on>yyyy<				
ProgressOctet	4				
	ProgressDescription>0000110<				
ProgressIndicator ProgressOctet	ProgressIndicator				
	CodingStandard>00<				
	on>0000<				
ProgressOctet4					
ProgressDescripti	ProgressDescription>PI_value<				
DSS1 parameter values					
SETUP:					
Progress Indicator Coding standard ='00', Location ='0000', Progress description='0000yyyy'					
Message flow	Message flow				
End device Test equipment					
SETUP					
SETUP ACKNOWLEDGE		←	407 Proxy Auther	ntication Required	
		→	АСК	1	
		nnly nost to			
	A	pply post te	si 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'

TSS Orig_Establishment_of_an _early_dialogue	TP_201_211		e e 5.1.1.1.3 of § 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1
Test purpose				
No Progress Indicator recei	ved			
Ensure that if no progress in ProgressDescription value is		d in the a Prog	ressIndicator eleme	ent is is contained and the
SIP header values				
INVITE:				
xml version="1.0" encodi</td <td>ng="utf-8"?></td> <td></td> <td></td> <td></td>	ng="utf-8"?>			
PSTN				
ProgressIndicator				
ProgressOctet				
	Standard>00< on>yyyy<			
ProgressOctet				
	sDescription>0000)110<		
ProgressIndicator	-			
ProgressOctet				
	Standard>00< n>0000<			
ProgressOctet				
ProgressDescripti				
DSS1 parameter values	<u></u>			
SETUP:				
Message flow				
End d	evice		Т	'est equipment
SETUP		→	INVITE	
SETUP ACKNOWLEDGE		←		ntication Required
		→	ACK	
		→	INVITE	
		_		
	A	Apply post tes	i routine	

TSS Orig_Establishment_of_an _early_dialogue	TP_201_212	Reference clause 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 high layer comp	atibility information el	ement			
Ensure that the high layer co HighLayerCompatibility	mpatibility information	n element received in the SETUP is n	napped into the PSTN XML		
SIP header values					
INVITE:					
xml version="1.0" encoding</td <td>ng="utf-8"?></td> <td></td> <td></td>	ng="utf-8"?>				
PSTN					
HighLayerCompatibi	lity				
HLOctet3					
-	CodingStandard>00<				
Interpretation>100<					
	PresentationMethod>01<				
HLOctet4					
HighLayerCharac	teristics>HLC_value<				

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

wiessage now		
End device		Test equipment
SETUP	→	INVITE
SETUP ACKNOWLEDGE	÷	407 Proxy Authentication Required
	→	ACK
	→	INVITE
	Apply post tes	st routine

TSS Orig_Establishment_of_an _early_dialogue	TP_201_213		e e 5.3.1.5.3 of S 183 043]	Selection expression PICS 5.1.1/2 and 5.4/1	
Test purpose					
Two high layer compatibility	information elements	received			
	patibility and the secon	nd High lag	ver compatibility inf	SETUP is mapped into the first formation element received in the	
SIP header values					
INVITE:					
xml version="1.0" encoding</td <td>ng="utf-8"?></td> <td></td> <th></th> <td></td>	ng="utf-8"?>				
PSTN					
HighLayerCompatibi	lity				
HLOctet3	Standard>00<				
	etation>100<				
	ationMethod>01<				
HLOctet4					
	yerCharacteristics>00	00001<			
HighLayerCompatibi HLOctet3	iity				
	CodingStandard>00<				
1	etation>100<				
PresentationMethod>01<					
HLOctet4					
HighLayerCharacteristics>HLC_value<					
DSS1 Parameter values SETUP:					
First High layer compatibilit Telephony	y Coding standard='00	', Interpret	ation='100', High la	yer characteristics identification=	
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 d	End device Test equipment				
SETUP		→	INVITE		
SETUP ACKNOWLEDGE		÷	407 Proxy Auther	ntication 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	TP_201_214	Reference	Selection expression
Orig_Establishment_of_an		5.1.1.2.1, 5.1.1.2.1.1/	PICS 5.1.1/2 and 5.4/1
_early_dialogue		[ETSI TS 183 036]	

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	TP_201_215	Reference	Selection expression
Orig_Establishment_of_an		subclauses 5.1.1.2.1 and	PICS 5.1.1/2 and 5.4/1
_early_dialogue		5.1.1.2.1.1 of	
		[ETSI TS 183 036]	

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
Ap	oply 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 bo	dy present XML PI s	et to value 7	
		nd a PSTN XML body present the puipment does not contain a Progr	
SIP header values			
180 (Ringing): XML body			
xml version="1.0" encodi</td <td>ng="utf-8"?></td> <td></td> <td></td>	ng="utf-8"?>		
PSTN			
ProgressIndicator			
ProgressOctet3			
CodingStandard>	>00		
Location>yyyy<			
ProgressOctet4			
ProgressDescripti	on>0000111<		
DSS1 Parameter values ALERTING: No Progress Ir	dicator present		
Message flow			
End d	evice	Tes	st equipment
SETUP		→ INVITE	
SETUP ACKNOWLEDGE		← 407 Proxy Authent	ication 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
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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
Appl	y post te	st routine

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

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 I	oost 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, PS included	STN XML body pres	ent two XML PI set to value 7	and x a PROGRESS is sent PI value x
Indicator elements set to x a Information Element is prese	nd 7, a PROGRESS ent, the Progress des ator Information Ele	message is sent to the calling user to the value PI_V ment is present, the Progress de	esent, which contains two Progress user equipment. A Progress Indicator VA as indicated in Table 7.2.2.1.4-5 escription is set to the value 4 if a 180
SIP header values			
180 (Ringing) 1: no XML be	ody present		
180 (Ringing) 2: XML body			
xml version="1.0" encodi</td <td>ng="utf-8"?></td> <td></td> <td></td>	ng="utf-8"?>		
PSTN			
ProgressIndicator			
ProgressOctet3			
CodingStandard>	>00		
Location <yyyy></yyyy>			
ProgressOctet4			
ProgressDescripti	on>PI_value<		
ProgressIndicator			
ProgressOctet3			
CodingStandard>	00<		
Location>yyyy<			
ProgressOctet4			
ProgressDescripti	on>0000111<		
DSS1 Parameter values			
PROGRESS: Progress Indic	ator value x and a P	rogress Indicator value 4 Call h	has returned to the $5.1.1/2$
Message flow			
End d	evice		Test equipment
SETUP		→ INVITE	
SETUP ACKNOWLEDGE		← 407 Proxy Auth	nentication Required
		→ ACK	· · · · · · · · · · · · · · · · · · ·
		→ INVITE	
ALEDTINC			
ALERTING		-	
PROGRESS		100 Kinghig2	
	Α	pply 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'

TSS Orig_Establishment_of_an _early_dialogue	TP_201_220	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 bo	dy present XML PI set	to value 7			
) and a PSTN XML body present the			
		ipment and a Progress Indicator Info			
	t to the value 4 if a 180) (Ringing) was received without a P	SIN XML body.		
SIP header values					
180 (Ringing) 1: no XML bo	• 1				
180 (Ringing) 2: XML body					
xml version="1.0" encodin</td <td>ng="utf-8"?></td> <td></td> <td></td>	ng="utf-8"?>				
PSTN Dragonalization					
ProgressIndicator					
Ũ	ProgressOctet3				
CodingStandard>00<					
Location>yyyy< ProgressOctet4					
ProgressDescription>0000111<					
DSS1 Parameter values					
ALERTING: No Progress Indicator present					
PROGRESS: Progress Indicator value 4 Call has returned to the 5.1.1/2					
Message flow					
End de	wico	Test equ	inmont		
SETUP	evice	→ INVITE	npment		
		INVIIL			
SETUP ACKNOWLEDGE					
		→ ACK			
		→ INVITE			
ALERTING		← 180 Ringing1			
PROGRESS	PROGRESS				
Apply post test routine					

TSS Orig_Establishment_of_an _early_dialogue	TP_201_221	Reference subclause 5.1.1.2.1.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1	
Test purpose	•		· · ·	
	hLayerCompatibili	y fallback information an ALERTI	NG is sent	
Ensure that on receipt of a 1	80 (Ringing) that co	ntains a PSTN XML HighLayerCo	ompatibility element that indicates	
		e to the calling user equipment that	t does not contain a High Layer	
Compatibility Information E	lement.			
SIP header values				
180 (Ringing): XML body				
xml version="1.0" encodi</td <td>ng="utf-8"?></td> <td></td> <td></td>	ng="utf-8"?>			
PSTN				
HighLayerCompatibility				
HLOctet3	00			
CodingStandard>				
Interpretation>10 PresentationMeth				
	od>01<			
HLOctet4 HighLayerCharacteristics>HLC_value<				
· ·				
DSS1 parameter values		6		
audio	bility information tr	ansfer capability = Speech or 3.1 k	ΉZ	
	nability Information	transfer capability = Unrestricted		
	n with tones/announ			
ALERTING:	in white tones, announ	cements		
Message flow				
0		T.		
End d	evice	•	st equipment	
SETUP		→ INVITE		
SETUP ACKNOWLEDGE		← 407 Proxy Authent	ication Required	
		→ ACK		
		→ INVITE		
ALERTING		← 180 Ringing		
Apply post test routine				

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		ce e 5.1.1.2.1.3 of S 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 13 fallback, the SUT sends an A Capability Information Elem	LERTING messag						
SIP header values 180 (Ringing): XML body xml version="1.0" encodi</td <td>ng="utf-8"?></td> <td></td> <td></td> <td></td>	ng="utf-8"?>						
PSTN BearerCapability							
BCoctet3 CodingStandard>	00 <						
-	ferCabability>0000	0<					
BCoctet4	ter cububility > 0000						
TransferMode>00)<						
InformationTrans	ferRate>10000<						
BCoctet5>							
Layer1Identificati	on>01<						
UserInfoLayer1Pr	otocol>00011<						
DSS1 Parameter values							
SETUP: First Bearer Capa	bility Information t	ransfer capabi	lity = Speech or 3.1 h	кНz			
audio							
		-	ability = Unrestricted	l			
•	n with tones/annour	ncements					
ALERTING:							
Message flow							
End deviceTest equipment							
SETUP		→	INVITE				
SETUP ACKNOWLEDGE		+	407 Proxy Authent	tication 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 head	SIP header values				
DSS1 Pa	rameter values				
SETUP:	First Bearer Capability Information tr	ansfer capabi	lity = Speech or 3.1 kHz		
	audio				
	Second Bearer Capability Information	n transfer cap	ability = Unrestricted		
	digital information with tones/announ	cements			
ALERTI	NG:				
Message	flow				
	End device		Test equipment		
SETUP		→	INVITE		
SETUP A	ACKNOWLEDGE	←	407 Proxy Authentication Required		
		→	ACK		
		→	INVITE		
ALERTI	NG	÷	180 Ringing		
	Α	pply post te	st 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

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: send	lonly, SDP				
DSS1 Parameter values					
CALL PROCEEDING: Progress Indicator va	alue 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						

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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 te	st routine

TSS Orig_Establishment_of_an _early_dialogue	TP_201_226	Reference subclause [ETSI TS	5.1.1.2.2.1 of	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 CALL PROCEEDING is sent PI value x						
Ensure that on receipt of a 18 elements set to x and 7, a CA Information Element is prese	LL PROCEEDING is	sent to the c	alling user equipment and	d a Progress Indicator			
SIP header values							
180 (Ringing): XML body							
xml version="1.0" encoding</td <td>ng="utf-8"?></td> <td></td> <td></td> <td></td>	ng="utf-8"?>						
PSTN	C						
ProgressIndicator							
ProgressOctet3							
CodingStandard>	>00						
Location>yyyy<							
ProgressOctet4							
ProgressDescription	on>PI_VA<						
ProgressIndicator							
ProgressOctet3							
CodingStandard>	>00						
Location>yyyy<							
ProgressOctet4							
ProgressDescription	on>0000111<						
DSS1 Parameter values							
CALL PROCEEDING: Prog	gress Indicator value x						
Message flow							
End d	evice		Test equ	ipment			
SETUP		→	INVITE				
SETUP ACKNOWLEDGE		←	407 Proxy Authentication	n Required			
		→	ACK	1			
		→	INVITE				
CALL PROCEEDING		€	183 (Session Progress)				
Apply post test routine							

TSS Orig_Establishment_of_an _early_dialogue	TP_201_227	Reference subclause 5 [ETSI TS 18		Selection expression PICS 5.1.1/2 and 5.4/1	
Test purpose	·				
183 received, PSTN XML bo	dy present XML PI s	set to value 7			
to value 7 a CALL PROCEE	EDING is sent toward	d the calling user	equipment, and a	dy is present, the XML PI is set Progress Indicator Information was received without a PSTN	
SIP header values					
180 (Ringing): XML body					
xml version="1.0" encodi</td <td>ng="utf-8"?></td> <td></td> <td></td> <td></td>	ng="utf-8"?>				
PSTN					
ProgressIndicator					
ProgressOctet3					
CodingStandard>	>00				
Location>yyyy<					
ProgressOctet4					
ProgressDescripti	on>0000111<				
DSS1 Parameter values					
CALL PROCEEDING: Prog	gress Indicator not pi	resent			
Message flow					
End d	evice		Tes	t equipment	
SETUP		→ II	NVITE		
SETUP ACKNOWLEDGE		← 4	07 Proxy Authenti	cation Required	
		→ A	CK	-	
		→ II	NVITE		
CALL PROCEEDING			83 (Session Progre	(224	
Apply post test routine					
	A	ppry post test re			

TSS	TP_201_228	Reference	Selection expression
Orig_Establishment_of_an		subclause 5.1.1.2.2.1 of	PICS 5.1.1/2 and 5.2/16
_early_dialogue		[ETSI TS 183 036]	

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 te	st 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							
	dy present two XML	PI set to value 7 and x a PROGRI	ESS is sent PI value x				
) that contains a P-Early-Media he esent in the sent PROGRESS mess					
SIP header values							
180 (Ringing):							
xml version="1.0" encoding</td <td>ng="utf-8"?></td> <td></td> <td></td>	ng="utf-8"?>						
PSTN	C						
ProgressIndicator							
ProgressOctet3							
CodingStandard>	>00						
Location>yyyy<							
ProgressOctet4							
ProgressDescripti	on>0000111<						
183 (Session Progress): XM	L body						
xml version="1.0" encoding</td <td>ng="utf-8"?></td> <td></td> <td></td>	ng="utf-8"?>						
PSTN							
ProgressIndicator							
ProgressOctet3							
CodingStandard>	>00						
Location>yyyy<							
ProgressOctet4							
ProgressDescripti	on>PI_VA<						
ProgressIndicator							
ProgressOctet3							
CodingStandard>	>00						
Location>yyyy<							
U	ProgressOctet4						
ProgressDescripti	on>0000111<						
DSS1 Parameter values							
ALERTING: No Progress In	-						
PROGRESS: Progress Indicator value 8							
Comments							
The Location value is don't c	care. The order of the	received ProgressIndicator eleme	nts 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 bo	ody present two XML	PI set to value 7 and x a PROGRE	ESS is sent PI value x
), a PSTN XML body is present, w	
		GRESS is sent to the calling user e gress description is set to the value	
Table 7.2.2.1.4-6. In the prev			11_VA as indicated in
SIP header values		· · · ·	
180 (Ringing):			
xml version="1.0" encodi</td <td>ng="utf-8"?></td> <td></td> <td></td>	ng="utf-8"?>		
PSTN	6		
ProgressIndicator			
ProgressOctet3			
CodingStandard>	00<		
Location>yyyy<			
ProgressOctet4			
ProgressDescripti	on>0000111<		
183 (Session Progress): XM	L body		
xml version="1.0" encodi</td <td>ng="utf-8"?></td> <td></td> <td></td>	ng="utf-8"?>		
PSTN			
ProgressIndicator			
ProgressOctet3			
CodingStandard>	00<		
Location>yyyy<			
ProgressOctet4			
ProgressDescripti	on>PI_VA<		
ProgressIndicator			
ProgressOctet3			
CodingStandard>	00<		
Location>yyyy<			
ProgressOctet4	0000111		
ProgressDescripti	on>000111<		
DSS1 Parameter values			
PROGRESS: Progress Indic	ator value x		
Comments			
The Location value is don't of	care. The order of the	e received ProgressIndicator eleme	nts 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_a n_early_dialogue	TP_201_231		ce e 5.1.1.2.2.1 of S 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1
Test purpose		L		
	oody present two XML P	PI set to val	ue 7 and x a PROGRESS	S is sent PI value x and value 4
elements set to x and 7, a P Element is present, the Prop	ROGRESS is sent to the gress description is set t Indicator Information E	e calling us to the value Element is p	er equipment and a Prog PI_VA as indicated in 7	
SIP header values				
180 (Ringing):				
183 (Session Progress): XN	/IL body			
xml version="1.0" encod</td <td></td> <td></td> <td></td> <td></td>				
PSTN				
ProgressIndicator				
ProgressOctet3				
CodingStandard	>00<			
Location>yyyy<				
ProgressOctet4				
ProgressDescrip	tion>PI_VA<			
ProgressIndicator				
ProgressOctet3				
CodingStandard	>00<			
Location>yyyy<	·			
ProgressOctet4				
ProgressDescrip	tion>0000111<			
DSS1 Parameter values				
PROGRESS: Progress Indi	cator value x and 4			
Comments				
The Location value is don't	care. The order of the r	received Pr	ogressIndicator elements	s is accidental
Message flow				
End	device		Test e	equipment
SETUP		→	INVITE	* * * *
SETUP ACKNOWLEDGE	3	(407 Proxy Authentica	tion Required
	-	÷	ACK	
		÷	INVITE	
ALERTING		÷	180 Ringing	
		← ←		
PROGRESS			183 Session Progress	
	App	oly post tes	at routine	

TSS	TP_201_232		ce se 5.1.1.2.2.1 of 'S 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1
Test purpose				· ·
183 received, PSTN XML	body present XML P	I set to value 7	,	
	essage is sent to the c	alling user equ		resent and the XML PI is set to ndicator Information Element is
SIP header values	ription is set to the va	aruc 4.		
183 (Session Progress):				
xml version="1.0" enco</td <td>oding="utf-8"?></td> <td></td> <td></td> <td></td>	oding="utf-8"?>			
ProgressIndicator	ang au o n			
ProgressOctet3				
CodingStandar	d>00<			
Location>yyyy	<			
ProgressOctet4				
ProgressDescri	ption>0000111<			
DSS1 Parameter values				
PROGRESS: Progress Inc	dicator value x and 4			
Message flow				
Enc	l device		Te	st equipment
SETUP		→	INVITE	
SETUP ACKNOWLEDO	Έ	÷	407 Proxy Authent	tication Required
		→	ACK	
		→	INVITE	
ALERTING		÷	180 Ringing	
PROGRESS		+	183 Session Progre	ess
		Apply post te	e	

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		ce 1.1.2.2.3 of S 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1
Test purpose				
Receiving of PSTN XML Be	earerCapability falll	oack informati	on in a 183 a CALI	L PROCEEDING is sent.
				earerCapability element that indicates
Capability Information Ele		G message to	the canning user equ	ipment that does not contain a Bearer
SIP header values				
183 (Session Progress):				
xml version="1.0" encod</td <td>ling="utf-8"?></td> <td></td> <td></td> <td></td>	ling="utf-8"?>			
BearerCapability				
BCoctet3				
CodingStandard	>00<			
InformationTran	sferCabability>0000	>0(
BCoctet4				
TransferMode>0	>00<			
InformationTran	sferRate>10000<			
BCoctet5>				
Layer1Identifica				
UserInfoLayer11	Protocol>00011<			
DSS1 Parameter values				
SETUP: First Bearer Cap	ability Information t	ransfer capabi	lity = Speech or 3.1	1 kHz
audio				
	Capability Informatic	-	ability = Unrestrict	ed
0	on with tones/annou	ncements		
CALL PROCEEDING				
Message flow				
End	device		r	Fest equipment
SETUP		→	INVITE	
SETUP ACKNOWLEDGE	Ξ	+	407 Proxy Authe	entication Required
		→	ACK	
		→	INVITE	
CALL PROCEEDING		÷	183 Session Prog	Press
		Apply post te		<u> </u>
	1	zhhià host te		

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

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>0000	00<
BCoctet4	
TransferMode>00<	
InformationTransferRate>10000<	
BCoctet5>	
Layer1Identification>01<	
UserInfoLayer1Protocol>00011<	
DSS1 Parameter values SETUP: First Bearer Capability Information a audio Second Bearer Capability Information digital information with tones/annou PROGRESS:	on transfer capability = Unrestricted
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
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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	r capability = Speech or 3.1 kHz audio
Second Bearer Capability Information trans tones/announcements	sfer capability = Unrestricted digital information with
CALL PROCEEDING:	
Message flow	
End device	Test equipment
SETUP	→ INVITE
SETUP ACKNOWLEDGE	← 407 Proxy Authentication Required
	→ ACK
	→ INVITE
CALL PROCEEDING	 183 Session Progress
	105 56551011 1051655
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
---	--

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 transfectores/announcements	er capability = Unrestricted digital information with			
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 p	ost test routine			

	expression .1/2 and 5.4/1	
--	------------------------------	--

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 te	st routine

TSS	TP_201_238	Reference subclause 5.1.1.2. [ETSI TS 183 036		Selection expression PICS 5.1.1/2 and 5.4/1
Test purpose				
No fallback info	rmation received, CLEARMO	DE codec not the preferr	ed codec in th	ne 183
				the first codec in the codec list ndicated. No BC is sent in the
SIP header valu	ies			
183 (Session Pro	ogress): CLEARMODE not the	e first codec		
DSS1 Paramete	r values			
	r values Bearer Capability Information	transfer capability = Spe	ech or 3.1 kH	z audio
SETUP: First I Secon tones/	Bearer Capability Information			
SETUP: First I Secon	Bearer Capability Information d Bearer Capability Informati			
SETUP: First I Secon tones/	Bearer Capability Information d Bearer Capability Informati			
SETUP: First I Secon tones/ PROGRESS	Bearer Capability Information d Bearer Capability Informati		Unrestricted d	
SETUP: First I Secon tones/ PROGRESS Message flow	Bearer Capability Information d Bearer Capability Informati announcements		Unrestricted d	igital information with
SETUP: First I Secon tones/ PROGRESS	Bearer Capability Information d Bearer Capability Informati announcements End device	on transfer capability = U → INVIT	Unrestricted d Test E	igital information with
SETUP: First I Secon tones/ PROGRESS Message flow SETUP	Bearer Capability Information d Bearer Capability Informati announcements End device	on transfer capability = U → INVIT	Unrestricted d Test E	igital information with equipment
SETUP: First I Secon tones/ PROGRESS Message flow SETUP	Bearer Capability Information d Bearer Capability Informati announcements End device	on transfer capability = U → INVITI ← 407 Pro	Unrestricted d Test E oxy Authentic	igital information with equipment
SETUP: First I Secon tones/ PROGRESS Message flow SETUP	Bearer Capability Information d Bearer Capability Informati announcements End device	on transfer capability = U → INVITI ← 407 Pro → ACK	Unrestricted d Test E Dxy Authentic E	igital information with equipment
SETUP: First I Secon tones/ PROGRESS Message flow SETUP SETUP ACKNO	Bearer Capability Information d Bearer Capability Informati announcements End device	 → INVITI ← 407 Pro → ACK → INVITI ← 180 (Ri 	Unrestricted d Test E Dxy Authentic E	igital information with equipment ation Required

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	17.1.	rence ons 12.2.1.1, 13.2.2.4, 1.3 and Figure 5 of F RFC 3261]	Selection expression
Test purpose				
200 OK received, an ACK is	s sent			
Ensure that when an early dis sent.	ialogue is not establis	hed that on r	receipt of a Success (200	OK) response an ACK request
SIP header values				
Message flow				
End d	levice		Test ec	quipment
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 ACKNOWLED	OGE	→	ACK	
	Ap	oply post tes	t 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 te	st routine

TSS Orig_Establishment_of_a _confirmed_dialogue	TP_202_003	Figure 1	ce 12.2.1.1, 13.2.2.4, and 5 7.1.1.3 of FC 3261]	Selection expression
Test purpose				
200 OK received in an earl	ly dialogue an ACK i	ssent		
Ensure that when an early sent.	dialogue is establish	ed that on rec	eipt of a Success (200 OK)) response an ACK request is
SIP header values				
Message flow				
End	device		Test equ	uipment
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	TP_202_004	Reference		Selection expression
Orig_Establishment_of_a _confirmed_dialogue			3.1.3.3 and Figure 5 of FC 3261]	
Test purpose				
200 OK containing two Vie	a header discarded			
Ensure that when the client with more than one Via her				uccess (200 OK) response
SIP header values				
200 OK INVITE				
Via:SIP 2.0 <transport< td=""><td>> <ip address="">;branch</ip></td><td>n= <any bran<="" td=""><td>ch value></td><td></td></any></td></transport<>	> <ip address="">;branch</ip>	n= <any bran<="" td=""><td>ch value></td><td></td></any>	ch value>	
Via:SIP 2.0 <transport< td=""><td>> <ip address="">;branch</ip></td><td>n= <any bran<="" td=""><td>ch value></td><td></td></any></td></transport<>	> <ip address="">;branch</ip>	n= <any bran<="" td=""><td>ch value></td><td></td></any>	ch value>	
Message flow				
End	device		Test eq	luipment
Interworking POTS				
Off hook				
Dial number		→	INVITE	
		÷	100 Trying	
		÷	200 OK INVITE	
ISDN interworking				
SETUP		→	INVITE	
SETUP ACKNOWLEDGI	Ξ	←	100 Trying	
CONNECT		÷	200 OK INVITE	
	A	pply post tes	t routine	

TSS	TP_202_005	Reference	Selection expression
Orig_Establishment_of_a		section 12.2.1.1, 13.2.2.4,	_
_confirmed_dialogue		17.1.1.3 and Figure 5 of	
_		[IETF RFC 3261]	

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 te	st routine	

TSS Orig_Establishment_of_a _confirmed_dialogue	TP_202_006	17.1.1.3	ce 12.2.1.1, 13.2.2.4, and Figure 5 of FC 3261]	Selection expression	
Test purpose					
ACK is sent with the same	To header filed as rece	vived in the	200 OK		
Ensure that when the clien ACK request is sent with t					
SIP header values					
200 OK INVITE					
To: <any uri="">;tag=<a< td=""><td>ny 200 OK tag value></td><td></td><td></td><td></td></a<></any>	ny 200 OK tag value>				
АСК					
To: <any uri="">;tag=<a< td=""><td>ny 200 OK tag value></td><td></td><td></td><td></td></a<></any>	ny 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	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	TP 202 007	Referen	Reference Selection expression					
Orig_Establishment_of_a		section 12.2.1.1, 13.2.2.4,		1				
_confirmed_dialogue		Figure 5 and 17.1.1.3 of [IETF RFC 3261]						
Test purpose			are 5201]					
ACK is sent with the same	To header filed as re	ceived in the	180					
Ensure that when the client ACK request is sent with the				Success (200 OK) response an sional response.				
SIP header values			1					
200 OK INVITE								
To: <any uri="">;tag=<ar< td=""><td>ny 200 OK tag value</td><td>></td><td></td><td></td></ar<></any>	ny 200 OK tag value	>						
ACK								
To: <any uri="">;tag=<ar< td=""><td>ny 200 OK tag value</td><td>></td><td></td><td></td></ar<></any>	ny 200 OK tag value	>						
Message flow	<u> </u>							
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					
	l	Apply post te	st 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 to	lg				
Ensure that when the client including a To header with				s (200 OK) response	
SIP header values					
200 OK INVITE					
To: <any uri=""></any>					
ACK					
To: <any uri=""></any>					
Message flow					
End	End device Test equipment				
Interworking POTS	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 ACKNOWLE	DGE	→	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
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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 1="" tag=""></any></any>			
ACK 1			
To: <any uri="">;tag=<any 1="" tag=""></any></any>			
200 OK INVITE 2			
To: <any uri="">;tag=<any 2="" tag=""></any></any>			
ACK 2			
To: <any uri="">;tag=<any 2="" tag=""></any></any>			
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 te		

TSS Orig_Establishment_of_a _confirmed_dialogue	TP_202_010	Reference section 12.2.1.1, 13.2.2.4, Figure 5 and 17.1.1.3 of [IETF RFC 3261]	Selection expression
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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 call="" id="" invite=""></any>		
From: <any invite="" uri="">;tag=<any invite="" tag=""></any></any>		
ACK 1		
Call-ID: <any call="" id="" invite=""></any>		
From: <any invite="" uri="">;tag=<any invite="" tag=""></any></any>		
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 tes	st 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 te	st 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	Test purpose									
ACK is sent to the address	indicated in the Record-	Route header with 'lr' parameter								
Record-Route header set to Request-URI set to the Cor	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 20<="" th=""><td colspan="8">Record-Route: <any 200="" ok="" route="" uri1;lr=""> ,<any 200="" ok="" route="" uri2;lr=""></any></any></td></any>	Record-Route: <any 200="" ok="" route="" uri1;lr=""> ,<any 200="" ok="" route="" uri2;lr=""></any></any>									
Contact: <any 200="" contact="" ok="" uri=""></any>										
ACK 1										
ACK sine conv 200 of route LIBIN SID 2.0										

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

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	Figure 5	ce 12.2.1.1, 13.2.2.4, and 17.1.1.3 of FC 3261]	Selection expression
Test purpose				
ACK is sent to the address	indicated in the Record	d-Route hee	ader without 'lr' parameter	
	a list in which the last element and a Route h	t element do neader set to	es not contain lr parameter	ss (200 OK) response with a c, an ACK request is sent with erse order of the received
SIP header values				
200 OK				
Record-Route: <any 20<="" td=""><td></td><td>y 200 ok rou</td><td>te URI2;lr></td><td></td></any>		y 200 ok rou	te URI2;lr>	
Contact: <any 200="" c<="" ok="" td=""><td>ontact URI></td><td></td><td></td><td></td></any>	ontact URI>			
ACK				
ACK sip: <any 200="" ok<="" td=""><td>route URI1> SIP 2.0</td><td></td><td></td><td></td></any>	route URI1> SIP 2.0			
Route: <any 200="" ok="" rou<="" td=""><td>te URI2;lr>,<any 200<="" td=""><td>ok contact U</td><td>JRI></td><td></td></any></td></any>	te URI2;lr>, <any 200<="" td=""><td>ok contact U</td><td>JRI></td><td></td></any>	ok contact U	JRI>	
Message flow				
End	device		Test eq	uipment
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 ACKNOWLE	DGE	→	ACK 1	
	An	oply post tes	st routine	

TSS Orig_Establishment_of_a _confirmed_dialogue	TP_202_014		ce 13.2.1 of RFC 3261]	Selection expression
Test purpose SDP offer sent in the initial	l INVITE request			
Ensure that while the client answer the initial offers giv			ssion description of	fer is sent in the INVITE request to
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 INVITH	Ξ
ISDN interworking				
SETUP		→	INVITE	
		←	200 OK INVITI	Ξ
		Apply post tes	st 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 init	tial 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	'a' line							
ACK 1								
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 te	st routine

TSS	TP_202_016	Referen	ce	Selection expression
Orig_Establishment_of_a			13.2.2.3, 17, 17.1.1.2,	_
_confirmed_dialogue			. and Figure 5 of	
		[IETF RFC 3261]		
Test purpose				
Update Record-Route head	ler			
Ensure that when the client	transaction is in the	Calling state	that on receipt of a Succe	ess (200 OK) response, with a
different Record-Route as i	in previous response,	, but with the	same Via branch paramete	er and CSeq header method as
in the INVITE request, an	ACK request is sent	with a Route	header set according to thi	is new Record-Route.
SIP header values				
200 OK 2				
Record-Route: <any 20<="" td=""><td>0 ok route></td><td></td><td></td><td></td></any>	0 ok route>			
ACK 2				
Route: <any 200="" ok="" rou<="" td=""><td>ite></td><td></td><td></td><td></td></any>	ite>			
Message flow				
End	device		Test eq	quipment
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 ACKNOWLE	DGE	÷	ACK 1	
		÷	200 OK INVITE 2	
		÷	ACK 2	
		-		

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		L	
Content-Length header pre	esont in the initial INVIT	F request	
Comeni Lengin neuder pre	Seni in me innui nvviii	5 request	
Ensure that when the client the message that contains the		ng a call, a Content-Length header se sent.	et to the size of the body in
SIP header values			
INVITE			
Content-Length: <any td="" v<=""><td>/alue></td><td></td><td></td></any>	/alue>		
SDP			
'm' line			
'a' line			
Message flow			
End	device	Test equ	ipment
Interworking POTS			- r
Off hook			
Dial number		→ INVITE	
ISDN interworking			
SETUP		→ INVITE	
	App	ly post test routine	
			I
TSS	TP_202_018	Reference	Selection expression
Orig_Establishment_of_a _confirmed_dialogue		section 20.14 and 13.2.1 of [IETF RFC 3261]	
Test purpose			
Content Type header indice	ated SDP		
session description is sent.	isaction is establishing a	call, a Content-Type header in the me	essage that contains the
_			
SIP header values			
INVITE	· /. 1		
Content-Type: applicati	ion/sdp		
SDP			
'm' line			
'a' line			
Message flow			
End	device	Test equ	lipment
Interworking POTS			
Off hook			
Dial number		→ INVITE	
		← 200 OK INVITE 1	
		 ← 200 OK INVITE 1 → ACK 1 	

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		ce se 6.3.2.4 of [S 183 043]	Selection expression
Test purpose	I	1		
Modifying SDP in confirm	ed dialogue			
Ensure that the SUT is able UPDATE with SDP answe		ATE request to	o modify the SDP in	confirmed dialogue. A 200 OK
SIP header values				
INVITE/UPDATE				
SDP offer 2				
200 OK (INVITE/UPDAT	E)			
SDP answer 2	,			
Message flow				
End	device		Т	est equipment
Off hook				
Dial number		→	INVITE	
		+	•	ntication Required
		→	ACK	
		→	INVITE	
		÷	180 Ringing	
			200 IK INVITE	
			ACK	
CASE A				
		←	INVITE	
		→	200 OK INVITE	
		←	ACK	
CASE B				
		÷	UPDATE	
		→	200 OK UPDATI	Ξ
		Apply post te	st 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 XM	AL ProgressIndicator pr	esent value x, a CONNECT is	sent PI x		
PI_VA, a CONNECT mes	sage is sent to the calling		nt, ProgressIndicator value is set to ss Indicator Information Element is 2.2.2.3-1.		
SIP header values					
200 OK (INVITE): XML b	oody				
xml version="1.0" encod</td <td>•</td> <td></td> <td></td>	•				
PSTN					
ProgressIndicator					
ProgressOctet3					
CodingStandard	>00<				
Location>yyyy<					
ProgressOctet4					
ProgressDescrip	tion>PI_VA<				
ProgressIndicator					
ProgressOctet3					
CodingStandard					
Location>yyyy<					
ProgressOctet4					
ProgressDescrip	tion>0000111<				
DSS1 Parameter values CONNECT:					
Message flow					
	device	•	l'est equipment		
SETUP		→ INVITE			
		•	ntication Required		
		→ _{ACK}			
		→ INVITE			
ALERTING		← 180 (Ringing)			
CONNECT		← 200 OK (INVITE	Ξ)		
		→ ACK	·		
	Am	bly 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	$\leftarrow 200 \text{ OK (INVITE)}$
	\rightarrow ACK
Apply j	post test routine

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

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 7.2.2.3-1.

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 Orig_Establishment_of_a _confirmed_dialogue	TP_202_204		ce se 5.1.1.3 of [S 183 036]	Selection expression PICS 5.1.1/2 and 5.4/1	
Test purpose					
200 OK received PSTN XM	IL ProgressIndicator	present value	e 7, a CONNECT is	sent.	
		•			
				nt, ProgressIndicator value is set to 7	
and a CONNECT message	is sent to the calling	user equipme	ent as described in T	able 7.2.2.3-1.	
SIP header values					
180 (Ringing): PSTN XMI	• • •)			
200 OK (INVITE): XML b	•				
xml version="1.0" encod</td <td>ling="utf-8"?></td> <td></td> <td></td> <td></td>	ling="utf-8"?>				
PSTN					
ProgressIndicator					
ProgressOctet3	× 00 <i>c</i>				
CodingStandard Location>yyyy<					
ProgressOctet4					
ProgressDescrip	tion>PI VA<				
DSS1 Parameter values					
CONNECT:					
Message flow			_		
	device	•		'est equipment	
SETUP		→	INVITE		
		÷	407 Proxy Auther	ntication Required	
		→	ACK		
		→	INVITE		
ALERTING		←	180 (Ringing)		
CONNECT		←	200 OK (INVITE		
		→	ACK		
	Apply post test routine				

TSS Orig_Establishment_of_a _confirmed_dialogue	TP_202_205	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 XM	L ProgressIndicator pro	esent value	e 7, a CONNECT is sent.			
	is sent to the calling use	er equipme	nt. A Progress Indicator In	gressIndicator value is set to 7 formation Element is present efore in a ALERTING		
SIP header values						
180 (Ringing): PSTN XML	body present (PI #7)					
200 OK (INVITE): XML b	•					
xml version="1.0" encod</td <td>ling="utf-8"?></td> <td></td> <td></td> <td></td>	ling="utf-8"?>					
PSTN						
ProgressIndicator						
ProgressOctet3						
CodingStandard						
Location>yyyy<						
ProgressOctet4	tion > 0000111					
ProgressDescrip						
DSS1 Parameter values						
ALERTING: PI #1 CONNECT: PI #4						
Message flow			—	•		
	device	→	-	uipment		
SETUP			INVITE			
		(407 Proxy Authentication	on Required		
	→ ACK					
	→ INVITE					
ALERTING		←	180 (Ringing)			
CONNECT		←	200 OK (INVITE)			
		→	ACK			
	Apply post test routine					

Table 7.2.2.3-1 – 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		ce se 5.1.1.3 of [S 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 7.2.2.3-2.						
SIP header values						
200 OK (INVITE): PSTN	XML MIME body					
xml version="1.0" encod</td <td>ling="utf-8"?></td> <td></td> <td></td> <td></td>	ling="utf-8"?>					
PSTN						
LowLayerCompatibility	y>					
LLOctet3>						
CodingStandard						
	sferCapability>ITC_	VA<				
LLOctet4>	20 4					
TransferMode>	oo< sferRate>10000<					
	Isterikate>10000<					
DSS1 Parameter values CONNECT: HLC						
Message flow						
End	device		Т	est equipment		
SETUP		→	INVITE			
		+	407 Proxy Auther	ntication Required		
		→	ACK			
		→	INVITE			
ALERTING		÷	180 (Ringing)			
CONNECT		÷	200 OK (INVITE)		
		→	ACK			
	Α	pply post te	st routine			

Table 7.2.2.3-2 – 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 TP_202_207 Orig_Establishment_of_a _confirmed_dialogue	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 pres	sent
	*	
Fallback connection type indicated in the SETUP. E PSTN XML BearerCapability indicating fallback, a Bearer Capability Information Element is present se described in Table 5.2.1.3-5.	CONNECT message is sent to	the calling user equipment and a
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 trans	fer capability = Speech or 3.1 k	кНz
audio		
Second Bearer Capability Information tra	ansfer capability = Unrestricted	digital
information with tones/announcements		
CONNECT: BC		
Message flow		
End device	_	st equipment
SETUP	→ INVITE	
	← 407 Proxy Authent	tication Required
	→ _{ACK}	
	→ INVITE	
ALERTING	← 180 (Ringing)	
CONNECT	← 200 OK (INVITE)	
COMPLET	-	
	nen	
Appl	ly post test routine	

Table 7.2.2.3-3 – Mapping of beare	r canability to PSTN XMI	BearerCanability
Table 7.2.2.2.5-5 Mapping of Dearch	a capability to 1 0 1 1 2 1 11	J Dearer Capability

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

TSS	TP_202_208	Reference	Selection expression
Orig_Establishment_of_a		subclause 5.1.1.3 of	
_confirmed_dialogue		[ETSI TS 183 036]	
Test purpose			
200 OK received no PSTN	XML ProgressIndic	ator present, a CONNECT	BC included is sent.
Fall-back connection type	indicated in the SET	UP. Ensure that on receipt	of a 200 OK (INVITE) and no PSTN XM
body is present, a CONNE	CT message is sent t	to the calling user equipme	nt. A Bearer Capability Information
Element is present set to th	e value speech or au	idio 3.1 kHz	
SIP header values			
200 OK (INVITE): no XML body present			
no AME body present			
SDP:			
SDI.			
DSS1 Parameter values			
	pability Information	transfer capability = Speec	h or 3.1 kHz audio
DSS1 Parameter values SETUP: First Bearer Cap Second Bearer C	Capability Information		h or 3.1 kHz audio restricted digital information with
DSS1 Parameter values SETUP: First Bearer Cap	Capability Information		
DSS1 Parameter values SETUP: First Bearer Cap Second Bearer C	Capability Information nents		
DSS1 Parameter values SETUP: First Bearer Cap Second Bearer C tones/announcer	Capability Information nents		
DSS1 Parameter values SETUP: First Bearer Cap Second Bearer C tones/announcer CONNECT: BC speech or Message flow	Capability Information nents		
DSS1 Parameter values SETUP: First Bearer Cap Second Bearer C tones/announcer CONNECT: BC speech or Message flow	Capability Information nents audio 3.1 kHz		restricted digital information with
DSS1 Parameter values SETUP: First Bearer Cap Second Bearer C tones/announcer CONNECT: BC speech or Message flow End	Capability Information nents audio 3.1 kHz	on transfer capability = Un	restricted digital information with
DSS1 Parameter values SETUP: First Bearer Cap Second Bearer C tones/announcer CONNECT: BC speech or Message flow End	Capability Information nents audio 3.1 kHz	on transfer capability = Un	restricted digital information with Test equipment
DSS1 Parameter values SETUP: First Bearer Cap Second Bearer C tones/announcer CONNECT: BC speech or Message flow End	Capability Information nents audio 3.1 kHz	on transfer capability = Un → INVITE ← 407 Prox	restricted digital information with Test equipment
DSS1 Parameter values SETUP: First Bearer Cap Second Bearer C tones/announcer CONNECT: BC speech or Message flow End	Capability Information nents audio 3.1 kHz	 → INVITE ← 407 Proxy → ACK 	restricted digital information with Test equipment y Authentication Required
DSS1 Parameter values SETUP: First Bearer Cap Second Bearer C tones/announcer CONNECT: BC speech or Message flow End SETUP	Capability Information nents audio 3.1 kHz	 → INVITE ← 407 Proxy → ACK → INVITE 	restricted digital information with Test equipment (Y Authentication Required
DSS1 Parameter values SETUP: First Bearer Cap Second Bearer C tones/announcer CONNECT: BC speech or Message flow End SETUP	Capability Information nents audio 3.1 kHz	 → INVITE ← 407 Proxy → ACK → INVITE ← 180 (Ring 	restricted digital information with Test equipment (Y Authentication Required

TSS	TP_202_209	Reference	Selection expression
Orig_Establishment_of_a		subclause 5.1.1.3 of	PICS 5.1.1/2 and 5.4/1
_confirmed_dialogue		[ETSI TS 183 036]	
Test purpose			
Fallback information recei	ived as PSTN XML body	y HLC in the 200 a CONNECT	contains the fall-back information
to HLC_VA and ProgressI equipment. The CONNEC	ndicator No. 5 PSTN X T message contains a H Progress Indicator Info	ML body, a CONNECT message LC Information Element derived formation Element derived form	ed from the received PSTN XML
SIP header values			
xml version="1.0" encod</td <td>ding="utf-8"?></td> <td></td> <td></td>	ding="utf-8"?>		
PSTN	C		
HighLayerCompatibi	lity		
HLOctet3	•		
CodingStandard	>00<		
Interpretation>1	00<		
PresentationMet	:hod>01<		
HLOctet4			
HighLayerChara	acteristics>HLC_VA<		
ProgressIndicator			
ProgressOctet3			
CodingStandard	>00<		
Location>yyyy<	<		
ProgressOctet4			
ProgressDescrip	ntion>0000101<		
ProgressIndicator			
ProgressOctet3			
CodingStandard			
Location>yyyy<			
ProgressOctet4			
ProgressDescrip	tion>0000111<		
DSS1 Parameter values			
1	Capability Information t	nsfer capability = Speech or 3.1 ransfer capability = Unrestricte	
CONNECT: BC speech or	audio 3.1 kHz		
Message flow			
End	device	Т	est equipment
SETUP		→ INVITE	
		-	ntication Required
		→ ACK	
		nen -	
		-	
ALERTING		roo (runging)	
CONNECT		← 200 OK (INVITE	
		→ ACK	
	Ар	ply post test routine	

Table 7.2.2.3-4 – 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		nce 1 12.2.1.1 and 15 of RFC 3261]	Selection expression
Test purpose		•		
To header in BYE is used from	the last final respons	se		
Ensure that once a dialogue ha			ds a BYE request to relea	ase it where a To header is set
to the same value as in the last	received final respon	se.		
SIP header values				
200 OK				
To: <any 200="" ok="" t<="" td="" to="" uri;=""><td>ag=<any 200ok="" tag=""></any></td><td></td><td></td><td></td></any>	ag= <any 200ok="" tag=""></any>			
BYE	age const 200 alt tags			
To: <any 200="" ok="" t<="" td="" to="" uri;=""><td>ag=<any 2000k="" tag=""></any></td><td></td><td></td><td></td></any>	ag= <any 2000k="" tag=""></any>			
Message flow				
End dev	rice		Test e	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	
DIGONNECT		→ →	ACK	
DISCONNECT		→ ~	BYE	
RELEASE COMPLETE		+ _	200 OK BYE	
RELEASE COMPLETE		→		

TSS Orig_Orig_Release_initiated _by_the_terminating_user	TP_203_002		rence on 12.2.1.1 and 15 of F RFC 3261]	Selection expression	
Test purpose		•			
BYE is sent without To tag					
Ensure that if a dialog had bee release it the IUT sends a BYE				To header was omitted, to	
SIP header values					
200 OK					
To: <any 200="" ok="" to="" uri=""></any>					
BYE					
To: <any 200="" ok="" to="" uri=""></any>					
Message flow					
End dev	vice		Test equ	lipment	
Interworking POTS					
Off hook		•			
Dial number		→	INVITE		
		+	200 OK INVITE		
		→	ACK		
On hook		_	→ BYE		
		÷	200 OK BYE		
ISDN interworking		•			
SETUP		→ <	INVITE		
CONNECT		(200 OK INVITE		
		→	ACK		
DISCONNECT		→	BYE		
RELEASE		÷	200 OK BYE		
RELEASE COMPLETE		→			

TSS Orig_Orig_Release_initiated _by_the_terminating_user	TP_203_003	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 respons	se						
the same value as in the last re-	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.							
	SIP header values							
200 OK								
To: <any 200="" ok="" t<="" td="" to="" uri;=""><td colspan="7">To: <any 200="" ok="" tag="" to="" uri;=""></any></td></any>	To: <any 200="" ok="" tag="" to="" uri;=""></any>							
BYE	BYE							
To: <any 200="" ok="" t<="" td="" to="" uri;=""><td>ag=<any 2000k="" tag=""></any></td><td></td><td></td></any>	ag= <any 2000k="" tag=""></any>							

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	TP_203_004	Refere	ence	Selection expression
Orig_Orig_Release_initiated			n 12.2.1.1 and 15 of	_
_by_the_terminating_user		[IETF	RFC 3261]	
Test purpose				
Call-ID is set to the value as in	the original INVIT	E		
Ensure that the IUT, once a dia	logua has been este	bliched to	ralaasa it sands o DVE r	aquast with the same Call ID
From headers as in the original			ielease, it selius a D I E I	equest with the same Can-ID,
SIP header values	in () II D message.			
200 OK				
Call-ID: <any 200="" call-<="" ok="" td=""><td>ID value></td><td></td><td></td><td></td></any>	ID value>			
BYE				
Call-ID: <any 200="" call-<="" ok="" td=""><td>ID value></td><td></td><td></td><td></td></any>	ID value>			
Message flow				
End device			Test e	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_005		nce 12.2.1.1 of RFC 3261]	Selection expression
Test purpose				
CSeq in BYE is incremented				
Ensure that the IUT, once a dia CSeq value, a method field in			release, it sends a BYE requ	uest with an increment of one
SIP header values				
INVITE				
CSeq: <any invite="" seque<="" td=""><td>nce number> INVITE</td><td></td><td></td><td></td></any>	nce number> INVITE			
BYE				
CSeq: <any invite="" seque<="" td=""><td>nce number> BYE</td><td></td><td></td><td></td></any>	nce 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		→		

Orig_Orig_Release_initiated se	eference Selection expression ection 12.2.1.1 and 15 of ETF RFC 32611
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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 nurnose			

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	→

Orig_Orig_Release_initiated _by_the_terminating_usersection 12.2.1.1 and 15 of [IETF RFC 3261]
--

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	\rightarrow 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	TP_203_009	Reference	Selection expression	
Orig_Orig_Release_initiated		section 12.2.1.1 and 15 of		
_by_the_terminating_user		[IETF RFC 3261]		
Test purpose				
Session in confirmed dialogue	e is terminated by ser	iding a BYE message		
Ensure that the IUT, when a cresponse considers the session		ablished, having sent a BYE reques rminated.	st, on receipt of a (200 OK)	
SIP header values				
Message flow				
End device		Test	Test equipment	
Interworking POTS				
Off hook				
Dial number		\rightarrow INVITE 1		
		← 180 Ringing		
		← 200 OK INVITE		
		→ _{ACK}		
On hook		\rightarrow BYE		
	St	art timer K		
		 ← 200 OK BYE 		
	Timeo	out timer K		
		← BYE		

ISDN interworking		
SETUP	→	\rightarrow INVITE 1
ALERTING	←	← 180 Ringing
CONNECT	←	← 200 OK INVITE
		→ _{ACK}
DISCONNECT	→ Start timer K	→ BYE
RELEASE	←	 ← 200 OK BYE
RELEASE COMPLETE	→ Timeout timer K	
	← _{BY}	Έ
	→ 482	1 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 [IETF RFC 3261]	of Selection expression
Test purpose			
BYE received after 200 OK BY	'E was sent		
Ensure that the IUT, when a d Leg/Transaction Does Not Exit terminated.			uest, on receipt of a Call ders the session and the dialogue
SIP header values			
Message flow			
End dev	vice	Те	est equipment
Interworking POTS			
Off hook			
Dial number		\rightarrow 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	s Not Exist
ISDN interworking			
SETUP	→	→ INVI	ГЕ 1
ALERTING	÷	← 180 R	inging
CONNECT	+	← 200 O	OK INVITE
		→ ACK	
DISCONNECT	→ Sta	art timer K \rightarrow BYE	
RELEASE	←		ОК ВҮЕ
RELEASE COMPLETE	→ Tim	eout timer K	
		← BYE	
		→ 481 Call Leg Does	s Not Exist

TSS Orig_Orig_Release_initiated _by_the_terminating_user	TP_203_011		ce 15 and 9.1, Figure 5 of RFC 3261]	Selection expression
Test purpose				
Session in early dialogue is ter	rminated by sendin	ag a CANCE	L or BYE message	
Ensure that the IUT, having re a CANCEL/BYE request.	ceived a Trying (1	00 Trying) r	esponse to its INVITE rec	uest, to give up the call, sends
SIP header values				
Message flow				
End dev	vice		Test e	quipment
Interworking POTS				
Off hook				
Dial number		→	INVITE	
		+	100 Trying	
On hook		→	CANCEL	
		÷	200 OK CANCEL	
		+	487 Request Terminate	ed
		→	ACK	
ISDN interworking				
SETUP	→		➔ INVITE	
			← 100 Trying	5
RELEASE	→		→ CANCEL/	BYE
RELEASE COMPLETE	÷		← 200 OK C	ANCEL/BYE
			← 487 Reque	est Terminated
			→ ACK	

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

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	TP_203_013	Referen	ce 15, 9.1 and Figure 5 of	Selection expression	
_by_the_terminating_user			RFC 3261]		
Test purpose	I				
CSeq method parameter in CA	NCEL or BYE				
Ensure that the IUT, having re request with the same numeric header set to "CANCEL/BYE	part of CSeq as in				
SIP header values					
INVITE					
Call-ID: <invite cseq#=""> IN</invite>	IVITE				
CANCEL					
Call-ID: <invite cseq#=""> C.</invite>	ANCEL				
Message flow					
End dev	vice		Test equipment		
Interworking POTS					
Off hook					
Dial number		→	INVITE		
		+	100 Trying		
On hook		→	CANCEL/BYE		
		+	200 OK CANCEL/BY		
		+	487 Request Terminat	ted	
		→	ACK		
ISDN interworking					
SETUP	→		→ INVITE		
			← 100 Tryir	ng	
RELEASE	→		→ CANCEL	6	
RELEASE COMPLETE	+		← 200 OK 0	CANCEL/BYE	
			← 487 Requ	lest Terminated	
			→ ACK		

TSS Orig_Orig_Release_initiated _by_the_terminating_user	TP_203_014		ce 15, 9.1 and Figure 5 of RFC 3261]	Selection expression	
Test purpose CANCEL or BYE contains the	Via header from th	ne initial IN	VITE request		
Ensure that the IUT, having re request with a single Via head					
SIP header values INVITE					
Via: SIP/2.0/ <any proto<="" td="" via=""><td>ocol> <any uri="" via=""></any></td><td>;branch=<a< td=""><td>ny via branch</td><td></td></a<></td></any>	ocol> <any uri="" via=""></any>	;branch= <a< td=""><td>ny via branch</td><td></td></a<>	ny via branch		
CANCEL					
Via: SIP/2.0/ <any proto<="" td="" via=""><td>col> <any uri="" via=""></any></td><td>;branch=<a< td=""><td>ny via branch</td><td></td></a<></td></any>	col> <any uri="" via=""></any>	;branch= <a< td=""><td>ny via branch</td><td></td></a<>	ny via branch		
Message flow					
End dev	ice		Test equipment		
Interworking POTS					
Off hook					
Dial number		→	INVITE		
		(100 Trying		
On hook		→ (CANCEL/BYE		
		(200 OK CANCEL/BY		
		(487 Request Terminate	ed	
		→	ACK		
ISDN interworking					
SETUP	→		→ INVITE		
			← 100 Trying	g	
RELEASE	→		→ CANCEL	/BYE	
RELEASE COMPLETE	÷		← 200 OK C	ANCEL/BYE	
			← 487 Reque	est Terminated	
			→ ACK		

TSS Orig_Orig_Release_initiated _by_the_terminating_user	TP_203_015	Reference section 15, 9.1 and Figure 5 of [IETF RFC 3261]	Selection expression			
Test purpose <i>The CANCEL or BYE does not contain a Require or Proxy Require header</i> 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. SIP header values						
CANCEL						
Require: [not present]						
Proxy-Require: [not preser	ıt]					

Message flow				
End device		Test equipment		
Interworking POTS				
Off hook				
Dial number		→ INVITE		
		← 100 Trying		
On hook		\rightarrow 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_016	Reference section 15, 9.1 and Figure 5 of [IETF RFC 3261]	Selection expression		
Test purpose A CANCEL or BYE is sent after	180 was received				
	eived a 180 Ringing	provisional response to its INVITE	request, to give up the call,		
Message flow					
End device		Test eq	Test equipment		
Interworking POTS					
Interworking POTS Off hook					
Ū.		→ INVITE			
Off hook		→ INVITE← 180 Ringing			
Off hook		IIII III			
Off hook Dial number		← 180 Ringing	1		
Off hook Dial number		 ← 180 Ringing → CANCEL/BYE 			

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		ce 15, 9.1 and Figure 5 of FC 3261]	Selection expression
Test purpose A CANCEL or BYE is sent after	181 was received			
Ensure that the IUT, having rece give up the call, sends a CANC		s Being Forv	warded provisional respon	nse to its INVITE request, to
Message flow				
End devie	e		Test e	quipment
Interworking POTS				
Off hook				
Dial number		→	INVITE	
		÷	Call Is Being Forwarde	ed
On hook		→ CANCEL/BYE		
		+	200 OK CANCEL/BY	E
		+	487 Request Terminate	ed
		→	ACK	
ISDN interworking				
SETUP	→		➔ INVITE	
			← 180 Ringin	ng
RELEASE	→		→ CANCEL	/BYE
RELEASE COMPLETE	÷		← 200 OK C	ANCEL/BYE
			← 487 Reque	est Terminated
			→ ACK	

TSS Orig_Orig_Release_initiated_ by_the_terminating_user	TP_203_018		ce 15, 9.1 and Figure 5 of FC 3261]	Selection expression
Test purpose A CANCEL or BYE is sent after	182 was received			
Ensure that the IUT, having rece sends a CANCEL/BYE request.		d provision	al response to its INVITE 1	request, to give up the call,
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	

ISDN interworking		
SETUP	→	→ INVITE
		← 182 Queued
RELEASE	→	→ CANCEL/BYE
RELEASE COMPLETE	÷	← 200 OK CANCEL/BYE
		← 487 Request Terminated
		→ ACK

TSS	TP_203_019	Referen	ce	Selection expression	
Orig_Orig_Release_initiated_		section 15, 9.1 and Figure 5 of			
by_the_terminating_user		[IETF F	RFC 3261]		
Test purpose					
A CANCEL or BYE is sent after	· 183 was received				
Ensure that the IUT, having rec the call, sends a CANCEL/BYI		on Progress	provisional response to it	s INVITE request, to give up	
Message flow	Tequest				
End devi	ce		Test e	quipment	
Interworking POTS					
Off hook					
Dial number		→	→ INVITE		
		← 183 Session Progress			
On hook		\rightarrow CANCEL/BYE			
		+	← 200 OK CANCEL/BYE		
		+	487 Request Terminate	ed	
		→	ACK		
ISDN interworking					
SETUP	→		➔ INVITE		
			← 183 Sessio	on Progress	
RELEASE	→		→ CANCEL	/BYE	
RELEASE COMPLETE	÷		← 200 OK C	ANCEL/BYE	
			← 487 Reque	est 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	X BYE is sent				
Ensure that the IUT, in the c terminated.	onfirmed state when	a BYE is re	ceived, a 200 OK BYE is s	sent. The connection is	
Message flow					
End d	evice		Test ea	uipment	
Off hook			1	r	
Dial number		→	INVITE		
		←	← 407 Proxy Authentication Required		
		→	ACK	-	
		→	INVITE		
		+	180 Ringing		
Conversation		+	200 OK INVITE		
		→	ACK		
Terminated + BYE			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 b	efore a provisional resp	ponse was	received				
Ensure that a final response a 180 Ringing was received ar				alogue was established, a			
SIP header values							
Message flow							
End d	evice		Test equ	iipment			
Interworking POTS							
Off hook							
Dial number		→	INVITE				
		←	407 Proxy Authentication Required				
		→	ACK				
		→	INVITE				
		←	180 Ringing				
		←	SIP_FINAL_RESPONS	E			
		→	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 a	fter a181 Call Is Being	g Forward	ed provisional response wa	s received		
Ensure that a final response a 181 Call Is Being Forwarded				alogue was established, a		
SIP header values						
Message flow						
End d	evice		Test equ	iipment		
Interworking POTS						
Off hook						
Dial number		→	INVITE			
		←	407 Proxy Authentication Required			
		→	ACK			
		→	INVITE			
		←	181 Call Is Being Forwa	rded		
		→	SIP_FINAL_RESPONS	E		
→			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 a	fter a 182 Queued prov	visional re	sponse was received				
Ensure that a final response a 182 Queued was received an				alogue was established, a			
SIP header values							
Message flow							
End d	evice		Test equ	ipment			
Interworking POTS							
Off hook							
Dial number		→	INVITE				
		÷	407 Proxy Authentication Required				
		→	ACK				
		→	INVITE				
		←	182 Queued				
		→	SIP_FINAL_RESPONS	E			
		→	ACK				

TSS	TP_204_005	Reference section 13.2.2 of		Selection expression			
Orig_Release_initiated_by _the_terminating_user			RFC 3261]	PICS 5.1.1/1			
Test purpose							
Receipt of a final response a	fter a 183 Session Pro	ogress prov	isional response was recei	ved			
Ensure that a final response a 183 Session Progress was re-				alogue was established, a			
SIP header values							
Message flow							
End d	evice		Test equipment				
Interworking POTS							
Off hook							
Dial number		→	INVITE				
		←	407 Proxy Authentication Required				
		→	ACK				
		→	INVITE				
	+		183 Session Progress				
		→	SIP_FINAL_RESPONS	E			
	→						

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

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

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 [ETSI TS 183 03		Selection expression PICS 5.1.1/2
Test purpose <i>Receipt of BYE, a DSS1 DIS</i>	CONNECT is sent ca	use value 16		
Ensure that on receipt of a B value of the DISCONNECT point.				
SIP header values				
DSS1 Parameter values DISCONNECT: Cause IE ca	use value=16			
Message flow				
End d	evice		Test equ	ipment
SETUP	→	→	INVITE	
		÷	407 Proxy Auth	entication Required
		→	ACK	
	-	→	INVITE	
ALERTING	+	+	180 Ringing	
CONNECT	+	÷	200 OK INVIT	E
		→	ACK	
DISCONNECT	+	+	BYE	
RELEASE	→	→	200 OK BYE	
RELEASE COMPLETE	→			

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

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	ue = 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	TP_204_103	Reference	Selection expression
Orig_Release_initiated_by		subclause 5.1.1.4 of	PICS 5.1.1/2
_the_terminating_user		[ETSI TS 183 036]	

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
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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 val	ue = 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	TP_204_105	Reference	Selection expression
Orig_Release_initiated_by		subclause 5.1.1.4 of	PICS 5.1.1/2
_the_terminating_user		[ETSI TS 183 036]	

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	TP_204_106	Reference	Selection expression
Orig_Release_initiated_by		subclause 5.1.1.4 of	PICS 5.1.1/2
		[ETSI TS 183 036]	

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 valu	e = 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 [ETSI TS 183 03	*-	Selection expression PICS 5.1.1/2	
Test purpose DISCONNECT received, a E	Test purpose DISCONNECT received, a BYE is sent				
Ensure that the IUT, while the request.	ne dialogue is in a conf	irmed state, of an I	SDN DISCONNE	CT message sends a BYE	
SIP header values					
DSS1 Parameter values DISCONNECT: Cause IE ca	ause value= <appropriat< td=""><td>te value></td><td></td><td></td></appropriat<>	te value>			
Message flow					
End d	End device Test equipment			iipment	
Interworking POTS					
SETUP	→	→	INVITE		
		+	407 Proxy Auth	entication Required	
		→	ACK		
		→	INVITE		
	÷	+			
DISCONNECT	÷	÷	SIP_FINAL_RI	ESPONSE	
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 1*T1	► INVITE
Dial number	Timeout timer A	• INVITE
ISDN interworking		
SETUP →	Start timer A 1*T1	→ INVITE
	Timeout timer A	➤ INVITE
	Apply post test routing	e

TSS Orig_Timers	TP_205_002	Reference section 17.1. Annex A of [IETF RFC 3		Selection expression PICS 5.1.2/2
Test purpose				
TCP: INVITE is not repeat	ted if timer T1 expires			
	an INVITE client transaction is ner A set with a value of T1 if a			
Message flow				
End	device		Test equ	iipment
Interworking POTS				
Off hook				
Dial number	Start tin 1*7		INVITE	
	Timeout	-		
ISDN interworking				
SETUP	→ Start tin 1*7	1	INVITE	
	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 rou	tine	

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 <i>UDP: the INVITE is r</i>	etransmitted with intervals that doub	le after each transmission	
	vhen an INVITE client transaction is fter each transmission if an unreliabl		mits its INVITE request with
Message flow			
	End device	Test e	quipment
Interworking POTS			
Off hook			
Dial number	Start timer	A 1*T1 → INVITE	
	Timeout t Start timer	\rightarrow INVITE	
	Timeout t Start timer	\rightarrow INVITE	
	Timeout t Start timer	\rightarrow INVITE	
	Timeout t Start timer	\rightarrow 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	TP_205_005	Reference	:	Selection expression
Orig_Timers		section 17		
		Annex A o		
		[IETF RF	C 3261]	
Test purpose				
Timer B expires no A	CK is sent			
Ensure that the IUT, we expire that the second second	when an INVITE client transa l an ACK.	action is in the Calli	ng state, when tim	er B set to a value of 64*T1
Message flow				
	End device		Test equ	iipment
Interworking POTS				
Off hook				
Dial number		Start timer A	→ INVITE	
		Start timer B		
	Т	imeout timer B		
			€ 200 OK INVI	TE
			200 OK IIII	
ISDN interworking				
SETUP	→	Start timer A	→ INVITE	
	7	Start timer B		
	_			
	Т	imeout timer B		
			E 200 OK INVI	

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. Annex A of [IETF RFC 3		Selection expression PICS 5.1.2/1
Test purpose		I		
100 received, the INVITE i	s not repeated			
	an INVITE client transaction is	in the Proceed	ing state, does n	ot 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 Annex [IETF]	17.1.1 A of		Selection expression PICS 5.1.2/1
Test purpose					
486 received, timer D	is started ACK is repeated				
					receipt of final responses that second expires if an unreliable
Message flow					
	End device			Test eq	uipment
Interworking POTS					
Off hook					
Dial number			→	INVITE	
	S	tart timer D	←	486 Busy Her	re
			→	ACK	
			←	486 Busy Her	re
			→	ACK	
ISDN interworking					
SETUP	→		→	INVITE	
	S	tart timer D	←	486 Busy Her	re
			→	ACK	
			←	486 Busy Her	re
			→	ACK	

TSS Orig_Timers	TP_205_009	Reference section 17.1. Annex A of	1.1 and	Selection expression PICS 5.1.2/2
		[IETF RFC 3	261]	
Test purpose				
486 received, timer D is sto	arted ACK is not repeated			
	an INVITE client transaction i bes not repeat its ACK request if			eccipt of a final response that
Message flow				
0				
End	device		Test equ	ipment
End Interworking POTS	device		Test equ	ipment
	device		Test equ	ipment
Interworking POTS	device	→	Test equ INVITE	ipment
Interworking POTS Off hook	device Start tin	-	-	-
Interworking POTS Off hook		-	INVITE	-
Interworking POTS Off hook		ner D 🗲	INVITE 486 Busy Here	-

ISDN interworking		
SETUP	→	→ INVITE
	Start time	D 🗲 486 Busy Here
		→ ACK
		← 486 Busy Here

TSS Orig_Timers	TP_205_010			Selection expression PICS 5.1.2/1	
Test purpose					
486 repeated Via wit	h different values, ACK is not se	ent			
	when an INVITE client transac ranch parameter value, does no le transport is used.				
SIP header values					
486 1					
Via: <any td="" via<=""><td>URI>;branch=<any branch="" td="" valu<=""><td>ie 1></td><td></td><td></td></any></td></any>	URI>;branch= <any branch="" td="" valu<=""><td>ie 1></td><td></td><td></td></any>	ie 1>			
486 2					
	URI>;branch= <any branch="" td="" valu<=""><td>ie 2></td><td></td><td></td></any>	ie 2>			
Message flow					
	End device		Test equipment		
Interworking POTS	5				
Off hook					
Dial number			→ INVITE		
	St	art timer D	← 486 Busy He	re 1	
			→ ACK		
			← 486 Busy He	re 2	
ISDN interworking					
SETUP	→		→ INVITE		
	St	art timer D	← 486 Busy He	re 1	
			➔ ACK		
			← 486 Busy He	re 2	
	Apply	v post test routin	le		

TSS Orig_Timers	TP_205_011	section 1 Annex A	Reference section 13.2.2.4 and Annex A of [IETF RFC 3261]		Selection expression		
Test purpose							
200 OK received, ACK	is sent						
	nen an INVITE client transa onse sends an ACK request				, on receipt of a retransmitted		
Message flow							
E	nd device			Test equ	ipment		
Interworking POTS							
Off hook							
Dial number			→	INVITE			
	St	tart timer 64*T1	←	 200 OK INVITE 			
			→	ACK			
			←	200 OK INVI	ГЕ		
			→	ACK			
ISDN interworking							
SETUP	→		→	INVITE			
	St	tart timer 64*T1	←	200 OK INVI	ГЕ		
			→	ACK			
			←	200 OK INVI	ΓE		
			× →	ACK			
	→ ACK Apply post test routine						

TSS Orig_Timers	TP_205_012	Reference section 13.2.2 Annex A of [IETF RFC 3		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 equ	ipment		
Interworking POTS						
0.001 1						
Off hook						
Dial number		→	INVITE			
	Start timer	-	INVITE 200 OK INVI	ГЕ		
	Start timer	-		ГЕ		
	Start timer Timeout	r 64*T1	200 OK INVI	ГЕ		
		r 64*T1	200 OK INVI			

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

TSS Orig_Timers	TP_205_013	section Annex	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	l		
Ensure that the IUT, hav value expires if an unreli			l dialo	og, repeats its re	equest after timer E set to T1
Message flow					
Er	nd device			Test equ	iipment
Interworking POTS					
Off hook					
Dial number			→	INVITE	
			←	180 Ringing	
			←	200 OK INVI	TE
			→	ACK	
On hook		Start timer E	→	BYE	
		Timeout timer E	→	BYE	
ISDN interworking					
SETUP	→		→	INVITE	
ALERTING	+		←	180 Ringing	
CONNECT	+		←	200 OK INVI	TE
			→	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	→	ВҮЕ
		Timeout timer E Start timer E 4*T1+T2	→	ВҮЕ
		Timeout timer E	→	BYE
		Apply post test routine	e	

TSS Orig_Timers	TP_205_016	Annex	17.1.2.2 and	Selection expression PICS 5.1.2/1
Test purpose <i>Timeout timer F, the</i>	BYE is not repeated			
Ensure that the IUT d unreliable transport is		est on an establishe	d dialogue after time	er F set to 64*T1 expires, if an
SIP header values				
Message flow				
	End device		Test eq	uipment
Interworking POTS				
Off hook				
Dial number			→ INVITE	
			← 180 Ringing	
			← 200 OK INV	ITE
			➔ ACK	
On hook		Start timer F	→ BYE	
		Timeout timer F		
ISDN interworking				
SETUP	→		➔ INVITE	
ALERTING	+		← 180 Ringing	
CONNECT	+		← 200 OK INV	ITE
			→ 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 set to T1 value expires.	a BYE client transaction is	in the Proceedin	g state, repeats	its BYE request after timer E		
SIP header values						
Message flow						
End	End device Test equipment					
Interworking POTS						
Off hook						
Dial number		→	INVITE			
		÷	180 Ringing			
On hook	Star	t timer E \rightarrow	BYE			
	Time	but timer E \rightarrow	BYE			
ISDN interworking						
SETUP	→	→	INVITE			
ALERTING	÷	÷	180 Ringing			
DISCONNECT	→ Star	t timer E →	BYE			
	Time	out timer E 🗕 🗕	BYE			

TSS Orig_TimersTP_205_018Reference section 17.1.2.2 and Annex A of [IETF RFC 3261]Selection expression PICS 5.2/18	Annex A of	ession
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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	→ BYE	
	Start timer E T2	7 DIE	
	Timeout timer E	→ BYE	
ISDN interworking			
SETUP	→	→ INVITE	
ALERTING	← Start timer E	→ BYE	
DISCONNECT	\rightarrow Timeout timer E	→ BYE	
	Start timer E T2	→ BYE	
	Timeout timer E	→ BYE	

TSS Orig_Timers	TP_205_019	Anne		2.2 and 3261]	Selection expression PICS 5.2/18
Test purpose					
BYE is nit repeated after til	meout timer F				
Ensure that the IUT, when established dialogue, after			oceeding	g state , does no	t repeat a BYE request on an
SIP header values					
Message flow					
End	device			Test equ	ipment
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. Annex A of [IETF RFC 3		Selection expression PICS 5.2/18			
Test purpose							
Timeout timer F, the BYE i	s 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.						
Message flow							
End	device		Test equ	ipment			
Interworking POTS							
Off hook							
Dial number		→	INVITE				
On hook	Start tir	mer F 🗕 🗲	BYE				
	Timeout	timer F					

ISDN interworking				
SETUP	→		→	INVITE
RELEASE	→	Start timer F	→	BYE
		Timeout timer F		
7.2.2.5 Abnormal situations				

TSS	TP_206_001	Referen		Selection expression
Orig_Abnormal_situations		17.1.1.3	13.2.2.3, 17.1.1.2, and Figure 5, of FC 3261]	
Test numeros			10 5201]	
Test purpose 404 received in calling state	sotting of Call_ID	From header	es and Request_URI	
+0+ received in culling state	setting of Cutt-1D,	1 rom neuuer	s una Requesi-ORI	
Ensure that when the client the	ansaction is in the (Calling state	that on receipt of a Not F	Found (404 Not Found) response,
			and Request-URI as in	the original INVITE request and
the same Tag in the To head	er as in this response	е.		
SIP header values				
INVITE 1 sip: [Request UR]				
Call-ID: <any call<="" invite="" td=""><td></td><td></td><td></td><td></td></any>				
From: <any invite="" uri="">;</any>	tag= <any invite="" tag<="" td=""><td>></td><td></td><td></td></any>	>		
ACK 1 sip: [Request URI] S	IP/2.0			
Call-ID: <any call<="" invite="" td=""><td></td><td></td><td></td><td></td></any>				
From: <any invite="" uri="">;</any>	tag= <any invite="" tag<="" td=""><td>></td><td></td><td></td></any>	>		
Message flow				
End d	evice		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
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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 call="" id="" invite=""></any>		
From: <any invite="" uri="">;tag=<any invite="" tag=""></any></any>		
404 1		
To: <any 404="" uri="">;tag=<any 404="" tag=""></any></any>		
ACK 1 sip: [Request URI] SIP/2.0		
Call-ID: <any call="" id="" invite=""></any>		
From: <any invite="" uri="">;tag=<any invite="" tag=""></any></any>		
To: <any 404="" uri="">;tag=<any 404="" tag=""></any></any>		
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, Fre	om headers and Request-URI						
request is sent with the same	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 call<="" invite="" td=""><td colspan="8">Call-ID: <any call="" id="" invite=""></any></td></any>	Call-ID: <any call="" id="" invite=""></any>							
From: <any invite="" uri="">;tag=<any invite="" tag=""></any></any>								
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 te	st routine	

TSS Orig_Abnormal_situations	TP_206_004	17.1.1.3	ce 13.2.2.3, 17.1.1.2, and Figure 5 of FFC 3261]	Selection expression		
Test purpose 480 received in calling state	; setting of Call-ID, F	rom heade	rs and Request-URI			
Ensure that when the client (480 Temporarily Unavailab Request-URI as in the origin	ole) response, an ACK	request is	sent with the same Call-II	D, From headers and		
SIP header values						
INVITE 1 sip: [Request UR	-					
Call-ID: <any cal<="" invite="" td=""><td></td><td></td><td></td><td></td></any>						
From: <any invite="" uri=""></any>	;tag= <any invite="" tag=""></any>					
480 1						
To: <any 480="" uri="">;tag=</any>	<any 480="" tag=""></any>					
Call-ID: <any call<br="" invite="">From: <any invite="" uri=""> To: <any 480="" uri="">;tag= Message flow</any></any></any>	;tag= <any invite="" tag=""></any>					
End d	levice		Test e	quipment		
Interworking POTS						
Off hook						
Dial number		→	INVITE 1			
← 480 Temporarily Unavailable						
	\rightarrow ACK 1					
		7	ACK 1			
ISDN interworking		7	ACK 1			
ISDN interworking		→				
SETUP		-	INVITE 1	ailable		
-		→		ailable		

TSS Orig_Abnormal_situations	TP_206_005	17.1.1.3	ce 13.2.2.3, 17.1.1.2, and Figure 5 of FFC 3261]	Selection expression
Test purpose 486 received in calling state	; setting of Call-ID, F1	om heade	rs and Request-URI	
	the same Call-ID, Fro			lere (486 Busy Here) response original INVITE request and
SIP header values				
INVITE 1 sip: [Request UR]	[] SIP/2.0			
Call-ID: <any call<="" invite="" td=""><td></td><td></td><td></td><td></td></any>				
From: <any invite="" uri="">;</any>	tag= <any invite="" tag=""></any>			
486 1				
To: <any 486="" uri="">;tag=</any>	<any 486="" tag=""></any>			
ACK 1 sip: [Request URI] S Call-ID: <any call<br="" invite="">From: <any invite="" uri="">; To: <any 486="" uri="">;tag=</any></any></any>	ID> tag= <any invite="" tag=""></any>			
Message flow	_		_	
	evice		Test equ	uipment
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_00	Reference sections 13.2.2.3, 17.1.1.2, 17.1.1.3 and Figure 5 of [IETF RFC 3261]	Selection expression
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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 call="" id="" invite=""></any>		
From: <any invite="" uri="">;tag=<any invite="" tag=""></any></any>		
500 1		
To: <any 500="" uri="">;tag=<any 500="" tag=""></any></any>		
ACK 1 sip: [Request URI] SIP/2.0		
Call-ID: <any call="" id="" invite=""></any>		
From: <any invite="" uri="">;tag=<any invite="" tag=""></any></any>		
To: <any 500="" uri="">;tag=<any 500="" tag=""></any></any>		
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 Orig_Abnormal_situations	TP_206_007	Reference sections 13.2.2.3, 17.1.1.2, 17.1.1.3 and Figure 5 of [IETF RFC 3261]	Selection expression
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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 Orig_Abnormal_situations	TP_206_008	17.1.1.3	ce 13.2.2.3, 17.1.1.2, and Figure 5 of FC 3261]	Selection expression
				cline (603 Decline) response an
ACK request is sent with the same Tag in the To header a		neaders and	l Request-URI as in the o	riginal INVITE request and the
SIP header values	s in this response.			
INVITE 1 sip: [Request UR]] SIP/2.0			
Call-ID: <any call<="" invite="" td=""><td></td><td></td><td></td><td></td></any>				
From: <any invite="" uri="">;</any>	tag= <any invite="" tag=""></any>			
603				
To: <any 603="" uri="">;tag=</any>	<any 603="" tag=""></any>			
ACK 1 sip: [Request URI] S				
Call-ID: <any call<="" invite="" td=""><td></td><td></td><td></td><td></td></any>				
From: <any invite="" uri="">; To: <any 603="" uri="">;tag=</any></any>	2 2			
Message flow	, , , , , , , , , ,			
End d	evice		Test e	quipment
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		÷		

TSS Orig_Abnormal_situations	TP_206_009	Referen	ce 13.2.2.3, 17.1.1.2,	Selection expression
6		17.1.1.3	and Figure 5 of FC 3261]	
Test purpose				
603 received. The same bran	ich parameter value	is sent in the	ACK as received in the	e initial INVITE request
	nternal Error (500 Se	erver Internal		a Busy Everywhere (600 Busy different branch parameter value
SIP header values				
INVITE:				
Via: SIP 2.0 <transport></transport>	<any uri="" via="">; bra</any>	nch= <any inv<="" td=""><td>vite branch value></td><td></td></any>	vite branch value>	
ACK:				
Via: SIP 2.0 <transport></transport>	<any uri="" via="">; bra</any>	nch=< any in	vite branch value >	
Message flow				
End device			Test	equipment
Interworking POTS				
Off hook				
Dial number		→	INVITE	
		←	603 Decline	
		→	ACK	
ISDN interworking				
SETUP		→	INVITE	
DISCONNECT		←	603 Decline	
RELEASE		→	ACK	
RELEASE COMPLETE		+		

TSS Orig_Abnormal_situations	TP_206_010	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]	Selection expression
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.

SIP header values		
INVITE:		
INVITE sip: [invite Request URI] SIP/2.0		
Call-ID: <any call="" id="" invite=""></any>		
From: <any invite="" uri="">;tag=<any invite="" tag=""></any></any>		
699:		
To: <any uri="">;tag=<any 699="" tag="" value=""></any></any>		
ACK:		
INVITE sip: [invite Request URI] SIP/2.0		
Call-ID: <any call="" id="" invite=""></any>		
From: <any invite="" uri="">;tag=<any invite="" tag=""></any></any>		
To: <any uri="">;tag=<any 699="" tag="" value=""></any></any>		
Message flow		
End device		Test equipment
Interworking POTS		
Off hook		
Dial number	→	INVITE 1
	←	699 Unknown
	→	ACK 1
ISDN intomyonking		
ISDN interworking	→	INVITE 1
SETUP		
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=""></not>						

'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 te	est 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	TP_301_001	Reference	Selection expression
Term_Establishment_of_		sections 8, 8.2, and	
an_early_dialogue		13.3.1.1 of	
		[IETF RFC 3261]	
Test purpose			
The IUT sends a 180 Ringi	ing		
Ensure that the IUT on rec	eipt of an INVITE reques	t, sends a provisional (180 Ringi	ng) response.
Message flow	<u>r</u>	,,	8, 1,
Test ec	Juipment	En	nd device
Interworking POTS			
INVITE	→		
180 Ringing	÷	Ringing	
ISDN interworking			
INVITE	→	→ SETUP	
180 Ringing	+	← ALERTIN	G
	Appl	y post test routine	
TSS	TP_301_002	Reference	Selection expression
Term_Establishment_of_		sections 8, 8.2, and	
an_early_dialogue		13.3.1.1 of	
		[IETF RFC 3261]	
Test purpose			
The IUT sends a 200 OK I	NVITE		
Ensure that the IUT on rec	eipt of an INVITE reques	t, sends a Success (200 OK) resp	onse.

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	TD 201 002	Reference	Colortian annuacion	
Term_Establishment_of_	TP_301_003	section 8 and 8.2.6.1 of	Selection expression	
an_early_dialogue		[IETF RFC 3261]		
Test purpose	• .1 • •	7		
The Timestamp header value	ue in the response is incl	reasea		
		nest with a Timestamp header, when neader with an increased value of		
SIP header values				
INVITE:				
Timestamp: <timestamp< td=""><td>p_value></td><td></td><td></td></timestamp<>	p_value>			
100:				
Timestamp: <timestamp< td=""><td>p_value + any value></td><td></td><td></td></timestamp<>	p_value + any value>			
Message flow				
Test eq	uipment	End	device	
Interworking POTS				
INVITE	→	Ringing		
100 Trying	+			
ISDN interworking				
INVITE	→	→ SETUP		
	÷			
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 A	ACK			
Ensure that the IUT on rece initial offer session descrip		uding no message body, includ	es in its first 2xx response an		
SIP header values					
200 OK:					
SDP	SDP				
m – line offer					
a – line offer	a – line offer				
ACK	ACK				
SDP					
m – line answer	m – line answer				
a – line answer					

Message flow		
Test equi	pment	End device
Interworking POTS		
INVITE	→	Ringing
200 OK INVITE	+	On hook
ACK	→	
ISDN interworking		
INVITE	→	→ SETUP
200 OK INVITE	÷	← CONNECT
ACK	→	
	Apply pos	st 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 IN	IVITE, initial answer in t	he 200 OK INVITE	
Ensure that the IUT on reco includes the answer in its f	eipt of an INVITE reques irst 2xx response in a ses	t including an initial offer sessio sion description.	on description in its message body,
SIP header values			
INVITE:			
SDP			
m – line offer			
a – line offer			
200 OK:			
SDP			
m – line answer			
a – line answer			
Message flow			
Test eq	Test equipment End device		
Interworking POTS			
INVITE	→	Ringing	
200 OK INVITE	+	On hook	
ACK	→		
ISDN interworking			
INVITE	→	→ SETUP	
200 OK INVITE	÷	← CONNEC	CT
ACK	→		
	Appl	ly post test routine	

TSS Term_Establishment_of_	TP_301_006	Reference section 8.2.6.2 of	Selection expression
an_early_dialogue		[IETF RFC 3261]	
Test purpose			
From, Call-ID, CSeq and	Via headers copy from th	e INVITE	
Ensure that the IUT on red From, Call-ID, CSeq and V			99) response including the headers
SIP header values			
INVITE:			
From: <from_value_invite< td=""><td>></td><td></td><td></td></from_value_invite<>	>		
Call-ID: <callid_value_inv< td=""><td>vite></td><td></td><td></td></callid_value_inv<>	vite>		
CSeq: <cseq_value_invite:< td=""><td>></td><td></td><td></td></cseq_value_invite:<>	>		
Via: <via_value_invite></via_value_invite>			
180:			
From: <from_value_invite< td=""><td>></td><td></td><td></td></from_value_invite<>	>		
Call-ID: <callid_value_inv< td=""><td>vite></td><td></td><td></td></callid_value_inv<>	vite>		
CSeq: <cseq_value_invite< td=""><td>></td><td></td><td></td></cseq_value_invite<>	>		
Via: <via_value_invite></via_value_invite>			
Message flow			
Test ed	quipment		End device
Interworking POTS			
INVITE	→	Ringing	
180 Ringing	←		
	Арр	ly post test routine	
ISDN interworking			
INVITE	→	→ SETUP	
	→ ←		
180 Ringing		← ALERT	ING
	Арр	ly post test routine	
TSS	TP_301_007	Reference	Selection expression

TSS	TP_301_007	Reference	Selection expression		
Term_Establishment_of_		sections 8.2.6.2, 12.2.2 and			
an_early_dialogue		13.3.1 of [IETF RFC 3261]			
Test purpose					
To tag is sent in the respon	se				
Ensure that the IUT on rece	eipt of an INVITE request with	no TAG set on the To header, s	sends a provisional (101-199)		
response including the sam	e URI and an additional TAG f	or the To header.			
SIP header values	SIP header values				
INVITE:					
To: <sip:to-uri></sip:to-uri>					
180:					
To: <sip:to-uri>;tag=to_tag</sip:to-uri>					

Message flow		
Test equipment		End device
Interworking POTS		
INVITE	→	Ringing
180 Ringing	÷	
	Apply post test routine	
ISDN interworking		
INVITE	\rightarrow \rightarrow	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 reque	st		
Ensure that the IUT on rece	pipt of an INVITE reques	st with a TAG set on the To head	der, either:
• sends a provisional (10 (recommended for rob		ng the same URI and the same T	AG for the To header
		action does not exist (481 Call/T	ransaction does not exist).
SIP header values	•	· · · · · · · · · · · · · · · · · · ·	
INVITE:			
To: <sip:<i>to_uri_value>;</sip:<i>	tag=to_tag_value		
180:			
To: <sip:to_uri_value>;</sip:to_uri_value>	tag=to_tag_value		
Message flow			
Test eq	uipment	Ε	and device
Interworking POTS			
INVITE	→	Ringing	
CASE A			
180 Ringing	+		
	Appl	ly post test routine	
CASE B			
481 Call/Transaction does n	not exist 🗧 🗲		
ACK	→		
ISDN interworking			
INVITE	→	→ SETUP	
CASE A			
180 Ringing	+	← ALERTI	NG
		ly post test routine	
CASE B			
481 Call/Transaction does n	not exist 🗧 🗲		
ACK	→		

TSS Term_Establishment_of_	TP_301_009	Reference section 12.1.1 of	Selection expression
an_early_dialogue		[IETF RFC 3261]	
Test purpose			
Contact header in the resp	onse.		
Ensure that the IUT on rece header.	ipt of an INVITE request, ser	ds a provisional (101-199) res	ponse including a single Contact
SIP header values			
180:			
Contact: <sip: contact_<="" td=""><td>value></td><td></td><td></td></sip:>	value>		
Message flow			
Test eq	Test equipment End device		
Interworking POTS			
INVITE	→	Ringing	
180 Ringing	+		
	Apply po	ost test routine	
ISDN interworking			
INVITE	→	→ SETUP	
180 Ringing	+	← ALERTIN	G
	Apply po	ost test routine	
TSS	TP 301 010	Reference	Selection expression

155	TP_301_010	Reference	Selection expression
Term_Establishment_of_		section 12.1.1 of	
an_early_dialogue		[IETF RFC 3261]	
Test purpose			
Record-Route header copie	ed from the INVITE requ	est into the response	
		st including a Record-Route head	
response including a Recor	d-Route header copy fro	m the INVITE request, in the same	e order.
SIP header values			
INVITE:			
Record-Route: <sip:inv< td=""><td><i>ite_record_route></i>;lr</td><td></td><td></td></sip:inv<>	<i>ite_record_route></i> ;lr		
180:			
Record-Route: <sip:inv< td=""><td><i>ite_record_route</i>>;lr</td><td></td><td></td></sip:inv<>	<i>ite_record_route</i> >;lr		
Message flow			
Test eq	uipment	En	d device
Interworking POTS			
INVITE	→	Ringing	
180 Ringing	+		
	App	ly post test routine	
	r r		
ISDN interworking			
INVITE	→	→ SETUP	
180 Ringing	÷ +	← ALERTIN	G
100 Kinging	T		U
		ly post test routine	

TSS	TP_301_011	Reference	Selection expression
Term_Establishment_of_		section 12.1.1of	_
an_early_dialogue		[IETF RFC 3261]	
Test purpose			
Record-Route header with	unknown parameter		
			der with parameters that it does not
understand, sends a provis with the unknown paramet		including a Record-Route head	er copy from the INVITE request
SIP header values			
INVITE:			
Record-Route: <sip:red< td=""><td>cord-route_value_invite;u</td><td>nknown=etsi></td><td></td></sip:red<>	cord-route_value_invite;u	nknown=etsi>	
180:			
Record-Route: <sip:rec< td=""><td>cord-route_value_invite;u</td><td>nknown=etsi></td><td></td></sip:rec<>	cord-route_value_invite;u	nknown=etsi>	
Message flow			
Test ed	quipment	E	End device
Interworking POTS			
INVITE	→	Ringing	
180 Ringing	+		
	Appl	y post test routine	
ISDN interworking			
INVITE	→	→ SETUP	
180 Ringing	←	← ALERTI	NG
		y post test routine	

TSS	TP_301_012	Reference	Selection expression
Term_Establishment_of_		section 12.1.1 of	-
an_early_dialogue		[IETF RFC 3261]	
Test purpose			
From header without "tag	,,		
response including a From		t including From header with	out tag, sends a provisional (101-199)
SIP header values			
INVITE:			
From: <sip:from_value_< td=""><td>_invite></td><td></td><td></td></sip:from_value_<>	_invite>		
180:			
From: <sip:from_value_< td=""><td>_invite></td><td></td><td></td></sip:from_value_<>	_invite>		
Message flow			
Test eq	uipment		End device
Interworking POTS			
INVITE	→	Ringir	ng
180 Ringing	+		
	Арр	ly post test routine	
ISDN interworking			
INVITE	→	→ SETU	Р
180 Ringing	÷	← ALER	TING
	App	ly 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 INVIT	E received, the previous	sent response is repeated	
		ction is in the Proceeding state, on re rameter and sent-by value in the topr	
SIP header values			
INVITE1:			
•	alue;branch=any_invite1		
• •	n_value;tag=any_invite1	_from_tag_value	
To: <i>any_invite1_to_val</i>			
Call-ID: <i>any_invite1_ca</i> CSeq: <i>any_invite1_csed</i>			
CSeq. uny_invite1_cseq	1		
INVITE2:			
Via: any_invite1_via_v	alue;branch=any_invite1	_branch_value	
From: <i>any_invite1_from</i>	n_value;tag=any_invite1	_from_tag_value	
To: any_invite1_to_val	ue		
Call-ID: <i>any_invite1_ce</i>			
CSeq: any_invite1_csec	9		
Message flow			
Test eq	luipment	End d	levice
Interworking POTS			
INVITE1	→	Ringing	
180 Ringing	←		
INVITE2	→		
180 Ringing	←		
	Арр	ly post test routine	
ISDN interworking			
INVITE1	→	→ SETUP	
180 Ringing	÷	← ALERTING	
INVITE2	→		
180 Ringing	÷		
	Арр	ly post test routine	

TSS	TP_301_014	Reference	Selection expression
Term_Establishment_of_		section 17.2.1 and	_
an_early_dialogue		17.2.3 of [IETF RFC 3261]	

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_va	ulue;branch= z9hG4bK any_in	vite1_branch_value
From: <i>any_invite1_from</i>	n_value;tag=any_invite1_from	_tag_value
To: any_invite1_to_valu	ie	
Call-ID: any_invite1_ca	ll-id_value	
CSeq: any_invite1_cseq		
INVITE2:		
Via: any_invite1_via_va	ulue;branch= z9hG4bK_any_i	nvite1_branch_value
Via: any_invite2_via_v	alue	
From: <i>any_invite1_from</i>	_value;tag=any_invite1_from	_tag_value
To: any_invite1_to_vali		
Call-ID: any_invite1_ca		
CSeq: any_invite1_cseq		
Message flow		
Test eq	uipment	End device
Interworking POTS		
INVITE	→	Ringing
180 Ringing	←	
INVITE2	→	
	,	
180 Ringing		· · · · ·
	Apply pos	t test routine
ISDN interworking		
INVITE	→	→ SETUP
180 Ringing	←	← ALERTING
INVITE2	→	
180 Ringing	÷	
	Apply pos	t test routine

TSS	TP_301_015	Reference	Selection expression
Term_Establishment_of_		section 17.2.1 and	
an_early_dialogue		17.2.3 of [IETF RFC 3261]	

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: <i>any_invite1_via_va</i>	lue;branch= z9hG4bK any_inv	ite1_branch_value	
• •	_value;tag=any_invite1_from_	tag_value	
To: any_invite1_to_valu			
Call-ID: <i>any_invite1_ca</i>			
CSeq: any_invite1_cseq			
INVITE2:			
Via: any_invite1_via_va	lue;branch= z9hG4bK any_inv	ite1_branch_value	
Via: any_invite1_via_va	lue;branch= any_invite2_bran	ch_value	
From: <i>any_invite1_from</i>	_value;tag=any_invite1_from_	tag_value	
To: any_invite1_to_valu			
Call-ID: <i>any_invite1_ca</i>			
CSeq: any_invite1_cseq			
Message flow			
Test equ	upment	End device	
Interworking POTS			
INVITE	→	Ringing	
180 Ringing	+		
100 Kinging			
100 Kinging			
INVITE2	→		
INVITE2	→ ←		
	``	test routine	
INVITE2 180 Ringing	``	test routine	
INVITE2 180 Ringing ISDN interworking	``	test routine	
INVITE2	``	test routine → SETUP	
INVITE2 180 Ringing ISDN interworking	← Apply post		
INVITE2 180 Ringing ISDN interworking INVITE	← Apply post →	→ SETUP	
INVITE2 180 Ringing ISDN interworking INVITE 180 Ringing	← Apply post → ←	→ SETUP	

TSS	TP_301_016	Reference	Selection expression
Term_Establishment_of_		section 22.2 of	
an_early_dialogue		[IETF RFC 3261]	

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 <i>INVITE without Authoriza</i> <i>values</i>	tion header received a 4	01 is sent containing a WWW-	Authenticate header with proper
	orized) response, containing		orization header field, sends an r including proper value for realm
SIP header values			
401:			
WWW-Authenticate: I	Digest realm="[any value]	',nonce="[any value]",algorithn	n=MD5,qop="auth"
Message flow			
Test ec	luipment	E	nd device
Interworking POTS			
INVITE	→		
401 Unauthorized	+		
ACK	→		
	Apply	y post test routine	
ISDN interworking			
INVITE	→		
401 Unauthorized	+		
ACK	→		
	Apply	y 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 <i>INVITE with valid Authoriz</i>	zation header received, a	200 OK INVITE is sent		
		01 Unauthorized) response to an der field, sends a Success (200 O	INVITE request, on receipt of a K) response.	
SIP header values				
401:				
WWW-Authenticate: D	pigest 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 eq	uipment	En	End device	
Interworking POTS				
INVITE 1	→			
401 Unauthorized	←			
ACK	→			
INVITE 2	→	Ringing		
200 OK INVITE	←	Off hook		
ACK	→			
	Appl	y post test routine		
ISDN interworking				
INVITE	→			
401 Unauthorized	+			
ACK	→			
INVITE 2	→	→ SETUP		
200 OK INVITE	+	← CONNEC	Т	
ACK	→			
	Appl	y post test routine		

TSS	TP_301_019	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.4.1 of	PICS: 5.2/14 and 5.2/15
an_early_dialogue		[ETSI TS 124 229]	

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.

стр р 4 1

SIP header values		
INVITE:		
Supported: precondition,100r	e1	
Supported. precondition, roor		
SDP a=curr:qos local none		
a=curr:qos remote none		
a=des:qos mandatory/opti		
a=des:qos mandatory/opti	onal remote sendrecv	
183:		
Require: precondition,100rel		
SDP a=curr:qos local none		
a=curr:qos remote none		
a=des:qos mandatory/opti	onal local sendrecv	
a=des:qos mandatory/opti		
a=conf:qos remote sendre		
UPDATE		
SDP a=curr:qos local sendrecv		
a=curr:qos remote none a=des:qos mandatory/opti	onal local sandroov	
a=des:qos mandatory/opti a=des:qos mandatory/opti		
a-des.qos mandatory/opti	sharrenote sendreev	
200 OK UPDATE		
SDP a=curr:qos local sendrecv		
a=curr:qos remote sendred	2V	
a=des:qos mandatory/opti	onal local sendrecv	
a=des:qos mandatory/opti	onal remote sendrecv	
Message flow		
Test equipm	ent	End device
Interworking POTS		
INVITE	→	
183 Session Progress	÷	
PRACK	→	
200 OK PRACK	÷	
UPDATE	→	
200 OK UPDATE	+	
180 Ringing	÷	Ringing
	Apply po	ost test routine
ICDN intone altico		
ISDN interworking		
INVITE	→	
183 Session Progress	÷	
PRACK	→	
200 OK PRACK	+	
UPDATE	→	
200 OK UPDATE	÷	
		→ SETUP
180 Ringing	←	← ALERTING
180 Ringing	← Annly no	← ALERTING ost 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 <i>INVITE with support for pr</i>	reconditions received. Pr	reconditions supported.	
the Supported header, the S	UT sends a (reliable) 18	request containing the precondition 3 session Progress or 180 Ringing d. A Require header requesting pr	and the SDP that does not
SIP header values			
INVITE:			
Supported: precondition	n,100rel		
a=des:qos mandator			
183/180:			
Message flow			
	luipment	En	d device
Interworking POTS			
INVITE	→		
CASE A	-		
183 Session Progress	+		
CASE B			
180 Ringing	←	Ringing	
100 Kinging		ly post test routine	
	**PP	- post sele routine	
ISDN interworking			
INVITE	→	→ SETUP	
180 Ringing	÷	← ALERTIN	G
	Арр	ly 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	
L		-	

Term Establishment_of	TSS	TP_301_022	Reference	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 UPDAT request or ACK request SIP header values INVITE: SUP offer CASE A ACK SDP answer CASE A ACK SDP answer End device Interworking POTS Interworking POTS INVITE ⇒ 183 Session Progress € Ringing PRACK 200 OK INVITE ⇒ 200 OK INVITE ⇒ 200 OK UPDATE ⇒ 200 OK UPDATE ⇒ 200 OK UPDATE ⇒ 200 OK UPDATE ⇒ 200 OK INVITE 4CK ⇒ 200 OK UPDATE >		11_301_022	subclause 6.3.2.4 of	Selection expression
IVVITE vithout SDP offer received. Ensure that the IUT upon receipt of an INVITE request without an SDP offer and transport of reliable provisional 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 \Rightarrow 183 Session Progress \leftarrow Ringing PRACK \Rightarrow 200 OK INVITE \leftarrow Off hook ACK \Rightarrow CASE B UPDATE \leftarrow Off hook ACK \Rightarrow ESDN interworking INVITE \Rightarrow ALERTING PRACK \Rightarrow SETUP	an_early_dialogue		[ETSI TS 183 043]	
Ensure that the IUT upon receipt of an INVITE request without an SDP offer and transport of reliable provisional request or ACK request SIP header values INVITE: Supported: 100rel I83 SDP offer CASE A ACK SDP answer CASE B UPDATE SDP answer Message flow Test equipment Fat equipment Fat equipment Fat equipment CASE A COSE B UPDATE ACK				
responses, send a reliable 183 Session Progress containing an SDP offer. The SDP answer is received in an UPDAT request or ACK request 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 PRACK € CASE B UPDATE ⇒ 200 OK INVITE ♦ ACK ⇒ SDP answer CASE B UPDATE ÷ 200 OK UPDATE € SDP answer SDN interworking INVITE ⇒ 35 SETUP	INVITE without SDP offer	received.		
INVITE: Supported: 100rel I83 SDP offer CASE A ACK SDP answer CASE B UPDATE SDP answer Messed flow Test equipment Test	responses, send a reliable 1			
Supported: 100rel 183 SDP offer CASE A ACK SDP answer CASE B UPDATE SDP answer Message flow Test equipment Fest equipment ACK 200 OK PRACK CASE A 200 OK INVITE ACK POATE 200 OK INVITE Apply post test routine SDN interworking INVITE Apply post test routine INVITE Action Progress Factor Factor PACK	SIP header values			
183 SDP offer CASE A ACK SDP answer CASE B UPDATE SDP answer Message flow Test equipment Fast equipment Proversing CASE A 200 OK INVITE ACK ACK ACK ACK ACK Apply post test routine Supproversing INVITE Apply post test routine Supple post test	INVITE:			
SDP offer CASE A ACK SDP answer CASE B UPDATE SDP answer Message flow Message	Supported: 100rel			
SDP offer CASE A ACK SDP answer CASE B UPDATE SDP answer Message flow Test equipment End device Message flow Message f	100			
CASE A ACK SDP answer CASE B UPDATE SDP answer Message flow End device INUTE 183 Session Progress 200 OK PRACK 200 OK INVITE 4CK 200 OK INVITE 5 4CK 4CK 5 4CAK 4 4 4 4 4 5 4 4 5 4 4 5 5 5 4 5 5 5 5 <				
ACK SDP answer CASE B UPDATE SDP answer Message flow Test equipment End device Interworking POTS INVITE \Rightarrow 183 Session Progress \Leftarrow Ringing PRACK \Rightarrow 200 OK PRACK \Leftarrow CASE A 200 OK INVITE \Leftarrow ACK \Rightarrow 200 OK INVITE \Leftarrow ACK \Rightarrow DIPDATE \Rightarrow 200 OK UPDATE \Rightarrow 200 OK UPDATE \Rightarrow SDN interworking INVITE \Rightarrow \Rightarrow Apply post test routine SDN interworking INVITE \Rightarrow \Rightarrow SETUP 183 Session Progress \Leftarrow \Leftarrow ALERTING PRACK \Rightarrow	SDP offer			
ACK SDP answer CASE B UPDATE SDP answer Message flow Test equipment End device Interworking POTS INVITE \Rightarrow 183 Session Progress \Leftarrow Ringing PRACK \Rightarrow 200 OK PRACK \Leftarrow CASE A 200 OK INVITE \Leftarrow ACK \Rightarrow 200 OK INVITE \Leftarrow ACK \Rightarrow DIPDATE \Rightarrow 200 OK UPDATE \Rightarrow 200 OK UPDATE \Rightarrow SDN interworking INVITE \Rightarrow \Rightarrow Apply post test routine SDN interworking INVITE \Rightarrow \Rightarrow SETUP 183 Session Progress \Leftarrow \Leftarrow ALERTING PRACK \Rightarrow	CASE A			
CASE B UPDATE SDP answer Subset Message flow End device Message flow End device INVITE > INVITE > 183 Session Progress € 200 OK PRACK > 200 OK INVITE € 200 OK UPDATE € UPDATE > 200 OK UPDATE € Xappy post test routine F INVITE > > INVITE >				
UPDATE SDP answer Sease flow Fact equipment Test equipment NVITE NVITE A ACK 200 OK PRACK Off hook ACK 200 OK INVITE ACK A Off hook CASE B VIPDATE UPDATE Apply post test routine SETUP 183 Session Progress A FUP 183 Session Progress A A SETUP 183 Session Progress A A LERTING	SDP answer			
UPDATE SDP answer Server Message flow End device Test equipment Fend device INVITE > INVITE INVITE Off hook CASE B UPDATE INPOTE INPOTE INPOTE INPOTE INVITE INVITE INVITE INVITE <				
SDP answer Ressage flow End device Interworking POTS INVITE > 183 Session Progress € PRACK > 200 OK PRACK • 200 OK INVITE € ACK • 200 OK INVITE € ACK • 200 OK UPDATE € 200 OK UPDATE € UPDATE • 200 OK UPDATE • Subjointerworking • INVITE • \$ INVITE • \$ ACK • \$				
Message flow End device Interworking POTS Find device INVITE > 183 Session Progress € PRACK > 200 OK PRACK > CASE A 200 OK INVITE 200 OK INVITE € ACK > UPDATE 200 OK UPDATE > LUPDATE > Session Progress € INVITE > Apply post test routine SETUP 183 Session Progress € INVITE > > SETUP 183 Session Progress € FACK >				
Test equipmentEnd deviceINVITE>INVITE>183 Session Progress€PRACK>200 OK PRACK€200 OK PRACK€CASE A 200 OK INVITE€200 OK INVITE€ACK>CASE B UPDATE€SDN interworking>INVITE>Apply post test routineSDN interworkingINVITE>INVITE>IS3 Session Progress€PRACK>INVITE>ISA Session Progress€INVITE>ISA Session Progress€INVITE>INVITE>INVITE>INVITE>INVITE>INVITE>INVITE>INVITE>INTE <t< td=""><td></td><td></td><td></td><td></td></t<>				
Interworking POTS INVITE INVITE INVITE IN	-	• •		
INVITE + Ringing RACK + Ringing PRACK + Hone + Ho	-	uipment	End	l device
183 Session Progress PRACK 200 OK PRACK€RingingCASE A 200 OK INVITE ACK€Off hookCASE B UPDATE 200 OK UPDATE€Session ProgressSBDN interworking INVITE 183 Session Progress>€INVITE PRACK€•SEDN interworking PRACK€•	-	د		
PRACK ÷ 200 OK PRACK ÷ CASE A 200 OK INVITE ← ACK • Off hook ACK ÷ Off hook CASE B UPDATE ÷ 200 OK UPDATE ÷ Source constant of the second seco			Dinging	
200 OK PRACK€CASE A 200 OK INVITE ACKOff hookDUPDATE 200 OK UPDATE•Share 200 OK UPDATE•Share Approx test routine•ISDN interworking INVITE 183 Session Progress•PRACK•	-		Kinging	
CASE A 200 OK INVITE ACK ACK CASE B UPDATE 200 OK UPDATE Apply post test routine ISDN interworking INVITE INVITE 183 Session Progress PRACK Off hook Off hook Off hook ALERTING				
200 OK INVITE ACKConservationOff hookCASE B UPDATE 200 OK UPDATE * * * Apply post test routine > > > > Apply post test routineISDN interworking INVITE 183 Session Progress PRACK * <b< td=""><td>200 OK PRACK</td><td>t</td><td></td><td></td></b<>	200 OK PRACK	t		
200 OK INVITE ACKConservationOff hookCASE B UPDATE 200 OK UPDATE * * * Apply post test routine > > > > Apply post test routineISDN interworking INVITE 183 Session Progress PRACK * 				
ACK CASE B UPDATE 200 OK UPDATE INVITE INVITE INVITE PACK				
CASE B UPDATE 200 OK UPDATE Happly post test routine Apply post test routine ISDN interworking INVITE 183 Session Progress PRACK Setup +		_	Оп поок	
UPDATE 200 OK UPDATE> Apply post test routineISDN interworking>INVITE>183 Session ProgressPRACK>	ACK	7		
UPDATE 200 OK UPDATE→ K Apply post test routineISDN interworking→INVITE→183 Session Progress←K←ALERTINGPRACK→	CASE D			
200 OK UPDATE Apply post test routine ISDN interworking INVITE A SESSION Progress A C A A A A A A A A A A A A A A A A A		د		
Apply post test routineISDN interworkingINVITE>183 Session Progress€€€PRACK>		_		
ISDN interworkingINVITE→→SETUP183 Session Progress←←ALERTINGPRACK→	200 OK UPDATE			
INVITE>SETUP183 Session ProgressPRACK>		Apply	post test routine	
INVITE>SETUP183 Session ProgressPRACK>	ISDN interworking			
183 Session Progress ← ALERTING PRACK →		→	→ SETUP	
PRACK		_		Ч Т
	-			
	200 OK I NAUK	T		

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 <i>Modifying SDP in early did</i>	ılogue		
Ensure that the SUT is able UPDATE with SDP answe		request to modify the SDP in early	y dialogue. A 200 OK
SIP header values			
183/180			
SDP answer 1			
UPDATE			
SDP offer 2			
200 OK (UPDATE)			
SDP answer 2			
Message flow			
Test eq	uipment	Enc	d device
Interworking POTS			
INVITE	→		
180/183	÷	Ringing	
UPDATE	→		
200 OK UPDATE	←		
	Apply	y post test routine	
ISDN interworking			
INVITE	→	→ SETUP	
180 Ringing	÷	← ALERTING	3
UPDATE	→		
200 OK UPDATE	÷		
	Apply	y post test routine	

TSS Term_Establishment_of_	TP_301_101	Reference subclause 5.1.2.1 of	Selection expression PICS 5.1.1/2 AND 5.4/1
an_early_dialogue		[ETSI TS 183 036]	
Test purpose			
A BearerCapability speech	h is included in the PST	N XML element	
		ed in a received INVITE request an ETUP is sent where the Bearer Cap	
received PSTM XML Bea		ET OF is sent where the bearer Caj	padinity is derived from the
SIP header values			
INVITE: PSTN XML MIN	VE body		
xml version="1.0" enco</td <td>•</td> <td></td> <td></td>	•		
PSTN			
BearerCapability			
BCoctet3			
CodingStandard	1>00<		
•	nsferCabability>ITC_va	lue<	
BCoctet4			
TransferMode>	00<		
InformationTra	nsferRate>10000<		
BCoctet5			
Layer1Identification Layer11 Layer1Identification Layer1	ation>01<		
UserInfoLayer1	Protocol>00011<		
If ITC_value = '01000' the	SDP m line contains Cl	LEARMODE as preferred codec	
DSS1 Parameter values			
SETUP: Bearer Capability	= PSTM XML Bearer	Capability	
Message flow			
Test ed	quipment	En	nd device
INVITE	→	→ SETUP	
100 Trying	÷		
		oly post test routine	

TSS	TP_301_102	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.1 of	PICS 5.1.1/2 AND 5.4/1
an_early_dialogue		[ETSI TS 183 036]	

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.

SIP header va	lues			
INVITE: PSTN	I XML MIME body			
	="1.0" encoding="utf-8"?>			
PSTN	C			
BearerCapa	bility			
BCocter	-			
Cod	ingStandard>00<			
	rmationTransferCabability>00000<			
	or .			
Info	rmationTransferCabability>10000<			
BCocter	-			
Trar	sferMode>00<			
Info	rmationTransferRate>10000<			
BCocter	5			
Laye	er1Identification>01<			
User	InfoLayer1Protocol>00011<			
BearerCapa	bility			
BCocte	3			
Cod	ingStandard>00<			
Info	rmationTransferCabability>10001<			
BCocte	4			
Trar	sferMode>00<			
Info	rmationTransferRate>10000<			
BCocte	5			
Laye	er1Identification>01<			
User	InfoLayer1Protocol>00011<			
SDP: m line co	ntains as the first codec CLEARMODE a	and as the second codec an ITU-T G.711 codec		
DSS1 Parame	ter values			
SETUP: First	Bearer Capability Information transfer c	capability = Speech or 3.1 kHz audio		
	nd Bearer Capability Information transfe s/announcements	er capability = Unrestricted digital information with		
Message flow				
	Test equipment	End device		
INVITE	→	→ SETUP		
100 Trying	F			
·····B	Apply post test routine			
Apply post test fourne				

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

TSS Term_Establishment_of_ an_early_dialogue TP_301_1	Reference subclause 5.1.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2
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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.

SIP header values

INVITE: SDP m line = SDP_VA

DSS1 Parameter values

SETUP: Bearer Capability

Message flow

Message now						
Test equipment End device						
INVITE	→	→	SETUP			
100 Trying	÷					
	Apply post to	est routine				

Table 7.2.3.1-2 – Coding of from SDP: SIP to BC DSS1

SDP_VA		m= line		a= line	Bearer Ca	pability IE
	<media></media>	<transport></transport>	<fmt- list></fmt- 	Rtpmap: <dynamic-pt> <encoding name="">/<clock rate>/encoding parameters></clock </encoding></dynamic-pt>	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</dynamic-pt>	"3.1 kHz audio"	"G.711 A-law"
SDP_VA_04	Audio	RTP/AVP	Dynamic PT	rtpmap: <dynamic-pt> PCMU/8000</dynamic-pt>	"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</dynamic-pt>	"3.1 kHz audio"	"G.711 A-law"
SDP_VA_08	Audio	RTP/AVP	Dynamic PT	rtpmap: <dynamic-pt> PCMA/8000</dynamic-pt>	"3.1 kHz audio"	"G.711 μ-law"
SDP_VA_09	Audio	RTP/AVP	Dynamic PT,	rtpmap: <dynamic-pt> CLEARMODE/8000</dynamic-pt>	"Unrestricted digital inf. W/tone/ann.")	
SDP_VA_10	Audio	RTP/AVP	Dynamic PT	Rtpmap: <dynamic-pt> CLEARMODE/8000</dynamic-pt>	"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"

an early dialogue		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
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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

8				
	Test equipment		End device	
INVITE	→	→	SETUP	
100 Trying	÷			
	Apply pos	st 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 ></media 	<transport ></transport 	<fmt-list></fmt-list>	Rtpmap: <dynamic-pt> <encoding name="">/<clock rate>/encoding parameters></clock </encoding></dynamic-pt>	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	TP_301_105	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.1 of	PICS 5.1.1/2 AND 5.4/1
an_early_dialogue		[ETSI TS 183 036]	

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 =	', Location ='0000', Progress description= PI_value
Message flow	
Test equipment	End device
INVITE →	→ SETUP
100 Trying +	
	oply post test routine

TSS	TP_301_106	Reference	Selection expression				
Term_Establishment_of_		subclause 5.1.2.1 of	PICS 5.1.1/2 AND 5.4/1				
an_early_dialogue		[ETSI TS 183 036]					
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	PSTM XML MIME body						
xml version="1.0" encoding="utf-8"?							
PSTN							
ProgressIndicator							
ProgressOctet3							
CodingStandard	>00<						
Location>yyyy<							
ProgressOctet4							
ProgressDescrip	tion>0000110<						
DSS1 Parameter values							
Message flow							
Test eq	luipment	End d	levice				
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	+						
	Арр	ly 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	TP_301_108	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.1 of	PICS 5.1.1/2 AND 5.4/1
an_early_dialogue		[ETSI TS 183 036]	

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 \rightarrow 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	TP_301_109	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.1 of	PICS 5.1.1/2 AND 5.4/1
an_early_dialogue		[ETSI TS 183 036]	

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	Message flow		
	Test equipment		End device
INVITE	÷	→ →	SETUP
100 Trying	÷	-	
		Apply post test routine	

TSS	TP_301_110	Reference	Selection expression	
Term_Establishment_of_		subclause 5.1.2.1 of	PICS 5.1.1/2 AND 5.4/1	
an_early_dialogue		[ETSI TS 183 036]		
Test purpose				
Two High layer compatibil	ity information elemen	ts received		
High layer compatibility in	formation element in the ceived in the INVITE	patibility received in the INVITE re the sent SETUP and the second PS' request is mapped into the second	IN XML	
SIP header values				
INVITE:				
PSTN XML MIME body				
xml version="1.0" encod</td <td>ling="utf-8"?></td> <td></td> <td></td>	ling="utf-8"?>			
PSTN				
HighLayerCompatibilit	У			
HLOctet3				
CodingStandard				
Interpretation>1				
PresentationMet	hod>01<			
HLOctet4				
• •	cteristics>0000001<			
HighLayerCompatibilit HLOctet3	У			
CodingStandard	>00~			
Interpretation>1				
PresentationMethod>01<				
HLOctet4				
	cteristics>HLC_value	<		
DSS1 parameter values				
SETUP:				
First high layer compatib Telephony	ility Coding standard=	'00', Interpretation='100', High lay	er characteristics identification=	
Second high layer comparidentification=HLC_value		d='00', Interpretation='100', High l .2.3.1-6	ayer characteristics	
Message flow				
Test eq	uipment	Ε	nd device	
INVITE	→	→ SETUP		
100 Trying	+			
		ply 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

TSS	TP_301_111	Reference	Selection expression
Term_Establishment_of_ an_early_dialogue		subclause 5.1.2.1/ [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/1

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.

SIP header values

DSS1 Parameter values CALL PROCEEDING: Progress Indicator Progress Description = PI_value

	e	U	1	_		
Message flow						
	Test equipment				End device	
INVITE		→		→	SETUP	
100 Trying		←				
183 Session Progr	ess	÷		←	CALL PROCEEDING	
		Apply	post test rout	tine		

TSS	TP_301_112	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.2 of	PICS 5.1.1/2 AND 5.4/1
an_early_dialogue		[ETSI TS 183 036]	

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.

SIP header values			
183 Session Progress:			
PSTM XML MIME body	PSTM XML MIME body		
xml version="1.0" encoding="utf-8"?			
PSTN			
ProgressIndicator			
ProgressOctet3			
CodingStandard>00<			
Location>yyyy<			
ProgressOctet4			
ProgressDescription>0000111<			
ProgressIndicator	ProgressIndicator		
ProgressOctet3			
CodingStandard>00<			
Location>0000<			
ProgressOctet4			
ProgressDescription>PI_value<			
DSS1 Parameter values			
CALL PROCEEDING: Progress Indicator Progress Descri	ption = 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_dialogueTP_301_113	Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1
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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 equip	ment	End device
INVITE	→	→ SETUP
100 Trying	←	
183 Session Progress	+	← CALL PROCEEDING
	Apply p	ost test routine

TSS	TP_301_114	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.2 of	PICS 5.1.1/2 AND 5.2/4
an_early_dialogue		[ETSI TS 183 036]	

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 equip	nent	End d	levice
INVITE	→	→ SETUP	
100 Trying	÷		
183 Session Progress	÷	← CALL PROCE	EEDING
	Apply po	test routine	

TSS	TP_301_115	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.2 of	PICS 5.1.1/2 AND 5.4/1
an_early_dialogue		[ETSI TS 183 036]	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 b	ody			
P-Erly-Media: supported				
xml version="1.0" encoding=</th <th>"utf-8"?></th> <th></th> <th></th> <th></th>	"utf-8"?>			
PSTN				
BearerCapability				
BCoctet3				
CodingStandard>00<				
InformationTransfe	rCabability>00000<			
or				
InformationTransfe	rCabability>10000<			
BCoctet4				
TransferMode>00<				
InformationTransfer	late>10000<			
BCoctet5				
Layer1Identification>	·01<			
UserInfoLayer1Proto	col>00011<			
183 Session Progress: P-Early-	Media: <appropriate td="" valu<=""><td>ie></td><td></td><td></td></appropriate>	ie>		
SDP answer				
DSS1 Parameter values				
Message flow				
Test equipn	nent		End device	
INVITE	→	→	SETUP	
100 Trying	+			
183 Session Progress	+	+	CALL PROCEEDING	
-	Apply po	st test routine		

TSS	TP_301_116		Selection expression
Term_Establishment_of_		subclause 5.1.2.2 of	PICS 5.1.1/2 and 5.4/1
an_early_dialogue		[ETSI TS 183 036]	and 5.1.3/1

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>000	>0<
ProgressIndicator	
ProgressOctet4	
ProgressDescription>0000101<	
SDP: m= audio xxxx RTP/AVP 8	
DSS1 Parameter values	
CALL PROCEEDING: Bearer Capability: Inform description = 5	ation Transfer Capability = speech, Progress Indicator Progress
Message flow	
Test equipment	End device
INVITE →	→ SETUP
100 Trying ←	
183 Session Progress ←	← CALL PROCEEDING
Ар	ply post test routine

TSS	TP_301_117	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.2 of	PICS 5.1.1/2 and 5.4/1
an_early_dialogue		[ETSI TS 183 036]	and 5.1.3/1

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 7 description = 5	Transfer Capability = speech, Progress Indicator Progress
Message flow	
Test equipment	End device
INVITE →	→ SETUP
100 Trying ←	
183 Session Progress +	← CALL PROCEEDING
Apply po	st test routine

TSS	TP_301_118	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.2 of	PICS 5.1.1/2 and 5.4/1
an_early_dialogue		[ETSI TS 183 036]	and 5.1.3/1

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 be	ody		
xml version="1.0" encoding=</td <td>'utf-8"?></td> <td></td> <td></td>	'utf-8"?>		
PSTN			
BearerCapability			
BCoctet3			
CodingStandard>00<			
InformationTransferC	abability>00000<		
BearerCapability			
BCoctet3			
CodingStandard>00<			
InformationTransferC	abability>10001<		
183 Session Progress: HighLay DSS1 Parameter values	erCompatibility		
CALL PROCEEDING: High La	ver Compatibility High	Laver Character	istics <appropriate value=""></appropriate>
Message flow			
Test equipm	ent		End device
INVITE	→	→	SETUP
100 Trying	+		
183 Session Progress	÷	÷	CALL PROCEEDING
	Apply pos	st test routine	

TSS	TP_301_119	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.2 of	PICS 5.1.1/2 AND NOT 5.4/1
an_early_dialogue		[ETSI TS 183 036]	

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 equipme	nt		End device
INVITE	→	→	SETUP
100 Trying	+		
183 Session Progress	÷	÷	CALL PROCEEDING
183 Session Progress	÷	÷	PROGRESS
	Apply pos	st 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 <i>PROGRESS received Prog</i>	rass Indicator is present		
FROGRESS Teceived Flog	ress maicaior is present		
Table 7.2.3.1-4, a 183 Ses	sion Progress is sent. Th	d a Progress Indicator value PI_v e 183 Session Progress contains a gressIndicator element withvalue =	N PSTN XML ProgressIndicator
SIP header values			
183 Session Progress2			
PSTM XML MIME body			
xml version="1.0" enco</td <td>ding="utf-8"?></td> <td></td> <td></td>	ding="utf-8"?>		
PSTN			
ProgressIndicator			
ProgressOctet3			
CodingStandard			
Location>yyyy<	<		
ProgressOctet4			
ProgressDescrip	otion>0000111<		
ProgressIndicator			
ProgressOctet3			
CodingStandard			
Location>0000<	<		
ProgressOctet4			
ProgressDescrip	otion>PI_value<		
DSS1 Parameter values			
PROGRESS: Progress Ind	icator Progress Description	on = PI_value	
Message flow			
Test ec	quipment	End	d device
INVITE	→	→ SETUP	
100 Trying	+		
183 Session Progress1	+	← CALL PRO	DCEEDING
183 Session Progress2	÷	← PROGRES	
	App	ly post test routine	

TSS	TP_301_121	Reference	Selection expression
Term Establishment of		subclause 5.1.2.2 of	PICS 5.1.1/2 and 5.4/1
an_early_dialogue		[ETSI TS 183 036]	

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=</td <td>"utf-8"?></td> <td></td>	"utf-8"?>	
PSTN		
ProgressIndicator		
ProgressOctet3		
CodingStandard>00<		
Location>yyyy<		
ProgressOctet4		
ProgressDescription>	0000111<	
DSS1 Parameter values		
PROGRESS:		
Message flow		
Test equipn	nent	End device
INVITE	→	→ SETUP
100 Trying	+	
183 Session Progress1	÷	← CALL PROCEEDING
183 Session Progress2	+	← PROGRESS
	Apply pos	st test routine

TSS	TP 301 122	Reference	Selection expression
Term Establishment of	11_001_122	subclause 5.1.2.2 of	PICS 5.1.1/2 and 5.4/1
an_early_dialogue		[ETSI TS 183 036]	and 5.2/16

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: <a propriate value> SDP answer

DSS1 Parameter values			
PROGRESS:			
Message flow			
Test equipn	ent		End device
INVITE	→	→	SETUP
100 Trying	÷		
183 Session Progress1	+	←	CALL PROCEEDING
183 Session Progress2	+	←	PROGRESS
	Apply pos	st test routine	

		r	
TSS	TP_301_123	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.2 of	PICS 5.1.1/2 AND 5.4/1
an_early_dialogue		[ETSI TS 183 036]	AND 5.1.3/1

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	t	End device
INVITE	→	→ SETUP
100 Trying	+	
183 Session Progress1	+	← CALL PROCEEDING
183 Session Progress2	+	← PROGRESS
	Apply	post test routine

TSS Term_Establishment_of_	TP_301_124	Reference subclause 5.1.2.2 of	Selection expression PICS 5.1.1/2 AND 5.4/1
an_early_dialogue		[ETSI TS 183 036]	AND 5.1.3/1
Test purpose			
Handling of fallback inform	nation HLC (only appli	cable at T reference point)	
The AGCF/VGW receives	a PROGRESS. The Hig	gh Layer Compatibility IE is pre	not able to support the UDI/TA. esent. A 183 Session Progress is pped from the DSS1 High Layer
SIP header values			
INVITE: PSTN XML MIN	vIE body		
xml version="1.0" encod</td <td>ling="utf-8"?></td> <td></td> <td></td>	ling="utf-8"?>		
PSTN			
BearerCapability			
BCoctet3			
CodingStandard			
	sferCabability>10000<		
BearerCapability			
BCoctet3	> 00 <		
CodingStandard	>00< sferCabability>10001<		
BCoctet4	IsterCabability>10001<	~	
Desterr			
183 Session Progress2: Hi	ghLayerCompatibility		
DSS1 Parameter values			
PROGRESS: Bearer Capal	oility: Information Trar	sfer Capability = speech, Progre	ess Indicator Progress description =
5			
Message flow			
Test eq	luipment		End device
INVITE	→	→ SETUP	
100 Trying	+		
183 Session Progress1			PROCEEDING
183 Session Progress2			RESS
-	Ap	ply post test routine	
TSS	TP 301 125	Reference	Selection expression

an_early_dialogue		[ETSI TS 183 036]	PICS 5.1.1/2 AND NOT 5.4/1
TSS	TP_301_125	Reference	Selection expression
Term Establishment of		subclause 5.1.2.2 of	PICS 5.1.1/2 AND NOT 5.4/1

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 eo	luipment		End device
INVITE	→	→	SETUP
100 Trying	+		
180 Ringing	+	÷	ALERTING
	Apply pos	t test routine	

TSS	TP_301_126	Reference	Selection expression
Term_Establishment_of_	11_301_120	subclause 5.1.2.2 of	PICS 5.1.1/2 AND 5.4/1
an_early_dialogue		[ETSI TS 183 036]	
Test purpose	L		
ALERTING received Progr	ress Indicator is not pres	sent	
Ensure that on receipt of a Ringing contains a PSTN 2		if no Progress Indicator is present, ProgressDescription = 7	a 180 Ringing is sent. The 180
SIP header values			
180 Ringing:			
PSTM XML MIME body			
xml version="1.0" encod</td <td>ling="utf-8"?></td> <td></td> <td></td>	ling="utf-8"?>		
PSTN			
ProgressIndicator			
ProgressOctet3			
CodingStandard	>00<		
Location>yyyy<	Ś		
ProgressOctet4			
ProgressDescrip	tion>0000111<		
DSS1 Parameter values			
ALERTING:			
Message flow			
Test eq	uipment	En	d device
INVITE	→	→ SETUP	
100 Trying	÷		
180 Ringing	+	← ALERTIN	G
	Арр	ly post test routine	

TSS	TP_301_127	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.2 of	PICS 5.1.1/2 AND 5.4/1
an_early_dialogue		[ETSI TS 183 036]	AND 5.2/16

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=""></appropriate>				
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 \leftarrow				
180 Ringing	← ALERTING			
Appl	post test routine			

TSS Term_Establishment_of_ an_early_dialogue	_301_128 Reference subclause 5.1.2.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/1
--	---	--

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	JP			
100 Trying \leftarrow				
180 Ringing ← ← ALE	RTING			
Apply post test routine				

TSS	TP_301_129	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.2 of	PICS 5.1.1/2 AND 5.4/1
an_early_dialogue		[ETSI TS 183 036]	AND 5.1.3/1

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 In	dicator Progress description = 5			
Message flow				
Test equipment End	l device			
INVITE → SETUP				
100 Trying \leftarrow				
180 Ringing ← ← ALERTING	3			
Apply post test routine				

TSS	TP_301_130	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.2 of	PICS 5.1.1/2 AND 5.4/1
an_early_dialogue		[ETSI TS 183 036]	AND 5.1.3/1

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 MI	ME body			
xml version="1.0" enco</td <td>ding="utf-8"?></td> <td></td> <td></td> <td></td>	ding="utf-8"?>			
PSTN				
BearerCapability				
BCoctet3				
CodingStandard	>00<			
InformationTra	nsferCabability>00000<			
BearerCapability				
BCoctet3				
CodingStandard	>00<			
InformationTrai	nsferCabability>10001<			
180 Ringing: HighLayerC	ompatibility			
DSS1 Parameter values				
ALERTING: Bearer Capa	bility: Information Transfer Ca	pability = speec	h, Progress Indicator Progr	ess description $=$ 5
Message flow				
Test ec	Juipment		End device	
INVITE	→	→	SETUP	
100 Trying	←			
180 Ringing	+	+	ALERTING	
	Apply pos	st test routine		

TSS	TP_301_131	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.2 of	PICS 5.1.1/2 AND 5.4/1
an_early_dialogue		[ETSI TS 183 036]	AND 5.1.3/1

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	ity = 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 <i>PROGRESS received Progress Indicator is present</i> 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 equip	ment		End device
INVITE	→	→	SETUP
180 Ringing	+	←	ALERTING
183 Session Progress	+	←	PROGRESS
	Apply po	st 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 Prog	gress Indicator is present	t			
	× •				
Ensure that on receipt of	a DSS1 PROGRESS an	nd a Progress Indicator value PI_	value is present as described in		
		he 183 Session Progress contains			
ProgressDescription = PI_	value. An additional Pro	gressIndicator element is present v	value = 7		
SIP header values					
183 Session Progress:					
PSTM XML MIME body					
xml version="1.0" enco</td <td>ding="utf-8"?></td> <td></td> <td></td>	ding="utf-8"?>				
PSTN					
ProgressIndicator					
ProgressOctet3					
CodingStandard	l>00<				
Location>yyyy-	<				
ProgressOctet4					
ProgressDescrip	ption>0000111<				
ProgressIndicator					
ProgressOctet3					
CodingStandard	l>00<				
Location>0000-	<				
ProgressOctet4					
ProgressDescrip	ption>PI_value<				
DSS1 Parameter values					
PROGRESS: Progress In	dicator Progress Descript	tion = PI_value			
Message flow					
•	Test equipment End device				
INVITE	→	→ SETUP			
180 Ringing	÷ +	← ALERTIN	G		
	÷				
183 Session Progress	_	← PROGRES	00		
	Арр	oly post test routine			

TSS	TP_301_134	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

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:	183 Session Progress:				
PSTM XML MIME body					
xml version="1.0" encoding="ut</td <td>f-8"?></td> <td></td> <td></td>	f-8"?>				
PSTN					
ProgressIndicator					
ProgressOctet3					
CodingStandard>00<					
Location>yyyy<					
ProgressOctet4					
ProgressDescription>00	00111<				
DSS1 Parameter values					
PROGRESS:					
Message flow					
Test equipment End device					
INVITE	→	→ SETUP			
180 Ringing	← ← ALERTING				
183 Session Progress	Session Progress				
Apply post test routine					

TSS	TP_301_135	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.2 of	PICS 5.1.1/2 AND 5.4/1
an_early_dialogue		[ETSI TS 183 036]	AND 5.2/16

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: <a propriate value>

DSS1 Parameter values PROGRESS:

Message flow		
Test equipm	ent	End device
INVITE	→	→ SETUP
180 Ringing	←	← ALERTING
183 Session Progress	←	← PROGRESS
	Apply p	ost test routine

TSS	TP_301_136	Reference	Selection expression	
Term_Establishment_of_		subclause 5.1.2.2 of	PICS 5.1.1/2 AND 5.4/1	
an_early_dialogue		[ETSI TS 183 036]	AND 5.1.3/1	
Test purpose Handling of fallback inform				

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 Ca	pability = speech, Progress Indicator Progress description = 5
Message flow		
Test equip	ment	End device
INVITE	→	→ SETUP
180 Ringing	←	← ALERTING
183 Session Progress	+	← PROGRESS
	Apply pos	st test routine

TSS	TP_301_137	Reference	Selection expression	
Term_Establishment_of_		subclause 5.1.2.2 of	PICS 5.1.1/2 AND 5.4/1	
an_early_dialogue		[ETSI TS 183 036]	AND 5.1.3/1	
Test purpose				
Handling of fallback inform	nation BC audio (only appli	cable at T reference point)		
			equipment is not able to support	
			sent and the Information Transfer resent. A 183 Session Progress is	
			ne InformationTransferCabability	
		r element is present set to valu	are 5. The first stated codec in the	
SDP answer is not equal to	CLEARMODE.			
SIP header values				
INVITE : PSTN XML MIN				
xml version="1.0" encod</td <td>ling="utf-8"?></td> <td></td> <td></td>	ling="utf-8"?>			
PSTN				
BearerCapability				
BCoctet3	> 00 <			
CodingStandard	>00< sferCabability>10000<			
BearerCapability	Ister Cabability >10000<			
BCoctet3				
CodingStandard	>00<			
-	sferCabability>10001<			
	, second s			
183 Session Progress: PST	N XML MIME body			
xml version="1.0" encod</td <td>ling="utf-8"?></td> <td></td> <td></td>	ling="utf-8"?>			
PSTN				
BearerCapability				
BCoctet3				
CodingStandard>00<				
	InformationTransferCabability>10000<			
ProgressIndicator				
ProgressOctet4	tion > 0000101 <			
ProgressDescription>0000101< SDP: m= audio xxxx RTP/AVP				
DSS1 Parameter values	hilitry Information Trace for	Conchility - 2.1 LU 1' - T	Decences Indicator Dramos	
description = 5	ionity: information Transfer	Capability = 3,1 kHz audio, F	Togress indicator Progress	
Message flow				
-	winmont	T7	nd device	
	uipment			
INVITE	→	→ SETUP		
180 Ringing	(← ALERTIN		
183 Session Progress	+	← PROGRES	88	
	Apply p	oost 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			
	mation HLC (only appli	cable at T reference point)	
		allback if the user equipment is not	
		gh Layer Compatibility IE is presen L HighLayerCompatibility is mappe	
Compatibility IE.			a nom die 19551 fingli Layer
SIP header values			
INVITE : PSTN XML M	ME body		
xml version="1.0" enco</td <td>•</td> <td></td> <td></td>	•		
PSTN	C		
BearerCapability			
BCoctet3			
CodingStandar	d>00<		
InformationTra	nsferCabability>10000<		
BCoctet4			
TransferMode>	>00<		
InformationTra	nsferRate>10000<		
BCoctet5			
Layer1Identific			
•	Protocol>00011<		
BearerCapability			
BCoctet3			
CodingStandar			
	nsferCabability>10001<		
BCoctet4	00 <		
TransferMode>	soo< sinsferRate>10000<		
BCoctet5	InsterRate>10000<		
Layer1Identific	vation \01~		
•	1Protocol>00011<		
e ser interaction au ser			
183 Session Progress: Hi	ghLayerCompatibility		
DSS1 Parameter values	- •		
PROGRESS: High Layer	Compatibility High Lay	yer Characteristics <appropriate td="" val<=""><td>ue></td></appropriate>	ue>
Message flow			
-	quipment	En	d device
INVITE	→	→ SETUP	
180 Ringing	÷	← ALERTIN	G
	÷	← PROGRES	
183 Session Progress	τ		C C C C C C C C C C C C C C C C C C C

TSS Term_Establishment_of_ a_confirmed_dialogue	TP_302_001	Reference [IETF RFC 3261]	Selection expression
Test purpose			
From, Call-ID, CSeq and	Via headers copy from the	INVITE	
_			
Ensure that the IUT on rec CSeq and Via headers copy			uding the headers From, Call-ID,
SIP header values			
INVITE:			
From: < from_value_invite	>		
Call-ID: <callid_value_inv< td=""><td>ite></td><td></td><td></td></callid_value_inv<>	ite>		
CSeq: <cseq_value_invite< td=""><td>></td><td></td><td></td></cseq_value_invite<>	>		
Via: <via_value_invite></via_value_invite>			
200 OK:			
From: <from_value_invite< td=""><td><</td><td></td><td></td></from_value_invite<>	<		
Call-ID: <callid_value_inv< td=""><td></td><td></td><td></td></callid_value_inv<>			
CSeq: <cseq_value_invite></cseq_value_invite>			
Via: <via_value_invite></via_value_invite>			
Message flow			
0	uipment	Er	nd device
Interworking POTS			
INVITE	→	Ringing	
200 OK INVITE	+	Off hook	
ACK	→		
	A 1		
	Apply	y post test routine	
ISDN interworking			
INVITE	→	→ SETUP	
200 OK INVITE	←	← CONNEC	Т
ACK	→		
	Apply	y post test routine	

Establishment of a confirmed dialogue

Test purpose

TSS

7.2.3.2

To tag is sent in the response

Term_Establishment_of_

a_confirmed_dialogue

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.

Reference

and 13.3.1 of [IETF RFC 3261]

sections 8.2.6.2, 12.2.2

Selection expression

TP_302_002

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	TP_302_003	Reference	Selection expression
Term_Establishment_of_		section 8.2.6.2 of	-
a_confirmed_dialogue		[IETF RFC 3261]	
Test purpose			
To tag in the INVITE reque	est		
Ensure that the IUT on rec	eipt of an INVITE reques	st with a TAG set on the To head	er, either:
 sends a Success (200 O for robustness), 	K) response including th	e same URI and the same TAG f	for the To header (recommended
- or reject the INVITE re	equest with a Call/Transa	ction does not exist (481 Call/Tra	ansaction does not exist).
SIP header values			
INVITE:			
To: <sip:to_uri_value></sip:to_uri_value>	;tag=to_tag_value		
200 OK:			
To: <sip:to_uri_value></sip:to_uri_value>	;tag=to_tag_value		
Message flow			
0	quipment	E	nd device
Interworking POTS			
INVITE	→	Ringing	
CASE A	-	BB	
200 OK INVITE	+	Off hook	
ACK	→	On nook	
i cix	_	ly post test routine	
CASE B	App	ly post lest routille	
481 Call/Transaction does	not exist 🗲		
ACK			

ISDN interworking			
INVITE	→	→	SETUP
CASE A			
200 OK INVITE	←	÷	CONNECT
ACK	→		
	Apply j	post test routine	
CASE B			
481 Call/Transaction does not exist	←		
ACK	→		

TSS	TP_302_004	Reference	Selection expression
Term_Establishment_of_		section 12.1.1 of	
a_confirmed_dialogue		[IETF RFC 3261]	
Test purpose			
Contact header in the resp	onse.		
Ensure that the IUT on rec header.	ceipt of an INVITE reque	est, sends a Success (200 OK) res	sponse including a single Contact
SIP header values			
200 OK:			
Contact: <sip: contact_<="" td=""><td>value></td><td></td><td></td></sip:>	value>		
Message flow			
Test eq	quipment	E	nd device
Interworking POTS			
INVITE	→	Ringing	
200 OK INVITE	←	Off hook	
ACK	→		
	Арр	ly post test routine	
ISDN interworking			
INVITE	→	→ SETUP	
200 OK INVITE	÷	← CONNEC	CT
ACK	→		
	App	ly 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_< td=""><td><i>record_route</i>>;lr</td><td></td><td></td></sip:invite_<>	<i>record_route</i> >;lr		
200 OK:			
Record-Route: <sip:invite_< td=""><td><i>record_route</i>>;lr</td><td></td><td></td></sip:invite_<>	<i>record_route</i> >;lr		
Message flow			
Test equip	ment	End device	
Interworking POTS			
INVITE	→	Ringing	
200 OK INVITE	+	Off hook	
ACK	→		
	Apply pos	st test routine	
ISDN interworking			
INVITE	→	→ SETUP	
200 OK INVITE	+	← CONNECT	
ACK	→		
	Apply pos	st 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		
	ccess (200 OK) response		header with parameters that it does neader copy from the INVITE request,
SIP header values			
INVITE:			
Record-Route: <sip:rec< td=""><td>ord-route_value_invite;</td><td>unknown=etsi></td><td></td></sip:rec<>	ord-route_value_invite;	unknown=etsi>	
200 OK:			
	ord-route_value_invite;	inknown=etsi>	
Message flow			
-	uipment		End device
Interworking POTS	[~- F		
INVITE	→	Ring	ing
200 OK INVITE	+	Off h	•
ACK	→		
	Арр	ly post test routine	
ISDN interworking			
INVITE	→	→ SET	UP
200 OK INVITE	+	← CON	INECT
ACK	→		
	Арр	ly post test routine	

TSS Term_Establishment_of_ a_confirmed_dialogue	TP_302_007	Reference section 12.1.1 of [IETF RFC 3261]	Selection expression
Test purpose			·
From header without "tag	,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,		
Ensure that the IUT on rec response including a From		est including From header witho	out tag, sends a Success (200 OK)
SIP header values			
INVITE:			
From: <sip:from_value< td=""><td>_invite></td><td></td><td></td></sip:from_value<>	_invite>		
200 OK:			
From: <sip:from_value< td=""><td>_invite></td><td></td><td></td></sip:from_value<>	_invite>		
Message flow			
Test eq	luipment	E	and device
Interworking POTS			
INVITE	→	Ringing	
200 OK INVITE	÷	Off hook	
ACK	→		
	Appl	y post test routine	
ISDN interworking			
INVITE	→	→ SETUP	
200 OK INVITE	+	← CONNEC	СТ
ACK	→		
	Appl	y post test routine	

TSS	TP_302_008	Reference	Selection expression
Term_Establishment_of_		section 13.3.1.4 of	_
a_confirmed_dialogue		[IETF RFC 3261]	
Test purpose			
Allow and a Supported he	aders sent in response		
Ensure that the IUT having	g received an INVITE requ	est, sends a Success (200 OK) incl	uding an Allow and a Supported
headers.			
SIP header values			
200 OK:			
Allow:			
Supported:			
Message flow			
Test e	quipment	Enc	l device
Interworking POTS			
INVITE	→	Ringing	
200 OK INVITE	←	Off hook	
ACK	→		
	Apply	y post test routine	

ISDN interworking			
INVITE	→	→	SETUP
200 OK INVITE	+	←	CONNECT
ACK	→		
	Apply po	st 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 <i>Modifying SDP in confirm</i>	ed dialogue		
Ensure that the SUT is ab UPDATE with SDP answe		TE request to modify the SDP i	n confirmed dialogue. A 200 OK
SIP header values			
200 OK:			
Allow:			
Supported:			
Message flow			
Test ec	Juipment	E	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	÷		
	Арр	ly post test routine	
ISDN interworking			
INVITE	→	→ SETUP	
180 Ringing	+	← ALERTI	
200 OK INVITE	+	← CONNE	СТ
ACK	``		
CASE A			
INVITE	→		
200 OK INVITE	÷		
ACK	→		

CASE B			
UPDATE	→		
200 OK UPDATE	÷		
	App	ly post test routine	
7.2.3.2.2 Test purpos	es for ISDN		
TSS Term_Establishment_of_	TP_302_101	Reference subclause 5.1.2.3 of	Selection expression PICS 5.1.1/2 AND 5.4/1

Term_Establishment_of_ a_confirmed_dialogue	subclause 5.1.2.3 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/1
Test purpose		
Sending of 200 OK (INVITE) PSTN XML ProgressIndicat	or is present	
Senaing of 200 OK (IIVVIIE) I SIIV XME I Togressinaica	or is present	
Ensure that on receipt of a DSS1 CONNECT message a S Ensure that if a Progress Indicator IE is present in the COI present in the 200 OK (INVITE) the ProgressDescription indicated in Table 7.2.3.2.2-1. Also, an additional PSTN X ProgressDescription value set to 7	NNECT message, a PSTN XM is derived from the Progress I	AL ProgressIndicator is Description indicator as
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	

←

→

CONNECT

CONNECT ACK

←

→

Apply post test routine

200 OK (INVITE)

ACK

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	TP_302_102	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.3 of	PICS 5.1.1/2 AND 5.4/1
a_confirmed_dialogue		[ETSI TS 183 036]	

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 p	ost 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'

TSS	TP_302_103	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.3 of	PICS 5.1.1/2 AND 5.4/1
a_confirmed_dialogue		[ETSI TS 183 036]	
Test purpose			
Sending of 200 OK (INVIT	E) PSTN XML HighLay	yerCompatibility is present	
			s sent to the calling user equipment.
		esent in the CONNECT messag	e, a PSTN XML cteristics element is derived from
the High Layer Characteris			cteristics element is derived from
SIP header values		lou in Tuble 7.2.3.2.2 3.	
200 OK (INVITE):			
xml version="1.0" encod</td <td>ding="utf-8"?></td> <td></td> <td></td>	ding="utf-8"?>		
PSTN	ung- un o .>		
HighLayerCompatibi	lity		
HLOctet3			
CodingStandard	>00<		
Interpretation>1			
PresentationMet	thod>01<		
HLOctet4			
HighLayerChara	acteristics>HLC_value	*	
DSS1 Parameter values			
CONNECT: High Layer O	Compatibility High Lay	er Characteristics HLC_value	
Message flow			
Test ec	quipment		End device
INVITE	→	→ SETUR	
180 Ringing	←	← ALER	ГING
200 OK (INVITE)	÷	← CONN	ECT
ACK	→	→ CONN	ECT ACK
	An	ply post test routine	

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'

TSS	TP_302_104	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.3 of	PICS 5.1.1/2 AND 5.4/1
a_confirmed_dialogue		[ETSI TS 183 036]	

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.

SIP header values				
200 OK (INVITE):				
xml version="1.0" encoding</td <td>="utf-8"?></td> <td></td> <td></td> <td></td>	="utf-8"?>			
PSTN				
BearerCapability				
BCoctet3				
CodingStandard>00	<			
InformationTransfer	Cabability>ITC_value<			
BCoctet4				
TransferMode>00<				
InformationTransfer	Rate>10000<			
BCoctet5				
Layer1Identification	n>01<			
UserInfoLayer1Prot	ocol>00011<			
DSS1 Parameter values				
CONNECT: Bearer Capability	y Information Transfer Cap	pability ITC_va	lue	
Message flow				
Test equip	ment		End device	
INVITE	→	→	SETUP	
180 Ringing	÷	+	ALERTING	
200 OK (INVITE)	÷	÷	CONNECT	
ACK	→	→	CONNECT ACK	
	Apply pos	st test routine		

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 7.2.3.2.2-4 – Mapping of Bearer Capability to PSTN XML BearerCapability

TSS Term_Establishment_of_	TP_302_105	Reference subclause 5.1.2.3 of	Selection expression PICS 5.1.1/2 AND 5.4/1
a_confirmed_dialogue		[ETSI TS 183 036]	
Test purpose	ator value 7 is sent		
PSTN XML ProgressIndice	nor value / is seni		
		resent in the 200 OK (INVITE) it	
	ECT message. The Progr	essDescription element is set to v	value 7
SIP header values			
200 OK (INVITE):			
PSTM XML MIME body			
xml version="1.0" encod</td <td>ding="utf-8"?></td> <td></td> <td></td>	ding="utf-8"?>		
PSTN			
ProgressIndicator			
ProgressOctet3			
CodingStandard			
Location>yyyy<	<		
ProgressOctet4			
ProgressDescrip	tion>0000111<		
DSS1 Parameter values			
Message flow			
Test eq	quipment	En	d device
INVITE	→	→ SETUP	
180 Ringing	÷	← ALERTIN	G
200 OK (INVITE)	÷	← CONNEC	Г
ACK	→	➔ CONNEC [*]	Т АСК
	Appl	y post test routine	

TSS	TP_302_106	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.3 of	PICS 5.1.1/2 AND 5.4/1
a_confirmed_dialogue		[ETSI TS 183 036]	AND 5.1.3/1

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</td <td colspan="6"><?xml version="1.0" encoding="utf-8"?></td>	xml version="1.0" encoding="utf-8"?					
PSTN						
BearerCapability						
BCoctet3						
CodingStandard>00<						
InformationTransferCabab	ility>00000<					
or						
InformationTransferCabab	ility>10000<					
BearerCapability						
BCoctet3						
CodingStandard>00<						
InformationTransferCabab						
SDP: m line contains as the first code	c CLEARMODE	and as the second	codec a G.711 codec			
200 OK (INVITE):						
<pre><?xml version="1.0" encoding="utf-8</pre></pre>	3"?>					
PSTN						
BearerCapability						
BCoctet3						
CodingStandard>00<						
InformationTransferCabab	ility> 10001 <					
SDP: m line contains as the first code	ec CLEARMOD	E and as the secor	nd codec a G.711 codec			
DSS1 Parameter values						
Message flow						
Test equipment End device						
INVITE						
180 Ringing \leftarrow \leftarrow ALERTING						
200 OK (INVITE) \leftarrow CONNECT						
ACK	→	→	CONNECT ACK			
Apply post test routine						

TSS	TP_302_107	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.3 of	PICS 5.1.1/2 AND 5.4/1
a_confirmed_dialogue		[ETSI TS 183 036]	AND 5.1.3/2

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 bod	у			
xml version="1.0" encoding="u</td <td></td> <td></td> <td></td> <td></td>				
PSTN				
BearerCapability				
BCoctet3				
CodingStandard>00<				
InformationTransferCa	bability>00000<			
or				
InformationTransferCa	bability>10000<			
BearerCapability				
BCoctet3				
CodingStandard>00<				
InformationTransferCa	•			
SDP: m line contains as the first c	odec CLEARMODE a	and as the second	codec a G.711 codec	
200 OK (INVITE): xml version="1.0" encoding="u<br PSTN BearerCapability	ttf-8"?>			
BCoctet3				
CodingStandard>00<				
InformationTransferCa	bability> 00000 <			
DSS1 Parameter values CONNECT : Bearer Capability In	formation Transfor Co	anability – anagal		
Message flow		ipaointy – speeci	1	
-	- 4			
Test equipme			End device	
INVITE	→	→	SETUP	
180 Ringing	+	+	ALERTING	
200 OK (INVITE)	←	4	CONNECT	
ACK	` →	``````````````````````````````````````	CONNECT ACK	
AUN		-	CONTRECT ACK	
	Appiy po	ost test routine		

TSS	TP_302_108	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.3 of	PICS 5.1.1/2 AND 5.4/1
a_confirmed_dialogue		[ETSI TS 183 036]	AND 5.1.3/2

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 be	ody			
xml version="1.0" encoding=</td <td>"utf-8"?></td> <td></td> <td></td> <td></td>	"utf-8"?>			
PSTN				
BearerCapability				
BCoctet3				
CodingStandard>00<				
InformationTransfer	Cabability>00000<			
or				
InformationTransfer	Cabability>10000<			
BearerCapability				
BCoctet3				
CodingStandard>00<				
InformationTransfer				
SDP: m line contains as the first	codec CLEARMODE as	nd as the second	codec a G.711 codec	
200 OK (INVITE): xml version="1.0" encoding=<br PSTN BearerCapability BCoctet3 CodingStandard>00< InformationTransferC DSS1 Parameter values CONNECT: Bearer Capability	Cabability> 10000 <	pability = 3.1 kH	z audio	
Message flow				
Test equipm	nent		End device	
INVITE	→	→	SETUP	
180 Ringing	<	÷	ALERTING	
200 OK (INVITE)	÷	÷	CONNECT	
ACK	→	→	CONNECT ACK	
	Annly no	st test routine		

TSS	TP_302_109	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.3 of	PICS 5.1.1/2 AND 5.4/1
a_confirmed_dialogue		[ETSI TS 183 036]	AND 5.1.3/2

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	7			
xml version="1.0" encoding="u</td <td></td> <td></td> <td></td> <td></td>				
PSTN				
BearerCapability				
BCoctet3				
CodingStandard>00<				
InformationTransferCat	ability>00000<			
or				
InformationTransferCat	ability>10000<			
BearerCapability				
BCoctet3				
CodingStandard>00<				
InformationTransferCat	ability>10001<			
SDP: m line contains as the first co	odec CLEARMODE	and as the second	l codec a G.711 codec	
200 OK (INVITE):				
xml version="1.0" encoding="u</td <td>tf-8"?></td> <td></td> <td></td> <td></td>	tf-8"?>			
PSTN				
BearerCapability				
BCoctet3				
CodingStandard>00<				
InformationTransferCat	ability>10000<			
DSS1 Parameter values				
Message flow				
Test equipmer	nt		End device	
INVITE	→	→	SETUP	
180 Ringing	÷	÷	ALERTING	
200 OK (INVITE)	÷	÷	CONNECT	
ACK	→	→	CONNECT ACK	
	Apply p	ost test routine		

TSS	TP_302_110	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.3 of	PICS 5.1.1/2 AND 5.4/1
a_confirmed_dialogue		[ETSI TS 183 036]	AND 5.1.3/2

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 bod	ly			
xml version="1.0" encoding="1</td <td></td> <td></td> <td></td> <td></td>				
PSTN				
BearerCapability				
BCoctet3				
CodingStandard>00<				
InformationTransferCa	bability>00000<			
or				
InformationTransferCa	bability>10000<			
BearerCapability				
BCoctet3				
CodingStandard>00<				
InformationTransferCa	bability>10001<			
SDP: m line contains as the first c	codec CLEARMODE a	nd as the second	codec a G.711 codec	
200 OK (INVITE):				
xml version="1.0" encoding="1</td <td>utf-8"?></td> <td></td> <td></td> <td></td>	utf-8"?>			
PSTN				
HighLayerCompatibility				
HLOctet3				
CodingStandard>00<				
Interpretation>100<				
PresentationMethod>0	1<			
HLOctet4				
HighLayerCharacterist	ics>HLC_VA<			
DSS1 Parameter values				
CONNECT: High Layer Compat	tibility High Layer Cha	racteristics=HL0	C_VA	
Message flow				
Test equipme	ent		End device	
INVITE	→	→	SETUP	
180 Ringing	÷	+	ALERTING	
200 OK (INVITE)	+	←	CONNECT	
	-	-	· · · -	
ACK	→	→	CONNECT ACK	

TSS	TP_302_111	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.3 of	PICS 5.1.1/2 AND 5.4/1
a_confirmed_dialogue		[ETSI TS 183 036]	AND 5.1.3/2

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

INVITE: PSTN XML MIME	E body		
xml version="1.0" encodin</td <td>-</td> <td></td> <td></td>	-		
PSTN			
BearerCapability			
BCoctet3			
CodingStandard>0	>00<		
InformationTransf	erCabability>00000<		
or			
InformationTransf	erCabability>10000<		
BearerCapability			
BCoctet3			
CodingStandard>0	>0(
InformationTransf	erCabability>10001<		
HighLayerCompatibility	y		
HLOctet4			
HighLayerCharact	teristics>HLC_VA<		
SDP: m line contains as the fi	irst codec CLEARMODE a	ad as the second codec a G.711 codec	
200 OK (INVITE):			
<pre><?xml version="1.0" encodin</pre></pre>	ng-"utf 8"?		
<	ig- uti-0 :>		
PSTN			
PSTN HighLaverCompatibility	V		
PSTN HighLayerCompatibility	y		
HighLayerCompatibility	y		
HighLayerCompatibility HLOctet4			
HighLayerCompatibility HLOctet4 HighLayerCharact	y teristics>HLC_VA<		
HighLayerCompatibility HLOctet4 HighLayerCharact DSS1 Parameter values			
HighLayerCompatibility HLOctet4 HighLayerCharact DSS1 Parameter values Message flow	teristics>HLC_VA<		
HighLayerCompatibility HLOctet4 HighLayerCharact DSS1 Parameter values Message flow Test equi	teristics>HLC_VA<	End device	
HighLayerCompatibility HLOctet4 HighLayerCharact DSS1 Parameter values Message flow Test equi INVITE	teristics>HLC_VA<	→ SETUP	
HighLayerCompatibility HLOctet4 HighLayerCharact DSS1 Parameter values Message flow Test equi	teristics>HLC_VA<		
HighLayerCompatibility HLOctet4 HighLayerCharact DSS1 Parameter values Message flow Test equi INVITE	teristics>HLC_VA<	→ SETUP	
HighLayerCompatibility HLOctet4 HighLayerCharact DSS1 Parameter values Message flow Test equi INVITE 180 Ringing	teristics>HLC_VA<	→ SETUP← ALERTING	

TSS	TP_302_112	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.3 of	PICS 5.1.1/2 AND 5.4/1
a_confirmed_dialogue		[ETSI TS 183 036]	AND 5.1.3/1

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" encodin</td <td>ug="utf-8"?></td> <td></td> <td></td> <td></td>	ug="utf-8"?>			
PSTN				
BearerCapability				
BCoctet3				
CodingStandard>0	>00<			
InformationTransfe	erCabability>00000<			
or				
InformationTransfe	erCabability>10000<			
BearerCapability				
BCoctet3				
CodingStandard>0				
	erCabability>10001<			
SDP: m line contains as the fi	irst codec CLEARMODE a	and as the second	codec a G.711 codec	
SDP: m line contains as the fi DSS1 Parameter values	erCabability> 10001 < irst codec CLEARMODE a		codec a G.711 codec nudio, Progress Indicator Progr	ess descriptior
Message flow				
Test equi	pment		End device	
INVITE	→	→	SETUP	
180 Ringing	÷	÷	ALERTING	
	←	+	CONNECT	
200 OK (INVITE)	x			
200 OK (INVITE) ACK	× →	→	CONNECT ACK	

TSS	TP_302_113	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.3 of	PICS 5.1.1/2 AND 5.4/1
a_confirmed_dialogue		[ETSI TS 183 036]	AND 5.1.3/1

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 bod	y			
xml version="1.0" encoding="</td <td>utf-8"?></td> <td></td> <td></td> <td></td>	utf-8"?>			
PSTN				
BearerCapability				
BCoctet3				
CodingStandard>00<				
InformationTransferCa	bability>00000<			
or				
InformationTransferCa	bability>10000<			
BearerCapability				
BCoctet3				
CodingStandard>00<				
InformationTransferCa	bability>10001<			
SDP: m line contains as the first of	codec CLEARMODE	and as the second	l codec a G.711 codec	
200 OK (INVITE):				
xml version="1.0" encoding="</td <td>utf-8"?></td> <td></td> <td></td> <td></td>	utf-8"?>			
BearerCapability				
BCoctet3				
CodingStandard>00<				
InformationTransferCa	bability> 00000 <			
ProgressIndicator				
ProgressOctet4				
ProgressDescription>0	000101<			
DSS1 Parameter values				
CONNECT: Bearer Capability In	nformation Transfer C	apability = speecl	h, Progress Indicator Progress descrip	tion $= 5$
Message flow				
Test equipme	ent		End device	
INVITE	→	→	SETUP	
180 Ringing	←	+	ALERTING	
200 OK (INVITE)	←	←	CONNECT	
ACK	`` →	÷	CONNECT ACK	
AUN	-	-	CONNECT ACK	
	Apply p	ost test routine		

TSS	TP_302_114	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.3 of	PICS 5.1.1/2 AND 5.4/1
a_confirmed_dialogue		[ETSI TS 183 036]	AND 5.1.3/1

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</td <td></td> <td></td> <td></td>			
PSTN	-		
BearerCapability			
BCoctet3			
CodingStandard>00)<		
InformationTransfe	erCabability>00000<		
or			
InformationTransfe	rCabability>10000<		
BearerCapability			
BCoctet3			
CodingStandard>00)<		
InformationTransfe	rCabability>10001<		
SDP: m line contains as the fin	rst codec CLEARMODE an	nd as the second codec a G.711 codec	
200 OK (INVITE):			
xml version="1.0" encoding</td <td>g="utf-8"?></td> <td></td> <td></td>	g="utf-8"?>		
BearerCapability			
BCoctet3			
CodingStandard>00)<		
_	rCabability> 10000 <		
ProgressIndicator	J		
ProgressOctet4			
ProgressDescription	n> 0000101 <		
DSS1 Parameter values			
	y Information Transfer Cap	bability = speech, Progress Indicator Progress descri	otion $= 5$
Message flow	· · · · ·		
Test equip	oment	End device	
INVITE	→	→ SETUP	
180 Ringing	+	← ALERTING	
200 OK (INVITE)	←	← CONNECT	
,	_		
ACK	→	\rightarrow CONNECT ACK	

TSS	TP_302_115	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.3 of	PICS 5.1.1/2 AND 5.4/1
a_confirmed_dialogue		[ETSI TS 183 036]	AND 5.1.3/1

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 b	ody			
xml version="1.0" encoding=</td <td>="utf-8"?></td> <td></td> <td></td> <td></td>	="utf-8"?>			
PSTN				
BearerCapability				
BCoctet3				
CodingStandard>00-	<			
InformationTransfer	Cabability>00000<			
or				
InformationTransfer	Cabability>10000<			
BearerCapability				
BCoctet3				
CodingStandard>00	<			
InformationTransfer	•			
SDP: m line contains as the first	at codec CLEARMODE and	nd as the second	codec a G.711 codec	
200 OK (INVITE):				
xml version="1.0" encoding:</td <td>="utf-8"?></td> <td></td> <td></td> <td></td>	="utf-8"?>			
BearerCapability				
BCoctet3				
CodingStandard>00	<			
InformationTransfer	Cabability> 10000 <			
ProgressIndicator				
ProgressOctet4				
ProgressDescription	>0000101<			
DSS1 Parameter values				
Message flow				
Test equip	ment		End device	
INVITE	→	→	SETUP	
180 Ringing	<	+	ALERTING	
200 OK (INVITE)	+	+	CONNECT	
ACK	→	→	CONNECT ACK	
		st test routine		

TSS	TP_302_116	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.3 of	PICS 5.1.1/2 AND 5.4/1
a_confirmed_dialogue		[ETSI TS 183 036]	AND 5.1.3/1

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	<				
InformationTransfer	Cabability>00000<				
or					
InformationTransfer	Cabability>10000<				
BearerCapability					
BCoctet3					
CodingStandard>00	<				
InformationTransfer	Cabability>10001<				
SDP: m line contains as the first	st codec CLEARMODE a	nd as the second codec a G.711 codec			
200 OK (INVITE):					
xml version="1.0" encoding</td <td>="utf-8"?></td> <td></td>	="utf-8"?>				
HighLayerCompatibility					
HLOctet3					
CodingStandard>00	<				
Interpretation>100<					
PresentationMethod					
HLOctet4					
HighLayerCharacter	ristics>HLC VA<				
DSS1 Parameter values	—				
CONNECT : High Layer Com	natibility High Laver Ch	aracteristics=HLC VA			
Message flow	pullollily, lligh 249 cl chi				
Test equip	ment	End device			
INVITE	→	→ SETUP			
180 Ringing	÷	← ALERTING			
200 OK (INVITE)	←	← CONNECT			
ACK	→	→ CONNECT ACK			
	-				

TSS	TP_302_117	Reference	Selection expression
Term_Establishment_of_		subclause 5.1.2.3 of	PICS 5.1.1/2 AND 5.4/1
a_confirmed_dialogue		[ETSI TS 183 036]	AND 5.1.3/1

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</td <td>g="utf-8"?></td> <td></td> <td></td>	g="utf-8"?>		
PSTN			
BearerCapability			
BCoctet3			
CodingStandard>0			
InformationTransfe	erCabability>00000<		
or			
	erCabability>10000<		
BearerCapability			
BCoctet3			
CodingStandard>0			
InformationTransfe SDP: m line contains as the fi	-	d as the second codec a G.711 codec	
SDP: m line contains as the fi 200 OK (INVITE): DSS1 Parameter values	rst codec CLEARMODE an		
SDP: m line contains as the fi 200 OK (INVITE): DSS1 Parameter values CONNECT: High Layer Con	rst codec CLEARMODE an		
SDP: m line contains as the fi 200 OK (INVITE): DSS1 Parameter values CONNECT: High Layer Con	rst codec CLEARMODE an		
SDP: m line contains as the fi 200 OK (INVITE): DSS1 Parameter values CONNECT: High Layer Con Message flow	rst codec CLEARMODE an	acteristics= HLC_VA	
SDP: m line contains as the fi 200 OK (INVITE): DSS1 Parameter values CONNECT: High Layer Con Message flow Test equij INVITE	rst codec CLEARMODE an npatibility, High Layer Char pment	acteristics=HLC_VA End device	
SDP: m line contains as the fi 200 OK (INVITE): DSS1 Parameter values CONNECT: High Layer Con Message flow Test equip	rst codec CLEARMODE an npatibility, High Layer Char pment →	acteristics=HLC_VA End device → SETUP	
SDP: m line contains as the fi 200 OK (INVITE): DSS1 Parameter values CONNECT: High Layer Con Message flow Test equij INVITE 180 Ringing	rst codec CLEARMODE an npatibility, High Layer Char pment ★ €	acteristics=HLC_VA End device → SETUP ← ALERTING	

7.2.3.3.1 Release initiated by the originating user

TSS	TP_303_001	Reference	Selection expression
Term_Release_initiated_		section 15 of	
$by_the_originating_user$		[IETF RFC 3261]	
		and	
		5.1.2.4 of [ETSI TS 183 036]	
Test purpose			
BYE received in the confi	irmed dialogue		
Ensure that that the IUT, response.	while a session has been	established, on receipt of a BYE requ	est sends a Success (200 OK)
SIP header values			
Message flow			
Test	equipment	End	levice
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	TP_303_002	Reference	Selection expression
Term_Release_initiated_		sections 15 and 12 of	-
by_the_originating_user		[IETF RFC 3261]	
Test purpose			
BYE received in the early of	lialogue		
Ensure that the IUT, while	the dialogue is in an early	stage, on receipt of a BYE reque	est sends a response.
SIP header values			
Message flow			
Test ec	luipment	Enc	d device
Interworking POTS			
INVITE	→	Ringing	
180 Ringing	+		
BYE	→		
200 OK BYE	+		
487 Request Terminated	+		
ACK	→		
ISDN interworking			
INVITE	→	→ SETUP	
180 Ringing	+	← ALERTING	G
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: testi	ng		

Message flow			
Test equip	ment	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	TP_303_004	Reference	Selection expression
Term_Release_initiated_		sections 8.2.6.2 and 15.1.2	
by_the_originating_user		of [IETF RFC 3261]	

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

0		
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 numeros			

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 <i>BYE received in the confirm</i>	ned 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='101	10' (network beyond interw	orking point)	
Message flow			
Test equip	oment		End device
INVITE	→	→	SETUP
180 Ringing	←	÷	ALERTING
200 OK (INVITE)	←	←	CONNECT
ACK	→	→	CONNECT ACK
BYE	→	→	DISCONNECT
200 OK BYE	+	←	RELEASE
		→	RELEASE COMPLETE
	Apply pos	st 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 equipmen	t	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 <i>CANCEL received before a</i>	an early dialogue is establ	ished	
		an early dialogue is established ment. The location is coded '10	, a RELEASE or RELEASE 10' network beyond interworking
SIP header values RELEASE or RELEASE O	COMPLETE: location='10	10' (network beyond interworki	ng point)
DSS1 Parameter values			
Message flow			
Test eq	luipment	E	nd device
INVITE	→	→ SETUP	
100 Trying	+		
CANCEL	→		
200 OK CANCEL	÷		
CASE A		→ RELEAS	E
		← RELEASI	E COMPLETE
CASE B			
		→ RELEASI	E 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 <i>BYE received after an earl</i>	Test purpose BYE received after an early dialogue is established						
	r RELEASE COMPLETE	y dialogue by means of sending message is sent to the called use					
SIP header values RELEASE or RELEASE O	SIP header values RELEASE or RELEASE COMPLETE: location='1010' (network beyond interworking point)						
DSS1 Parameter values							
Message flow							
Test eq	uipment	En	nd device				
INVITE	→	→ SETUP					
183 Session Progress	<	← CALL PR	OCEEDING				
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
	early dialogue is established		
	r RELEASE COMPLETE mes		ding a 183 Session Progress is er equipment. The location is
SIP header values			
	COMPLETE: location='1010' (network beyond interworkir	ng point)
DSS1 Parameter values			
Message flow			
	uipment		d device
INVITE	→	→ SETUP	
183 Session Progress	+	← CALL PR	OCEEDING
CANCEL	→		
200 OK CANCEL	÷		
CASE A		→ RELEASI	E
		← RELEASE	COMPLETE
CASE B			
		→ RELEASE	E 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	l 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
197 Dequest Termineted	÷		
487 Request Terminated	-		
ACK	→		

erence Selection expression PICS 5.1.1/2 SI TS 183 036] Subseque by means of sending a 180 Ringing is sent to the called user equipment. The location is	
SI TS 183 036] ogue by means of sending a 180 Ringing is	
k beyond interworking point)	
End device	
→ SETUP	
← CALL PROCEEDING	
→ RELEASE	
← RELEASE COMPLETE	
\rightarrow RELEASE COMPLETE	
k	

→

ACK

TSS	TP_304_001	Reference		Selection expression	
Term_Release_initiated_		sections 17.2	.3.1, 13.3.1.4	-	
by_the_terminating_user		and Figure 7			
		[IETF RFC 3	3261]		
Test purpose					
The transaction enters the	completed state				
Ensure that the IUT when a	a server INVITE transaction is	in the Proceedi	ng state, after se	ending a unsuccessful final	
response, enters in the Con	pleted state.				
SIP header values					
Message flow					
Test eq	Test equipment End device				
Interworking POTS					
INVITE	→		Ringing		
180 Ringing	÷				
4xx response	+		Reject call		
	Verify con	npleted state			
ISDN interworking					
INVITE	→	7	SETUP		
180 Ringing	+	+	ALERTING	~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~	
4xx response	+	+	RELEASE_C	OMPLETE	
	Verify co	npleted state			

TSS	TP_304_002	Reference	Selection expression
Term_Release_initiated_		sections 17.2.3.1, 13.3.1.4	
by_the_terminating_user		and Figure 7of	
		[IETF RFC 3261]	

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 equ	ipment	End device
Interworking POTS		
INVITE	→	Ringing
180 Ringing	+	
4xx_response	+	Reject call
ACK	→	
	Verify con	nfirmed state
ISDN interworking		
INVITE	→	→ SETUP
180 Ringing	+	← ALERTING
4xx_response	+	← RELEASE_COMPLETE
ACK	→	
	Verify co	nfirmed state

7.2.3.3.2 Release initiated by the terminating user

TSS	TP_304_003	Reference	Selection expression	
Term_Release_initiated_		section 12.2.1.1 of	-	
by_the_terminating_user		[IETF RFC 3261]		
Test purpose				
To header in the BYE is se	t to the From header in th	ne previous received request		
Ensure that the IUT, once the same value as in the Fr			E request with a To header set to	
SIP header values	on neader of the previou	s received request.		
INVITE				
	value;tag=any_invite1_j	from tag value		
		<u> </u>		
BYE:				
To: any_invite_from_v	alue;tag=any_invite1_fro	m_tag_value		
Message flow				
Test e	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	+	← ALERTIN	١G	
200 OK INVITE	+	← CONNEC	T	
ACK	→			
BYE	+	← RELEAS	E	
200 OK BYE	→	→ RELEAS	E COMPLETE	

TSS	TP_304_004	Reference	Selection expression
Term_Release_initiated_		section 12.2.1.1/	
by_the_terminating_user		[IETF RFC 3261]	

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	TP_304_005	Reference	Selection expression
Term_Release_initiated_		section 12.2.1.1 of	_
by_the_terminating_user		[IETF RFC 3261]	

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

Wiessage now		
Test equipment		End device
Interworking POTS		
INVITE	→	Ringing
180 Ringing	F	
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		L J	
	E is set to the value of the	e Contact header in the received	request
		lished with an INVITE request in in a lr parameter to release the ca	
		ader set to the list in a reverse or	
in the original INVITE req	uest.		
SIP header values			
INVITE:			
Contact: < <i>any_invite_c</i>			
Record-Route: <sip:an< td=""><td></td><td></td><td></td></sip:an<>			
Record-Route: <sip:an< td=""><td>y_invite_value2;lr></td><td></td><td></td></sip:an<>	y_invite_value2;lr>		
BYE:			
BYE sip: <i>any_invite_co</i>	ontact_value SIP/2.0		
Route: <sip:<i>any_invite_</sip:<i>			
Route: <sip:any_invite_< td=""><td>_value1></td><td></td><td></td></sip:any_invite_<>	_value1>		
Message flow			
Test ec	quipment	Er	nd 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	+	← ALERTIN	IG
200 OK INVITE	÷	← CONNEC	Т
ACK	→		
BYE	÷	← RELEASE	3

TSS	TP_304_007	Reference	Selection expression
Term_Release_initiated_		section 12.2.1.1 of	_
by_the_terminating_user		[IETF RFC 3261]	

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_cont< td=""><td>tact_value></td><td></td><td></td></any_invite_cont<>	tact_value>		
Record-Route: <sip:any_i< td=""><td>nvite_value1;lr></td><td></td><td></td></sip:any_i<>	nvite_value1;lr>		
Record-Route: <sip:any_in< td=""><td>nvite_value2></td><td></td><td></td></sip:any_in<>	nvite_value2>		
BYE:			
BYE sip: any_invite_value	e1 SIP/2.0		
Route: <sip:any_invite_va< td=""><td>lue2></td><td></td><td></td></sip:any_invite_va<>	lue2>		
Route: <sip: any_invite_co<="" td=""><td>ontact_value></td><td></td><td></td></sip:>	ontact_value>		
Message flow			
Test equi	pment	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
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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_userTP_304_102Reference 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	
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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		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 di	ialogue is established		
Session terminated by the C	uncu purty before un curty u	alogue is established		
received from the called u Status code to be sent is o	ser equipment before a provi	uipment if a DSS1 RELEASE sional response was sent to th Cause code value received in 2.	e calling user equipment. The	
SIP header values				
Status code				
DSS1 Parameter values RELEASE: Cause - Cause value or RELEASE COMPLETE: Cause - Cause value				
Message flow				
	uipment		device	
INVITE	→	→ SETUP		
100 Trying	+	← ALERTING		
CASE A				
SIP_final_response_VA	+	← RELEASE		
ACK	→	→ RELEASE C	COMPLETE	
CASE B	_	_		
SIP_final_response_VA	+	← RELEASE C	COMPLETE	
ACK	→			

	←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 7.2.3.3.2.2-1 – 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	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)Cause value in the Class 010 (resource unav Cause value No 34)else 503 Service UnavailableCause 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)

Table 7.2.3.3.2.2-2 – Interworking of Release causes to SIP status codes

7.2.3.4 Timers

TSS Term_Timers	TP_305_001	Reference section 17.2.1 and Annex A of [IETF RFC 3261]	Selection PICS 5.1.2/1	expression
Test purpose				

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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 dev	vice	Test equipment
Interworking POTS		
INVITE	→	Ringing
180 Ringing	+	
200 OK INVITE	← Start time	G On hook
200 OK INVITE	← Timeout tim	er 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	Referent section Annex	17.2.1 and	Selection expression PICS 5.1.2/2
		[IETF I	RFC 3261]	
Test purpose				·
Timeout timer G 200	OK is not repeated			
	le transport (TCP) is used ponse on the timeout con			nsaction is in the Completed state
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	+		← ALERTIN	١G
200 OK INVITE	+	Start timer G	← CONNEC	CT
		Timeout timer G		
ACK	→			
		Apply post test rout	tine	

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 <i>Timer G is started 3 time</i>				
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	Annex	17.2.1 and	Selection expression
Test purpose				
Timeout timer H an ACK	is sent			
Ensure that the IUT, when the timer H, which is set t			Completed state, it	enters the Terminated state after
SIP header values				
Message flow				
End	l 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	s Not Exist			
ISDN interworking				
INVITE	→		→ SETUP	
180 Ringing	÷		← ALERTIN	G
200 OK INVITE	÷	Start timer H	← CONNEC	Т
		Timeout timer H		
ACK	→			
481 Call/Transaction does	s Not Exist 🗧 🗲		➔ DISCONN	NECT
			← RELEASE	E
			→ RELEASE	ECOMPLETE

TSS Term_Timers	TP_305_006	Reference section 17.2.1 and Annex A of [IETF RFC 3261]	
Test purpose			
Timeout timer H an ACK is	s sent		
Ensure that the IUT, when timer H, which is set to 64		action is in the Completed	state, does not repeat its response after
SIP header values			
Message flow			
End	device		Test equipment
Interworking POTS			
INVITE	→	Rin	ging
180 Ringing	+		
200 OK INVITE	← 5	Start timer H On	hook
200 OK INVITE	+		
	Tin	meout timer H	
ACK	→		
481 Call/Transaction does	Not Exist 🗲		
ISDN interworking			
INVITE	→	→ SET	ГИР
180 Ringing	+	← AL	ERTING
200 OK INVITE	← .5	Start timer H 🗧 🗲 CO	NNECT
200 OK INVITE	+		
	Ti	meout timer H	
ACK	→	→ DIS	SCONNECT
481 Call/Transaction does	Not Exist 🗧 🗲	← REI	LEASE
		→ RE	LEASE 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, v Terminated state. SIP header values	when an INVITE serve	r transaction is in the Confirme	ed state, enters immediately in the
Message flow			
E	nd device	Т	lest equipment
Interworking POTS			
INVITE	→	Ringin	ng
180 Ringing	←		
200 OK INVITE	←	On hoo	ok
ACK	→	Start timer I	
		Timeout timer I	
ACK	→		
481 Call/Transaction do	es Not Exist 🗧 🗲		
for cull runsaction do			

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	TP_305_009	Reference	ce	Selection expression
Term_Timers			3.3.1.4 and	
		Annex A		
		[IETF R	FC 3261]	
Test purpose				
200 OK INVITE recei	ved Timer T1 is started			
Ensure that the IUT, expires without receiv		an INVITE request	with 2XX respons	e, repeats it after T1 duration
SIP header values				
Message flow				
	End device		Test eq	uipment
Interworking POTS				
INVITE	→		Ringing	
180 Ringing	+			
200 OK INVITE	+	Start timer T1	On hook	
200 OK INVITE	← ′	Timeout timer T1		
ACK	→			
	Α	pply post test routi	ne	
ISDN interworking				
INVITE	→		→ SETUP	
	,			
180 Ringing		G		
200 OK INVITE	+	Start timer T1	← CONNECT	
200 OK INVITE	_	Timeout timer T1		
ACK	→			
	A	pply post test routi	ne	

TSS Term_Timers	TP_305_010	Annex	13.3.1.4 and	Selection expression
Test purpose 200 OK INVITE receiv	ed Timer T1 is started 2	time		
	then it has already answe without receiving an AC		VITE request with a 2	2XX response, repeats it after
SIP header values				
Message flow				
I	End device		Test equ	uipment
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	→			
	P	Apply post test rou	tine	
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	→			
	A	Apply post test rou	tine	

TSS Term_Timers	TP_305_011	Reference section 13.3.1.4 and Annex A of [IETF RFC 3261]	Selection expression			
Test purpose <i>Timer T1 was started 64 tin</i>	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 6	54*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	
ВҮЕ	+	➔ DISCONNECT
200 OK BYE	→	← RELEASE
		→ RELEASE COMPLETE

TSS Term_Timers	TP_305_013	Reference section 1 Annex A	7.2.2 and	Selection expression PICS 5.1.2/1
			FC 3261]	
Test purpose				
BYE is retransmitted a	fter exiry of 64*T1			
		saction is in the Comple er J set to 64*T1 expire		pt of the repetitions of the BYE
SIP header values				
Message flow				
1	End device		Test e	quipment
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	+		← ALERTIN	٩G
200 OK INVITE	+		← CONNEC	T
ACK	→			
BYE	→	Start timer J 64*T1	➔ DISCONI	NECT
200 OK BYE	÷		← RELEAS	E
			→ RELEAS	E COMPLETE
BYE	→	Wait 31.5 sec		
200 OK BYE	÷			

TSS T Term_Timers	P_305_014	Reference section 17.2.2 and Annex A of [IETF RFC 3261]	Selection expression PICS 5.1.2/1
Test purpose <i>Timeout timer J</i> Ensure that the IUT, when a B request, retransmits its respon not retransmitted.	BYE server transactions use until the timer J set	n is in the Completed state, o et to 64*T1 expires. After exp	on receipt of the repetitions of the BYE piry of timer J the 200 OK response is
SIP header values			
Message flow			
End de	vice		Test equipment
Interworking POTS			
INVITE	→	Rin	nging
180 Ringing	←		
200 OK INVITE	←	On	hook
ACK	→		
BYE	→ Sta	rt timer J 64*T1	
200 OK BYE	←		
	Т	imeout timer J	
BYE	→		
Case A			
CASE B			
481 Call/Transaction does No	ot Exist 🗲		
ISDN interworking			
INVITE	→	→ SE	TUP
180 Ringing	÷	← AL	ERTING
200 OK INVITE	÷	← CO	NNECT
ACK	→		
BYE	→ Sta	rt timer J 64*T1 → DIS	SCONNECT
200 OK BYE	÷	← RE	LEASE
		→ RE	LEASE COMPLETE
	Т	imeout timer J	
BYE	→		
Case A			
CASE B			
481 Call/Transaction does No	ot Exist 🗲		

7.2.3.5 Abnormal situations

TSS	TP_306_001	Reference	Selection expression
Term_Abnormal_situations		section 8.2.2.1 of [IETF RFC 3261]	
Test purpose		·	·
A call setup is rejected due to a	n unknown IRU schem	e	
Ensure that the IUT on receipt sends an Unsupported URI sche			a scheme that it does not support,
SIP header values			
INVITE:			
INVITE got: <any destin<="" td=""><td>ation URI> SIP/2.0</td><td></td><td></td></any>	ation URI> SIP/2.0		
Message flow			
Test equipr	nent	Ε	nd device
Interworking POTS			
INVITE	→		
416 Unsupported URI scheme	÷		
ACK	→		
ISDN interworking			
INVITE	→		
416 Unsupported URI scheme	←		
ACK	→		
	Apply	post test routine	

TSS Term_Abnormal_situations	TP_306_002	Reference section 8.2.2.1 of [IETF RFC 3261]	Selection expression
Test purpose			
A call setup is rejected due to	an unknown destinatio	n user address	
Ensure that the IUT on receip sends a Not Found (404 Not F		t with a Request-URI set with a	an address that it does not accept,
SIP header values			
INVITE			
INVITE sip: <any td="" unkr<=""><td>nown destination URI></td><td>SIP/2.0</td><td></td></any>	nown destination URI>	SIP/2.0	
Message flow			
Test equi	pment	E	nd device
Interworking POTS			
INVITE	→		
404 Not Found	+		
ACK	→		
ISDN interworking			
INVITE	→		
404 Not Found	+		
ACK	→		
Apply post test routine			

TSS	TP_306_003	Reference	Selection expression
Term_Abnormal_situations		section 13.3.1 of [IETF RFC 3261]	
Test purpose		·	·
A call setup is rejected due to	an Expires header set	to 0	
Ensure that the IUT on receipt (487 Request Terminated) res		including an Expires header set	to 0, sends a Request Terminated
SIP header values			
INVITE			
Expires: 0			
Message flow			
Test equi	oment	Er	nd device
Interworking POTS			
INVITE	→		
487 Request Terminated	←		
ACK	→		
ISDN interworking			
INVITE	→		
487 Request Terminated	+		
ACK	→		
	Apply	y post test routine	

TSS Term_Abnormal_situations	TP_306_004	Reference sections 8.2.3, 13.2.1, 13.3.1 and 20.11 of [IETF RFC 3261]	Selection expression
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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.

SIP header values

INVITE Content-Language: by Content-Disposition: session

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 routi	ine

TSS	TP_306_005	Reference	Selection expression
Term_Abnormal_situations		sections 8.2.3, 21.4.13,	
		13.2.1, 13.3.1 and 20.11 of [IETF RFC 3261]	
T			
Test purpose	a Contant Dispositio	on handlingparameter set to require	d
A cuit setup is rejected due to	a Content-Dispositio	m nunuingpurumeier sei io require	u
Ensure that the IUT on rece	ipt of an INVITE re	equest including a Content-Langua	ge header value that it cannot
understand, a Content-Dispos	ition with a handling	set to "required" and a disposition	n-types set to session, sends an
Unsupported Media Type (41 bodies it understands.	5 Unsupported Medi	ia Type) response with an Accept l	neader that lists the types of all
SIP header values			
INVITE Content-Language: by			
Content-Disposition: se	ession handling – reg	uired	
Content Disposition. S		uneu	
415:			
Accept:			
Accept-Language:			
Message flow			
Test equij	oment	Enc	l device
Interworking POTS			
INVITE	→		
415 Unsupported Media Type	÷		
ACK	→		
ISDN interworking			
INVITE	→		
415 Unsupported Media Type	_		
ACK	× →		
	Ann	ly 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, reje	ected with 415		
	ds an Unsupported Medi	uding a Content-Encoding heade ia Type (415 Unsupported Me t it understands.	
SIP header values			
INVITE			
Content-Encoding: unsupp	orted		
415:			
Accept-Encoding:			
Message flow			
Test equip	oment	End d	levice
Interworking POTS		2000	
INVITE	→		
415 Unsupported Media Type	÷		
ACK	→		
ISDN interworking			
INVITE	→		
415 Unsupported Media Type	÷		
ACK	→		
	Apply po	st 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 set to session value, includes in			ontent-Disposition header not
SIP header values			
INVITE	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 pos	t test routine
ISDN interworking		
INVITE	→	→ SETUP
200 OK INVITE	÷	← CONNECT
ACK	→	
	Apply pos	t test routine

TSS	TP_306_008	Reference	Selection expression
Term_Abnormal_situations		sections 13.2.1, 13.3.1 and	
		20.11 of [IETF RFC 3261]	

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

SDP			
Message flow			
Test equi	pment	End device	
Interworking POTS			
INVITE	→	Ringing	
200 OK INVITE	÷	On hook	
ACK	→		
	Apply pos	t test routine	
ISDN interworking			
INVITE	→	→ SETUP	
200 OK INVITE	←	← CONNECT	
ACK	→		
	Apply pos	t test routine	

TSS	TP_306_009	Reference	Solaction expression
Term_Abnormal_situations	TP_300_009	section 13.3.1.3 and	Selection expression
Term_Abilormal_situations		21.4.13 of	
		[IETF RFC 3261]	
Test purpose			
SDP not acceptable, rejected v	with 188		
SD1 nor acceptable, rejected (viin 400		
Ensure that the IUT on receipt	of an INVITE request in	cluding a session description the	at it can understand but it cannot
			pes set to session, sends a Not
Acceptable Here (488 Not Acc			
SIP header values			
INVITE			
Content- Disposition: sessi	on:handling=required		
SDP not acceptable	,		
488:			
Warning:			
Message flow			
Test equip	mont	End	l device
Interworking POTS	ment	En	l'uevice
INVITE	→		
488 Not Acceptable Here	÷		
ACK	+		
ACK	7		
ISDN interworking			
INVITE	→		
	+ +		
488 Not Acceptable Here			
ACK	→		

TSS	TP_306_010	Reference	Selection expression
Term_Abnormal_situations	11_000_010	sections 17.2.3, 8.2.2.3	
		and Figure 7 of	
		[IETF RFC 3261]	
Test purpose			
Proxy-Require header with an	unsupported option-t	ag in ACK	
		n is in the Completed state, on recei	
a Proxy-Require header set wi	th an option-tag that i	t does not support, enters in the Con	nfirmed transaction state.
SIP header values			
ACK			
Proxy-Require: unsupporte	ed		
Message flow			
Test equip	oment	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	→		
		ïrmed dialogue st test routine	2

TSS Term_Abnormal_situatio ns	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 un	supported option-tag in ACK	ζ.	
		s in the Completed state, on rec support, enters in the Confirm	eipt of an ACK request including led transaction state.
SIP header values			
ACK			
Require: unsupported			
Message flow			
Test eq	luipment	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	Г
ACK	→		
		nfirmed dialogue ost test routine	

TSS Term_Abnormal_situatio ns	TP_306_012	Reference section 17.2.1 and 17.2.3 of [IETF RFC 3261]	Selection expression
Test			l.

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_valu	e;branch= z9hG4bK any_in	nvite1_branch_value
<pre>From: any_invite1_from_v</pre>	value;tag=any_invite1_fron	n_tag_value
To: any_invite1_to_value		
Call-ID: any_invite1_call-	id_value	
CSeq: any_invite1_cseq		
INVITE2:		
Via: any_invite1_via_valu	e;branch= z9hG4bK_any_a	invite1_branch_value
<pre>From: any_invite1_from_v</pre>	value;tag=any_invite1_fron	n_tag_value
To: any_invite1_to_value		
Call-ID: any_invite1_call-	id_value	
CSeq: any_invite1_cseq		
Message flow		
Test equi	pment	End device
Interworking POTS		
INVITE1	→	Ringing
3xx – 6xx response	←	
INVITE2	→	
3xx – 6xx response	←	
АСК	→	
ISDN interworking		
INVITE1	→	→ SETUP
3xx – 6xx response	←	← RELEASE_COMPLETE
INVITE2	→	
3xx – 6xx response	←	
ACK	→	

TSS Term_Abnormal_situationsTP_306_013	Reference section 17.2.1 and 17.2.3 of [IETF RFC 3261]	Selection expression
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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: <i>any_invite1_via_value</i> ;	oranch= z9hG4bK any	invite1 branch value	
From: <i>any_invite1_from_val</i>	•		
To: any_invite1_to_value	, , , , , , ,	_ 0_	
Call-ID: any_invite1_call-id	_value		
CSeq: any_invite1_cseq			
INVITE2:			
Via: any_invite1_via_value;	oranch= z9hG4bK_any	invite1_branch_value	
Via: any_invite2_via_value			
<pre>From: any_invite1_from_val</pre>	ue;tag=any_invite1_fro	pm_tag_value	
To: any_invite1_to_value			
Call-ID: any_invite1_call-id	_value		
CSeq: any_invite1_cseq			
Message flow			
Test equipn	nent	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_situationsT	TP_306_014	Reference section 17.2.1 and 17.2.3 of [IETF RFC 3261]	Selection expression
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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: <i>any_invite1_via_value</i>	•	
From: <i>any_invite1_from_v</i>	alue;tag=any_invite1_fr	om_tag_value
To: any_invite1_to_value	• 1 1	
Call-ID: any_invite1_call-	d_value	
CSeq: any_invite1_cseq		
INVITE2:		
Via: any_invite1_via_value	e;branch= z9hG4bK any	_invite1_branch_value
Via: any_invite1_via_value	e;branch= any_invite2_b	pranch_value
From: <i>any_invite1_from_v</i>	alue;tag=any_invite1_fr	om_tag_value
To: any_invite1_to_value		
Call-ID: any_invite1_call-	id_value	
CSeq: any_invite1_cseq		
Message flow		
Test equip	ment	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
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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:				
	e;branch= z9hG4bK any_ir	wite1 branch value		
•	value;tag=any_invite1_from			
To: any_invite1_to_value				
Call-ID: any_invite1_call-	id_value			
CSeq: any_invite1_cseq				
INVITE2:				
Via: any_invite1_via_valu	e;branch= z9hG4bK_any_a	invite2_branch_value		
From: <i>any_invite1_from_</i>	value;tag=any_invite1_from	n_tag_value		
To: any_invite1_to_value				
Call-ID: any_invite1_call-	id_value			
CSeq: any_invite1_cseq				
Message flow				
Test equi	pment	End device		
Interworking POTS				
INVITE1	→	Ringing		
180 Ringing	+			
INVITE2	→			
482 Loop Detected	÷			
ACK	→			
	Apply pos	st test routine		
ISDN interworking				
INVITE1	→	→ SETUP		
180 Ringing	←	← ALERTING		
INVITE2	→			
482 Loop Detected	←	→ RELEASE		
ACK	$\leftarrow \text{ RELEASE_COMPLETE}$			

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
		[IETF KFC 3201]	

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_valu	e;branch= z9hG4bK any_	_invite1_branch_value
<pre>From: any_invite1_from_v</pre>	alue;tag=any_invite1_fro	om_tag_value
To: any_invite1_to_value		
Call-ID: any_invite1_call-	id_value	
CSeq: any_invite1_cseq		
INVITE2:		
Via: any_invite2_via_valu	e;branch= z9hG4bK_any	_invite1_branch_value
<pre>From: any_invite1_from_v</pre>	alue;tag=any_invite1_fre	om_tag_value
To: any_invite1_to_value		
Call-ID: any_invite1_call-	id_value	
CSeq: any_invite1_cseq		
Message flow		
Test equip	oment	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
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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_valu	e;branch= z9hG4bK any_i	wite1_branch_value
<pre>From: any_invite1_from_v</pre>	alue;tag=any_invite1_fror	n_tag_value
To: any_invite1_to_value		
Call-ID: any_invite1_call-	id_value	
CSeq: any_invite1_cseq		
INVITE2:		
Via: any_invite2_via_valu	e;branch= z9hG4bK_any_	invite2_branch_value
<pre>From: any_invite1_from_v</pre>	alue;tag=any_invite1_from	n_tag_value
To: any_invite1_to_value		
Call-ID: any_invite1_call-	id_value	
CSeq: any_invite1_cseq		
Message flow		
Test equip	oment	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
102 Loop Detterted	-	

TSS	TP_306_018	Reference	Selection expression
Term_Abnormal_situations		section 8.2.2.3 of	L L
		[IETF RFC 3261]	

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	TP_306_019	Reference	Selection expression
Term_Abnormal_situations		section 8.2.2 and 15.1.2 of [IETF RFC 3261]	

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 equipm	nent	End device
Interworking POTS		
INVITE	→	
180 Ringing	+	Ringing
200 OK INVITE	←	Off hook
ACK	→	
BYE	→	
420 Bad Extension	←	
ISDN interworking		
	2	→ SETUP
INVITE	→	
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 exis	at a start star			
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 equip	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 receive	ed, 481 is sent		
Ensure that the IUT, while a disends a Call/Transaction does			est without TAG in the To header,
SIP header values			
INVITE:			
From: <i>any_invite1_from_v</i>	alue;tag=any_invite1_	from_tag_value	
To: any_invite1_to_value			
BYE:			
From: <i>any_invite1_from_v</i>	alue;tag=any invite1	from tag value	
To: any_invite1_to_value		• _ 0_	
Message flow			
Test equip	oment	Ε	nd 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	+	← ALERTII	NG
200 OK INVITE	+	← CONNEC	CT
ACK	→		
BYE	→		
481 Call/Transaction does not	exist 🗲		

TSS Term_Abnormal_situations	TP_306_022	Reference section 12.2.2 of [IETF RFC 3261]	Selection expression
Test purpose			
BYE CSeq value in BYE high	er than in INVITE req	uest	
			quest including a CSeq header set OK) response with the same CSeq
SIP header values			
INVITE1:			
CSeq: any_invite1_cseq			
BYE:			
CSeq: any_invite1_cseq+	-x		
200 OK BYE			
CSeq: <i>any_invite1_cseq</i> +	- <i>x</i>		
Message flow	<i>x</i>		
Test equ	ipment	Е	nd device
Interworking POTS	- r		
INVITE	→		
180 Ringing	÷	Ringing	
200 OK INVITE	÷	Off hook	
ACK	→		
BYE	→		
200 OK BYE	÷		
ISDN interworking			
INVITE	→	→ SETUP	
180 Ringing	+	← ALERTIN	NG
200 OK INVITE	+	← CONNEC	CT
ACK	→		
BYE	→	→ DISCON	NECT
200 OK BYE	+	← RELEAS	Е
		→ RELEAS	E_COMPLETE

TSS Term_Abnormal_situations	TP_306_023	Reference section 9.2 of [IETF RFC 3261]	Selection expression	
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.				

SIP header values		
INVITE1:		
Via: <i>any_invite1_via_value</i> ;branch= <i>z9h</i>	ıG4bK any in	wite1 branch value
From: <i>any_invite1_from_value</i> ;tag= <i>any_</i>		
To: any_invite1_to_value	•	
Call-ID: any_invite1_call-id_value		
CSeq: any_invite1_cseq		
INVITE2:		
Via: <i>any_invite2_via_value</i> ;branch= z9h	1G4bK_any_c	cancel_branch_value
From: any_invite1_from_value;tag=any_	_invite1_from	n_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: <i>any_invite1_via_value</i> ;branch= <i>z9h</i>	G4bK anv i	invite1 branch value
From: <i>any_invite1_from_value</i> ;tag= <i>any_</i>		
To: any_invite1_to_value		
Call-ID: any_invite1_call-id_value		
CSeq: any_invite1_cseq		
INVITE2:		
Via: <i>any_cancel_via_value</i> ;branch= z9h	G4bK_any_	invite1_branch_value
From: any_invite1_from_value;tag=any_	_invite1_from	n_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 nurnose			

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: <i>any_invite1_via_value</i> ;branch= <i>z9</i>	1G4bK any_in	vite1_branch_value
From: any_invite1_from_value;tag=any	•	
To: any_invite1_to_value		
Call-ID: any_invite1_call-id_value		
CSeq: any_invite1_cseq		
INVITE2:		
Via: <i>any_cancel_via_value</i> ;branch= z9h	G4bK_any_ c	ancel_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
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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
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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	+		

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:				
	than in INVITE request	than in INVITE request of a BYE request with a CSeq number set to a lower valu		

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 Abasemal situations	TP_306_029	Reference section 8.2.2.3 of	Selection expression
Term_Abnormal_situations		[IETF RFC 3261]	
Test purpose	·		
CANCEL received Require he	eader contains not supp	orted value	
		ion is in the Proceeding state, t it does not support, sends a Su	on receipt of a CANCEL request access (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 ← ALERTIN	
180 Ringing	+	-	
CANCEL	→	✓ RELEAS	E/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-Requ	ire header contains n	ot supported value	
CANCEL received 110xy-Kequ	re neuler contains no	n supported value	
			on receipt of a CANCEL request ds a Success (200 OK) response.
SIP header values			
Proxy-Require: 100unsupported	1		
Message flow			
Test equip	nent	E	nd device
Interworking POTS			
INVITE	→	Ringing	
180 Ringing	+		
CANCEL	→		
200 OK CANCEL	+		
487 Request Terminated	+		
ACK	→		
ISDN interworking			
INVITE	→	→ SETUP	
180 Ringing	÷	← ALERTIN	NG
CANCEL	→	_	E/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 dialogu	e 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 equij	oment	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 rel	eased			
Ensure that the IUT, while a sess Exist (481 Call/transaction Does		on receipt of a BYE reques	st, sends a Call/transaction Does Not	
SIP_CC_TE_CR_I_005	riot Emili, response.			
SIP header values				
Magaza flow				
Message flow	ant		End device	
Test equipm Interworking POTS				
INVITE	→			
	÷	Ringing	T	
180 Ringing←200 OK INVITE←		Ringing Off hook		
ACK	• •			
BYE1	→ Start ti	mer J		
200 OK BYE				
	Timeou	ıt timer J		
BYE2	→			
481 Call/transaction Does Not E	xist 🗲			
ISDN interworking				
INVITE	→	→ SETUP	0	
180 Ringing	←	← ALERT	ΓING	
200 OK INVITE	+	← CONN	ECT	
ACK	→			
BYE1	→ Start tin	mer J \rightarrow DISCO	NNECT	
200 OK BYE	÷	← RELEA	ASE	
	Timeou	it timer J \rightarrow RELEA	ASE_COMPLETE	
BYE2	→			
481 Call/transaction Does Not E	xist 🗲			

7.2.4 Emergency service

TSS Emergency_service	TP_401_001	Referen	ce se 5.1.6.8.4 of	Selection expression
Emergency_service			S 124 229]	
Test purpose		_		
Emergency session setup w	vithin a non-emergency	registratio	n	
contains:		ll and nor	n-emergency registration so	ends an INVITE request that
• a service URN in the R	equest-URI			
	the same emergency serv		•	
	-	•	or the tel URI associated w	-
• if available to the SUT, field	and if defined for the acc	cess type, t	he SUT shall insert in the P	P-Access-Network-Info header
• one P-Preferred-Identit user identity	y header field that includ	de the pub	lic user identity or the tel U	JRI associated with the public
• if the UE has its location include a Geolocation h		, or a URI	that points to the location i	nformation, then the UE shall
SIP header values INVITE				
Request Line: urn:service: control/sos.fire/sos.marine/			olice	
or				
operator specific emerg	gency digits			
To: urn:service:{sos/sos.ar	nbulance/sos.animal-cor	ntrol/sos.fi	re/sos.marine/sos.mountain	/sos.physician/sos.police
or				
operator specific emerg	gency digits			
From: public user identity				
P-Access-Network-Info				
Message flow				
End	device		Test equ	uipment
Interworking POTS				
Off hook				
Dial number		→	INVITE	
		+	407 Proxy Authentication	on Required
		→	ACK	
		→	INVITE	
ISDN interworking				
SETUP		→	INVITE	
		÷	407 Proxy Authentication	on Required
		→	ACK	
		→	INVITE	
	Арр	oly post te	st 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	TP_501_101	Reference	Selection expression
OIP_OIR		subclause C.1.2.1 and C5 of [TS183 043]	5 PICS 5.1.1/1
Test purpose		· ·	
User Identification i	information received in an l	NVITE request	
User Identification i	nformation received in an I	NVITE request ("P-Asserted-Id" and	"From" headers) is used by the
SUT. The SUT requ	ests the media gateways to	send this message over the analogue l	ine using the Display Data Bloc
parameter of the Dis	splay.		
SIP header values			
INVITE:			
From: [A user identi	ification]		
P-Asserted-Identific	ation: [A user identification	1]	
Message flow			
,	Test equipment	En	nd device
INVITE	→		
180 Ringing	+	Ringing, A	user identification
	1	Apply post test routine	
	TD 5 01 102	Reference	Selection expression
TSS	TP_501_102		
TSS OIP_OIR	TP_501_102	subclause C.1.2.1, C5 of [TS183 043]	PICS 5.1.1/1

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 te	st routine

TSS OIP_OIR	TP_501_103		ce se C.1.2.1 and C5 of [S183 043]	Selection expression PICS 5.1.1/1
Test purpose				
User Name information red	eived in an INVITE	request		
	nerate the appropria	te Call Setup	message. The SUT request	der /or "P-Asserted-Identity") s the media gateways to send
SIP header values				
From: [A user Name]				
P-Asserted-Identification:	A user Name]			
Message flow				
End	device		Test equ	ipment
Off hook				
Dial number		→	INVITE	
		÷	407 Proxy Authenticatio	n Required
		→	ACK	*
		→	INVITE	
	ŀ	Apply post te	st 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-A	Asserted-Identity and no Pr	ivacy header present	
		P-Asserted-Identity is not present ation Element is not included.	and no Privacy header is present,
DSS1 Parameter values	5		
Message flow			
Test	equipment	Er	nd device
INVITE	→	→ SETUP	
183 Ringing	+	← ALERTIN	ſĠ
	App	ly post test routine	

TSS	TP_501_202	Reference	Selection expression
OIP_OIR		subclause 5.2.3.1 of	PICS 5.1.1/2
		[ETSI TS 183 036]	
Test purpose			
INVITE received, no P-A	Asserted-Identity present an	d Privacy header present From h	eader Anonymous
		P-Asserted-Identity is not preser	
		rom header is set to an Anonymo	ous value, a SETUP is sent and a
Caning party number in	formation Element is includ	led and coded as follows:	
Calling party number			
Type of number = Unkne	own		
Numbering plan identified	cation = Unknown		
Presentation indicator =	Presentation restricted		
Screening indicator = Ne	etwork provided		
Number digits not presen	nt		
SIP header values			
INVITE: From: <sip:an< td=""><td>onymous@anonymous.inv</td><td>alid>; tag=</td><td></td></sip:an<>	onymous@anonymous.inv	alid>; tag=	
DSS1 Parameter values	5		
SETUP: Calling party	number		
Presentatio	on indicator = Presentation	restricted	
Message flow			
Test	equipment	En	d device
INVITE	→	→ SETUP	
183 Ringing	+	← ALERTIN	G
	App	y 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
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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 set 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	TP_501_204	Reference	Selection expression
OIP_OIR		subclause 5.2.3.1 of	PICS 5.1.1/2
		[ETSI TS 183 036]	

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

8			
Т	est equipment		End device
INVITE	→	→	SETUP
183 Ringing	÷	←	ALERTING
	Apply post tes	st 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-Asser E.164 address	rted-Identity present and	l no Privacy header present From	header not in the format of an				
address in the global numb	per format and a no Priva	ne P-Asserted-Identity is present, the cy header is present, the URI of the I party number Information Element is	From header is not in the format				
Calling party number							
Type of number = Nation	al number						
Numbering plan identifica	ntion = ISDN/Telephony	numbering plan					
Presentation indicator = P	resentation allowed						
Screening indicator = Use	r provided, verified and j	passed					
Number digits:							
		P-Asserted-identity. In case for glob ted, the country code is removed fro					
SIP header values							
INVITE: P-Asserted-Identity: userinfo in local number format							
DSS1 Parameter values							
SETUP: Calling party n	umber						
Type of nur	nber = National number						
Number digits derived from the P-Asserted-Identity							
Message flow							
Test e	quipment	End	device				
INVITE	→	→ SETUP					
183 Ringing	+	← ALERTING	ł				
	Apr	bly post test routine					

TSS	TP_501_206	Reference	Selection expression
OIP_OIR		subclause 5.2.3.1 of [ETSI TS 183 036]	PICS 5.1.1/2

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	SIP header values						
INVITE: P-Asserted-Ident	ity: userinfo in global number	format					
DSS1 Parameter values							
SETUP: Calling party nur	nber						
Type of num	ber = International number						
Number digit	Number digits derived from the P-Asserted-Identity						
Message flow							
Test eq	Test equipment End device						
INVITE	→	→	SETUP				
183 Ringing ← ← ALERTING							
	Apply pos	t test routine					

OIP_OIR subclause 5.2.3.1 of [ETSI TS 183 036] PICS 5.1.1/2	TSS OIP_OIR	TP_501_207		Selection expression PICS 5.1.1/2
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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 End device **Test equipment** INVITE SETUP → → ALERTING 183 Ringing ← ← 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-Asse	erted-Identity not equal F	rom header not in the format of a	n E.164 address Privacy header			
present						
address and a no Privacy userinfo is not equal to	header is present the UR	the P-Asserted-Identity is present the AI of the From header is in the form serted-Identity, a SETUP is sent allows:	mat of an E.164 address and the			
1 st Calling party numbe	r					
· ·	ation = ISDN/Telephony	numbering plan				
Presentation indicator = I		8 F				
	er provided, not verified	L				
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						
2 nd Calling party number	er					
• •	ation = ISDN/Telephony	numbering plan				
Presentation indicator $=$ I						
•	er provided, verified and	l passed				
Number digits:						
		e P-Asserted-Identity header . In ISDN line is located, the userinfo				
SIP header values						
INVITE: From:						
P-Asserted-Ide	entity:					
DSS1 Parameter values						
SETUP: Calling party r						
Screening						
	rovided, not verified					
Calling party r						
Screening indicator						
-	rovided, verified and pas	ssed				
Message flow						
Test	equipment	En	d device			
INVITE	→	→ SETUP				
183 Ringing	←	← ALERTIN	G			
	Apr	ly 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 he	eader present no Privacy header	present
E.164 address and a the userinfo is equ	ipt of an INVITE request and if a no Privacy header is present, the al to the userinfo of the P-Asse atts is included and coded as follow	URI of the From header is in the erted-Identity, a SETUP is sent	e format of an E.164 address and
Calling party num	ber		
Numbering plan ide	entification = ISDN/Telephony nu	umbering plan	
Presentation indicat	or = Presentation allowed		
-	= User provided, verified and j	passed	
Number digits:			
U	re derived from userinfo of the l code is not located the same as th	ĩ	ç
SIP header values			
INVITE: From:			
P-Asserte	ed-Identity:		
DSS1 Parameter v	alues		
SETUP: Calling p	arty number		
	ning indicator		
U	ser provided, verified and passo	ed	
Message flow			
	Test equipment	En	d device
INVITE	+	→ SETUP	
183 Ringing	+	← ALERTIN	G
	Apply	y post test routine	
TSS	TP_501_210	Reference	Selection expression
OIP_OIR		subclause 5.2.3.1 of [ETSI TS 183 036]	PICS 5.1.1/2
Test purpose INVITE received, P	-Asserted-Identity present and Fr	rom header present Privacy head	ler present
E.164 address and a	ipt of an INVITE request and if a Privacy header is present, the a Calling party number Informati	URI of the From header is in th	e format of an E.164 address, a
Calling party num Type of number $= U$			

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 n	umber			
Presentation	n indicator			
Present	ation restricted			
Message flow				
Test e	quipment		End device	
INVITE	→	→	SETUP	
183 Ringing	+	÷	ALERTING	
	Apply pos	st test routine		

TSS OIP_OIR	TP_501_211	Referen	ce se 5.2.3.2 of	Selection expression PICS 5.1.1/2 AND 5.1.3/2
			S 183 036]	11C5 5.1.1/2 AND 5.1.5/2
Test purpose		L.		
SETUP received, Call	ing party number nationa	l significant H	Presentation Restric	tion Indicator is set to restricted
of number is set to 'Na request is sent. If the C header field is present, derived from the addre	tional number', Presentati Calling party number digit	on Restriction s matches with identified Pub rty number IE	h Indicator is set to h a registered Public lic user identity. The in the local number	
SIP header values	-			
	ved from the Calling part	•	dress digits or	
	ymous@anonymous.inva			
	Identity: <matched public<="" td=""><td>c user identity</td><td>></td><td></td></matched>	c user identity	>	
Privacy: id,	header, user			
DSS1 Parameter valu				
SETUP: Calling part	-			
	number = National numb			
	tion Restriction Presentat	ion restricted		
Address	digits present			
Message flow				
]	End device		r.	Fest equipment
SETUP		→	INVITE	
		÷	407 Proxy Authe	entication Required
		→	ACK	
		→	INVITE	
SETUP ACKNOWLE	DGE	÷	100 Trying	
	A	apply post tes	t 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 P	party number nationa	l significant Presentation Restrie	ction Indicator is set to restricted				
of number is set to 'Nation request is sent. If the Callir Identity header field is pre header is derived from the	al number', Presentating party number digits sent, the userinfo is se address digits of the o		tity. The Userinfo of the From cal number format or set to				
SIP header values							
	• •	y number Address digits or					
•	ous@anonymous.inva						
	ntity: <default public<="" td=""><td>user identity></td><td></td></default>	user identity>					
Privacy: id, hea	der, user						
DSS1 Parameter values							
SETUP: Calling party nu	inder iber = National numb	or					
• •	Restriction Presentat						
Address dig							
Message flow							
End	device		Test equipment				
SETUP		→ INVITE					
		← 407 Proxy Author	entication Required				
		→ _{ACK}					
		→ INVITE					
SETUP ACKNOWLEDG	E	← 100 Trying					
	Apply post test routine						

TSS	TP_501_213	Reference	Selection expression
OIP_OIR		subclause 5.2.3.2 of	PICS 5.1.1/2 AND 5.1.3/2
		[ETSI TS 183 036]	

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.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 te	st routine

TSS OIP_OIR	TP_501_214	Reference subclause	5.2.3.2 of	Selection expression PICS 5.1.1/2 AND 5.1.3/2			
		[ETSI TS	183 036]				
Test purpose							
SETUP received, Calling p	party number Subscri	ber Presentation	n Restriction Indi	cator is set to restricted			
number is set to 'Subscribe sent. If the Calling party nu present, the userinfo is der	r', Presentation Restr umber digits matches ived from the Addres number IE in the loca	iction Indicator with a registere s digits. The Us l number forma	is set to Presenta ed Public identity, serinfo of the Fro	ddress digits present and the Type of tion restricted, an INVITE request is a P Preferred Identity header field is m header is derived from the address nous@anonymous.invalid. A Privacy			
SIP header values							
INVITE: From: <derived< td=""><td colspan="7">INVITE: From: <derived address="" calling="" digits="" from="" number="" or<="" party="" td="" the=""></derived></td></derived<>	INVITE: From: <derived address="" calling="" digits="" from="" number="" or<="" party="" td="" the=""></derived>						
anonymous@anonymous.invalid>							
P Preferred Identity: <matched identity="" public="" user=""></matched>							
Privacy: id, head	ler, user						
DSS1 Parameter values							
SETUP: Calling party nu							
• •	ber = Subscriber						
	Restriction Presentat	tion restricted					
Address digi	ts present						
Message flow							
End	device]	lest equipment			
SETUP		→	INVITE				
		÷	407 Proxy Authe	ntication Required			
		→	ACK				
		→	INVITE				
SETUP ACKNOWLEDGI	Ξ	÷	100 Trying				
	A	Apply post test	routine				

TSS OIP_OIR	TP_501_215		ce e 5.2.3.2 of S 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2		
Test purpose SETUP received, Calling p	party number Subscri	ber Presentati	on Restriction Indi	cator is set to restricted		
number is set to 'Subscribe sent. If the Calling party nu field is present, the userinfo	er', Presentation Restr umber digits does not to is set to the Default lling party number IE	iction Indicate match with a Public user ic in the local m	or is set to Presenta registered Public i lentity. The Userint	address digits present and the Type of ation restricted, an INVITE request is dentity, a P Preferred Identity header fo of the From header is derived from t to anonymous@anonymous.invalid.		
SIP header values						
INVITE: From: <derived address="" calling="" digits="" from="" number="" or<="" party="" td="" the=""></derived>						
anonymous@anonymous.invalid>						
	tity: <default public<="" td=""><td>user identity></td><td>•</td><td></td></default>	user identity>	•			
Privacy: id, head	ier, user					
DSS1 Parameter values						
SETUP: Calling party nu	lber = Subscriber					
	Restriction Presentat	ion restricted				
Address digi						
Message flow	-					
End	device]	Fest equipment		
SETUP		→	INVITE			
		+	407 Proxy Authe	entication Required		
		→	ACK	1		
		→	INVITE			
SETUP ACKNOWLEDGE	Ξ	←	100 Trying			
	A	Apply post tes				

TSS	TP_501_216	Reference	Selection expression
OIP_OIR		subclause 5.2.3.2 of	PICS 5.1.1/2 AND 5.1.3/2
		[ETSI TS 183 036]	

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 tes	st routine

TSS OIP_OIR	TP_501_217	Reference subclause 5.2.3.2 of	Selection expression PICS 5.1.1/2 AND 5.1.3/2
		[ETSI TS 183 036]	
Test purpose			
SETUP received, Ca	lling party number Unknow	n Presentation Restriction India	cator is set to restricted
number is set to 'Un sent. If the Calling p field is present, the u the address digits of	known', Presentation Restri arty number digits does not serinfo is set to the Default	ction Indicator is set to Presenta match with a registered Public Public user identity. The Userin in the local number format or set	Address digits present and the Type of ation restricted, an INVITE request is identity, a P Preferred Identity header of the From header is derived from et to anonymous@anonymous.invalid.
SIP header values	<u> </u>		
INVITE: From: <de< td=""><td>erived from the Calling part</td><td>y number Address digits or</td><td></td></de<>	erived from the Calling part	y number Address digits or	
	onymous@anonymous.inva		
	d Identity: <default public<="" td=""><td>user identity></td><td></td></default>	user identity>	
Privacy: i	d, header, user		
DSS1 Parameter va			
SETUP: Calling pa	-		
	of number = Unknown		
	tation Restriction Presentat ss digits present	ion restricted	
Message flow	ss digns present		
Message now	F 11 '		
	End device	→ INVITE	Test equipment
SETUP			
		Novi Toxy Hum	entication Required
		ACK	
		➔ INVITE	
SETUP ACKNOWL	LEDGE	← 100 Trying	
	A	pply post test routine	

TSS OIP_OIR	TP_501_218	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, Co	alling party number Presente	ation Restriction Indicator is set to	prestricted Address digits not present
the Type of number INVITE request is s	is set to 'National number', sent. If a P Preferred Identition fo of the From header is set	Presentation Restriction Indicatory header field is present, the user	f Address digits are not present and or is set to Presentation restricted, and info is set to the default Public user d, and if a Privacy header is present.
P Preferre	ip: unavailable@unknown.i ed Identity: <default public<br="">id, header, user</default>		
DSS1 Parameter v			
SETUP: Calling p			
• •	of number = National numb	er	
	ntation Restriction Presentat		
Addre	ess digits not present		
Message flow			
	End device	Т	'est equipment
SETUP		→ INVITE	
	← 407 Proxy Authentication Required		ntication Required
		→ ACK	1
		→ INVITE	
SETUP ACKNOW	LEDGE	← 100 Trying	
	I	Apply post test routine	
TSS	TD 501 210	Reference	Solootion arression
100	TP_501_219	Reference	Selection expression

	1F_301_219	whaleves 5.2.2.2	of PICS 5.1.1/2 AND 5.1.3/2
OIP_OIR		subclause 5.2.3.2 ([ETSI TS 183 036	
)
Test purpose			
SETUP received, Calling p	oarty number absent,	Privacy header is prese	ent
			nber IE is absent, an INVITE request is sent. If
	1 .		efault Public user identity. The Userinfo of the
	allable@unknown.in	valid. If a Privacy heade	er is present set to 'id' and 'header' and 'user'.
SIP header values			
INVITE: From: <sip: td="" una<=""><th></th><td></td><td></td></sip:>			
	tity: <default public<="" th=""><td>user identity></td><td></td></default>	user identity>	
Privacy: id, head	der, user		
DSS1 Parameter values			
SETUP: Calling party nu	mber not present		
Message flow			
End	device		Test equipment
SETUP		→ INVITE	E
		← 407 Pro	oxy Authentication Required
		→ ACK	
		→ INVITE	E
SETUP ACKNOWLEDG	E	← 100 Try	ving
		Apply post test routine	

	A	apply post tes	st routine	
SETUP ACKNOWLEDGE	SETUP ACKNOWLEDGE + 100 Trying			
		→	INVITE	
		→	ACK	
		←	407 Proxy Authenti	ication Required
SETUP		→	INVITE	
End	device		Tes	st equipment
Message flow				
SETUP: Calling party nu	mber not present			
DSS1 Parameter values	tity: <default public<="" td=""><td>user identity?</td><td>></td><td></td></default>	user identity?	>	
INVITE: From: <sip: td="" unav<=""><td></td><td></td><td></td><td></td></sip:>				
SIP header values				
From header is set. A Priva	cy header is not pres	ent.		
				c user identity. The Userinfo of the
Ensure that on receipt of a	SETUP message and	if the Calling	narty number IF is ab	osent, an INVITE request is sent. If
Test purpose SETUP received, Calling p	arty number absent,	Privacy head	er is absent	
		[ETSI T	S 183 036]	
OIP_OIR	11_501_220	subclaus	se 5.2.3.2 of	PICS 5.1.1/2 AND 5.1.3/2
TSS	TP_501_220	Referen	ce	Selection expression

TSS OID OID	TP_501_221	Reference subclause 5.2.3.2 of	Selection expression PICS 5.1.1/2 AND 5.1.3/2
OIP_OIR		[ETSI TS 183 036]	PICS 5.1.1/2 AND 5.1.5/2
Test purpose			
SETUP received, Ca	lling party number NOA n	ational significant Presentation Re	estriction Indicator is set to allowed
number is set to 'Na request is sent. If the header field is prese	tional number', Presentation Calling party number dig nt, the userinfo is set to the lress digits of the calling p	on Restriction Indicator is set to P its matches with a registered Public e identified Public user identity. I	ddress digits present and the Type of Presentation allowed and an INVITE ic identity and a P Preferred Identity f the Userinfo of the From header is er format, and if a Privacy header is
SIP header values			
INVITE: From: <de< td=""><td>erived from the Calling par</td><td>ty number Address digits></td><td></td></de<>	erived from the Calling par	ty number Address digits>	
P Preferre	d Identity: <identified pub<="" td=""><td>lic user identity></td><td></td></identified>	lic user identity>	
Privacy: n	one		
DSS1 Parameter va			
SETUP: Calling pa			
• •	of number = National number		
	tation Restriction Presenta	tion allowed	
	ss digits present		
Message flow			
	End device	Т	est equipment
SETUP		→ INVITE	
		← 407 Proxy Auther	ntication Required
		→ ACK	
		→ INVITE	
SETUP ACKNOWL	EDGE	← 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, Call	ing party number NOA n	ational significant Presentation Re	estriction Indicator is set to allowed
of number is set to 'Na INVITE request is sen Preferred Identity head	tional number', Presentat t. If the Calling party nur ler field is present, the us l from the address digits	ion Restriction Indicator is set to l nber digits does not match with a	registered Public identity and a P user identity. If the Userinfo of the
SIP header values			
INVITE: From: <der< td=""><td>ived from the Calling par</td><td>ty number Address digits></td><td></td></der<>	ived from the Calling par	ty number Address digits>	
P Preferred	Identity: < Default Public	user identity>	
Privacy: no	ne		
DSS1 Parameter valu	ies		
SETUP: Calling part	y number		
Type of	number = National numb	ber	
Presenta	tion Restriction Presenta	tion allowed	
Address	digits present		
Message flow			
	End device	Т	est equipment
SETUP		→ INVITE	
		← 407 Proxy Auther	ntication Required
		→ ACK	-
		→ INVITE	
SETUP ACKNOWLE	DGE	← 100 Trying	
SETUP AUMOWLE			

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 pur	pose			
SETUP r	eceived, Calling p	oarty number NOA i	nternational Presentation Restrict	ion Indicator is set to allowed
number is request is header fie	s set to 'Internation's sent. If the Call eld is present, the	hal number', Presenta ing party number d userinfo is set to th	ation Restriction Indicator is set to igits matches with a registered P is identified Public user identity.	ddress digits present and the Type of Presentation allowed and an INVITE ublic identity, a P Preferred Identity If the Userinfo of the From header is at and a Privacy header is present, set
SIP head	er values			
INVITE:	From: <derived< td=""><th>from the Calling par</th><td>rty number Address digits></td><td></td></derived<>	from the Calling par	rty number Address digits>	
	P Preferred Iden	tity: <matched publ<="" th=""><td>ic user identity></td><td></td></matched>	ic user identity>	
	Privacy: none			
DSS1 Pa	rameter values			
SETUP:	Calling party nu	mber		
	Type of num	ber = International i	number	
	Presentation	Restriction Presenta	ation allowed	
	Address digi	ts present		

Message flow		
End device		Test equipment
SETUP	→	INVITE
	÷	407 Proxy Authentication Required
	→	ACK
	→	INVITE
SETUP ACKNOWLEDGE	÷	100 Trying
	Apply post te	st routine

TSS	TP_501_224	Reference	Selection expression
OIP_OIR		subclause 5.2.3.2 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.1.3/2
Test purpose			
SETUP received, Ca	lling party number NOA in	ternational Presentation Restricti	on Indicator is set to allowed
number is set to 'Inter request is sent. If the Identity header field	rnational number', Presenta e Calling party number dig is present, the userinfo is se iddress digits of the calling	tion Restriction Indicator is set to light does not match with a register to the Default Public user identit	ddress digits present and the Type of Presentation allowed and an INVITE ered Public identity, if a P Preferred ty. If the Userinfo of the From header mber format and a Privacy header is
SIP header values			
INVITE: From: <de< td=""><td>erived from the Calling part</td><td>y number Address digits></td><td></td></de<>	erived from the Calling part	y number Address digits>	
	d Identity: <default public<="" td=""><td>• •</td><td></td></default>	• •	
Privacy: n		,	
DSS1 Parameter va	lues		
SETUP: Calling pa	rty number		
• •	of number = International n	umber'	
• •	tation Restriction Presentat		
Addre	ss digits present		
Message flow			
-	End device	Т	est equipment
SETUP		→ INVITE	
		← 407 Proxy Auther	ntication Required
		→ ACK	
		→ INVITE	
	EDGE		
SETUP ACKNOWL	EDGE	← 100 Trying	
	A	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, Callin	g party number NOA s	ubscriber Presentation Restriction	n Indicator is set to allowed
number is set to 'Subscr is sent. If the Calling pa field is present, the user	iber', Presentation Rest rty number digits matc info is set to the identi	riction Indicator is set to Presenta thes with a registered Public ident fied Public user identity. If the Us	ddress digits present and the Type of tion allowed and an INVITE request ity and a P Preferred Identity header serinfo of the From header is derived t and a Privacy header is present, set
SIP header values			
INVITE: From: <derive< td=""><td>ed from the Calling par</td><td>ty number Address digits></td><td></td></derive<>	ed from the Calling par	ty number Address digits>	
P Preferred Id	lentity: <matched publ<="" td=""><td>ic user identity></td><td></td></matched>	ic user identity>	
Privacy: none	;		
DSS1 Parameter value	s		
SETUP: Calling party	number		
Type of n	umber = Subscriber		
Presentati	on Restriction Presenta	tion allowed	
Address d	igits present		
Message flow			
E	nd device	ſ	lest equipment
SETUP		→ INVITE	
		← 407 Proxy Authe	ntication Required
		→ ACK	-
		→ INVITE	
SETUP ACKNOWLED	GE	← 100 Trying	
		Apply post test routine	

TSS TP_50 OIP_OIR	226Reference subclause 5.2.3.2 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/2
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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 te	st routine

TSS	TP_501_227	Reference	Selection expression
OIP_OIR		subclause 5.2.3.2 of	PICS 5.1.1/2 AND 5.1.3/2
		[ETSI TS 183 036]	
Test purpose			
SETUP received, Calling	party number NOA u	nknown Presentation Restriction	n Indicator is set to allowed
number is set to 'Unknown sent. If the Calling party nu is present, the userinfo is s	', Presentation Restri umber digits matches set to the identified P	ction Indicator is set to Presenta with a registered Public identity public user identity. If the Userin	Address digits present and the Type of ation allowed and an INVITE request is and a P Preferred Identity header field afo of the From header is derived from and a Privacy header is present, set to
SIP header values			
INVITE: From: <derived< td=""><td>from the Calling par</td><td>ty number Address digits></td><td></td></derived<>	from the Calling par	ty number Address digits>	
P Preferred Ider	ntity: <matched publi<="" td=""><td>ic user identity></td><td></td></matched>	ic user identity>	
Privacy: none			
DSS1 Parameter values			
SETUP: Calling party nu	ımber		
Type of num	nber = Unknown		
Presentation	Restriction Presenta	tion allowed	
Address dig	its present		
Message flow			
End	device		Test equipment
SETUP		→ INVITE	
		← 407 Proxy Aut	hentication Required
		→ _{ACK}	Ĩ
SETUP ACKNOWLEDG	F	← 100 Trying	
SETULACING WEEDO.	_	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			
	arty number NOA n	ational significant Presentation R	estriction Indicator is set to allowed
number is set to 'Unknown', sent. If the Calling party nu header field is present, the	Presentation Restri umber digits does nuserinfo is set to the	iction Indicator is set to Presentation not match with a registered Public he Default Public user identity. If	ddress digits present and the Type of on allowed and an INVITE request is c identity and a P Preferred Identity f the Userinfo of the From header is nber format and a Privacy header is
SIP header values			
INVITE: From: <derived fr<="" td=""><td>om the Calling par</td><td>ty number Address></td><td></td></derived>	om the Calling par	ty number Address>	
	ty: <default public<="" td=""><td>•</td><td></td></default>	•	
Privacy: none	2	-	
DSS1 Parameter values			
SETUP: Calling party num	ıber		
Type of numb	er = Unknown		
Presentation F	Restriction Presenta	tion allowed	
Address digits	present		
Message flow			
End d	levice	Т	lest equipment
SETUP		→ INVITE	
		← 407 Proxy Authe	ntication Required
		→ ACK	1
		→ INVITE	
SETUP ACKNOWLEDGE		\leftarrow 100 Trying	
		100 119115	

TSS	TP_501_229	Reference	Selection expression
OIP_OIR		subclause 5.2.3.2 of	PICS 5.1.1/2 AND 5.1.3/1
		[ETSI TS 183 036]	

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.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 Pre-	esentation restricte	ed		
Address digits present				
Message flow				
End device			Test equipment	
SETUP	→	INVITE		
SETUP ACKNOWLEDGE	÷	100 Trying		
	Apply post te	st routine		

TSS	TP_501_230	Reference	Selection expression
OIP_OIR		subclause 5.2.3.2 of	PICS 5.1.1/2 AND 5.1.3/1
		[ETSI TS 183 036]	

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**

	Test equipment
→ INVITE	
← 100 Tryii	ng
	IIII III

Apply post test routine

TSS	TP_501_231	Reference	Selection expression
OIP_OIR		subclause 5.2.3.2 of	PICS 5.1.1/2 AND 5.1.3/1
		[ETSI TS 183 036]	

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.invalid and Privacy header is present, set to 'id' and 'header' and 'user'.

SIP head	er values					
INVITE:	ITE: From: <derived address="" calling="" digits="" from="" number="" or<="" party="" th="" the=""></derived>					
	anonymo	ous@anonymous.inva	ulid>			
	P Asserted Iden	tity: <matched public<="" td=""><td>user identity:</td><td>></td><td></td><td></td></matched>	user identity:	>		
	Privacy: id, head	ler, user				
DSS1 par	rameter values					
SETUP:	Calling party nu	mber				
	Type of num	ber = International n	umber			
	Presentation	Restriction Presentat	tion restricted			
	Address digi	ts present				
Message	flow					
	End	device			Test equ	ipment
SETUP			→	INVITE		
SETUP A	CKNOWLEDGI	Ξ	÷	100 Trying		
		P	Apply post tes	t routine		
TSS		TP_501_232	Referen	ce		Selection expression

Test 1	ourpose
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OIP_OIR

Mw interface. SETUP received, Calling party number international Presentation Restriction Indicator is set to restricted

subclause 5.2.3.2/

[ETSI TS 183 036]

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.invalid, a Privacy header is present, set to 'id' and 'header' and 'user'.

SIP header values

INVITE:	From: <derived calling="" from="" nun<="" party="" th="" the=""><th>nber Ada</th><th>dress digits or</th><th></th></derived>	nber Ada	dress digits or	
	anonymous@anonymous.invalid>		C	
	P Asserted Identity: < Default Public user id	dentity>		
	Privacy: id, header, user			
DSS1 Pa	rameter values			
SETUP:	Calling party number			
	Type of number = International number	r		
	Presentation Restriction Presentation re	stricted		
	Address digits present			
Message	flow			
	End device			Test equipment
SETUP		→	INVITE	
SETUP A	ACKNOWLEDGE	←	100 Trying	
	Apply	post tes	st routine	

PICS 5.1.1/2 AND 5.1.3/1

TSS	TP_501_233	Referen	ce	Selection expression
OIP_OIR			se 5.2.3.2 of	PICS 5.1.1/2 AND 5.1.3/1
		[ETSI T	TS 183 036]	
Test purpose				
Mw interface. SETUP rece	eived, Calling party ni	umber Subscr	iber Presentation Re	striction Indicator is set to restricted
number is set to 'Subscribt sent. If the Calling party n present, the userinfo is set	er', Presentation Restrumber digits matches to the identified Publing party number IE in	riction Indicat with a register ic user identit the local num	tor is set to Presentat ered Public identity, by. If the Userinfo of	ddress digits present and the Type of tion restricted, an INVITE request is a P-Asserted-Identity header field is the From header is derived from the anonymous@anonymous.invalid, a
SIP header values				
INVITE: From: <derived< td=""><td>from the Calling part</td><td>ty number Ad</td><td>dress digits or</td><td></td></derived<>	from the Calling part	ty number Ad	dress digits or	
anonyme	ous@anonymous.inva	alid>		
P Asserted Iden	tity: <matched public<="" td=""><td>c user identity</td><td>></td><td></td></matched>	c user identity	>	
Privacy: id, hea	der, user			
DSS1 Parameter values				
SETUP: Calling party nu	umber			
Type of num	nber = Subscriber			
Presentation	Restriction Presentat	tion restricted		
Address dig	its present			
Message flow				
End	device		Т	est equipment
SETUP		→	INVITE	
SETUP ACKNOWLEDG	E	+	100 Trying	
		Apply post te		

TSS	TP_501_234	Reference	Selection expression
OIP_OIR		subclause 5.2.3.2 of	PICS 5.1.1/2 AND 5.1.3/1
		[ETSI TS 183 036]	

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 <u>anonymous@anonymous.invalid</u>, 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 = Subscriber

Presentation Restriction Presentation restricted

Address digits present

Test equipment VITE 0 Trying Itine
0 Trying
itine
Selection expression
PICS 5.1.1/2 AND 5.1.3/1
3 036]

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 <u>anonymous@anonymous.invalid</u>, 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 ide<="" public="" th="" user=""><th>ntity></th></matched>	ntity>

Privacy: id, header, user

DSS1 Parameter values

DSSI Parameter values				
SETUP: Calling party number				
Type of number = Unknown				
Presentation Restriction Presentation	ion restricted			
Address digits present				
Message flow				
End device			Test equipment	
SETUP	→	INVITE		
SETUP ACKNOWLEDGE	+	100 Trying		

Apply post test routine

TSS	TP_501_236	Reference	Selection expression
OIP_OIR		subclause 5.2.3.2 of	PICS 5.1.1/2 AND 5.1.3/1
		[ETSI TS 183 036]	

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 <u>anonymous@anonymous.invalid</u>, 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	l		
Presentation Restriction Pres	sentation restricted		
Address digits present			
Message flow			
End device			Test equipment
SETUP	→	INVITE	
SETUP ACKNOWLEDGE	+	100 Trying	
	Apply post tes	t routine	

-	1		
TSS	TP_501_237	Reference	Selection expression
OIP_OIR		subclause 5.2.3.2 of	PICS 5.1.1/2 AND 5.1.3/1
		[ETSI TS 183 036]	

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				

	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 te	st routine	

TSS OIP_OIR	TP_501_239		ce se 5.2.3.2 of [S 183 036]	Selection expression PICS 5.1.1/2 AND 5.1.3/1
Test purpose			.5 105 050]	
	P received, Calling party n	umber is not	present	
Presentation, an INV	TTE request is sent. If a P A	Asserted Identi	ty header field is pr	ber Information Element not present esent, the userinfo is set to the Default nown.invalid'. A Privacy header is not
SIP header values				
INVITE: From: <u< td=""><td>navailable@unknown.inval</td><td>id></td><td></td><td></td></u<>	navailable@unknown.inval	id>		
P Asserte	d Identity: <default public<="" td=""><td>user identity></td><td>></td><td></td></default>	user identity>	>	
DSS1 Parameter va	alues			
SETUP: Calling pa	arty number not present			
Message flow				
	End device		r	Fest equipment
SETUP		→	INVITE	
SETUP ACKNOWI	LEDGE	←	100 Trying	
	1	Apply post te	st routine	

TSS		TP_501_240	Reference	Selection expression			
OIP_OIR			subclause 5.2.3.2 of	PICS 5.1.1/2 AND 5.1.3/1			
			[ETSI TS 183 036]				
Test pur	pose						
SETUP r	eceived, Calling p	arty number NOA n	ational significant Presentation R	estriction Indicator is set to allowed			
number is is sent. If is present	s set to 'National n' the Calling party t, the userinfo is se	umber', Presentation number digits mate et to the identified P	a Restriction Indicator is set to Pres hes with a registered Public identi Public user identity. If the Userinf	ddress digits present and the Type of sentation allowed, an INVITE request ty, a P Asserted Identity header field o of the From header is derived from wacy header is present set to 'none'.			
SIP head	er values						
INVITE:	From: <derived< td=""><th>from the Calling par</th><td>rty number Address digits></td><td></td></derived<>	from the Calling par	rty number Address digits>				
	P Asserted Identity: <identified identity="" public="" user=""></identified>						
	Privacy: none						
DSS1 Pa	rameter values						
SETUP:	Calling party nu	mber					
	Type of number = National number						
	Presentation	Restriction Presenta	tion allowed				
Address digits present							

Message flow					
End	device			Test equipment	
SETUP		→	INVITE		
SETUP ACKNOWLEDGI	E	÷	100 Trying		
		Apply post test	routine		
TSS OIP_OIR	TP_501_241	Reference subclause [ETSI TS	5.2.3.2 of	Selection expression PICS 5.1.1/2 AND 5	
Test purpose		[215115	100 000]		
	partv number NOA	national significa	nt Presentation	n Restriction Indicator is set to a	llowed
request is sent. If the Callin header field is present, the	ng party number dig e userinfo is set to	its does not match the Default Publi	with a register c user identity	o Presentation allowed and an II red Public identity, a P Asserted I . If the Userinfo of the From he nber format, a Privacy header is	dentity ader is
SIP header values					
INVITE: From: <derived< td=""><th>from the Calling pa</th><td>arty number Addre</td><td>ess digits></td><th></th><td></td></derived<>	from the Calling pa	arty number Addre	ess digits>		
	tity: <default publi<="" th=""><td>c user identity></td><td></td><th></th><td></td></default>	c user identity>			
Privacy: none					
DSS1 Parameter values	_				
SETUP: Calling party nu		1			
• 1	ber = National num Restriction Present				
Address digi		ation anowed			
Message flow	no present				
	device			Test equipment	
SETUP	uevice	→	INVITE	i est equipment	
SETUP SETUP ACKNOWLEDGI	F	-	100 Trying		
SETUL ACKING WEEDOI		-	100 Hymg		
		Annly nost tost	nouting		
		Apply post test	routine		
		Apply post test	routine		

OIP_OIR

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'.

subclause 5.2.3.2 of

[ETSI TS 183 036]

PICS 5.1.1/2 AND 5.1.3/1

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 = Internation	nal number			
Presentation Restriction Pres	entation allowed			
Address digits present				
Message flow				
End device			Test equipment	
SETUP	→	INVITE		
SETUP ACKNOWLEDGE	÷	100 Trying		
	Apply post tes	st routine		

	r	1	
TSS	TP_501_243	Reference	Selection expression
OIP_OIR		subclause 5.2.3.2 of	PICS 5.1.1/2 AND 5.1.3/1
		[ETSI TS 183 036]	

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 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
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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: <deriv< th=""><th>ved from the Calling par</th><th>ty number Address</th><th>digits></th><th></th></deriv<>	ved from the Calling par	ty number Address	digits>	
	lentity: <matched public<="" th=""><th></th><th>C</th><th></th></matched>		C	
Privacy: none	e			
DSS1 Parameter value	28			
SETUP: Calling party	number			
	number = Subscriber			
Presentat	ion Restriction Presenta	tion allowed		
Address of	ligits present			
Message flow				
Ε	nd device		T	est equipment
SETUP		→ IN	VITE	
SETUP ACKNOWLED	DGE	← 100) Trying	
		Apply post test rou		
	TTD 501 045	D		
TSS OIP_OIR	TP_501_245	Reference subclause 5.2	3.2 of	Selection expression PICS 5.1.1/2 AND 5.1.
on_ont		[ETSI TS 183		1100 0111/2 11(0 011
				Indicator is set to allowed
SETUP received, Callin Ensure that on receipt on number is set to 'Subscrissent. If the Calling pa field is present, the user	f a SETUP message cor riber', Presentation Rest rty number digits does r info is set to the Default	ntaining a Calling pa riction Indicator is s not match with a reg t Public user identit	arty number Ad set to Presentat istered Public i y. The Userinfo	dress digits present and the Typ ion allowed aand n INVITE rec dentity, a P Asserted Identity he o of the From header is derived t
SETUP received, Callin Ensure that on receipt on number is set to 'Subscription is sent. If the Calling patifield is present, the user the address digits of the SIP header values INVITE: From: <deriv< td=""><td>f a SETUP message cor riber', Presentation Rest rty number digits does r info is set to the Default calling party number If yed from the Calling par lentity: <default public<="" td=""><td>ntaining a Calling pa riction Indicator is s not match with a reg t Public user identity E in the local number ty number Address</td><td>arty number Ac set to Presentat stered Public i y. The Userinfo r format. A Pri</td><td>dress digits present and the Typ ion allowed aand n INVITE rec dentity, a P Asserted Identity he</td></default></td></deriv<>	f a SETUP message cor riber', Presentation Rest rty number digits does r info is set to the Default calling party number If yed from the Calling par lentity: <default public<="" td=""><td>ntaining a Calling pa riction Indicator is s not match with a reg t Public user identity E in the local number ty number Address</td><td>arty number Ac set to Presentat stered Public i y. The Userinfo r format. A Pri</td><td>dress digits present and the Typ ion allowed aand n INVITE rec dentity, a P Asserted Identity he</td></default>	ntaining a Calling pa riction Indicator is s not match with a reg t Public user identity E in the local number ty number Address	arty number Ac set to Presentat stered Public i y. The Userinfo r format. A Pri	dress digits present and the Typ ion allowed aand n INVITE rec dentity, a P Asserted Identity he
SETUP received, Callin Ensure that on receipt on number is set to 'Subscrission is sent. If the Calling part field is present, the user the address digits of the SIP header values INVITE: From: <deriv P Asserted Io Privacy: none DSS1 Parameter value SETUP: Calling party Type of m Presentat</deriv 	f a SETUP message cor riber', Presentation Rest rty number digits does r info is set to the Default calling party number If yed from the Calling par lentity: <default public<br="">e</default>	ntaining a Calling pa riction Indicator is s not match with a reg t Public user identity E in the local number ty number Address user identity>	arty number Ac set to Presentat stered Public i y. The Userinfo r format. A Pri	dress digits present and the Typ ion allowed aand n INVITE rec dentity, a P Asserted Identity he o of the From header is derived t
SETUP received, Callin Ensure that on receipt on number is set to 'Subscription is sent. If the Calling patifield is present, the user the address digits of the SIP header values INVITE: From: <deriv P Asserted Ic Privacy: none DSS1 Parameter values SETUP: Calling party Type of m Presentation Address of</deriv 	f a SETUP message cor riber', Presentation Rest rty number digits does r info is set to the Default calling party number If yed from the Calling par lentity: <default public<br="">e es number number sumber = Subscriber ion Restriction Presenta</default>	ntaining a Calling pa riction Indicator is s not match with a reg t Public user identity E in the local number ty number Address user identity>	arty number Ac set to Presentat stered Public i y. The Userinfo r format. A Pri	dress digits present and the Typ ion allowed aand n INVITE rec dentity, a P Asserted Identity he o of the From header is derived t
SETUP received, Callin Ensure that on receipt on number is set to 'Subscriss sent. If the Calling par field is present, the user the address digits of the SIP header values INVITE: From: <deriv P Asserted Io Privacy: none DSS1 Parameter value SETUP: Calling party Type of m Presentati Address of Message flow</deriv 	f a SETUP message cor riber', Presentation Rest rty number digits does r info is set to the Default calling party number If yed from the Calling par lentity: <default public<br="">e es number number sumber = Subscriber ion Restriction Presenta</default>	ntaining a Calling pa riction Indicator is s not match with a reg t Public user identity E in the local number ty number Address user identity>	arty number Ad set to Presentat istered Public i y. The Userinfo r format. A Pri digits>	dress digits present and the Typ ion allowed aand n INVITE rec dentity, a P Asserted Identity he o of the From header is derived t
SETUP received, Callin Ensure that on receipt on number is set to 'Subscriss sent. If the Calling par field is present, the user the address digits of the SIP header values INVITE: From: <deriv P Asserted Io Privacy: none DSS1 Parameter value SETUP: Calling party Type of m Presentati Address of Message flow</deriv 	f a SETUP message cor riber', Presentation Rest rty number digits does r info is set to the Default calling party number II yed from the Calling par dentity: <default public<br="">e es number number = Subscriber ion Restriction Presenta digits present</default>	ntaining a Calling pa riction Indicator is s not match with a reg t Public user identity E in the local number ty number Address user identity>	arty number Ad set to Presentat istered Public i y. The Userinfo r format. A Pri digits>	dress digits present and the Typ ion allowed aand n INVITE rec dentity, a P Asserted Identity he o of the From header is derived b vacy header is present set to 'no
SETUP received, Callin Ensure that on receipt on number is set to 'Subscription is sent. If the Calling particle field is present, the user the address digits of the SIP header values INVITE: From: <derive P Asserted Io Privacy: none DSS1 Parameter value SETUP: Calling party Type of m Presentation Address of Message flow</derive 	f a SETUP message cor riber', Presentation Rest rty number digits does r info is set to the Default calling party number If yed from the Calling par lentity: <default public<br="">e es number number = Subscriber ion Restriction Presenta digits present</default>	taining a Calling parition Indicator is so not match with a reg t Public user identity E in the local number ty number Address user identity> tion allowed	arty number Ad set to Presentat istered Public i y. The Userinfor r format. A Pri digits>	dress digits present and the Typ ion allowed aand n INVITE rec dentity, a P Asserted Identity he o of the From header is derived b vacy header is present set to 'no

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		· · ·	
	alling party number NOA u	nknown Presentation Restriction I	ndicator is set to allowed
number is set to 'Un sent. If the Calling p present, the userinfo	known', Presentation Restri party number digits matches o is set to the identified Pub	ction Indicator is set to Presentations with a registered Public identity,	ddress digits present and the Type of on allowed and an INVITE request is a P Asserted Identity header field is the From header is derived from the cy header is present set to 'none'.
SIP header values			
INVITE: From: <de< td=""><td>erived from the Calling par</td><td>ty number Address digits></td><td></td></de<>	erived from the Calling par	ty number Address digits>	
P Asserte	d Identity: <matched public<="" td=""><td>c user identity></td><td></td></matched>	c user identity>	
Privacy: r	none		
DSS1 Parameter va	alues		
SETUP: Calling pa	arty number		
Type	of number = Unknown		
Preser	ntation Restriction Presenta	tion allowed	
Addre	ess digits present		
Message flow			
	End device	Т	est equipment
SETUP		→ INVITE	
SETUP ACKNOWI	EDGE	← 100 Trying	
SET OF MERICO WI		Apply post test routine	
	1	appry 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			
	alling party number NOA n	ational significant Presentation Re	estriction Indicator is set to allowed
Ensure that on receiv	pt of a SETUP message cor	ntaining a Calling party number Ad	ddress digits present and the Type of

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	t routine	

7.2.5.2 Terminating identification presentation and terminating identification restriction)

7.2.5.2.1 Test purposes for ISDN

TSS TIP_TIR	TP_502_101	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 no Connec	ted number or no valid Con	nected number received, 200 OK	(INVITE) is sent
the value saved from t	he P-Called-Party-ID heade	ONNECT message was received. r that was received in the INVITE eived CONNECT. No "from-chan	E request if no Connected number
SIP header values			
INVITE:			
P-Called-Party-ID = <	called number>		
200 OK (INVITE):			
P-Asserted-Identity = \cdot	<called number=""></called>		
DSS1 Parameter valu			
Message flow			
Te	st equipment	E	nd device
INVITE	→	→ SETUP	
180 Ringing	+	← ALERTIN	NG
200 OK INVITE	+	← CONNEC	۳
ACK	÷	→ CONNEC	
		ply post test routine	
TSS	TP_502_102	Reference	Selection expression
TIP_TIR	1F_302_102	subclause 5.2.2.1.1 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.1.3/1
Test purpose	•		
CONNECT	or Subscriber number receiv		

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

SIP header values

200 OK (INVITE): P-Asserted-Identity = <registered number>

DSS1 Parameter values

CONNECT: Connected number

Nature of address = unknown or Subscriber number

Number Digits = <connected number>

Message flow		
Test equip	ment	End device
INVITE	→	→ SETUP
180 Ringing	÷	← ALERTING
200 OK INVITE	+	← CONNECT
ACK	→	➔ CONNECT ACK
	Apply pos	at test routine

TSS	TP_502_103	Reference	Selection expression
TIP_TIR		subclause 5.2.2.1.1 o	f PICS 5.1.1/2 AND 5.1.3/1
	[ETSI T		
Test purpose			
CONNECT National n	umber received, 200 OK	(INVITE) is sent	
(INVITE) is sent whe including the display n	n a CONNECT messag ame if previously stored	e was received. The P-A during registration repres	he Supported header, ensure that a 200 OK sserted-Identity is present with that value, enting the terminating user indicated in the tag in the supported header.
SIP header values			
200 OK (INVITE):			
P-Asserted-Identity = <	<registered number=""></registered>		
DSS1 Parameter valu	es		
CONNECT: Connected	l number		
Nature of	f address = National num	ıber	
Number	Digits = <connected nun<="" td=""><td>nber></td><td></td></connected>	nber>	
Message flow			
Tes	st equipment		End device
INVITE	→	→	SETUP
180 Ringing	+	÷	ALERTING
200 OK INVITE	+	÷	CONNECT
ACK	→	→	CONNECT ACK
	A	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
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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 num	ber			
Nature of addre	ess = international number			
Number Digits	= <connected number=""></connected>			
Message flow				
Test equi	pment		End device	
INVITE	→	→	SETUP	
180 Ringing	<	+	ALERTING	
200 OK INVITE	+	÷	CONNECT	
ACK	ACK \rightarrow CONNECT ACK			
	Apply pos	st 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
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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 equipmen

Test equip	ment	End device
INVITE	→	→ SETUP
180 Ringing	+	← ALERTING
200 OK INVITE	÷	← CONNECT
ACK	→	➔ CONNECT ACK
	Apply pos	st test routine

TSS	TP_502_106	Reference	Selection expression
TIP_TIR		subclause 5.2.2.1.1 of [ETSI TS 183 036]	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 = < regist	ered number>			
DSS1 Parameter values				
CONNECT: Connected num	ber			
Nature of addre	ess = international number			
Number Digits	= <connected number=""></connected>			
Message flow				
Test equi	pment		End device	
INVITE	→	→	SETUP	
180 Ringing	+	← ALERTING		
200 OK INVITE	+	+	CONNECT	
ACK	\rightarrow \rightarrow CONNECT ACK			
	Apply pos	st test routine		

TSS	TP_502_107	Reference	Selection expression
TIP_TIR		subclause 5.2.2.1.1 of	PICS 5.1.1/2 AND 5.1.3/1
		[ETSI TS 183 036]	AND 5.4/2

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

in the second se			
Test equipme	ent	End device	
INVITE	→	→ SETUP	
180 Ringing	+	← ALERTING	
200 OK INVITE	+	← CONNECT	
ACK	→	→ CONNECT ACK	
UPDATE	÷		
200 OK UPDATE	→		
	Apply pos	st test routine	

TSS TIP_TIR	TP_502_108	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			
	al number received, 200 C	OK (INVITE) is sent	
sent when a CONNEC display name if previo number type of number	T message was received. usly stored during registration r	contained in the Supported header, If the P-Asserted-Identity is prese ation, representing the terminating from-change" tag is present in the ved in the Connected Number.	ent with that value, including the g user indicated in the connected
SIP header values			
INVITE:			
Supported: from-change	e		
200 OK (INVITE):			
P-Asserted-Identity = <	registered number>		
UPDATE:			
From: <connected num<="" td=""><td>ber></td><td></td><td></td></connected>	ber>		
DSS1 Parameter value	es		
CONNECT: Connected	l number		
Nature of	address = international nu	umber	
Number I	Digits = <connected numb<="" th=""><th>er></th><th></th></connected>	er>	
Message flow			
Tes	t equipment	E	nd device
INVITE	→	→ SETUP	
180 Ringing	÷	← ALERTIN	NG
200 OK INVITE	+	← CONNEC	T
ACK	→	→ CONNEC	
UPDATE	+		
200 OK UPDATE	→		
	Ap	oply post test routine	
TSS	TP 502 109	Reference	Selection corresponden
TIP_TIR	17_302_109	subclause 5.2.2.1.1 of	Selection expression PICS 5.1.1/2 AND 5.1.3/1

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

[ETSI TS 183 036]

AND 5.4/2

SIP header values 200 OK (INVITE):

P-Asserted-Identity = <registered number>

DSS1 Parameter values				
CONNECT: Connected num	ber			
Nature of addre	ess = national number			
Number Digits	= <connected number=""></connected>			
Message flow				
Test equi	pment		End device	
INVITE	→	→	SETUP	
180 Ringing	÷	+	ALERTING	
200 OK INVITE	←	+	CONNECT	
ACK	→	→	CONNECT ACK	
	Apply pos	t test routine		

TSS	TP_502_110	Reference	Selection expression
TIP_TIR		subclause 5.2.2.1.1 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.1.3/2

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 equip	ment	End device	
INVITE	→	→ SETUP	
180 Ringing	÷	← ALERTING	
200 OK INVITE	÷	← CONNECT	
ACK	→	→ CONNECT ACK	
	Apply po	st test routine	

TSS	TP_502_111	Reference	Selection expression
TIP_TIR		subclause 5.2.2.1.1 of	PICS 5.1.1/2 AND 5.1.3/2
		[ETSI TS 183 036]	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			
Te	st equipment		End device
INVITE	→	→	SETUP
180 Ringing	<	÷	ALERTING
200 OK INVITE	÷	÷	CONNECT
ACK	→	→	CONNECT ACK
	Apply post t	test routine	

TSS	TP_502_112	Reference	Selection expression
TIP_TIR		subclause 5.2.2.1.1 of	PICS 5.1.1/2 AND 5.1.3/2
		[ETSI TS 183 036]	AND 5.4/2
Test purpose			
'from-change' tag in the l	NVITE received, CONNI	ECT received, 200 OK (INVITE) is s	eent no 'from-change' tag present
	orted header in the receive	neader in the sent 200 OK INVITE ed INVITE and no or invalid connec	
No P-Asserted-Identity o	r P-Preferred-Identity he	ader is present in the 200 OK INVI	TE. No UPDATE is sent.
SIP header values			
INVITE : 'from-change'			
200 OK INVITE: no 'fro	m-change' tag		
DSS1 Parameter values			
CONNECT: no or invali	id connected number info	ormation element	
Message flow			
Test	equipment	En	d device
INVITE	→	→ SETUP	
180 Ringing	÷	← ALERTIN	G
200 OK INVITE	+	← CONNEC	Γ
ACK	→	→ CONNEC [*]	Г АСК
	Ар	ply 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 IN	VITE received, CONN	ECT received, 200 OK (INVITE) is	s sent 'from-change' tag present
was received in the support set to Nature_of_address a	rted header in the receiv s indicated in Table 7.2 P-Preferred-Identity he	red INVITE and the nature of adda 5.2.1-2 ader is present in the 200 OK INV	E is present if a 'from-change' tag ress of the received CONNECT is TTE. An UPDATE is sent and the
SIP header values			
INVITE: Supported: from	n-change		
200 OK (INVITE):			
	ed: from-change		
UPDATE : From: <connec< td=""><td>cted number></td><td></td><td></td></connec<>	cted number>		
DSS1 Parameter values			
CONNECT: Connected n			
		er or international number	
6	tits = <connected number<="" td=""><td></td><td></td></connected>		
Message flow	_		
	quipment		nd device
INVITE	→	→ SETUP	
180 Ringing	+	← ALERTI	NG
200 OK INVITE	←	← CONNEC	Т
АСК	→	→ CONNEC	
UPDATE	,		
200 OK UPDATE	→ →		
Loo on or prile	-	ply post test routine	
	Ар	pry post test routile	

TSS TIP_TIRTP_502_114Reference subclause 5.2.2.1.1 of [ETSI TS 183 036]Selection PICS 5.1. AND 5.4/
--

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 equipm	ent		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	TP_502_115	Reference	Selection expression
TIP_TIR		subclause 5.2.2.1.1 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.1.3/2 AND 5.4/3
Test purpose			
CONNECT Connected num	nber Presentation restri	cted received, 200 OK (INVITE) i	s sent
	icator is set to 'Presenta	e containing a Connected numb tion restricted', a 200 OK INVITE	
SIP header values			
200 OK: Privacy: id, head	ler, user		
DSS1 Parameter values			
CONNECT: Connected n	umber		
Presentat	ion restriction = Presen	tation restricted	
Message flow			
Test eq	luipment	E	nd device
INVITE	→	→ SETUP	
180 Ringing	÷	← ALERTIN	١G
200 OK INVITE	+	← CONNEC	Т
ACK	→	→ CONNEC	T ACK

TSS	TP_502_116	Reference	Selection expression
TIP_TIR		subclause 5.2.2.1.2 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/2

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-ch	ange		
DSS1 Parameter values			
Message flow			
End dev	ice	Test equipment	
SETUP	→	→ INVITE	
ALERTING	+	← 180 Ringing	
CONNECT	÷	← 200 OK INVITE	
CONNECT ACK	→	→ ACK	
	Apply pos	st test routine	

TSS	TP_502_117	Reference	Selection expression
TIP_TIR		subclause 5.2.2.1.2 of	PICS 5.1.1/2 AND 5.1.3/1
		[ETSI TS 183 036]	

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			
E	nd device		Test equipment
SETUP	→	→	INVITE
ALERTING	+	÷	180 Ringing
CONNECT	+	÷	200 OK INVITE
CONNECT ACK	→	→	ACK
	Apply post t	est routine	

TSS	TP_502_118	Reference		Selection expression
TIP_TIR		subclause 5.2.2.1 [ETSI TS 183 030		PICS 5.1.1/2 AND 5.4/3
Test purpose	÷	·		
	P-Asserted-Identity is sent in ag present in the 200 OK (I		lo Privacy hea	der present or Privacy value
Privacy header is press in the From header in sent to the calling use Connected number	ent, the sending of the CON	NECT message is held un m of a tel URI containing	ntil the UPDAT g a E.164 num	the Supported header and no FE is received. If the Userinfo ber a CONNECT message is
Type of number				
National number		at @la a tra a rt'		
sip: local-num International num	ber-digits; phone-context=n	atenostportion; user=ph	ione	
	nber-digits@hostportion; us	er-nhone		
	ification = ISDN/Telephony			
• •	r = Presentation allowed	, numbering plan		
Screening indicator =	User provided, not verified			
Number digits derived	l from the userinfo of the Fi	rom header.		
	global number and the coun number of the Type of num			is located, the country code is
SIP header values				
UPDATE:From				
DSS1 Parameter val CONNECT: Connec				
Message flow				
	End device		Test equ	iipment
SETUP	→	→	INVITE	
ALERTING	÷	÷	180 Ringing	
CONNECT	+	÷	200 OK INVI	TE
CONNECT ACK	→	→	ACK	
UPDATE	+			
200 OK UPDATE	→			

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			I
		nnected number. Privacy header vo	alue 'id' or 'header' 'from-change'
Privacy header is press a tel URI containing a present coded as descr	ent, the priv-value set to 'id E.164 number and a CONN	re the 'from-change' tag is not pres ' or 'header' the Userinfo of the P-4 NECT message is sent to the calling	Asserted-Identity is in the form of
Connected number			
Type of number = Unl Numbering plan identi			
• •	= Presentation restricted		
Screening indicator =			
•	from the userinfo of the P-	Asserted-Identity.	
Number digits not	present	-	
SIP header values			
200 OK:			
P-Asserted-Identity			
Privacy: id, header, use	r		
DSS1 Parameter valu	ies		
CONNECT: Connect	ed number		
Message flow			
	End device	Test	t equipment
SETUP	→	→ INVITE	
ALERTING	÷	← 180 Ringi	ng
CONNECT	÷	← 200 OK I	NVITE
CONNECT ACK	→	→ ACK	
	Ap	ply 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
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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

 $\label{eq:presentation} Presentation\ indicator = Presentation\ restricted$

Screening indicator = Network provided

Number digits not present

SIP header values		
UPDATE:From: anonymous		
Privacy: id, header		
DSS1 Parameter values CONNECT: Connected numb	er	
Message flow		
End dev	ice	Test equipment
SETUP	→	→ INVITE
ALERTING	÷	← 180 Ringing
		← 200 OK INVITE
		→ ACK
CONNECT	÷	← UPDATE
CONNECT ACK	→	→ 200 OK UPDATE
	Apply pos	t 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 nurnose			

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	TP_502_122	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		·	
No P-Asserted-Identit	y header and no Privacy id	or header received, no Connec	cted number is sent
present, a CONNECT	message is sent to the callin	ng user and no Connected num s not subscribed to the COLP	-
DSS1 Parameter value	ues		
Message flow			
	End device		Test equipment
SETUP	→	→ INVI	TE
ALERTING	+	← 180 F	Ringing
CONNECT	+	← 200 0	DK INVITE
CONNECT ACK	→	→ ACK	
	Ар	ply post test routine	

TSS	TP_502_123	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			
No P-Asserted-Identity	y header and no Privacy id	or header received, no Connected	l number is sent
a CONNECT message	e is sent to the calling user a	no P-Asserted-Identity and no Priv and no Connected number is prese s not subscribed to the COLP servi	
SIP header values			
DSS1 Parameter valu	ies		
Message flow			
	End device	Tes	st equipment
SETUP	→	→ INVITE	
ALERTING	÷	← 180 Ring	ging
CONNECT	←	← 200 OK	INVITE
CONNECT ACK	→	→ ACK	
		oply post test routine	

TSS	TP_502_124	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			
'from-change' tag in 200 O	K (INVITE) received, e	expiry of timer T_{TIR1}	
header is present, timer TT	IR1 is started and the C	CONNECT message is held. Afte	he Supported header and no Privacy er expiry of TTIR1, the CONNECT cted number and coded as described
Connected number			
Type of number			
National number			
sip: local-number-d	igits; phone-context=na	at@hostportion; user=phone	
International number			
sip: global-number-	digits@hostportion; us	er=phone	
Numbering plan identificat	1 *	numbering plan	
Presentation indicator = \mathbf{Pr}	esentation allowed		
Screening indicator = User	-		
Number digits derived			
			or line is located, the country code is
	ber of the Type of num	ber is set to "national number.	
SIP header values			
DSS1 Parameter values			
CONNECT: Connected n	umber		
Message flow			
End	device	Те	est equipment
SETUP	→	➔ INVITE	Ξ
ALERTING	←	← 180 Rin	ging
CONNECT	CT \leftarrow 200 OK INVITE		
CONNECT ACK	→ ACK		
	Ap	ply post test routine	
TSS	TP_502_125	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 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.

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 o	levice	Test equip	ment
SETUP	→	➔ INVITE	
ALERTING	÷	← 180 Ringing	
CONNECT	÷	← 200 OK INVITE	2
CONNECT ACK	→	→ ACK	
	Apply pos	test routine	

TSS	TP_502_126	Reference	Selection expression
TIP_TIR		subclause 5.2.2.1.2 of	PICS 5.1.1/2 AND 5.4/3
		[ETSI TS 183 036]	
Test purpose			
Mapping of isub para	meter in the UPDATE From	n header into Connected subaddre	SS
Ensure that on receipt	of source D Assorted Iden	tity booders in different responses	the Connected number cent to the
	d from the latest received P	tity headers in different responses, -Asserted header.	, the Connected number sent to the
SIP header values			
180: P-Asserted	-Identity identity 1		
200 OK: P-Asserted	-Identity identity 2		
DSS1 Parameter val	ues		
CONNECT: Connec	ted number identity 2		
Message flow			
	End device	Tes	st equipment
SETUP	→	→ INVITE	
ALERTING	+	← 180 Ring	ging
CONNECT	+	← 200 OK	INVITE
CONNECT ACK	→	→ ACK	
	A	oply post test routine	

7.2.5.3 Communication HOLD

7.2.5.3.1 Test purposes for POTS

TSS	TP_503_101	Reference	Selection expression
HOLD		subclause B.4.2.2 and C.1.2.1 of [ETSI TS 183 043]	PICS 5.1.1/1
Test purpose			
Originating user set	s the remote party on hold		
		quest is received, the SDP description fo	r the active media stream is set
to a=sendonly, a 200) OK is sent the a line in th	e SDP is set to "recvonly".	
SIP header values			
INVITE/UPDATE			
SDP			
a=sendonly			
200 OK			
SDP			
a=recvonly			

Message flow	
Test equipn	ent End device
	A confirmed dialogue exists
CASE A	
INVITE	→
200 OK INVITE	+
ACK	→
CASE B	
UPDATE	→
200 OK UPDATE	÷ +
	Apply post test routine

TSS	TP_503_102	Reference	Selection expression
HOLD	11_505_102	subclause B.4.2.2 and C.1.2.1 of [ETSI TS 183 043]	PICS 5.1.1/1
Test purpose			
User resumes the remote p	party		
		receipt of a ReINVITE or UPDATE re 0 OK is sent and the a line in the SDP i	
NOTE – sendrecvis the det		O OK is sent and the a line in the SDP i	is set to sendrecv.
	launt value.		
SIP header values INVITE			
SDP			
a=sendrecv (or absent)			
a sendree, (or assent)			
200 OK			
SDP			
a=sendrecv (or absent)			
Message flow			
Test ec	quipment	End	device
	A c	onfirmed dialogue exists	
CASE A			
INVITE1	→		
200 OK INVITE1	+		
ACK	→		
INVITE2 200 OK INVITE2	→ ←		
ACK	←		
ACK			
CASE B			
UPDATE1	→		
200 OK UPDATE1	+		
UPDATE2	→		
200 OK UPDATE3	+		
	A	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 of	on hold		
Ensure that when flash-hoo description for the active m		GC component is received a ReINV donly.	ITE request is sent. The SDP
SIP header values			
INVITE			
SDP			
a=sendonly			
200 OK			
SDP			
a=recvonly			
Message flow			
End	device	Test equ	ipment
	A confirm	ned dialogue exists	
flash-hook		→ INVITE	
		← 200 OK INVITE	
		→ ACK	
	Apply	post test routine	

TSS	TP_503_104	Referen		Selection expression
HOLD		subclaus	se 5.3.1.4 of [TS183 043]	PICS 5.1.1/1 AND 5.3/6
Test purpose				
User resumes the remote h	eld party			
Ensure that if the remote	party is on hold when f	lash-hook	notification from the MC	GC component is received a
ReINVITE request is sent.	The SDP description for	the active	e media stream is set to a=s	endrecv
SIP header values				
INVITE2				
SDP				
a=sendrecv				
200 OK2				
SDP				
a=sendrecv				
Message flow				
End	device		Test equ	ipment
	A confi	irmed dia	logue exists	-
flash-hook		→	INVITE1	
		←	200 OK INVITE1	
		→	ACK	
flash-hook		→	INVITE2	
	← 200 OK INVITE2			
		→	ACK	
	Appl	ly post tes	st routine	

7.2.5.3.2 Test purposes for ISDN

TSS	TP_503_201	Reference	Selection expression
HOLD		subclause 5.2.1.1.1 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/4
Test purpose			
HOLD requested by the	e calling party in an reIN	/ITE request (a=sendonly)	
in the SDP set to 'sendo	only'. The SUT sends a 20	munication on HOLD. The received re 0 OK (INVITE) containing an a attribu- ng user equipment and the Notification	the in the SDP set to 'recvonly'.
SIP header values			
INVITE: SDP a=sendo	nly		
200 OK (INVITE): SD	P a=recvonly		
DSS1 Parameter valu	es		
NOTIFY: Notification	indicator = Remote hold		
Message flow			
Tes	st equipment	End	device
INVITE	→	→ SETUP	
183 Ringing	+	← ALERTING	
200 OK INVITE	+	← CONNECT	
ACK	→		
INVITE (sendonly)	→	→ NOTIFY (Reference)	emote hold)
200 OK (INVITE)	+		
ACK	→		
	A	pply 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			

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 equip	ment	End device
INVITE	→	→ SETUP
180 Ringing	+	← ALERTING
200 OK INVITE	+	← CONNECT
ACK	→	
UPDATE (sendonly)	→	→ NOTIFY (Remote hold)
200 OK (UPDATE)	+	
	Apply pos	st test routine

TSS HOLD	TP_503_203	Reference subclause 5.2.1.1.1 of	Selection expression PICS 5.1.1/2 AND 5.4/4
		[ETSI TS 183 036]	
Test purpose			
HOLD requested by the	calling party in an reINV	ITE request (a=inactive)	
in the SDP set to 'inactiv A DSS1 NOTIFY messa	ve'. The SUT sends a 200 ge is sent to the terminatin	OK (INVITE) containing an a	ived reINVITE contains an a attribute attribute in the SDP set to 'inactive'. ication indicator information element called party.
SIP header values			
INVITE: SDP a=inactiv	e		
200 OK (INVITE): SDP	a=inactive		
DSS1 Parameter value	s		
NOTIFY: Notification in	ndicator = Remote hold		
Message flow			
Test	equipment		End device
INVITE	→	→ SETU	Р
180 Ringing	←	← ALER	TING
200 OK INVITE	+	← CONN	VECT
ACK	→		
		← NOTI	FY (Remote hold)
CASE A	_		
INVITE (sendonly)	+		
200 OK INVITE	→		
ACK	+		
CASE B			
UPDATE (sendonly)	+		
200 OK (UPDATE)	→		
	2		
INVITE (inactive)	→	→ NOTI	FY (Remote hold)
200 OK INVITE	+		
ACK	→		
	Ap	ply post test routine	

TSS	TP_503_204	Reference	Selection expression
HOLD		subclause 5.2.1.1.1 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/4
Test purpose			
HOLD requested by the c	calling party in an UPDA	TE request ($a=inactive$)	
in the SDP set to 'inactive A DSS1 NOTIFY message	b'. The SUT sends a 200 (ge is sent to the terminating	OK (UPDATE) containing an a	ved UPDATE containes an a attribute a attribute in the SDP set to 'inactive'. ication indicator information element called party.
SIP header values			
UPDATE: SDP a=inactiv	/e		
200 OK (UPDATE): SDI	P a=inactive		
DSS1 Parameter values NOTIFY: Notification in			
Message flow			
Test	equipment		End device
INVITE	→	→ SETU	Р
180 Ringing	←	← ALER	TING
200 OK INVITE	←	← CONN	IECT
ACK	→		
		← NOTI	FY (Remote hold)
CASE A			
INVITE (sendonly)	←		
200 OK INVITE	→		
ACK	÷		
CASE B			
UPDATE (sendonly)	÷		
200 OK (UPDATE)	→		
UPDATE (inactive)	→	→ NOTI	FY (Remote hold)
200 OK (UPDATE)	÷		
	Арг	oly 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 <i>RETRIEVE requested by the calling party in an INVITE request</i>				

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 indicate	or = Remote retrieval	
Message flow		
Test equip	ment	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 po	st test routine

T 00	TD 502 006		
TSS HOLD	TP_503_206	Reference subclause 5.2.1.1.1 of	Selection expression
HOLD		[ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/4
Test purpose			
	the calling party in an U	PDATE request	
UPDATE contains an a attribute in the SDP set Notification indicator in	ttribute in the SDP set to	an earlier communication previou o 'sendrecv'. The SUT sends a 200 OTIFY message is sent to the terr o 'Remote retrieval'.	0 OK (INVITE) containing an a
SIP header values			
UPDATE: SDP a=sendr			
200 OK (UPDATE): SD	P a=sendrecv		
DSS1 Parameter value	s		
NOTIFY: Notification in	ndicator = Remote retrieva	al	
Message flow			
Test	equipment	Er	nd device
INVITE	→	→ SETUP	
180 Ringing	+	← ALERTIN	ΙG
200 OK INVITE	+	← CONNEC	Т
ACK	→		
UPDATE (sendonly)	→	→ NOTIFY	(Remote hold)
• • •	, ←		(Remote hold)
200 OK (UPDATE)	2		
UPDATE (sendrecv)	→	→ NOTIFY	(Remote retrieval)
200 OK (UPDATE)	+		
	Ар	ply post test routine	

TSS HOLD	TP_503_207	Reference subclause 5.2.1. [ETSI TS 183 02		Selection expression PICS 5.1.1/2 AND 5.4/4
Test purpose				
HOLD requested by t	he called party a reINVITE o	r UPDATE is sent		
party. The SUT sends		PDATE request and the	e a attribute in	age is received from the called the SDP is set to 'sendonly'. A set to 'recvonly'.
SIP header values				
INVITE/UPDATE: S	•			
200 OK (INVITE/UP	DATE): SDP a=recvonly			
DSS1 Parameter val	ues			
HOLD				
Message flow				
T	est equipment		End	device
INVITE	→	→	SETUP	
180 Ringing	÷	÷	ALERTING	
200 OK INVITE	+	+	CONNECT	
ACK	→			
		4	HOLD	
CASE A		-	noub	
INVITE (sendonly)	+			
200 OK INVITE	→			
ACK	÷			
CASE B				
UPDATE (sendonly)	+			
200 OK (UPDATE)	→			
	App	ly post test routine		

TSS	TP_503_208	Reference	Selection expression
HOLD		subclause 5.2.1.1.3 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/4 AND 5.1.3/1

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 to	est routine

TSS	TP_503_209	Reference	Selection expression
HOLD		subclause 5.2.1.1.3 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/4 AND 5.1.3/1
Test purpose			
	e called party in a private	network	
received and the Notific request is sent and the a	cation indicator informatio attribute in the SDP is set	n element is set to 'Remote hold'. A	on HOLD. A NOTIFY message is A reINVITE request or an UPDATE) or 200 OK (UPDATE) is received a HOLD by the calling user.
SIP header values			
INVITE/UPDATE: SD	P a= sendrecv		
200 OK (INVITE/UPD	ATE): SDP a= sendrecv		
DSS1 Parameter valu	es		
NOTIFY: Notification	indicator = Remote retriev	zal	
Message flow			
Tes	t equipment]	End device
INVITE	→	→ SETUP	
180 Ringing	+	← ALERT	ING
200 OK INVITE	+	← CONNE	СТ
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 tes	t routine

TSS	TP_503_210	Reference	Selection expression
HOLD		subclause 5.2.1.2.1 [ETSI TS 183 036]	of PICS 5.1.1/2 AND 5.4/4
Test purpose			
HOLD requested by the	called party. INVITE wa	s received	
	n INVITE request was re	eceived and the a attribute	otification indicator information element is e in the SDP is set to 'sendonly'. A 200 OK
SIP header values		-	
INVITE: SDP a=sendon	ly		
200 OK (INVITE) SDP	a=recvonly		
DSS1 Parameter values	5		
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
	A	pply post test routine	

TSS HOLD	TP_503_210	Reference subclause 5.2.1.2.1 of	Selection expression PICS 5.1.1/2 AND 5.4/4
		[ETSI TS 183 036]	1105 5.1.1/2 11(1) 5.4/4
Test purpose			
HOLD requested by the ca	lled party. UPDATE w	as received	
	PDATE request was re	TIFY message and the Notification eceived and the a attribute in the SI 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 eq	uipment	Test	equipment
SETUP	→	→ INVITE	
		← 407 Proxy	Authentication Required
		→ ACK	
		→ INVITE	
ALERTING	+	← 180 Ringi	ng
CONNECT	+	← 200 OK II	NVITE
		→ ACK	
NOTIFY (Remote hold)	+	← UPDATE	(sendonly)
		→ 200 OK (UPDATE)
	An	ply 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 equipm	ent	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 po	st 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
maa	TTD 700 010	D 6	

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 po	st 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 c	alled party. INVITE was rec	reived		
set to 'Remote retrieval' if	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	INVITE: SDP a=sendrecv			
200 OK (INVITE) SDP a=	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 po	est test routine

TSS	TP_503_214	Reference	Selection expression
HOLD		subclause 5.2.1.2.1 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/4
Test purpose			
Retrieve requested	by the called party. UPDATE w	as received	
set to 'Remote retri		FY message and the Notification is s received and the a attribute in the DP is set to 'sendrecv.	
SIP header values	1		
UPDATE: SDP a=	sendrecv		
200 OK (UPDATE	E) SDP a=sendrecv		
DSS1 Parameter	values		
NOTIFY: Remote	retrieval		
Message flow			
	Test equipment	Test	equipment
SETUP	→	→ INVITE	
			Authentication Required
		→ ACK	
		→ INVITE	
ALERTING	←	← 180 Ringin	0
CONNECT	+	← 200 OK IN	VITE
		→ ACK	
NOTIFY(Remote h	nold) 🗲	← UPDATE	(condonly)
NOTIT'I (Remote I		 → 200 OK (U 	•
		200 OK (0	1 <i>D</i> (11 L)
NOTIFY (Remote	retrieval)	← UPDATE	sendrecy)
NOTIFY (Remote	retrieval)	 ← UPDATE (→ 200 OK (U 	,

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		
	Γ is able to send a reINVITE 1 HOLD message was received		the a attribute in the SDP is set to
SIP header values			
INVITE/UPDATE:	•		
200 OK (INVITE/U	PDATE) SDP a=recvonly		
DSS1 Parameter v	alues		
Message flow			
	Test equipment	Tes	st equipment
SETUP	→	→ INVITE	
		 ← 407 Prox 	xy Authentication Required
		→ ACK	
		➔ INVITE	
ALERTING	+	← 180 Ring	ging
CONNECT	+	← 200 OK	INVITE
		→ ACK	
HOLD	→		
CASE A		→ INVITE	(sendonly)
		← 200 OK	INVITE
		→ ACK	
CASE B			
		→ UPDAT	E (sendonly)
			(UPDATE)
	An	ply 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 o	calling party			
Ensure that the SUT is abl	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	TP_503_217	Reference	Selection expression
HOLD		subclause 5.2.1.2.3 of	PICS 5.1.1/2 AND 5.4/4
		[ETSI TS 183 036]	AND 5.1.3/1

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	\rightarrow \rightarrow	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 b	y the calling party		
	NOTIFY message was receive	request or UPDATE request and the d from an user in a private network	
SIP header values	1		
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 Ringir	ıg
CONNECT	+	← 200 OK IN	0
		→ ACK	
NOTIFY (Remote	Hold) -		

CASE A	→ INVITE(sendonly)
	← 200 OK INVITE
	➔ ACK
CASE B	
	→ UPDATE (sendonly)
	 ← 200 OK (UPDATE)
	Apply post test routine

TSS	TP_503_219	Reference	Selection expression
HOLD		subclause 5.2.1.2.3 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/4 AND 5.1.3/1
Test purpose			
Retrieve requested by	the calling party.		
	emote retrieval'. A SIP reIN	ession if a DSS1 NOTIFY message VITE or UPDATE request is sent to	
SIP header values			
INVITE/UPDATE: SI	DP a= sendrecv		
200 OK (INVITE/UPI	DATE) SDP a= sendrecv		
DSS1 Parameter valu			
NOTIFY: Remote retr	ieval		
Message flow			
Te	st equipment	Test	equipment
SETUP	→	→ INVITE	
		 ← 407 Proxy 	Authentication Required
		→ ACK	
		→ INVITE	
ALERTING	+	 ← 180 Ringin 	ng
CONNECT	+	← 200 OK IN	IVITE
		➔ ACK	
NOTIFY (Remote Hol	ld) 🗲		
CASE A		\rightarrow INVITE(set	endonly)
		← 200 OK IN	IVITE
		→ ACK	
CASE B			
		➔ UPDATE	(sendonly)
		← 200 OK (U	
NOTIFY (remote retri	eval) →		
CASE A		\rightarrow INVITE(set	endrecv)
		← 200 OK IN	IVITE
		→ ACK	

CASE B

→ UPDATE (sendrecv)

← 200 OK (UPDATE)

Apply post test routine

7.2.5.4 Communication Diversion (CDIV)

7.2.5.4.1 Test purposes for POTS

TSS	TP_504_101	Reference	Selection expression
CDIV		clause C.7 and D.2 of [ETSI TS 183 043]	PICS 5.1.1/1 AND 5.3/7
		[E13113185045]	
Test purpose			
First call line identity			
		in an INVITE request ("history-info be delivered to the called terminal,	
SIP header values			
INVITE:			
History-Info: [<first call="" fo<="" td=""><td>orwarding URI];user=ph</td><td>none>;index=1, [<last call="" forwardin<="" td=""><td>ng URI];user=phone>;index=1.1</td></last></td></first>	orwarding URI];user=ph	none>;index=1, [<last call="" forwardin<="" td=""><td>ng URI];user=phone>;index=1.1</td></last>	ng URI];user=phone>;index=1.1
Message flow			
Test eq	quipment	En	d device
INVITE	→		
180 Ringing	÷	Ringing, information	display call forwarding
	An	ply post test routine	

TSS CDIV	TP_504_102	Referen clause C	ce 2.7 of [ETSI TS183 043]	Selection expression PICS 5.1.1/1	
Test purpose Activate, deactivate or interrogate the call forwarding service					
Ensure that the SUT is able	to send an INVITE requ	lest to activ	vate, deactivate or interroga	te the call forwarding service.	
SIP header values	SIP header values				
INVITE					
Request-Line: PX SC (SR	SI) SX@pes-scc.operato	or.com			
Message flow					
End	device		Test equ	ipment	
Off hook					
Dial service code command	1	→	INVITE		
		÷	407 Proxy Authenticatio	n Required	
→			ACK		
		→	INVITE		
	Apply post test routine				

7.2.5.4.2 Test purposes for ISDN

TSS CDIV	TP_504_201	Reference subclause 5.2.5.1 of	Selection expression PICS 5.1.1/2 AND 5.4/5
		[ETSI TS 183 036]	
Test purpose			
181 received, a N	OTIFY is sent		
	ceipt of a 181 Call Being Forward The Notification indicator is set to		FY message is sent to the DSS
SIP header value	S		
DSS1 Parameter	values		
NOTIFY: Notifica	ation indicator		
Call	l is diverting		
Message flow			
	Test equipment	Test	equipment
SETUP	→	➔ INVITE	
		 ← 407 Proxy 	Authentication Required
		➔ ACK	
		➔ INVITE	
NOTIFY	+	← 181 (Call I	Being Forwarded)
		y post test routine	

TSS	TP_504_202	Reference	Selection expression
CDIV		subclause 5.2.5.1 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/5
Test purpose			
181 received, a PROGRE	ESS is sent		
a P-Early-Media header a		d provisional response where a 18 s previously received, a PROGRE 'Call is diverting'.	
SIP header values			
DSS1 Parameter values			
PROGRESS: Notificatio	n indicator		
Call is	diverting		
Message flow			
Test	equipment	Test	equipment
SETUP	→	➔ INVITE	
		← 407 Proxy	Authentication Required
		→ ACK	
		→ INVITE	
CALL PROCEEDING	+	← 183 (Sessio	on Progress)
PROGRESS	÷	← 181 (Call E	Being Forwarded)
	Appl	y 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 PROGI	RESS is sent		
			a 180 (Ringing) was previously ication indicator is set to 'Call is
SIP header values			
DSS1 Parameter value	es		
PROGRESS: Notificati	on indicator		
Call is	s diverting		
Message flow			
Tes	t equipment	Test	equipment
SETUP	→	➔ INVITE	
		← 407 Proxy	Authentication Required
		→ ACK	
		➔ INVITE	
ALERTING	÷	← 180 (Ring	ging)
PROGRESS	+	← 181 (Call	Being Forwarded)
	Appl	y 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			
	n cause 487 and 408, a notificat	tion is sent	
parameter value 487		onal response indicating a subset 408 (No reply) in the History-Inf	
SIP header values			
181: History-Info:	<appropriate value="">; index=1,</appropriate>		
	<appropriate cause="487</td" value;=""><td>>; index=1.1</td><td></td></appropriate>	>; index=1.1	
	or		
	<appropriate cause="408</td" value;=""><td>>; index=1.1</td><td></td></appropriate>	>; index=1.1	
DSS1 Parameter v	alues		
PROGRESS: Notifi			
Call i	s diverting		
Message flow			
	Test equipment	Test	equipment
SETUP	→	→ INVITE	
		 ← 407 Proxy 	Authentication Required
		→ ACK	
	_	→ INVITE	
ALERTING	+	← 180 (Ringi	0
PROGRESS	+		Being Forwarded)
	Appl	y post test routine	

TSS	TP_504_205	Reference	Selection expression
CDIV		subclause 5.2.5.1 of	PICS 5.1.1/2 AND 5.4/5
		[ETSI TS 183 036]	
Test purpose			
Subsequent diversion	n cause not equal to 487 and 40	08, no notification is sent	
is diverting' is sent to		nal response indicating a subseque ne cause parameter value other than cated in Table 7.2.5.4.2-1.	
SIP header values			
181 2: History-Info:			
<appropri< td=""><td>ate value>; index=1, <appropri< td=""><td>ate value; cause=Reason>; index=</td><td>=1.1</td></appropri<></td></appropri<>	ate value>; index=1, <appropri< td=""><td>ate value; cause=Reason>; index=</td><td>=1.1</td></appropri<>	ate value; cause=Reason>; index=	=1.1
DSS1 Parameter va	alues		
NOTIFY: Notificati	on indicator		
Call is	diverting		
Message flow			
]	Fest equipment	Test	equipment
SETUP	→	→ INVITE	
		← 407 Proxy	Authentication Required
		→ ACK	-
		➔ INVITE	
ALERTING	+	← 180 (Ringi)	ng)
NOTIFY	÷	← 181 (Call H	Being Forwarded) 1
		← 181 (Call H	Being Forwarded) 2
	Арр	ly post test routine	

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	TP_504_206	Reference	Selection expression
CDIV		subclause 5.2.5.1 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/5
Test purpose			·
181 received. Map entry	ping of latest History-Info entry	into the Redirection number if no	Privacy value is present in the
and no Privacy hea		rded) provisional response where htry, a NOTIFY is sent to the DSS t coded as described below:	
	er if Userinfo of latest entry is in	the local number format or if in gl	lobal
	•	JRI is equal to the country where t	
SUT is loca	ted.		
International nu	umber if Userinfo of latest entry	is in the global number format and	the
	-	country where the SUT is located	
• •	entification = ISDN (telephony)	numbering plan	
	tor = Presentation allowed		
Number digits			T
-		try; if the country code of the URI	
•	oal number and the country code intry code is removed from the n	is the same as the AGCF/VGW or	r line is
	•	uniter.	
SIP header values	5		
181: History-Info:	av-1 divarted to UDI: causa-s	any appropriate value>; index=1.1	
		my appropriate value>, mdex=1.1	
DSS1 Parameter NOTIFY: Redirect			
	tor = Presentation allowed		
Message flow	tor – rresentation anowed		
wiessage now		The second se	•
	Test equipment		equipment
SETUP	→	→ INVITE	
		•	Authentication Required
		→ ACK	
		➔ INVITE	
NOTIFY	+	← 181 (Call E	Being Forwarded)
	Арр	ly post test routine	

TSS	TP_504_207	Reference	Selection expression
CDIV		subclause 5.2.5.1 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/5
Test purpose			
180 received. Mappin entry	g of latest History-Info entry	n into the Redirection number if no	Privacy value is present in the
header is present in the a Redirection number		nal response where a History-Info h ING is sent to the DSS1 User equip as described below:	
Type of number			
	•	the local number format or if in gl	
number format SUT is located	•	URI is equal to the country where t	he
	•	is in the global number format and	the
	•	country where the SUT is located	the
•	ification = ISDN (telephony)	•	
• •	= Presentation allowed		
Number digits			
-		ntry; if the country code of the URI	
•	•	the same as the AGCF/VGW or lin	e is
located, the countr	y code is removed from the r	lumber.	
SIP header values			
180: History-Info:	1 d'ant la UDL		
		any appropriate value>; index=1.1	
DSS1 Parameter valu			
ALERTING: Redirec	= Presentation allowed		
Message flow			
-	· · ·	T (• •
	est equipment		equipment
SETUP	→	→ INVITE	
		•	Authentication Required
		→ ACK	
	_	→ INVITE	
ALERTING	+	← 180 (Ringin	ng)
	Арр	ly post test routine	

TSS CDIV	TP_504_208	Reference subclause 5.2.5.1 of	Selection expression PICS 5.1.1/2 AND 5.4/5
		[ETSI TS 183 036]	
Test purpose			
200 OK received. M. the entry	lapping of latest History-Info en	try into the Redirection number	if no Privacy value is present in
Privacy header is pr		visional response where a Histor CONNECT is sent to the DSS1 U coded as described below:	
Type of number			
	_	he local number format or if in gl	
	-	I is equal to the country where th	le
SUT is locate			
	of the URI is not equal to the co	s in the global number format and	I the
•	ntification = ISDN (telephony) r	•	
• •	or = Presentation allowed		
Number digits			
•	eived in the URI of the latest ent	ry; if the country code of the URI	I: In
case for global n	umber and the country code is th	ne same as the AGCF/VGW or lin	ne is
located, the cour	try code is removed from the nu	mber.	
SIP header values			
200: History-Info:			
		ny appropriate value>; index=1.1	
DSS1 Parameter va			
CONNECT: Redire			
	or = Presentation allowed		
Message flow		_	
	Fest equipment		equipment
SETUP	→	→ INVITE	
		•	Authentication Required
		→ ACK	
		→ INVITE	
ALERTING	+	← 180 (Ringin	ng)
CONNECT	÷	← 200 OK IN	IVITE
		→ ACK	
	Apply	y post test routine	

	TP_504_209	Reference	Selection expression
CDIV		subclause 5.2.5.1 of	PICS 5.1.1/2 AND 5.4/5
		[ETSI TS 183 036]	AND 5.1.3/1
Test purpose			
181 received. Mappin the entry	ng of latest History-Info entry	into the Redirection number if a P	rivacy value history is present in
and a Privacy header	r is present in the latest hist-	rded) provisional response where entry value set to 'history', a NO mber Information Element coded	TIFY is sent to the DSS1 User
Type of number = Un			
Numbering plan ident			
	r = Presentation restricted		
Number digits not pre	esent		
SIP header values			
181: History-Info:			
•	=1, <diverted-to cause="a</td" uri;=""><td>ny appropriate value>?Privacy=h</td><td>istory; index=1.1</td></diverted-to>	ny appropriate value>?Privacy=h	istory; index=1.1
<uri>; index= DSS1 Parameter val</uri>	lues	ny appropriate value>?Privacy=h	istory; index=1.1
<uri>; index= DSS1 Parameter val NOTIFY: Redirection</uri>	l ues n number		istory; index=1.1
<uri>; index= DSS1 Parameter val NOTIFY: Redirection</uri>	lues		istory; index=1.1
<uri>; index= DSS1 Parameter val NOTIFY: Redirection</uri>	l ues n number		istory; index=1.1
<uri>; index= DSS1 Parameter val NOTIFY: Redirection Present Message flow</uri>	l ues n number	restricted	istory; index=1.1 equipment
<uri>; index= DSS1 Parameter val NOTIFY: Redirection Present Message flow</uri>	lues n number ation indicator = Presentation	restricted	
<uri>; index= DSS1 Parameter val NOTIFY: Redirection Present Message flow</uri>	lues n number ation indicator = Presentation est equipment	restricted Test → INVITE	
<uri>; index= DSS1 Parameter val NOTIFY: Redirection Present Message flow</uri>	lues n number ation indicator = Presentation est equipment	restricted Test → INVITE	equipment
<uri>; index= DSS1 Parameter val NOTIFY: Redirection Present Message flow</uri>	lues n number ation indicator = Presentation est equipment	restricted Test → INVITE ← 407 Proxy	equipment
<uri>; index= DSS1 Parameter val NOTIFY: Redirection Present Message flow</uri>	lues n number ation indicator = Presentation est equipment	restricted Test → INVITE ← 407 Proxy → ACK → INVITE	equipment

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
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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
		[E13113183030]	AND 5.1.3/1
Test purpose			
* *	ıg of latest History-Info entry i	nto the Redirection number if a F	rivacy value history is present in
the entry			
Ensure that on receipt	t of a 200 OK (INVITE) provisi	onal response where a History-In	fo header is present and a Privacy
header is present in t	he latest hist-entry value set to		the DSS1 User equipment. The
Type of number = Ur	ıknown		
Numbering plan iden	tification = Unknown		
Presentation indicator	r = Presentation restricted		
Number digits not pro	esent		
SIP header values			
200: History-Info:			
<uri>; index</uri>	=1, <diverted-to cause="a</td" uri;=""><td>ny appropriate value>?Privacy=h</td><td>istory; index=1.1</td></diverted-to>	ny appropriate value>?Privacy=h	istory; index=1.1
DSS1 Parameter val	lues		
CONNECT: Redired	ction number		
Present	tation indicator = Presentation	restricted	
Message flow			
Т	est equipment	Test	equipment
SETUP	→	→ INVITE	
		← 407 Proxy	Authentication Required
		→ ACK	
		→ INVITE	
ALERTING	+	← 180 (Ringi	nσ)
CONNECT	`````	 ← 200 OK IN 	0,
COMMECT	τ.		N VII L
		→ ACK	
	Appl	y post test routine	

TSS	TP_504_212	Reference	Selection expression
CDIV		subclause 5.2.5.1 of	PICS 5.1.1/2
		[ETSI TS 183 036]	
Test purpose			
181 received. Sendin	g of Redirection number 'not a	vailable' if no History-Info heade	r is present.
a History-Info heade	r is not present, a NOTIFY is	receipt of a 181 (Call Being Forverse sent to the DSS1 User equipment	
	Information Element coded as o	described below:	
Type of number = Un			
• •	tification = Unknown		
	r = Number not available due	e to interworking	
Number digits not pr	esent		
SIP header values			
DSS1 Parameter va	lues		
PROGRESS: Redire			
Presen	tation indicator = Presentation	restricted	
Message flow			
Т	est equipment	Test	equipment
SETUP	→	➔ INVITE	
		 ← 407 Proxy 	Authentication Required
		→ ACK	
		→ INVITE	
ALERTING	←	← 180 (Ringi	ng)
PROGRESS	÷		Being Forwarded)
I ICOULDD	× ×	×.	Joing I of Warded)
	Δ	y post test routine	

TSS	TP_504_213	Reference	Selection expression
CDIV		subclause 5.2.5.1 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/5
		[E13113163030]	AND 5.1.3/1

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 equipmen	ıt		Test equipment
SETUP	→	→	INVITE
		÷	407 Proxy Authentication Required
		→	ACK
		→	INVITE
	Apply post	test routine	

TSS	TP_504_214	Reference	Selection expression
CDIV		subclause 5.2.5.1 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1

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

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

	Test equipment		Test equipment
SETUP	→	→ INV	ITE
		← 407	Proxy Authentication Required
		→ ACI	X
		→ INV	ITE
	Арр	y 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	TP_504_215	Reference	Selection expression
CDIV		subclause 5.2.5.1 of	PICS 5.1.1/2 AND 5.4/5
		[ETSI TS 183 036]	AND 5.1.3/1
Test purpose			
181 received. Mapping of invoke component	the value of the Privacy para	neter in latest History Entry	into the DivertingLegInformation3
history entry are mapped		ment DivertingLegInformati	privacy requirements in the latest on3 presentationAllowedIndicator
SIP header values			
181: History-Info:			
< diverting	Nr >; index=1,		
<request td="" u<=""><td>RI>; cause= privacy_VA>; in</td><td>ndex=1.1</td><td></td></request>	RI>; cause= privacy_VA>; in	ndex=1.1	
DSS1 Parameter values			
FACILITY: Facility			
DivertingLe	egInformation3		
presenta	tionAllowedIndicator = priva	cy_VA	
Message flow			
Test e	quipment	Tes	st equipment
SETUP	→	➔ INVITE	
		← 407 Prox	y Authentication Required
		→ ACK	-
		➔ INVITE	

← 181 (Call Being Forward)	ed)
----------------------------	-----

Apply post test routine

(

TSS	TP_504_216	Reference	Selection expression
CDIV		subclause 5.2.5.1 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1

Test purpose

FACILITY

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:				
< dive	rtingNr >; index=1,			
<requ< td=""><td>est URI>; cause= privacy_VA></td><td>>; index=1.1</td><td></td><td></td></requ<>	est URI>; cause= privacy_VA>	>; index=1.1		
DSS1 Parameter va	lues			
ALERTING: Facilit	У			
Divert	ingLegInformation3			
pre	esentationAllowedIndicator = pr	rivacy_VA		
Message flow				
J	Fest equipment		Test e	quipment
SETUP	→	→	INVITE	
		÷	407 Proxy A	Authentication Required
		→	ACK	
		→	INVITE	
ALERTING	←	←	181 (Call B	eing Forwarded)
	_	y post test routine	101 (0001 2	
	1 PP	J Post test routille		
TSS	TP_504_217	Reference		Selection expression
CDIV		subclause 5.2		PICS 5.1.1/2 AND 5.4/5
		[ETSI TS 183 036] AND 5.1.3		AND 5.1.3/1

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</request>

DSS1 Parameter values

CONNECT:	Facility
----------	----------

CONNECT: Faci	lity			
Dive	ertingLegInformation3			
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	TP_503_218	Reference	Selection expression
CDIV		subclause 5.2.5.2.1 of	PICS 5.1.1/2 AND 5.4/5
		[ETSI TS 183 036]	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 v	alues			
ALERTING: Facili	ty			
D	ivertingLegInformation1			
	diversionReason = reason_VA			
	subscriptionOption = noNotification	n		
	divertedToNumber = diverted to nu	mber (PIXIT)		
Message flow				
	Test equipment		End device	
INVITE	→	→	SETUP	
180 Ringing	←	÷	ALERTING	
	Apply pos	st test routine		

TSS	TP_503_219	Reference	Selection expression
CDIV		subclause 5.2.5.2.1 of [ETSI TS 183 036]	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
------------------------	--	---

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 num	iber (PIXIT)
Message flow	
Test equipment	End device
INVITE SETUP	
	← FACILITY
Annly n	nast test routine

Apply post test routine

TSS CDIV	'P_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
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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				
Diverting	LegInformation1			
divers	ionReason = reason_VA			
subscr	iptionOption = notificationWi	ithoutDivertedT	CoNr	
divert	edToNumber = diverted to nur	mber (PIXIT)		
Message flow				
Test eq	uipment		End device	
INVITE	→	→	SETUP	
180 Ringing	←	÷	ALERTING	
	Apply pos	t 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
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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				
DivertingLegI	nformation1			
diversionR	eason = reason_VA			
subscriptio	nOption = notificationW	ithoutDivertedT	`oNr	
divertedTo	Number = diverted to nu	mber (PIXIT)		
Message flow				
Test equipm	ient		End device	
INVITE	→	→	SETUP	
183 (Session Progress)	+	÷	PROGRESS	
	Apply pos	st test routine		

TSS	TP_503_223	Reference	Selection expression
CDIV		subclause 5.2.5.2.1 of	PICS 5.1.1/2 AND 5.4/5
		[ETSI TS 183 036]	AND 5.1.3/1
Test purpose			
ALERTING received. M is sent.	apping of DivertingLegIn	formation1 into History-Info head	er a restricted History-Info header
diversionReason as 'notificationWithDivert Privacy header is presen First entry: URI not sign	indicated in Table edToNr', a 180 (Ringing) nt in the last entry coded a nificant; index=1	7.2.5.4.2-2 is present, the provisional response is sent and a	History-Info header is present no
SIP header values		uniber, cause=reason_vA, index-	-1.1.
180: History-Info:			
•	gnificant value; index=1,		
		cause=reason_VA; index=1.1	
DSS1 Parameter value	es		
ALERTING: Facility			
Diver	tingLegInformation1		
div	versionReason = reason_V	/A	
su	bscriptionOption = notific	ationWithDivertedToNr	
div	vertedToNumber = diverted	ed to number (PIXIT)	
Message flow			
Tes	t equipment	E	Ind device
INVITE	→	→ SETUP	
180 (Ringing)	←	← ALERTI	NG
	AI	oply 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
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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,

 $< diverted To Number ? Privacy=history; \ cause=reason_VA; \ index=1.1$

DSS1 Parameter values

FACILITY: Facility

DivertingLegInformation1 diversionReason = reason_VA subscriptionOption = notificationWithDivertedToNr divertedToNumber = diverted to number (PIXIT)

Message flow			
Test equipm	ent		End device
INVITE	→	→	SETUP
181 (Being forwarded)	+	←	FACILITY
	Apply po	ost test routine	

TSS	TP_503_225	Reference	Selection expression
CDIV		subclause 5.2.5.2.1 of	PICS 5.1.1/2 AND 5.4/5
		[ETSI TS 183 036]	AND 5.1.3/1
Test purpose			
	ed. Mapping of DivertingLegI	nformation1 into History-Info head	der a restricted History-Info header
is sent.			
Enguna that on m	agint of an DDOCDESS or	d a Easility Information Flows	nt Diverting LagInformation 1 the
			nt DivertingLegInformation1, the subscriptionOption is set to
			sent and a History-Info header is
	header is present in the last en	ntry coded as follows:	
•	t significant; index=1		
Second entry URI of	lerived from the 'divertedToN	Number'; cause=reason_VA; index	=1.1.
SIP header values			
183: History-Info			
	n significant value; index=1,		
< diverte	dToNumber?Privacy=history	; cause=reason_VA; index=1.1	
DSS1 Parameter v			
PROGRESS: Facil	•		
D	ivertingLegInformation1		
	diversionReason = reason_		
	subscriptionOption = notifi		
	divertedToNumber = divert	ted to number (PIXIT)	
Message flow			
	Test equipment]	End device
INVITE	→	→ SETUP	
183 (Session Progre	ess) +	← PROGR	ESS
	Α	pply post test routine	

TSS	TP_503_226	Reference	Selection expression
CDIV		subclause 5.2.5.2.1 of	PICS 5.1.1/2 AND 5.4/5
		[ETSI TS 183 036]	AND 5.1.3/1

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</any>				
<uri of="" re<="" set="" td="" the="" to="" value=""><td>equest URI; cause=any>; index=1.1</td></uri>	equest URI; cause=any>; index=1.1			
180: History-Info:				
<any uri="">index=1</any>				
<uri of="" reque<="" set="" td="" the="" to="" value=""><td>est URI; cause=any>; index=1.1</td></uri>	est URI; cause=any>; index=1.1			
DSS1 Parameter values				
ALERTING: Facility				
DivertingLegInformation3				
presentationAllowedIng	licator=true			
Message flow				
Test equipment	End device			
INVITE -	→ SETUP			
180 (Ringing) \leftarrow \leftarrow ALERTING				
	Apply post test routine			

TSS	TP_503_227	Reference		Selection expression
CDIV		subclause 5.2.5.2.1 of		PICS 5.1.1/2 AND 5.4/5
		[ETSI TS 183 036]		AND 5.1.3/1
Test purpose				
DivertingLegInformation3	received in an ALERT	TING and History-Info h	eader is send in	the 180
and the presentationAllow	edIndicator is set to prisent. A Privacy header	ivacy_VA in an ALERT	ING message,	ormation3 invoke component a 180 (Ringing) is sent and a y-Info header is processed as
SIP header values				
INVITE: History-Info:				
<any uri="">ii</any>	ndex=1			
<uri set="" td="" to<=""><td>the value of the Reques</td><td>st URI; cause=any>; ind</td><td>ex=1.1</td><td></td></uri>	the value of the Reques	st URI; cause=any>; ind	ex=1.1	
180: History-Info:				
<any uri="">inde</any>	x=1			
<uri set="" td="" the<="" to=""><td>value of the Request U</td><td>RI[privacy_VA]; cause</td><td>=any>; index=1</td><td>.1</td></uri>	value of the Request U	RI[privacy_VA]; cause	=any>; index=1	.1
DSS1 Parameter values				
ALERTING: Facility				
Diverting	gLegInformation3			
prese	ntationAllowedIndicate	or= privacy_VA		
Message flow				
Test ec	quipment		End d	levice
INVITE	→	→	SETUP	
180 (Ringing)	+	←	ALERTING	
	A	oply post test routine		

TSS	TP_503_228	Reference	Selection expression
CDIV		subclause 5.2.5.2.1 of	PICS 5.1.1/2 AND 5.4/5
		[ETSI TS 183 036]	AND 5.1.3/1
Test purpose	·		· · ·
DivertingLegInformation3	received in an FAC	ILITY and History-Info header is se	nd in the 181
and the presentation Allow	edIndicator is set to is present. A Privac	on Element containing a DivertingL privacy_VA in a FACILITY messa y header field in the latest entry in th	ge, a 181 (Being forwarded) is sent
SIP header values			
INVITE: History-Info:			
<any uri="">ii</any>	ndex=1		
<uri set="" td="" to<=""><td>the value of the Req</td><td>uest URI; cause=any>; index=1.1</td><td></td></uri>	the value of the Req	uest URI; cause=any>; index=1.1	
181: History-Info:			
<any uri="">inde</any>	x=1		
<uri set="" td="" the<="" to=""><td>value of the Reques</td><td>t URI[privacy_VA]; cause=any>; in</td><td>dex=1.1</td></uri>	value of the Reques	t URI[privacy_VA]; cause=any>; in	dex=1.1
DSS1 Parameter values			
FACILITY: Facility			
Diverting	gLegInformation3		
prese	ntationAllowedIndi	cator= privacy_VA	
Message flow			
Test ec	luipment		End device
INVITE	→	→ SETUP	
181 (Being forwarded)	←	← FACILI	TY
		Apply post test routine	

TSS	Reference	Selection expression
CDIV	subclause 5.2.5.2.1 of	PICS 5.1.1/2 AND 5.4/5
TP_503_229	[ETSI TS 183 036]	AND 5.1.3/1

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.

INVITE: History-Info:
<any uri="">index=1</any>
<uri cause="any" of="" request="" set="" the="" to="" uri;="" value="">; index=1.1</uri>
200 OK: History-Info:
<any uri="">index=1</any>
<uri cause="any" of="" request="" set="" the="" to="" uri[privacy_va];="" value="">;</uri>
index=1.1
DSS1 Parameter values
CONNECT: Facility
DivertingLegInformation3
presentationAllowedIndicator= privacy_VA

Message flow					
Г	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	Selection expression PICS 5.1.1/2 AND 5.4/5	
		[ETSI TS 183 036]		
Test purpose				
INVITE received History	-Info header contains two	o entries not restricted, Redirecting	number sent in SETUP	
entry is not escaped with from the URI of the first	an Privacy header, a SE at entry of the History-Ir	a History-Info header is present co TUP is sent and the Redirecting nu- tifo header and the Reason of dive ow and in Table 7.2.5.4.2-5:	umber address signals is derived	
Redirecting number				
Type of number				
National if the URI is	coded as follows sip: loc	cal-number@hostportion		
international if the UI	RI is coded as follows sip	: globalnumber@hostportion and	CC is	
not the	same as the country when	re the user or line is located		
Numbering plan identific	ation: ISDN numbering p	blan		
Presentation indicator: pr	Presentation indicator: presentation allowed			
Reason of diversion: reas	on_VA			
Number digits:				
		mber-digits: if the country code of the de is removed from the Userinfo	he URI is the same as the country	
SIP header values				
INVITE: History-Info:				
<any td="" uri<=""><td>1 (PIXIT)>index=1</td><td></td><td></td></any>	1 (PIXIT)>index=1			
<uri set="" td="" to<=""><td>the value of the Request</td><td>t URI; cause=reason_VA >; index=</td><td>1.1</td></uri>	the value of the Request	t URI; cause=reason_VA >; index=	1.1	
DSS1 Parameter values				
SETUP: Redirecting nu	mber			
Presentation indicator: presentation allowed				
Reason of	diversion: reason_VA			
Message flow				
Test	equipment	En	d device	
INVITE	→	→ SETUP		
	Ар	ply post test routine		

TSS	TP_503_231	Reference	Selection expression
CDIV		subclause 5.2.5.2.2 of	PICS 5.1.1/2 AND 5.4/5
		[ETSI TS 183 036]	
Test purpose			
IINVITE received Histo	ry-Info header contains tv	vo entries restricted, Redirecting n	umber sent in SETUP
entry is escaped with a		a History-Info header is present co story', a SETUP is sent and the Re able 7.2.5.4.2-5:	
Redirecting number			
Type of number; unkno	wn		
Numbering plan identif	ication: unknown		
Presentation indicator: j	presentation restricted		
Reason of diversion: re	ason_VA		
Number digits: not p	present		
SIP header values			
INVITE: History-Info:			
<any td="" ur<=""><td>[?Privacy=history >index</td><td>=1</td><td></td></any>	[?Privacy=history >index	=1	
<uri set<="" td=""><td>to the value of the Reques</td><td>t URI; cause=reason_VA >; index</td><td>=1.1</td></uri>	to the value of the Reques	t URI; cause=reason_VA >; index	=1.1
DSS1 Parameter value	es		
SETUP: Redirecting r	number		
Presentat	ion indicator: presentatio	n restricted	
Reason of	f diversion: reason_VA		
Message flow			
Tes	t equipment	E	nd device
INVITE	→	→ SETUP	
		ply post test routine	

TSS	TP_503_232	Reference	Selection expression
CDIV		subclause 5.2.5.2.2 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/5
Test purpose			
INVITE received H	istory-Info header contains on	e entry, Redirecting number sent in	n SETUP
		a History-Info header is present con address signals are absent and co	
Redirecting numb	er		
Type of number; un	ıknown		
Numbering plan ide	entification: unknown		
Presentation indicat	tor: number not available du	e to interworking	
Reason of diversion	n: reason_VA		
Number digits:	not present		
SIP header values			
INVITE: History-	Info:		
<ur< td=""><td>I set to the value of the Reques</td><td>st URI; cause=reason_VA >; index</td><td>x=1.1</td></ur<>	I set to the value of the Reques	st URI; cause= reason_VA >; index	x=1.1
DSS1 Parameter v	values		
SETUP: Redirect	ing number		
Prese	entation indicator: number not	t available due to interworking	

Reason of diversion: reason_VA

Message flow			
	Test equipment	I	End device
INVITE	→	→ SETUP	
	Ap	ply post test routine	
TSS	TP_503_233	Reference	Selection expression
CDIV		subclause 5.2.5.2.2/ [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/5
Test purpose			
	History-Info header contains mo	re than two entries not restricted,	Redirecting number sent in SETUP
the second last ent	try is not escaped with an Privac	y header, a SETUP is sent and the	ntaining more than two entries and e first Redirecting number address der and the Reason of diversion is
			ddress signals are derived from the
URI of the first en	ntry of the History-Info header a	and the Reason of diversion is de	rived from the cause parameter of
the second entry c	oded as described below and in	Table 7.2.5.4.2-5:	
First Redirecting	number		
Type of number	,		
• •	URI is coded as follows sip: loc	cal-number@hostportion	
		: global –number@hostportion a	nd CC is
	not the same as the country when	•	
	dentification: ISDN numbering p		
• •	ator: presentation allowed		
Reason of diversion	_		
Number digits:			
		bbal-number-digits: if the country puntry code is removed from the	code of the URI is the same as the Userinfo
Second Redirecti	ng number		
Type of number			
National if the	URI is coded as follows sip: loc	cal-number@hostportion	
	_	: global –number@hostportion a	nd CC is
	not the same as the country when	•	
	dentification: ISDN numbering p		
	ator: presentation allowed		
Number digits:			
		mber-digits: if the country code of ode is removed from the Userinfo	f the URI is the same as the country
SIP header value	es		
INVITE: History-	Info:		
<any td="" u<=""><td>RI 1 (PIXIT)>index=1</td><td></td><td></td></any>	RI 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			
Т	est equipment	E	nd device
INVITE			
	A	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 Hist	tory-Info header contains n	nore than two entries restricted, Red	lirecting number sent in SETUP
last entry is escaped v	with an Privacy header values and the Reason of diver	a History-Info header is present con ue 'history', a SETUP is sent and the rsion is derived from the cause par	e first Redirecting number address
1st Redirecting num	lber		
Type of number; unk	nown		
Numbering plan iden	tification: unknown		
Presentation indicator	r: presentation restricted		
Reason of diversion:	reason_VA		
Number digits: not pr	resent		
2nd Redirecting nur	nber		
SIP header values			
INVITE: History-Int	fo:		
•	RI 1 (PIXIT)>index=1		
•		tory; cause=any appropriate value>;	index=1.1
-		est URI; cause=reason_VA>; index:	
DSS1 Parameter val	ues		
SETUP: Redirectin	ng number		
Present	ation indicator: presentation	on restricted	
Reason	of diversion: reason_VA		
Redirecting	g number		
Message flow			
Т	est equipment	E	nd device
INVITE	→	→ SETUP	

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	TP_503_235	Reference	Selection expression	
CDIV		subclause 5.2.5.2.3 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/5 AND 5.1.3/1	
Test purpose				
	ry-Info header present w	ith more than two entries, a SETUP	is sent at the T Reference point	
SETUP contains a Fac diversionCounter is deriv The diversionReason is	ility Information Elem ed from the number of do derived from the cause on the URI of second las	d a History-Info header is present, nent with a DivertingLegInformat ots in the index parameter of the last parameter of the last entry as deso t entry and the originalCalledNr is o	ion2 invoke component. The entry of the History-Info header. cribed in Table 7.2.5.4.2-6, the	
SIP header values				
INVITE: History-Info:				
•	IXIT)>index=1			
2	IXIT); cause=any approp	oriate value>; index=1.1		
•		RI; cause= reason_VA >; index=1.1	.1	
DSS1 Parameter values				
SETUP: Facility				
•	egInformation2			
	onCounter: 1			
diversio	diversionReason: reason_VA			
divertingNr: derived from the first entry				
Message flow				
Test	Test equipment End device			
INVITE	→	→ SETUP		
	An	ply 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	TP_503_236	Reference	Selection expression
CDIV		subclause 5.2.5.2.3 of	PICS 5.1.1/2 AND 5.4/5
		[ETSI TS 183 036]	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-Inf	ō:		
•	(PIXIT)>index=1		
•		RI; cause=any appropriate value>;	
index=			
180: History-Info:			
	(PIXIT)>index=1		
<uri set="" t<="" th=""><th>to the value of the Request U</th><th>RI of the INVITE?Privacy=history</th><th>· · · · · · · · · · · · · · · · · · ·</th></uri>	to the value of the Request U	RI of the INVITE? Privacy=history	· · · · · · · · · · · · · · · · · · ·
cause=	any appropriate value>; inde	ex=1.1	
DSS1 Parameter va	lues		
Message flow			
Т	est equipment	En	d device
INVITE	→	→ SETUP	
180 Ringing	+	← ALERTIN	G
88		oply post test routine	-
	·		
TSS	TP 503 237	Reference	Selection expression
CDIV	11_505_257	subclause 5.2.5.2.3 of	PICS 5.1.1/2 AND 5.4/5
0211		[ETSI TS 183 036]	AND 5.1.3/1
	occurs a 183 is sent		
183 Session Progress value is the same as r	received and a History-Info	header is present, ensure that on rec t. The 183 Session Progress contair E, a Privacy header is escaped in the	ns a History-Info header and the
<i>No further diversion</i> When an INVITE is 183 Session Progress value is the same as r 'history'.	received and a History-Info	t. The 183 Session Progress contair	ns a History-Info header and the
No further diversion When an INVITE is 183 Session Progress value is the same as r 'history'. SIP header values	received and a History-Info s provisional response is sen received in the initial INVIT	t. The 183 Session Progress contair	ns a History-Info header and the
No further diversion When an INVITE is 183 Session Progress value is the same as r 'history'. SIP header values INVITE: History-Infe	received and a History-Info s provisional response is sen received in the initial INVIT	t. The 183 Session Progress contair	ns a History-Info header and the
No further diversion When an INVITE is 183 Session Progress value is the same as r 'history'. SIP header values INVITE: History-Info <any td="" uri<=""><td>received and a History-Info s provisional response is sen received in the initial INVITI o: (PIXIT)>index=1</td><td>t. The 183 Session Progress contair E, a Privacy header is escaped in the</td><td>ns a History-Info header and the</td></any>	received and a History-Info s provisional response is sen received in the initial INVITI o: (PIXIT)>index=1	t. The 183 Session Progress contair E, a Privacy header is escaped in the	ns a History-Info header and the
No further diversion When an INVITE is 183 Session Progress value is the same as r 'history'. SIP header values INVITE: History-Info <any td="" uri<=""><td>received and a History-Info s provisional response is sen received in the initial INVIT o: (PIXIT)>index=1 to the value of the Request U</td><td>t. The 183 Session Progress contair</td><td>ns a History-Info header and the</td></any>	received and a History-Info s provisional response is sen received in the initial INVIT o: (PIXIT)>index=1 to the value of the Request U	t. The 183 Session Progress contair	ns a History-Info header and the
No further diversion When an INVITE is 183 Session Progress value is the same as r 'history'. SIP header values INVITE: History-Info <any uri<br=""><uri set="" t<br="">index=</uri></any>	received and a History-Info s provisional response is sen received in the initial INVIT o: (PIXIT)>index=1 to the value of the Request U	t. The 183 Session Progress contair E, a Privacy header is escaped in the	ns a History-Info header and the
No further diversion When an INVITE is 183 Session Progress value is the same as r 'history'. SIP header values INVITE: History-Info <any uri<br=""><uri set="" t<br="">index= 180: History-Info:</uri></any>	received and a History-Info s provisional response is sen received in the initial INVIT o: (PIXIT)>index=1 to the value of the Request U 1.1	t. The 183 Session Progress contair E, a Privacy header is escaped in the	ns a History-Info header and the
No further diversion When an INVITE is 183 Session Progress value is the same as r 'history'. SIP header values INVITE: History-Info <any uri<br=""><uri set="" t<br="">index= 180: History-Info: <any td="" uri<=""><td>received and a History-Info s provisional response is sen received in the initial INVIT o: (PIXIT)>index=1 to the value of the Request U 1.1 (PIXIT)>index=1</td><td>t. The 183 Session Progress contair E, a Privacy header is escaped in the RI; cause=any appropriate value>;</td><td>as a History-Info header and the last entry and the value is set to</td></any></uri></any>	received and a History-Info s provisional response is sen received in the initial INVIT o: (PIXIT)>index=1 to the value of the Request U 1.1 (PIXIT)>index=1	t. The 183 Session Progress contair E, a Privacy header is escaped in the RI; cause=any appropriate value>;	as a History-Info header and the last entry and the value is set to
No further diversion When an INVITE is 183 Session Progress value is the same as r 'history'. SIP header values INVITE: History-Info <any uri<br=""><uri set="" t<br="">index= 180: History-Info: <any uri<br=""><uri set="" t<="" td=""><td>received and a History-Info s provisional response is sen received in the initial INVITI o: (PIXIT)>index=1 to the value of the Request U 1.1 (PIXIT)>index=1 to the value of the Request U</td><td>t. The 183 Session Progress contair E, a Privacy header is escaped in the RI; cause=any appropriate value>; RI of the INVITE?Privacy=history</td><td>as a History-Info header and the last entry and the value is set to</td></uri></any></uri></any>	received and a History-Info s provisional response is sen received in the initial INVITI o: (PIXIT)>index=1 to the value of the Request U 1.1 (PIXIT)>index=1 to the value of the Request U	t. The 183 Session Progress contair E, a Privacy header is escaped in the RI; cause=any appropriate value>; RI of the INVITE? Privacy=history	as a History-Info header and the last entry and the value is set to
No further diversion When an INVITE is 183 Session Progress value is the same as r 'history'. SIP header values INVITE: History-Info <any uri<br=""><uri set="" t<br="">index= 180: History-Info: <any uri<br=""><uri set="" t<br="">cause=</uri></any></uri></any>	received and a History-Info s provisional response is sen received in the initial INVIT o: (PIXIT)>index=1 to the value of the Request U 1.1 (PIXIT)>index=1 to the value of the Request U tany appropriate value>; inde	t. The 183 Session Progress contair E, a Privacy header is escaped in the RI; cause=any appropriate value>; RI of the INVITE? Privacy=history	as a History-Info header and the last entry and the value is set to
No further diversion When an INVITE is 183 Session Progress value is the same as r 'history'. SIP header values INVITE: History-Info <any uri<br=""><uri set="" t<br="">index= 180: History-Info: <any uri<br=""><uri set="" t<br="">cause= DSS1 Parameter va</uri></any></uri></any>	received and a History-Info s provisional response is sen received in the initial INVIT o: (PIXIT)>index=1 to the value of the Request U 1.1 (PIXIT)>index=1 to the value of the Request U tany appropriate value>; inde	t. The 183 Session Progress contair E, a Privacy header is escaped in the RI; cause=any appropriate value>; RI of the INVITE? Privacy=history	as a History-Info header and the last entry and the value is set to
No further diversion When an INVITE is 183 Session Progress value is the same as r 'history'. SIP header values INVITE: History-Info <any uri<br=""><uri set="" t<br="">index= 180: History-Info: <any uri<br=""><uri set="" t<br="">cause= DSS1 Parameter va Message flow</uri></any></uri></any>	received and a History-Info s provisional response is sen received in the initial INVIT o: (PIXIT)>index=1 to the value of the Request U 1.1 (PIXIT)>index=1 to the value of the Request U tany appropriate value>; inde	t. The 183 Session Progress contair E, a Privacy header is escaped in the RI; cause=any appropriate value>; RI of the INVITE? Privacy=history ex=1.1	as a History-Info header and the last entry and the value is set to
No further diversion When an INVITE is 183 Session Progress value is the same as r 'history'. SIP header values INVITE: History-Info <any uri<br=""><uri set="" t<br="">index= 180: History-Info: <any uri<br=""><uri set="" t<br="">cause= DSS1 Parameter va Message flow</uri></any></uri></any>	received and a History-Info s provisional response is sen received in the initial INVIT o: (PIXIT)>index=1 to the value of the Request U 1.1 (PIXIT)>index=1 to the value of the Request U any appropriate value>; inde lues	t. The 183 Session Progress contair E, a Privacy header is escaped in the RI; cause=any appropriate value>; RI of the INVITE? Privacy=history ex=1.1	hs a History-Info header and the last entry and the value is set to
No further diversion When an INVITE is 183 Session Progress value is the same as r 'history'. SIP header values INVITE: History-Info <any uri<br=""><uri set="" t<br="">index= 180: History-Info: <any uri<br=""><uri set="" t<br="">cause= DSS1 Parameter va Message flow T INVITE</uri></any></uri></any>	received and a History-Info s provisional response is sen received in the initial INVIT o: (PIXIT)>index=1 to the value of the Request U 1.1 (PIXIT)>index=1 to the value of the Request U any appropriate value>; inde lues	t. The 183 Session Progress contair E, a Privacy header is escaped in the RI; cause=any appropriate value>; RI of the INVITE? Privacy=history ex=1.1 En	hs a History-Info header and the e last entry and the value is set to "; d device
No further diversion When an INVITE is 183 Session Progress value is the same as r 'history'. SIP header values INVITE: History-Info <any uri<br=""><uri set="" t<br="">index= 180: History-Info: <any uri<br=""><uri set="" t<br="">cause= DSS1 Parameter va Message flow</uri></any></uri></any>	received and a History-Info s provisional response is sen received in the initial INVITI o: (PIXIT)>index=1 to the value of the Request U any appropriate value>; inde lues Yest equipment → €	t. The 183 Session Progress contair E, a Privacy header is escaped in the RI; cause=any appropriate value>; RI of the INVITE? Privacy=history ex=1.1 En → SETUP	hs a History-Info header and the e last entry and the value is set to "; d device G

	TP_503_238	Reference	Selection expression
CDIV		subclause 5.2.5.2.3 of	PICS 5.1.1/2 AND 5.4/5
		[ETSI TS 183 036]	AND 5.1.3/1
Test purpose			
No further diversion occurs of	a 200 OK is sent		
200 OK INVITE final respon	nse is sent. The 200 C	header is present, ensure that on re DK INVITE contains a History-Info er is escaped in the last entry and the	header and the value is the same
SIP header values			
INVITE: History-Info:			
<any (pixit)<="" td="" uri=""><td>)>index=1</td><td></td><td></td></any>)>index=1		
<uri set="" td="" the="" to="" va<=""><td>lue of the Request U</td><td>RI; cause=any appropriate value>;</td><td></td></uri>	lue of the Request U	RI; cause=any appropriate value>;	
index=1.1			
200 OK: History-Info:			
<any (pixit)<="" td="" uri=""><td></td><td></td><td></td></any>			
<any (pixit)<br="" uri=""><uri set="" td="" the="" to="" va<=""><td>lue of the Request U</td><td>RI of the INVITE?Privacy=histor</td><td>y;</td></uri></any>	lue of the Request U	RI of the INVITE? Privacy=histor	y;
<any (pixit)<br="" uri=""><uri set="" td="" the="" to="" va<=""><td></td><td>-</td><td>y;</td></uri></any>		-	y;
<any (pixit)<br="" uri=""><uri set="" td="" the="" to="" va<=""><td>lue of the Request U</td><td>-</td><td>y;</td></uri></any>	lue of the Request U	-	y ;
<any (pixit)<br="" uri=""><uri set="" the="" to="" va<br="">cause=any app</uri></any>	lue of the Request U	-	y ;
<any (pixit)<br="" uri=""><uri set="" the="" to="" va<br="">cause=any app DSS1 Parameter values</uri></any>	lue of the Request U ropriate value>; inde	ex=1.1	y; nd device
<any (pixit)<br="" uri=""><uri set="" the="" to="" va<br="">cause=any app DSS1 Parameter values Message flow</uri></any>	lue of the Request U ropriate value>; inde	ex=1.1	-
<any (pixit)<br="" uri=""><uri set="" the="" to="" va<br="">cause=any app DSS1 Parameter values Message flow Test equ</uri></any>	lue of the Request U ropriate value>; inde	ex=1.1	nd device
<any (pixit)<br="" uri=""><uri set="" the="" to="" va<br="">cause=any app DSS1 Parameter values Message flow Test equi INVITE</uri></any>	lue of the Request U ropriate value>; inde ipment →	ex=1.1 En → SETUP	nd device
<any (pixit)<br="" uri=""><uri set="" the="" to="" va<br="">cause=any app DSS1 Parameter values Message flow Test equi INVITE 180 Ringing</uri></any>	lue of the Request U ropriate value>; inde ipment → ←	ex=1.1 ► ► SETUP ← ALERTIN	nd device

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
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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				
Diverting	LegInformation1			
diversi	onReason			
subscr	iptionOption			
diverte	edToNumber			
Message flow				
Test equ	iipment		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	I		
	s in the private network a	183 is sent	
		o header is present, ensure that on re a DivertingLegInformation1 invoke	
Progress) provisional	response is sent. If the 18	3 (Session Progress) contains a Hist	tory-Info header the value is the
		her entry is added and the URI of the	
		er is derived from the diversionReas by is derived from the subscriptionO	
		et to 'noNotification'no History-Info	
(Session Progress).			
SIP header values			
INVITE: History-Info			
•	PIXIT)>index=1		
	1	URI; cause=any appropriate value>	;
index=1	.1		
183: History-Info:			
•	PIXIT)>index=1		
•	the value of the Request	URI in the INVITE;	
cause=a	ny appropriate value>; in	dex=1.1	
<uri deriv<="" td=""><td>ed from the divertedToNu</td><td>umber [Privacy_VA]; cause=reaso</td><td>n_VA>;index=1.1.1</td></uri>	ed from the divertedToNu	umber [Privacy_VA]; cause=reaso	n_VA >;index=1.1.1
DSS1 Parameter valu	ies		
PROGRESS: Facility			
Dive	ertingLegInformation1		
d	liversionReason		
	ubscriptionOption		
	livertedToNumber		
	livertedToNumber		
d Message flow	livertedToNumber st equipment]	End device
d Message flow		J → SETUP	End device
d Message flow Te	st equipment		

Apply post test routine

TSS CDIV	TP_503_241	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 t	he private network a	181 is sent	
containing a Facility Info Forwarded) provisional res same as received in the ini divertedToNumber element Table 7.2.5.4.2-7, the Priva	rmation Element wi ponse is sent. The 18 tial INVITE. A furth t, the cause parame acy header in the en	fo header is present,ensure that on the a DivertingLegInformation 1 in 31 (Being Forwarded) contains a H er entry is added and the URI of the ter is derived from the diversion try is derived from the subscription to 'noNotification' no provisional	nvoke component, a 181 (Being istory-Info header the value is the ne added entry is derived from the nReason element as described in onOption element as described in
SIP header values			
INVITE: History-Info:			
<any (pixi'<="" td="" uri=""><td>Γ)>index=1</td><td></td><td></td></any>	Γ)>index=1		
	value of the Request	URI; cause=any appropriate value>	· ,
index=1.1			
181: History-Info:			
<pre></pre>	T)>index=1		
•	value of the Request V	URI in the INVITE;	
	propriate value>; ind		
<uri derived="" from<="" td=""><td>om the divertedToNu</td><td>mber [Privacy_VA]; cause=reason</td><td>n_VA>;index=1.1.1</td></uri>	om the divertedToNu	mber [Privacy_VA]; cause=reason	n_VA >;index=1.1.1
DSS1 Parameter values			
FACILITY: Facility			
0	gInformation1		
diversion			
subscripti	-		
divertedT	oNumber		
Message flow			
Test eq	uipment		End device
INVITE	→	→ SETUP	
181 (Being Forwarded)	+	← FACILIT	ГҮ
	Α	pply post test routine	

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	TP_503_242	Reference	Selection expression
CDIV		subclause 5.2.5.2.4 of	PICS 5.1.1/2 AND 5.4/5
		[ETSI TS 183 036]	AND 5.1.3/1
Test purpose			
A further call diversion	beyond the private netw	vork	
and a Facility Informa presentationAllowedIn (Ringing) contains the	ation Element is presen dicator set to privacy_V History-Info header as	t containing a DivertingLegInform A as described in Table 7.2.5.4.2-9	receipt of an ALERTING message nation3 invoke component with a 9, a 180 (Ringing) is sent. The 180 ccording the privacy requirements to 'history' or not present.
SIP header values			
INVITE: History-Info:			
<any (f<="" td="" uri=""><td>PIXIT)>index=1</td><td></td><td></td></any>	PIXIT)>index=1		
<uri set="" td="" to<=""><td>the value of the Request</td><td>URI; cause=any appropriate value></td><td>>; index=1.1</td></uri>	the value of the Request	URI; cause=any appropriate value>	>; index=1.1
180: History-Info:	IVIT : day 1		
•	PIXIT)>index=1 d from the divertedToN	umber [Driveou VA]	
	ny appropriate value >; i		
DSS1 Parameter valu	• • • •		
ALERTING: Facility	65		
	rtingLegInformation3		
	resentationAllowedIndic	ator: privacy_VA	
Message flow			
-	st equipment		End device
INVITE	A equipment	→ SETUP	
180 Ringing	, ←	← ALERT	ING
100 Kinging	-	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' r	eceived, rerouteingCounter	r is set to 1 302 is sent	
component, a 302 (Move calledAddress element, th PSTN XML Low layer co	ed Temporarily) final respo ne PSTN XML BearerCapa	onse is sent. The URI of the C bility is derived from the q931 in the q931InfoElemen High lay	n Element CallRerouteing invoke Contact header is derived from the InfoElemen Bearer Capability, the yer compatibility, the User-to-User
SIP header values			
302: Contact: <derived< td=""><td>from the calledAddress></td><td></td><td></td></derived<>	from the calledAddress>		
PSTN XML MIM	E body		
xml version:</td <td>="1.0" encoding="utf-8"?></td> <td></td> <td></td>	="1.0" encoding="utf-8"?>		
PSTN			
BearerCapa	bility derived from the q93	31InfoElemen Bearer Capabilit	у
•	· ·	the q931InfoElemen Low laye	r
compati	•		
HighLayer compati	· ·	the q931InfoElemen High lay	er
User-to-User deriv	ed from the q931InfoElem	en User-user information	
History-Info:			
<derived fr<="" td=""><td>om lastRerouteingNr>index</td><td>x=1</td><td></td></derived>	om lastRerouteingNr>index	x=1	
<derived fr<="" td=""><td>om calledAddress; cause=r</td><td><pre>reason_VA>; index=1.1</pre></td><td></td></derived>	om calledAddress; cause= r	<pre>reason_VA>; index=1.1</pre>	
DSS1 Parameter values			
FACILITY: Facility			
CallRerout	eing invoke		
reroutei	ngReason: reason_VA		
calledA	ddress		
	ngCounter		
-	oElement		
	rer Capability		
	layer compatibility		
-	n layer compatibility		
	r-user information		
	outeingNr	_	
subscrip	otionOption: any appropriat	te value	
Message flow			
Test e	equipment	I	End device
INVITE	→	→ SETUP	
302 (Moved Temporarily) +	← FACILI	ГҮ
ACK	→ →		
		y post test routine	

Table 7.2.5.4.2-10 – Mapping of the CallRerouteing invoke component into added history entry

reason_VA	reason_VA rerouteingReason	
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

TSS 3PTY	TP_505_101	Reference subclause C.14.1.1 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/6
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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.

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
Termin	nating user is connec	eted with the initial party
	C	Flash hook
		Dial tone
		Dial switching order command
INVITE1 (initial party)	÷	
200 OK INVITE1	→	
ACK	÷	
ACK	-	
INVITE2 (additional party)	÷	
180 Ringing2/183 Session Progress2	→	
200 OK INVITE2	→	
	÷	
ACK	× ×	
INVITE3 (initial party)	÷	Three party communication
200 OK INVITE4	→	1 2 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1
	←	
ACK		

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
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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 th	ree 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	→	i no party ominianeation
ACK	€	
	Apply post test ro	utine

TSS	TP_505_103	Reference	Selection expression
3PTY		subclause C.14.2 of	PICS 5.1.1/1 AND 5.3/6
		[ETSI TS 183 043]	AND 5.3/9 AND 5.3/11

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=sendrecy. 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 h	old
Terminating user	is connected with the active pa	rty (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	

T 00	TD 505 104	Deferment	C. L. C. market	
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 reje	ect one of the parties(Loos	se coupling, INVITE method)		
Ensure that when the three ne	ty call is astablished with	the waiting party processing of	f the flesh book event is similar	
			f the flash-hook event is similar user wishes to reject one of the	
parties, the SUT performs the	following actions:			
• Send a BYE request towa				
• Request the media gatew	-			
	onding ephemeral terminat	tion		
SIP header values				
INVITE1 (additional party) SDP				
a=sendonly				
······································				
200 OK 1				
SDP				
a=recvonly				
INIVITE2 (initial party)				
INVITE2 (initial party) SDP				
a=sendrecv				
200 OK 2				
SDP				
a=sendrecv				
Message flow				
Test e	quipment		End device	
	-	oarty is set on hold		
	A three party con	versation is established		
			n hook	
INVITE1 (initial party)	+	Dial	tone	
200 OK INVITE1	→			
ACK	+			
	_	Dial	switching order command	
BYE (additional party)	÷			
200 OK BYE	→			
	F			
INVITE2 (initial party)	÷	Two	party ommunication	
200 OK INVITE2	_			
ACK	+			
Apply post test routine				

TSS	TP_505_105	Reference	Selection expression
ЗРТҮ		subclause C.14.2A of [ETSI TS 183 043]	PICS 5.1.1/1 AND 5.3/6 AND 5.3/9 AND 5.3/12

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 pa	rty is set on hold
Terminating use	r 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	TP_505_106	Reference	Selection expression
3PTY		subclause C.14.2A of	PICS 5.1.1/1 AND 5.3/6
		[ETSI TS 183 043]	AND 5.3/9 AND 5.3/12

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

Mossage fle

Message flow		
Test equipm	nent	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 rou	utine

TSS	TP_505_107	Reference	Selection expression	
3PTY		subclause C.14.2B of [ETSI TS 183 043]	PICS 5.1.1/1 AND 5.3/6 AND 5.3/9 AND 5.3/13	
Test purpose		[======	AND 5.5/7 AND 5.5/15	
	e party conversation (Loose	coupling INVITE request with UI	RI list)	
	r,			
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 cp:copycontrol="to" uri="B?Call ID=1a&From=A%3Btag%3Da&To=B%3Btag%3Db"></entry> </list> </resource>				
Message flow				
Test eq	uipment	End d	evice	
	The initial j	party is set on hold		
Termiı	nating user is connected wi	ith the active party (hold or call	waiting)	
		Flash hook		
INVITE1 (initial party, sen	-	Dial tone		
200 OK INVITE1	→ ←			
ACK	F			
INVITE? (Conference 1 d 1	_{Фе}) +	Dial switching	g order command	
INVITE2 (Conference brid 200 OK INVITE1	ge) ×			
ACK	÷			
	-	Three party co	ommunication	
	Apply p	ost test routine		

TSS 3PTY	TP_505_108	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 served user wishes to re	eject one of the parties (Loo	ose coupling INVITE request with	URI list)
 to the call waiting service. parties, the SUT performs the Send a BYE request to Request the media gate 	If the switching order comm he following actions: wards the held party	n the waiting party, processing of t mand indicates that the served us ation	
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			
-	equipment	E	nd device
	The initial]	party is set on hold	
	A three party con	nversation is established	
		Flash I	nook
	÷		n 0
INVITE1 (initial party)	C	Dial to	lie
INVITE1 (initial party) 200 OK INVITE1	÷	Dial to	ne
		Dial to	ne
200 OK INVITE1	→ ←		vitching order command
200 OK INVITE1	→ ← ←		
200 OK INVITE1 ACK	→ ←		
200 OK INVITE1 ACK BYE (additional party) 200 OK BYE	→ ← ←	Dial sv	vitching order command
200 OK INVITE1 ACK BYE (additional party)	→ ← ← →	Dial sv	
200 OK INVITE1 ACK BYE (additional party) 200 OK BYE INVITE2 (initial party)	→ ← → ←	Dial sv	vitching order command

TSS	TP_505_109	Reference	Selection expression
ЗРТҮ		clause C.14.3 of [ETSI TS 183 043]	PICS 5.1.1/1 AND 5.3/6 AND 5.3/9

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 he	bld
Terminating user	is connected with the active par	
		Flash hook
INVITE1 (initial party, sendonly)	+	
200 OK INVITE1	→ ←	
ACK	₹	
INVITE D2-1	←	
484 Address Incomplete	→	
ACK	(
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	

T CC	TD 505 110		0.1
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
5111		0.14.20 01 [2151 15 105 045]	AND 5.3/9 AND 5.3/13
Test purpose			111(12) 515/711(12) 515/15
	et one of the parties (Lo	oose coupling INVITE request with U	(RI list)
The served user wishes to rejec	i one of the purites (Le	ose coupling in the request with o	n <i>usu</i>
Ensure that when the three-part	y call is established wit	h the waiting party, processing of the	flash-hook event is similar
to the call waiting service. If the	ne switching order com	nmand indicates that the served user	
parties, the SUT performs the f	-		
• Send a BYE request towar			
• Request the media gateway			
 Remove the correspon 	ding ephemeral termin	ation	
SIP header values			
INVITE1 (additional party)			
SDP			
a=sendonly			
200 OK 1			
SDP			
a=recvonly			
a-ree vonity			
INVITE2 (initial party)			
SDP			
a=sendrecv			
200 OK 2			
SDP			
a=sendrecv			
Message flow			
Test equ	uipment	End	device
	The initial	party is set on hold	
	A three party co	onversation is established	
		Flash hoe	ok
INVITE1 (initial party)	+	Dial tone	•
200 OK INVITE1	→		
ACK	←		
		Dial swit	ching order command
BYE (additional party)	+		oning order communa
200 OK BYE	→		
200 OK DTE	2		
INVITEO (::+:-1+)	÷	T	
INVITE2 (initial party)	÷	I wo part	y ommunication
200 OK INVITE2	_		
ACK	÷	post test routine	

7.2.5.5.2 Test purposes for ISDN

TSS	TP_505_201	Reference	Selection expression
3PTY		subclause 5.2.13 of	PICS 5.1.1/2 AND 5.4/6
		[ETSI TS 183 036]	AND 5.6/7

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.

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

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 ← BYE S1 **DISCONNECT 1** ← 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** \leftarrow BYE S2 ← RELEASE COMPLETE → → 200 OK (BYE) Apply post test routine

TSS 3PTY	TP_505_202	Reference 5.2.13 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/6 AND 5.6/7
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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.

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 3Event contains conference; Subscription-State contains active
message/sipfrag contains SIP/2.0 200 OK
application/conference-info+xml contains (S1) connected, dialled-in
Event contains conference; Subscription-State contains active
message/sipfrag contains SIP/2.0 100 Trying

NOTIFY 5 Event contains conference; Subscription-State contains active

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 activ	ve 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	→	\rightarrow 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	→	\rightarrow BYE S2
		← 200 OK (BYE)
	Apply post test	routine

TSS	TP_505_203	Reference	Selection expression
3PTY	11_505_205	subclause 5.2.13 of [ETS	-
		TS 183 036]	AND 5.6/7
Test purpose			
Conference creation by th subscribed	hree-way session creatio	n. REFER request to the Focu	s, Conference notification service not
	a so-called three-way se	ession. The conference notific	e put on hold and wants to join two of ation service is not subscribed. The
	at the conference factory e conference URI in the 2		t with the conference factory URI.
with the Request UR		ly received conference URI an	y session, send two REFER requests d the Refer-To header indicating the
The remote instance disco	onnects the previously he	eld sessions.	
SIP header values INVITE1 SDP a=sendonly			
INVUTEO			
INVITE2 Request Line = confer	rence factory URI		
Contact:; isfocus			
REFER1			
Request URI indicatin	ng the conference URI		
Refer-to header contai	ins URI of remote user 1		
REFER2			
Request URI indicatin	ng the conference URI		
Refer-to header contai	ins URI of remote user 2		
DSS1 Parameter values			
FACILITY			
Begin3PTY-Inv			
Call reference of call	A-B		
Message flow			
Test e	equipment	,	Fest equipment
	Session	n 1 is in active hold state	
	Session	n 2 is in active idle state	
FACILITY	→	→ INVI	ГЕ1
		← 200 C	РК
		→ ACK	
		→ INVI	rF2
		 ✓ INVI ✓ 200 C 	
		← 200 C	Ж

DISCONNECT 2	÷	← BYE S1
RELEASE COMPLETE	→	→ 200 OK (BYE)
		→ REFER2
		← 200 OK REFER
DISCONNECT 2	←	← BYE S2
RELEASE COMPLETE	→	→ 200 OK (BYE)
	Apply po	st test routine

TSS	TP_505_204	Reference	Selection expression
3PTY		subclause 5.2.13 of	PICS 5.1.1/2 AND 5.4/6
		[ETSI TS 183 036]	AND 5.6/7

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 st	ate		
	Session 2 is in active idle st	ate		
FACILITY	\rightarrow \rightarrow	INVITE1		
	÷	200 OK		
	+	ACK		
	→	INVITE2		
	+	200 OK		
	+	ACK		
	+	REFER1		
	÷	200 OK REFER		
RELEASE COMPLETE1	\rightarrow \rightarrow	BYE S1		
	÷	200 OK (BYE)		
	``	REFER2		
	+	200 OK REFER		
RELEASE COMPLETE2	→ →	BYE S2		
	÷	200 OK (BYE)		
	Apply post test routine			

TSS	TP_505_205	Reference	Salastion aumossion	
3PTY	1F_303_203	subclause 5.2.13 of	Selection expression PICS 5.1.1/2 AND	
5111		[ETSI TS 183 036]	5.6/7AND 5.4/8	
Test numpers			5.0/ /AID 5.4/0	
Test purpose	o an ability isin a soufs			
The user equipment has the	e capability join a conjet	rence.		
	" parameter set to INVI" ng the received Referred	ER request that contains a Refer-T TE and contains a Referred-By hea I-By header.		
SIP header values				
REFER: Refer-To=confer	rence URI; method=INV	ITE		
•	mote user equipment U			
INVITE S2: Request URI	-			
Referred-By=Re	mote user equipment U	RI		
DSS1 Parameter values				
Message flow				
Test eq	uipment	En	d device	
A session is in active hold state				
REFER	→			
200 OK REFER	÷			
INVITE S2	÷			
200 OK	→			
ACK	÷			
BYE S1	←	← RELEASE	COMPLETE	
200 OK (BYE)	→			
	-	ly post test routine		

TSS	TP_505_206	Reference	Selection expression
		subclause 5.2.13 of	1100 5.1.1/2 1110
		[ETSI TS 183 036]	5.6/7AND 5.4/8
Test purpose			
The user equipment he	is the capability join a confere	ence.	
URI including the "me the conference URI in	thod" parameter set to INVIT cluding the received Referred	•	
	isconnects the previously held	session.	
SIP header values			
	onference URI; method=INV		
	y=Remote User Equipment Ul URI indicating the received c		
	y=Remote User Equipment U		
DSS1 Parameter valu			
Message flow			
Те	st equipment	End	device
	A session	is in active hold state	
REFER	→		
200 OK REFER	<		
INVITE S2	÷		
200 OK	→		
ACK	÷		
BYE S1	→	➔ DISCONNE	CT
200 OK (BYE)	←	← RELEASE (COMPLETE
	Appl	y post test routine	

TSS	TP_505_207	Reference	Selection expression
3PTY		subclause 5.2.13 of	PICS 5.1.1/2 AND 5.4/6
		[ETSI TS 183 036]	AND 5.6/7 AND 5.3/13

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 fa	ctory URI		
Contact:; isfocus			
Content-Type: application/resour			
Content-Disposition: recipient-lis	st		
xml version="1.0" encoding="</td <td>UTF-8"?></td> <td></td> <td></td>	UTF-8"?>		
<resource-lists xmlns="urn:ietf:p</td><td>arams:xml:ns:resource-l</td><td>lists" xmlns:cp="</td"><td>"urn:ietf:params:xml:ns:copyControl"></td></resource-lists>	"urn:ietf:params:xml:ns:copyControl">		
<list></list>			
<entry cp:c<="" td="" uri="S1 URI"><td>opyControl="to"/></td><td></td><td></td></entry>	opyControl="to"/>		
DSS1 Parameter values			
FACILITY			
Begin3PTY-Inv			
Call reference of call A-B			
Message flow			
Test equipm	ent		Test equipment
	Session 1 is in	active hold sta	ate
	Session 2 is in	n active idle sta	ite
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)
	Annly nos	st test routine	

TSS	TP_505_208	Reference	Selection expression
3PTY		subclause 5.2.13 of	PICS 5.1.1/2 AND 5.4/6
		[ETSI TS 183 036]	AND 5.6/7 AND 5.3/13

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-lis	sts+xml		
Content-Disposition: recipient-list			
xml version="1.0" encoding="UTF-</td <td>-8"?></td> <td></td> <td></td>	-8"?>		
<resource-lists xmlns="urn:ietf:parama</td><td>s:xml:ns:resource-</td><td>lists" xmlns:cp="urn:ietf:params:xml:ns:copyControl"></resource-lists>	>		
<list></list>			
<entry cp:copyc<="" td="" uri="S1 URI"><td>ontrol="to"/></td><td></td><td></td></entry>	ontrol="to"/>		
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 i	n active idle state	
FACILITY	→	→ INVITE1	
		← 200 OK	
		→ ACK	
		→ INVITE2	
		← 200 OK	
		→ ACK	
RELEASE COMPLETE1	→	\rightarrow BYE S1	
		← 200 OK (BYE)	
RELEASE COMPLETE2	→	\rightarrow BYE S2	
		← 200 OK (BYE)	
	Apply po	st test routine	

7.2.5.6 Closed User Group (CUG)

parameter.

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	Test purpose Setup for a CUG call with outgoing access request CUG index present					
invoke component, an SIP cugCallOperation elemen	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					

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	TP 505 002	Reference	Selection expression
CUG	11_000_002	subclasue 5.2.9.1 of	PICS 5.1.1/2 AND 5.4/9
		[ETSI TS 183 036]	
Test purpose			
Setup for a CUG call with	outgoing access requ	est CUG index not present	
invoke component, a SIP cugCallOperation element	INVITE request is solution that contains the that contains the second se	sent. The INVITE request c	rmation Element with CUGCallOperation contains an 'cug' XML MIME body. The element is mapped from the DSS1 arameter is not present.
SIP header values			
INVITE:			
cug			
cugCallOpe	ration		
outgoing	AccessRequest>true<	<	
DSS1 Parameter values			
SETUP:			
Facility			
-	eration invoke		
outgoing	AccessRequest TRU	E	
Message flow			
End	device		Test equipment
SETUP		→ INVITE	
		← 407 Proxy A	Authentication Required
		→ ACK	-
		→ INVITE	
SETUP ACKNOWLEDG	E	← 100 Trying	
	A	Apply post test routine	

TSS TP_50 CUG	05_003		ce se 5.2.9.1 of 'S 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/9
Test purpose				
Setup for a CUG call without outg	oing access	request CUG i	ndex present	
invoke component, a SIP INVITI cugCallOperation element that	E request is contains t	sent. The IN he outgoingA	VITE request cont ccessRequest eler	tion Element with CUGCallOperation ains a 'cug' XML MIME body. The ment is mapped from the DSS1 s mapped from the DSS1 cUGIndex
SIP header values				
INVITE:				
1) cug				
cugCallOperation				
outgoingAccessF	-			
cugIndex>[mapp	ed from cU	GIndex]<		
DSS1 Parameter values				
SETUP:				
Facility CUGCallOperation i	nuoko			
outgoingAccessF		SF		
cUGIndex config	-			
Message flow	, , uu un			
End device			,	Test equipment
SETUP		→	INVITE	
		←		entication Required
		→	ACK	
		→	INVITE	
SETUP ACKNOWLEDGE		÷	100 Trying	
SET OF ACKING WLEDGE		Apply post te		

TSS	TP_505_004	Reference	Selection expression
CUG		clause 5.2.9.1 of	PICS 5.1.1/2 AND 5.4/9
		[ETSI TS 183 036]	

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.

SIP header values

INVITE:

1)

cugCallOperation

cug

outgoingAccessRequest>false<

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 t	est 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 reques	t is rejected	
invoke component, the DS not present. If the AGCF/V	S1 outgoingAccessReq VGW receives a 403 Fo mation Element is pre	uest element is set to 'TRUE' an rbidden final response a DSS1 F sent. The CUGCallOperation	on Element with CUGCallOperation d the DSS1 cUGIndex parameter is RELEASE COMPLETE message is Return Error component is set to
SIP header values			
INVITE:			
Cug			
cugCallOper			
	AccessRequest>true<		
DSS1 Parameter values			
SETUP:			
Facility	· · · · · · · · · · · · · · · · · · ·		
CUGCallOperat	cessRequest TRUE		
outgoingAct	cesskequest IKUE		
RELEASE COMPLETE			
CUGCallOperation Re	turn Error		
inconsistencyInDes	ignatedFacilityAndSub	scriberClass	
Message flow			
End	device	Те	est equipment
SETUP		→ INVITE	
		← 407 Proxy Authen	tication Required
		→ ACK	1
		→ INVITE	
		← 403 Forbidden	
CASE A		+05 1 0101ddell	
		←	
RELEASE COMPLETE		Χ.	

CASE B		
DISCONNECT	\	
RELEASE COMPLETE	→	
	→ ACK	

TSS CUG	TP_505_006		ee e 5.2.9.1 of S 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/9
Test purpose			· - 1	
Setup for a CUG call with	out outgoing access requ	est is rejec	tea	
				nent with CUGCallOperation
				DSS1 cUGIndex parameter is ASE COMPLETE message is
sent and a Facility Inform	mation Element is pres	ent. The		Error component is set to
'inconsistencyInDesignated	lFacilityAndSubscriberC	'lass'		
SIP header values				
INVITE: Cug				
cugCallOper	ation			
	AccessRequest>false<			
DSS1 Parameter values				
SETUP:				
Facility				
	eration invoke			
outgoing.	AccessRequest FALSE			
RELEASE COMPLETE				
CUGCallOperation Ret	turn Error			
inconsistencyInDes	ignatedFacilityAndSubsc	criberClass	l i	
Message flow				
End	device		Test equ	ipment
SETUP		→	INVITE	
		←	407 Proxy Authenticatio	n Required
		→	ACK	
		→	INVITE	
		←	403 Forbidden	
CASE A				
RELEASE COMPLETE		+		
CASE B				
DISCONNECT		←		
RELEASE COMPLETE		→		
		→	ACK	

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 re	eceives indication "a call is w	vaiting"	
	ninating user in a confirmed a he user is informed that a call		uest to another dialogue, the SUT
SIP header values			
INVITE:			
Message flow			
Tes	t equipment	E	End device
INVITE1	→	Ringing	
180 Ringing	÷		
200 OK INVITE	+	Off hook	
ACK	→		
INVITE2	→		
180 Ringing	+	Call wait	ing indication
	Apply	y post test routine	
TSS	TP_507_102	Reference	Selection expression

122	TP_507_102	Reference	Selection expression
CW		clause C.9 of [ETSI TS 183 043]	PICS 5.1.1/1 AND 5.3/14
Test purpose			
	r receives indication "a call is t	waiting" ignored	
			quest to another dialogue, the SUT
sends a 180 Ringin, Here) response.	g. The user ignores the waiting	call, when the answer timer e	xpires the SUT sends a 486 (Busy
SIP header values			
INVITE:			
Message flow			
-	Test equipment	1	End device
INVITE1	Test equipment →	Ringing	
180 Ringing	+	Kinging	
200 OK INVITE	÷	Off hool	c
ACK	``````````````````````````````````````		x .
	-		
INVITE2	→		
180 Ringing	← Star	t T _{no answer} Call wai	ting indication
	Exp	biry T _{no answer}	
486 Busy Here2	(
ACK	→		
	App	ly 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 r	eceives indication "a call is wa	uiting". Terminal based commun	ication waiting (CW) is supported
	rminating user in a confirmed s The user is informed that a call		uest to another dialogue, the SUT
SIP header values			
180			
Alert-Info: <urn:alert:< td=""><td>service:call-waiting></td><td></td><td></td></urn:alert:<>	service:call-waiting>		
Message flow			
Те	est equipment	E	nd device
INVITE	→	Ringing	
180 Ringing	+		
200 OK INVITE	+	Off hook	
ACK	→		
INVITE	→		
180 Ringing	+	Call waiti	ng indication
	Apply	y 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
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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 equipme	nt	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	TP_507_105	Reference	Selection expression
CW		clause C.9 of [ETSI TS 183 043]	PICS 5.1.1/1 AND 5.3/14 AND 5.3/9

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		
200 OK 4 SDP		
a=sendrecv		
Message flow		
Test equipmer		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	←	
ACK	•	
	+	
603 Decline	→	
ACK	7	
INVITE4	+	
200 OKINVITE4	→	
ACK	←	
	Apply post to	est routine

TSS	TP_507_106	Reference	Selection expression
CW		clause C.9 of [ETSI TS 183 043]	PICS 5.1.1/1 AND 5.3/14 AND 5.3/9

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

Wiessage now			
Test equipment		End device	
Terminating user is connec		eted 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	TP_507_107	Reference	Selection expression
CW		clause C.9 of [ETSI TS 183 043]	PICS 5.1.1/1 AND 5.3/14 AND 5.3/10
Test purpose			
The terminating user con	firms the waiting call		
sends a 180 Ringing. The The SOC indicates that the	SUT evaluates the switch	session receives an INVITE require order command (SOC) based on e connected to the waiting party, t	
following actions:			
	-	quest received from the waiting p	party
Request the med	• •		
	e Remote Descriptor of the om the waiting party.	ephemeral termination according	g to the SDP information
The AGCF also sends a r	e-INVITE request on the i	nitial dialogue to hold the associa	ated media stream.
• The Request UR	I is structured as follows:		
- A user part and finish		prefix followed by the switching	order command without the start
the approp		ent information to the S-CSCF to Filter Criteria stored in the user pr	
- A P-Assert command.	ted-Identity containing the	public identity of the subscriber	issuing the switching control
- An SDP of	fer for a voice call.		
SIP header values			
INVITE3			
)@pes.operator.com		
Request-Line: SO (SR SI SDP)@pes.operator.com		
INVITE3 Request-Line: SO (SR SI SDP a=sendonly)@pes.operator.com		
Request-Line: SO (SR SI SDP a=sendonly)@pes.operator.com		
Request-Line: SO (SR SI SDP a=sendonly 200 OK 3)@pes.operator.com		
Request-Line: SO (SR SI SDP a=sendonly 200 OK 3 SDP)@pes.operator.com		
Request-Line: SO (SR SI SDP a=sendonly 200 OK 3 SDP a=recvonly)@pes.operator.com		
Request-Line: SO (SR SI SDP a=sendonly 200 OK 3 SDP a=recvonly Message flow			nd device
Request-Line: SO (SR SI SDP a=sendonly 200 OK 3 SDP a=recvonly Message flow Test o	equipment		nd device
Request-Line: SO (SR SI SDP a=sendonly 200 OK 3 SDP a=recvonly Message flow Test of INVITE1	equipment ->	Ea Ringing	nd device
Request-Line: SO (SR SI SDP a=sendonly 200 OK 3 SDP a=recvonly Message flow Test of INVITE1 180 Ringing	equipment → €	Ringing	nd device
Request-Line: SO (SR SI SDP a=sendonly 200 OK 3 SDP a=recvonly Message flow Test of INVITE1 180 Ringing 200 OK INVITE1	equipment ->		nd device
Request-Line: SO (SR SI SDP a=sendonly 200 OK 3 SDP a=recvonly Message flow	equipment → ← ←	Ringing	nd device
Request-Line: SO (SR SI SDP a=sendonly 200 OK 3 SDP a=recvonly Message flow Test of INVITE1 180 Ringing 200 OK INVITE1 ACK	equipment → ← ←	Ringing	nd device
Request-Line: SO (SR SI SDP a=sendonly 200 OK 3 SDP a=recvonly Message flow Test of INVITE1 180 Ringing 200 OK INVITE1 ACK INVITE2	equipment → ← →	Ringing Off hook	nd device
Request-Line: SO (SR SI SDP a=sendonly 200 OK 3 SDP a=recvonly Message flow Test of INVITE1 180 Ringing 200 OK INVITE1 ACK INVITE2	equipment → ← → →	Ringing Off hook	ng indication
Request-Line: SO (SR SI SDP a=sendonly 200 OK 3 SDP a=recvonly Message flow Test of INVITE1 180 Ringing 200 OK INVITE1 ACK INVITE2	equipment → ← → →	Ringing Off hook Call waiti	ng indication
Request-Line: SO (SR SI SDP a=sendonly 200 OK 3 SDP a=recvonly Message flow Test of INVITE1 180 Ringing 200 OK INVITE1 ACK INVITE2	equipment → ← → →	Ringing Off hook Call waiti Flash hoo Dial tone	ng indication
Request-Line: SO (SR SI SDP a=sendonly 200 OK 3 SDP a=recvonly Message flow Test of INVITE1 180 Ringing 200 OK INVITE1 ACK INVITE2 180 Ringing2	equipment → ← → →	Ringing Off hook Call waiti Flash hoo Dial tone	ng indication k
Request-Line: SO (SR SI SDP a=sendonly 200 OK 3 SDP a=recvonly Message flow Test of INVITE1 180 Ringing 200 OK INVITE1 ACK INVITE2 180 Ringing2	equipment ÷ ÷ ÷ ÷ ÷	Ringing Off hook Call waiti Flash hoo Dial tone	ng indication k
Request-Line: SO (SR SI SDP a=sendonly 200 OK 3 SDP a=recvonly Message flow Test of INVITE1 180 Ringing 200 OK INVITE1	equipment ← ← → ← ←	Ringing Off hook Call waiti Flash hoo Dial tone	ng indication k
Request-Line: SO (SR SI SDP a=sendonly 200 OK 3 SDP a=recvonly Message flow Test of INVITE1 180 Ringing 200 OK INVITE1 ACK INVITE2 180 Ringing2 INVITE3 200 OKINVITE3 200 OKINVITE3 ACK	equipment \div \leftarrow \Rightarrow \leftarrow \div \leftarrow \div	Ringing Off hook Call waiti Flash hoo Dial tone	ng indication k
Request-Line: SO (SR SI SDP a=sendonly 200 OK 3 SDP a=recvonly Message flow Test of INVITE1 180 Ringing 200 OK INVITE1 ACK INVITE2 180 Ringing2 INVITE3 200 OKINVITE3	equipment ← ← ÷ → ← ← ÷	Ringing Off hook Call waiti Flash hoo Dial tone	ng indication k

TSS	TP_507_108	Reference	Selection expression
CW		clause C.9 of [ETSI TS 183 043]	PICS 5.1.1/1 AND 5.3/14 AND 5.3/10

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 e	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	TP_507_109	Reference	Selection expression
CW		clause C.9 of [ETSI TS 183 043]	PICS 5.1.1/1 AND 5.3/14 AND 5.3/10

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.opera	tor.com	
SDP		
a=sendrecv		
200 OK 2		
200 OK 3 SDP		
a=sendrecv		
INVITE4 (waiting party)		
SDP		
a=sendonly		
200 OK 4		
SDP		
a=recvonly		
Message flow		
Test equipment		End device
Termin	nating 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 rout	ine

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 <i>Call waiting indication rec</i>	eived in a 183 Session P	rogress	

Ensure that on receipt of a 183 Session Progress containing an Alert-Info header set to the urn '<urn:alert:service:callwaiting>' 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 tes	st routine

TSS	TP 507 202	Reference	Selection expression
CW	11_507_202	subclause 5.2.11.1 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/10
Test purpose			
Call waiting indi	cation received in a 180 Ringi	ng	
			rn ' <urn:alert:service:call-waiting>', it is present set to 'Call is a waiting</urn:alert:service:call-waiting>
SIP header value	es		
180			
Alert-Info	<pre><urn:alert:service:call-waitin< pre=""></urn:alert:service:call-waitin<></pre>	g>	
DSS1 Parameter	· values		
ALERTING			
Notificatio	on indicator		
Call is	a waiting call		
Message flow			
	End device	Т	est equipment
SETUP		→ INVITE	
		← 407 Proxy Auther	tication Required
		→ ACK	
		→ INVITE	
ALERTING		$\leftarrow 180 \text{ (Ringing)}$	
		ico (iunghig)	

TSS	TP_507_203	Reference	Selection expression
CW		subclause 5.2.11.1 of	PICS 5.1.1/2 AND 5.4/10
		[ETSI TS 183 036]	

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	TP_507_204	Reference	Selection expression
CW		subclause 5.2.11.2 of	PICS 5.1.1/2 AND 5.4/10
		[ETSI TS 183 036]	AND 5.1.3/2
Test purpose			
CW indication received in	the INVITE busy cond	ition met a CW notification is sen	t
			cw' XML element where the busy The Channel identification is set to
Ensure that on receipt of 'urn:alert:service:call-waiti		sage a 180 Ringing is sent cont	aining an Alert-Info header set to
SIP header values			
INVITE			
xml version="1.0</td <td>)"?></td> <td></td> <td></td>)"?>		
<ims-cw xmlns="ur</td><td>rn:3gpp:ns:cw:1.0"></ims-cw>			
<communication< td=""><td>n-waiting-indication/></td><td></td><td></td></communication<>	n-waiting-indication/>		
180			
Alert-Info: <urn:ale< td=""><td>ert:service:call-waiting</td><td>></td><td></td></urn:ale<>	ert:service:call-waiting	>	
DSS1 Parameter values			
SETUP			
Channel identificati	ion		
no channel			
Message flow			
Test ec	quipment		End device
INVITE	→	→ SETUP	
180 Ringing	+	← ALERT	ING
	A	pply post test routine	

			-
TSS	TP_507_205	Reference	Selection expression
CW		subclause 5.2.11.2 of	PICS 5.1.1/2 AND 5.4/10
		[ETSI TS 183 036]	AND 5.1.3/2
Test purpose		-	
CW indication received in	the INVITE busy condition	not met a CW notification is not se	ent
Ensure that on receipt of a	CW indication in an INV	ITE request due to the 'ims-cw' X	ML element where the busy
condition is not met, a SET	UP is sent. The Channel ide	entification is not set to 'no channe	el'.
		180 Ringing is sent containing and	d the Alert-Info header set to
'urn:alert:service:call-waiti	ng' is not present.		
SIP header values			
INVITE			
xml version="1.0</td <td>"?></td> <td></td> <td></td>	"?>		
<ims-cw xmlns="ur</td><td>n:3gpp:ns:cw:1.0"></ims-cw>			
<communication< td=""><td>n-waiting-indication/></td><td></td><td></td></communication<>	n-waiting-indication/>		
DSS1 Parameter values			
Message flow			
Test eq	uipment	End d	levice
INVITE	→	→ SETUP	
180 Ringing	+	← ALERTING	
	Apply (post test routine	
		<u>.</u>	
			1
TSS	TP_507_206	Reference	Selection expression
CW		subclause 5.2.11.2 of	PICS 5.1.1/2 AND 5.4/10
		[ETSI TS 183 036]	AND 5.1.3/2
Test purpose			
	the INVITE busy condition 1	met a CW notification is sent Call	Waiting condition terminated
after expiry of T _{UE-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 Tue-cw		
	Expiry Tue-cw		
480 (Temporarily Unavailable)	←	→	RELEASE COMPLETE
ACK	→		
	Apply post test rou	tine	

TSS	TP_507_207	Reference	Selection expression
CW		subclause 5.2.11.2 of	PICS 5.1.1/2 AND 5.4/10
		[ETSI TS 183 036]	AND 5.1.3/2

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'.

	e		
SIP header values			
180			
Alert-Info: <urn:ale< th=""><th>rt:service:call-waiting></th><th></th><th></th></urn:ale<>	rt:service:call-waiting>		
DSS1 Parameter values			
SETUP			
Channel identification	on		
no channel			
Message flow			
Test eq	luipment	End device	
INVITE	→	→ SETUP	
180 Ringing	+	← ALERTING	
	Apply pos	t 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 nurnose	-		

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:ca< td=""><td>ll-waiting></td><td></td><td></td><td></td></urn:alert:service:ca<>	ll-waiting>			
183				
Alert-Info: <urn:alert:service:ca< td=""><td>ll-waiting></td><td></td><td></td><td></td></urn:alert:service:ca<>	ll-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 equipmen	t		End device	
INVITE	→	→	SETUP	
CASE A				
180 Ringing	+	÷	ALERTING	
CASE B				
183 (Session Progress)	←	←	PROGRESS	
105 (50551011 1 1081055)	× ×	¢.	I KOUKLOD	
CASE C				
183 (Session Progress)	←	÷	NOTIFY	
	Apply p	ost 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				
		e in the confirmed state an INVITE r	equest or UPDATE request is sent.	
A 'a' line in the SDP	is set to 'sendonly' or 'inac	ctive'		
SIP header values				
on neutral values				
INVITE				
INVITE	у			
INVITE SDP	y			

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			
		confirmed state and the ISDN conne a' line in the SDP is set to 'sendrecv'	ection is in the suspended state
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 rou	utine

TSS	TP_508_003	Reference	Selection expression		
TP	11_500_005	subclause 5.2.12.1.1 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/11 AND 5.1.3/2		
			1110 5.1.5/2		
Test purpose					
Suspend initiated T307 exp	iry				
-	•	mer T307 is started. After the ex is sent. A Reason header is present			
SIP header values					
INVITE					
SDP	SDP				
a= sendonly					
or	•				
a=inactive	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 ro	utine

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	SIP header values				
INVITE					
SDP	SDP				
a= sendonly					
DSS1 Parameter v	DSS1 Parameter values				
NOTIFY	NOTIFY				
Notification	Notification indicator				
Notificat	ion 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
	→	АСК
CASE B	+	UPDATE1
	+	200 OK UPDATE
	Apply post test routine	

TSS	TP_508_005	Reference	Selection expression		
TP		subclause 5.2.12.1.2 of	PICS 5.1.1/2 AND 5.4/11		
		[ETSI TS 183 036]	AND 5.1.3/1		
Test purpose					
Resumed initiated j	from the private network				
		e Notification indicator Information			
	ate, the ISDN connection is i e SDP is set to 'sendrecv'	n the suspended state and an INVIT	TE request or UPDATE request is		
SIP header values					
INVITE1/UPDAT	E 1				
SDP					
a=sendo	nly				
INVITE2/UPDAT	E 2				
SDP					
a=sendre	a=sendrecv				
DSS1 Parameter	values				
NOTIFY 1					
Notifica	tion indicator				
Noti	fication description				
U	User suspended				
NOTIFY 2					
Notifica	tion indicator				
Notit	fication description				
User resumed					

End de	evice	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	→	\rightarrow INVITE2
		← 200 OK INVITE
		→ _{ACK}
CASE B		
NOTIFY 1	→	→ UPDATE1
		← 200 OK UPDATE
NOTIFY 2	→	→ UPDATE2
		← 200 OK UPDATE
	Apply pos	t 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	DSS1 Parameter values				
SUSPEND					
Call identity					

Message flow		
Test equipme	nt	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 po	st 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			
	of an INVITE or UPDA ND message is sent to the		te the 'a' line in the SDP is set to
SIP header values			
INVITE3/UPDATE2			
SDP			
a= sendonly			
DSS1 Parameter value	es		
RESUME			
Call identity			
Message flow			
Test	t equipment	E	End device
INVITE1	→	→ SETUP	
180 Ringing	←	← ALERTI	NG
200 OK (INVITE)	←	← CONNE	СТ
ACK	→		
CASE A			
INVITE2	→	→ SUSPEN	-
200 OK (INVITE)	+	← SUSPEN	ID ACKNOWLEDGE
ACK	→		
INVITE3	+	→ RESUM	0
200 OK (INVITE)	→ ←		e ID ACKNOWLEDGE
ACK	← →		D ACKINOW LEDGE
AUN	7		

→	→ SUSPEND
+	← SUSPEND ACKNOWLEDGE
→	→ RESUME
÷	← SUSPEND ACKNOWLEDGE
	€ →

Apply post test routine

TSS TP	TP_508_008	Reference subclause 5.2.12.1 [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			
A V	*			
				e 'a' line in the SDP is set to User suspended' is sent to the
SIP header values				
INVITE2/UPDATE				
SDP				
a= sendonly				
DSS1 Parameter valu	es			
NOTIFY				
Notification				
	tion description			
	suspended			
Message flow				
Test equipment			End d	levice
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	÷	2		
Loo on or brill		pply 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	L		
Resumed initiated from	the private network		
			ate, the 'a' line in the SDP is set to s set to 'User resumed' is sent to the
SIP header values			
INVITE2/UPDATE			
SDP			
a= sendrecv			
DSS1 Parameter value	es		
NOTIFY 2 Notification	indiaatan		
	ion description		
	resumed		
Message flow			
-	t equipment		End device
INVITE1	→	→ SETU	P
180 Ringing	+	← ALER	TING
200 OK (INVITE)	+	← CONN	ECT
ACK	→		
CASE A			
INVITE2	→	→ NOTIF	FY1
200 OK (INVITE)	÷		
ACK	→		
INVITE2	→	→ NOTIF	FY2
200 OK (INVITE)	←		
ACK	→		
CASE B			
UPDATE	→	→ NOTIF	FY1
200 OK UPDATE	+		
UPDATE	→	→ NOTIF	FY2
200 OK UPDATE	+		
		ply 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	L		L
	T using the EctExecute invoke co	mponent in active state	
		l to an additional user is in active st	
		e containing a 'EctExecute' invoke	e component regarding the Cal
	user in held state a REFER is ser ine is set to the address of the hel		
-		the active user, the method parame	ter is set to 'invite', the Replace
	ne session identification values of		
• The Referred	-By header is set to the Public us	er identity of the served user	
ISDN equipment		g 'SIP/2.0 200 OK' a DISCONNEC' the Call Reference of the received	
SIP header valu	es		
REFER: [Conn	-		
		ethod=invite?Replaces=[Call CR 2]
Referr	ed-By: <sip:[public identity<="" td="" user=""><td>]></td><td></td></sip:[public>]>	
NOTIFY 1:			
	ntent-Type: message/sipfrag		
SII	P/2.0 100 Trying		
NOTIFY 2:			
	ntent-Type: message/sipfrag 2/2.0 200 OK		
DSS1 Parameter			
FACILITY:			
CF	81		
Ec	tExecute invoke		
DISCONNECT	1		
Discontinuer	EctExecute return result		
	End device	Test	equipment
		tablished in held state CR 1	
		tablished in active state CR2	
FACILITY	→		CR2, sendonly)
		← 200 OK IN	NVITE
		➔ ACK	
		→ _{REFER}	
		← 202 Accep	oted
		← NOTIFY	l
		→ 200 OK N	
		← NOTIFY 2	2
		- NUTIFI 2	

		➔ 200 OK NOTIFY
		 ← BYE (CR 2) → 200 OK BYE
DISCONNECT 1	÷	 → BYE (CR 1) ← 200 OK BYE
RELEASE	→	
RELEASE COMPLETE	÷	
DISCONNECT 2	+	
RELEASE	→	
RELEASE COMPLETE	+	← 200 OK UPDATE
	Apply p	ost test routine

TSS	TP_509_002	Reference	Selection expression
ECT		subclause 5.2.7.1 of	PICS 5.1.1/2 AND 5.4/12
		[ETSI TS 183 036]	

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 ret		
End devic		Test equipment
		red in held state CR 1
		rting 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)
		$\Rightarrow 200 \text{ OK BYE}$
		$\Rightarrow BYE (CR 1)$
DISCONNECT 1	+	 ← 200 OK BYE
RELEASE	→	
RELEASE COMPLETE	÷	
DISCONNECT 2	÷	
RELEASE	→	
RELEASE COMPLETE	+	← 200 OK UPDATE
	Apply po	st 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
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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 e	established CR 1
A call is	established CR2
FACILITY →	
Apply p	oost test routine

TSS	TP_509_004	Reference	Selection expression		
ECT		subclause 5.2.7.1 of	PICS 5.1.1/2 AND 5.4/12		
		[ETSI TS 183 036]	AND 5.1.3/1		
Test purpose					
Activation of ECT perform	med in a private networl	k using the EctInform invoke comp	ponent set to active, no mapping		
		tt on receipt of a DSS1 FACILI Active, no SIP message is sent and	ΓY message for a Call Reference I no SIP action occurs.		
SIP header values					
DSS1 Parameter values					
FACILITY:					
EctInform	invoke				
Active					
redirect	ionNumber				
En	d device	Tes	st equipment		
	A c	all 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 Subad	dress using the SubaddressTrans	sfer 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 cal	ll is established CR 1	
	A ca	ll is established CR2	
FACILITY	→		
	Apj	ply 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			
	VITE request to a call in held stat	e	
	n is in the held state.		
	ecceipt of an INVITE request, the 1 ment indicating the user is resume	ine in the SDP is set to 'sendrecv', a ed.	a DSS1 NOTIFY message is sent
SIP header valu	ies		
INVITE:			
SDP			
a=	sendrecv		
DSS1 Paramete	r values		
NOTIFY: Notifi	cation description		
Us	ser resumed		

Message flow

Message now				
	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
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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 equip	oment	End device
	A call is establ	lished in held state
UPDATE	→	→ NOTIFY
200 OK UPDATE	←	
	Apply po	st test routine

TSS	TP_509_008	Reference	Selection expression
ECT	TP_309_008	subclause 5.2.7.2 of	PICS 5.1.1/2 AND 5.4/12
201		[ETSI TS 183 036]	
Test purpose	·	·	·
Receipt of an INVIT	E request to a call in active sta	te	
A communication is	in the active state.		
Ensure that on receipt	pt of an INVITE request the SI	OP is not different from the previou	sly received SDP containing a:
P-Asserted-Ident	it.		
 Privacy absent or 	-		
•	n the P-Asserted-Identity		
1	,		
No Message is sent	to the DSS1 user equipment.		
SIP header values			
INVITE:			
	d-Identity		
Isub			
DSS1 Parameter va	alues		
Message flow			
0	Fest equipment	Fn	l device
		established in held state	
INVITE	A call is (totabiloncu ili nelu state	
200 OK INVITE			
ACK		•	
	Арр	ly 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 nurnese			

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	TP_510_001	Reference	Selection expression		
UUS		subclause 5.2.10.1.1 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/13		
Test purpose			L		
Call setup contains a	DSS1 User-user service Inj	formation Element implicit request			
discriminator in the U	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 value	ues				
SETUP					
User-user					
Protoco	Protocol discriminator				
User inf	ormation				
	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	Protocol discriminator			
User information	User information			
ALERTING				
User-user				
Protocol discriminator				
User information				
		Test estimates		
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
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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 discriminate	or	
User information		
PROGRESS		
User-user		
Protocol discriminate	or	
User information		
End device		Test equipment
SETUP	→	→ INVITE
		← 407 Proxy Authentication Required
		→ ACK
		→ INVITE1
ALERTING	÷	← 180 Ringing
PROGRESS	÷	← 183 (Session Progress)
	Apply pos	st test routine

TSS	TP_510_004	Reference	Selection expression
UUS		subclause 5.2.10.1.1 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/13
Test numeros			
Test purpose	DSS1 User user Informatic	on Element implicit request, respons	a in 200 OK
Can setup contains a	DSSI Oser-user injormano	m Element implicit request, respons	e <i>m</i> 200 OK
Ensure that on receipt	of a 200 OK INVITE a PI	ROGRESS is sent. The uuidata para	meter in the User-to-User header
		ocol discriminator in the User-user	
SIP header values			
INVITE			
User-to-User			
Uuidata			
180			
User-to-User			
uuidata			
DSS1 Parameter value	ues		
SETUP			
User-user			
	l discriminator		
User inf	ormation		
CONNECT			
User-user			
Protoco	l discriminator		
User inf	ormation		

	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
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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 po	st 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		L	
	to-user request service 1 ex	plicit request preferred rejected in	CALL PROCEEDING
			lement and a Facility Information
			ent. Upon receipt of a 183 (Session
		et to UserUserService return error	the requested User to User service rejectedBvNetwork
DSS1 Parameter val	1		
SETUP	ues		
User-user			
	l discriminator		
User in	formation		
Facility			
UserUs	erService request invoke		
Ser	vice 1 preferred		
CALL PROCEEDIN	3		
Facility			
UserUs	erService return error		
reje	ctedByNetwork		
	End device	Tes	t equipment
SETUP	→	→ INVITE	
		← 407 Prox	y Authentication Required
		→ ACK	-
		→ INVITE	l
CALL PROCEEDIN	G 🗲	-	sion Progress)
		pply post test routine	5 /

TSS	TP_510_007	Reference	Selection expression
UUS		5.2.10.1.2 of	PICS 5.1.1/2 AND 5.4/13
		[ETSI TS 183 036]	

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 discrim	inator	
User information	1	
Facility		
UserUserService	e request invoke	
Service 1 pre	ferred	
ALERTING		
Facility		
UserUserService	e return error	
rejectedByN	etwork	
End dev	ice	Test equipment
SETUP	→	→ INVITE
		← 407 Proxy Authentication Required
		→ ACK
		→ INVITE
ALERTING	+	← 180 Ringing
	Apply pos	t 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 exp	licit request preferred rejected in C	CONNECT
Element UserUserServi INVITE final response	ce request invoke Service a CONNECT message	taining a user-user Information Ele e 1 preferred an INVITE request is is sent. To reject the requested rvice return error rejectedByNetwo	sent. Upon receipt of a 200 OK User to User service a Facility
DSS1 Parameter value	es		
SETUP			
User-user			
Protocol	discriminator		
User info	rmation		
Facility			
UserUser	Service request invoke		
Servic	ce 1 preferred		
CONNECT			
Facility			
UserUser	Service return error		
rejecte	rejectedByNetwork		

	End device	Test equipment			
SETUP	→	→ INVITE			
		← 407 Proxy Authentication Required			
		→ ACK			
		→ INVITE			
ALERTING	←	← 180 Ringing			
CONNECT	←	← 200 OK INVITE			
		→ ACK			
	App	Apply post test routine			

TSS UUS	TP_510_009	Reference	Selection expression PICS 5.1.1/2 AND 5.4/13
		5.2.10.1.2 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/13
Test purpose	, . .		
Call setup with user-to-use	er request service I ex	plicit request required rejected in DISC	CONNECT
Element UserUserService : User to User service a Fac	request invoke Service ility Information elem	taining an User-user Information Elem e 1 required a DISCONNECT message ent is present set to UserUserService re ested facility not implemented)	is sent. To reject the requested
DSS1 Parameter values			
SETUP			
User-user			
Protocol disc	criminator		
User information	ation		
Facility			
	vice request invoke		
Service 1	required		
DISCONNECT			
Facility			
UserUserSer	vice return error		
rejectedE	3 yNetwork		
Cause			
Value 69 (re	equested facility not in	nplemented)	
End	device	Test eq	uipment
SETUP	→		
DISCONNECT	+		
RELEASE	→		
RELEASE COMPLETE	←		
RELEASE COMILETE	-		

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			
	to-user request service 2 e	explicit request preferred rejected in	CALL PROCEEDING
Service 2 preferred a PROCEEDING mes	n INVITE request is sent.	taining a Facility Information Eleme Upon receipt of a 183 (Session Prog equested User to User service a Fac Network	ress) provisional response a CALI
DSS1 Parameter va	lues		
SETUP			
Facility			
UserU	serService request invoke		
Sei	rvice 2 preferred		
CALL PROCEEDIN	IG		
Facility			
UserU	serService return error		
rej	ectedByNetwork		
	End device	Те	st equipment
SETUP	→	\rightarrow INVITE	
		← 407 Proz	xy Authentication Required
		→ _{ACK}	
		→ INVITE	
CALL PROCEEDIN	IG F	← 183 (Ses	ssion Progress)
		Apply post test routine	

TSS	TP_510_011	Reference	Selection expression
UUS		subclause 5.2.10.1.3 of	PICS 5.1.1/2 AND 5.4/14
		[ETSI TS 183 036]	

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	

T CC	TD 510 010		
TSS UUS	TP_510_012	Reference subclause 5.2.10.1.3 of	Selection expression PICS 5.1.1/2 AND 5.4/14
005		[ETSI TS 183 036]	FICS 5.1.1/2 AND 5.4/14
Test purpose			
	er-to-user request service 2 exp	olicit request preferred rejected in (CONNECT
Service 2 preferred	an INVITE request is sent. Up the requested User to User serv	ining a Facility Information Elemer on receipt of a 200 OK INVITE fina ice a Facility Information element	al response a CONNECT message
DSS1 Parameter	values		
SETUP			
Facility			
User	UserService request invoke		
S	Service 2 preferred		
CONNECT			
Facility			
User	UserService return error		
r	ejectedByNetwork		
	End device	Test	t equipment
SETUP	→	→ INVITE	
		← 407 Proxy	y Authentication Required
		→ _{ACK}	
		→ INVITE	
CONNECT	+	← 200 OK I	NVITE
→ ACK			
	Aj	oply post test routine	

,	TSS	TP_510_013	Reference	Selection expression
1	UUS		subclause 5.2.10.1.3 of	PICS 5.1.1/2 AND 5.4/14
			[ETSI TS 183 036]	

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 re	quest invoke	
Service 2 requir	ed	
DISCONNECT		
Facility		
UserUserService re	turn error	
rejectedByNetw	vork	
Cause		
Value 69 (requeste	d facility not implemented)	
End device		Test equipment
SETUP	→	
DISCONNECT	+	
RELEASE	→	
RELEASE COMPLETE	÷	

TSS	TP_510_014	Reference	Selection expression
UUS		subclause 5.2.10.1.3 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/15
Test purpose			
Call setup with user-to-use	er request service 3 e	xplicit request preferred rejected	in CALL PROCEEDING
Service 3 preferred an INV	TTE request is sent. s sent. To reject the r	Upon receipt of a 183 (Session Pr equested User to User service a F	ment UserUserService request invoke rogress) provisional response a CALL acility Information element is present
DSS1 Parameter values			
SETUP			
Facility			
	rvice request invoke		
Service 3	3 preferred		
CALL PROCEEDING			
Facility			
UserUserSer	vice return error		
rejectedI	ByNetwork		
End	device	,	Test equipment
SETUP	→	→ INVI	ГЕ
		← 407 P	roxy Authentication Required
		→ _{ACK}	
		→ INVI	ГЕ
CALL PROCEEDING	+	-	Session Progress)
	1	Apply post test routine	

		Apply post test routine	
ALERTING	+	← 180 Ringing	
		\rightarrow INVITE	
		\rightarrow ACK	automicution required
			uthentication Required
SETUP	→	→ INVITE	
E	nd device	Test ec	quipment
rejecte	dByNetwork		
•	Service return error		
Facility			
ALERTING			
Service	e 3 preferred		
UserUserS	Service request invoke		
Facility			
SETUP	3		
DSS1 Parameter value			
Service 3 preferred an II	NVITE request is sent. ect the requested User	taining a Facility Information Element U Upon receipt of a 180 (Ringing) provis to User service a Facility Information rk	sional response an ALERTINC
Call setup with user-to-i	iser request service 3 e.	xplicit request preferred rejected in ALI	ERTING
Test purpose			
UUS		5.2.10.1.3 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/15
TSS	TP_510_015	Reference	Selection expression

TSS	TP_510_016	Reference	Selection expression
UUS		5.2.10.1.3 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/15

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	Selection expression PICS 5.1.1/2 AND 5.4/15
005		[ETSI TS 183 036]	PICS 5.1.1/2 AIND 5.4/15
Test purpose			
	to-user request service 3 ex	cplicit request required rejected in A	DISCONNECT
Ensure that on receipt	t of a SETUP message cont	aining a Facility Information Eleme	ent UserUserService request invoke
Service 3 required a I	DISCONNECT message is	sent. To reject the requested user to	o user service a facility information
		eturn error rejectedByNetwork and	the Cause Information Element set
to 69 (requested facili	ity not implemented).		
DSS1 Parameter val	lues		
SETUP			
Facility			
UserUs	serService request invoke		
Ser	vice 3 required		
DISCONNECT			
Facility			
UserUs	serService return error		
reje	ectedByNetwork		
Cause			
Value	69 (requested facility not in	mplemented)	
	End device	Te	st equipment
SETUP	→		
DISCONNECT	+		
RELEASE	→		
RELEASE COMPLE	TF +		

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 contai	ns a user-to-user hea	der 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 usidata 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	TP_510_019	Reference	Selection expression
UUS		subclause 5.2.10.2.1 of	PICS 5.1.1/2 AND 5.4/13
		[ETSI TS 183 036]	

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 equip	oment		End device
INVITE1	→	→	SETUP
180 Ringing	+	÷	ALERTING
200 OK (INVITE)	+	÷	CONNECT
ACK	→		
	Apply pos	t test routine	

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.

is mapped into the utildata para	infecter of the user-to-user	fiedder in the sent BTE request.
SIP header values		
INVITE		
User-to-User		
uuidata		
BYE		
User-to-User		
uuidata		
DSS1 Parameter values		
SETUP		
User-user		
Protocol discrim		
User information	1	
RELEASE		
User-user	• .	
Protocol discrim		
User information	1	
Message flow		
Test equip	ment	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 cont	ains a user-to-user heade	r response in RELEASE COMPI	LETE in confirmed dialogue
message in confirmed of	lialogue where a user-use		receipt of a RELEASE COMPLETE , the Protocol discriminator and user e sent BYE request.
SIP header values	-		-
INVITE			
User-to-User			
uuidata			
BYE			
User-to-User			
uuidata			
DSS1 Parameter value	es		
SETUP			
User-user			
	discriminator		
User info	rmation		
RELEASE			
User-user			
Protocol	discriminator		
User info	rmation		
Message flow			
Tes	t equipment		End device
INVITE1	→	→ SETUL	2
180 Ringing	←	← ALER	TING
200 OK (INVITE)	+	← CONN	ECT
ACK	→		
BYE	+	← RELEA	ASE COMPLETE
200 OK BYE	→		

TSS UUSTP_510_022Reference subclause 5.2.10.2.1 of [ETSI TS 183 036]	Selection expression PICS 5.1.1/2 AND 5.4/13
---	---

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	TP_511_001	Reference	Selection expression
SUB	11_511_001	5.2.8.1:	PICS 5.1.1/2 AND 5.4/16
~		Table: 5.2.8.1-1 of	
		[ETSI TS 183 036]	
Test purpose	·	· ·	· · ·
Mapping of Called po	arty sub-address into IN	/ITE	
			present, an INVITE request is sent and
the To header contain	is an 'isub' parameter set	to the 'Subaddress information' pa	arameter as received in the SETUP.
SIP header values			
INVITE:			
To: <any uri="">; i</any>	sub= Subaddress informa	ation; isub-encoding=nsap-ia5	
DSS1 Parameter val	lues		
SETUP:			
Called par	ty sub-address		
Туре о	f subaddress = NSAP		
Subado	lress information		
	End device		Test equipment
SETUP	→	→ INVI	ITE
		← 4071	Proxy Authentication Required
		→ _{ACK}	-

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 INVI	ΤΕ		
				nt, an INVITE request is sent parameter as received in the
SIP header values				
INVITE:				
From: <any uri="">; isut</any>	= Subaddress informa	tion; isub-encoding=nsa	ap-ia5	
or				
P-Preferred-Identity: <	any URI>; isub= Suba	ddress information; isub	o-encoding=nsa	p-ia5
DSS1 Parameter values				
SETUP:				
Calling party subaddres				
Type of subaddress				
Subaddress informa	ation			
End	device		Test equ	iipment
SETUP	→	→	INVITE	
		+	407 Proxy Au	thentication Required
		→	ACK	
		→	INVITE1	
	A	pply 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 INVI	TE		1

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 th="" u<=""><th>RI>; isub= Subaddress i</th><th>nformation; isub-encoding=nsap-ia5</th></any>	RI>; isub= Subaddress i	nformation; isub-encoding=nsap-ia5
DSS1 Parameter values		
SETUP:		
Calling party subaddress		
Type of subaddress $=$ NS	AP	
Subaddress information		
End devic	e	Test equipment
SETUP	→	→ INVITE
		← 407 Proxy Authentication Required
		→ ACK
		→ INVITE1
	Apply po	ost 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 par	ty sub-address from	200 OK	
CONNECT message is sen	t to the DSS1 user	equipment and a Connected subadd	in the P-Asserted-Identiy header, a ress. The Type of subaddress is set ation is set to isub value received in
SIP header values			
UPDATE:			
P-Asserted-Ident	tity: <any uri="">; isu</any>	b= Subaddress information; isub-en	coding=ToSUB_VA
DSS1 Parameter values			
CONNECT:			
Connected subac			
• •	ddress = NSAP		
Subaddress in	nformation		
End	device	_	est equipment
SETUP	→	→ INVITE	3
			xy Authentication Required
		→ _{ACK}	
		→ INVITE	51
ALERTING	÷	← 180 Rin	ging
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
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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 = ToSUB_V		
Subaddress i	nformation = isub va	llue	
Message flow			
Test ec	Juipment		End device
INVITE1	→	→ SETUP)
	1	Apply post test routine	
TSS	TP_511_006	Reference	Selection expression
SUB		subclause 5.2.8.2 and	PICS 5.1.1/2 AND 5.4/16
		Table 5.2.8.2-1 of [ETSI TS 183 036]	
		[E13113185030]	
Test purpose		UD.	
Mapping of calling party s	ubaddress into SEI C)P	
	ess is present. The T	ype of subaddress is set according	To header, a SETUP message is sent to variable ToSUB_VA indicated in
SIP header values			
INVITE:	tive conv LIDIN in	= isub value; isub-encoding = ToS	
	ury. <arr uki="">, Isut</arr>	= 1sub value, isub-encounig = 103	SUB_VA
DSS1 Parameter values SETUP:			
Calling party su	baddress		
Type of sub	address = ToSUB_V	/A	
Subaddress i	nformation		
Message flow			
Test ec	luipment		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)

TSS	TP_511_007	Reference	Selection expression
SUB		subclause 5.2.8.2 and	PICS 5.1.1/2 AND 5.4/16
		Table 5.2.8.2-1 of	
		[ETSI TS 183 036]	
Test purpose			
Mapping of Connected	d subaddress from CON	INECT	
			ress and the Type of subaddress is set
		parameter is present in the P-Asser onnected subaddress. The isub-end	red-Identity header set to the value of coding parameter is set to nsap-ia5
SIP header values			
200 OK INVITE:			
P-Asserted-	-Identity: <any uri="">; is</any>	sub= isub value; isub-encoding =ns	ap-ia5
DSS1 Parameter value	ues		
CONNECT:			
Connected	subaddress		
Type of	subaddress = NSAP		
Subaddi	ress information		
Message flow			
Te	est equipment		End device
INVITE1	→	→ SETU	JP
180 Ringing	+	← ALE	RTING
200 OK (INVITE)	+	← CON	NECT
ACK	→		
		Apply post test routine	

7.2.5.12 Malicious communication identification

7.2.5.12.1 Test purposes for POTS

TSS MCID	TP_512_101	Reference subclause C.11.4 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/15
Test purpose		[E151 15 165 045]	
	register of details of the last in	ncoming call in a special record	
The user requires the r	egister of details of the tast if	teoming ean in a special record	
		quest after a session was termina contains the special service code of	
SIP header values			
INVITE2:			
Request-Line: Public u	iser identity of SUT@pes.ope	erator.com	
Message flow			
Te	st equipment	En	nd device
INVITE1	→	Ringing	
180 Ringing	+		
200 OK INVITE	+	Off hook	
ACK	→		
CASE A			
BYE	→		
200 OK BYE	+		

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 TP_512_102 MCID	Reference subclause B.4.2.2.2.4 of [ETSI TS 183 043]	Selection expression PICS 5.1.1/1 AND 5.3/15
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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 t	est routine	

MCID subclause B.4.2.2.2 of PI [ETSI TS 183 043]	11C5 5.1.1/1 AI(D 5.5/15
---	--------------------------

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	←	
	l	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

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>
454
          Rec. ITU-T Q.4014.2 (04/2019)
```

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
-------------	------------	--	---

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	+	1			
200 OK INVITE	→				
ACK	+				
INVITE3	÷				
200 OK INVITE	→				
ACK	÷				
	Apply post test routine				

7.2.512.2 Test purposes for ISDN

TSS	TP_512_201	Reference	Selection expression
MCID		5.2.6.2 of	PICS 5.1.1/2 AND 5.4/17
		[ETSI TS 183 036]	AND 5.4/18
Test purpose			
MCID Invocation dur	ing the active state		
		taining a Facility Information Elen ML MIME body is present and th	
SIP header values			
INVITE2:			
MCID XML MIME b	oody		
	.0" encoding="utf-8"?>		
mcid			
request>			
	estIndicator>1 </td <td></td> <td></td>		
HoldingInd	licator>0 </td <td></td> <td></td>		
DSS1 Parameter val	ues		
FACILITY: Facility	,		
McidRe	equest invoke		
Message flow			
Te	est equipment	En	d device
INVITE1	→	→ SETUP	
180 Ringing	+	← ALERTIN	G
200 OK (INVITE)	←	← CONNEC	Г
ACK			
INVITE2	+	← FACILITY	Z
200 OK (INVITE)	→	→ FACILITY	Z
ACK	←		
	Appl	y post test routine	

TSS	TP_512_202	Reference	Selection expression
MCID		subclause 5.2.6.2 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/17 AND NOT 5.4/18
Test purpose		·	
MCID Invocation during th	he active state		
			ement with a McidRequest invoke ne McidRequestIndicator element
SIP header values INVITE2:			
DSS1 Parameter values			
FACILITY: Facility			
McidReques	t invoke		
Message flow			
Test eq	luipment	Er	nd device
INVITE1	→	→ SETUP	
180 Ringing	+	← ALERTIN	IG
200 OK (INVITE)	+	← CONNEC	Т
ACK			
INVITE2	+	← FACILIT	Y
200 OK (INVITE)	→	→ FACILIT	Y
ACK	÷		
	Apply	y post test routine	

7.2.5.13 Message waiting indication (MWI)

7.2.5.13.1 Test purposes for POTS

-					
TSS	TP_513_101	Reference	Selection expression		
MWI		clause C.12.1 of	PICS 5.1.1/1 AND 5.3/19		
		[ETSI TS 183 043]			
Test purpose					
Update of message w	vaiting information				
			licated. On receipt of a NOTIFY		
request reporting the	"message-summary" event, th	e SUT requests the following act	ions from the media gateway:		
• Modify the defau	It dial tone.				
• Send a Message V	Waiting Indicator message using	ng the ITU-T H.248 andisp/data s	ignal.		
SIP header values					
NOTIFY					
Event: message-sum	mary				
Subscription-State: a	ctive				
Content-Type: applic	cation/simple-message-summa	ry			
		-			
MIME body:					
Messages-Waiting: y	ves				
Message-Account: si	p:served_user@Server				
Voice-Message: 4/1	(2/0)				
Video-Message: 3/1	(1/0)				
Fax-Message: 2/1 (0/	Fax-Message: 2/1 (0/1)				

Message flow				
Test equipment		E	End device	
NOTIFY	→			
200 OK (NOTIFY)	+	Display of	f waiting messages	
	A	pply post test routine		
7.2.5.13.2 Test pur	noses for ISDN			
TSS	TP_513_201	Reference	Selection expression	
MWI	11_515_201	clause 5.2.17 of [ETSI TS 183 036]	PICS 5.1.1/2 AND 5.4/19	
Test purpose				
Update of message wa	uiting information			
Modify the defaultSend a facility info	• •	the SUT requests the following acti	ions from the media gateway.	
SIP header values NOTIFY				
Event: message-summ	ary			
Subscription-State: act				
-	tion/simple-message-sumr	mary		
MIME hadre				
MIME body: Messages-Waiting: ye	s			
Message-Account: sip				
Voice-Message: 4/1 (2				
Video-Message: 3/1 (1	1/0)			
Fax-Message: 2/1 (0/1)			
DSS1 Parameter valu	ies			
FACILITY: Facility				
	licate invoke	1		
	ingUserProvidedNr= serve	ed_user		
	cService			
	peech elefaxGroup2-3			
	rideotelephony			
Message flow	1 2			
-	st equipment	E	nd device	
		as subscribed the MWI Service		
NOTIFY	→	→ FACILIT`	Y	
NOTIFY 200 OK (NOTIFY)	→ ←	→ FACILIT	Y	

7.2.5.14 Completion of communications to busy subscriber (CCBS) and completion of communications by no reply (CCNR)

TSS TP_514_101 Reference Selection expression CCBS_CCNR PICS 5.1.1/1 clauses C.18.5 and C.18.6 of [ETSI TS 183 043] **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' SIP header values **INVITE** Request-Line URI <served user>; m=BS or URI <served user>; m=NR Message flow **Test equipment End device** → **INVITE1** Ringing 180 Ringing ← 200 OK INVITE Off hook → ACK Apply post test routine TSS TP_514_102 Reference Selection expression CCBS CCNR subclause C.18.1 of PICS 5.1.1/1 [ETSI TS 183 043] **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 Message flow **End device Test equipment** Off hook → Dial number INVITE ← 407 Proxy Authentication Required → ACK → INVITE 4 **183 Session Progress CCBS** activation in-band ← 486 Busy Here

→

ACK

TSS CCBS_CCNR	TP_514_103	Reference subclause [ETSI TS	C.18.1 of	Selection expression PICS 5.1.1/1
Test purpose				
CCNR invocation				
Session Progress is receiv	ved and in succession	n an announcei	ment with the pron	is possible (in the network) a 183 npt to activate the call completion by receiving a 199 Early Dialogue
SIP header values				
Message flow				
End	device		Те	est equipment
Off hook				
Dial number		→	INVITE	
		÷	407 Proxy Authentication Required	
		→	ACK	
		→	INVITE	
		÷	183 Session Progr	ess
	С	CBS activation	ı in-band	
		←	199 Early Dialogu	

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 w	as receivea			
Ensure that SUT sends a S line contained a 'm' param		equipment after an INVITE reque	st was received and the Request	
SIP header values				
INVITE				
Request-Line URI <serve< td=""><td>d user>; m=BS</td><td></td><td></td></serve<>	d user>; m=BS			
or				
URI < serve	d user>; m=NR			
DSS1 Parameter values				
Message flow				
Test equipment		En	End device	
INVITE	→	→ SETUP		
180 Ringing	+	← ALERTIN	3	
200 OK (INVITE)	+	← CONNECT]	
ACK	→			
	Appl	y 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			
Progress and in succession	n an announcement with	r is busy and CCBS is possible (the prompt to activate the call c ated by receiving a 486 Busy Here	ompletion procedure. After the
SIP header values			
DSS1 Parameter values			
Message flow			
Test ed	quipment	Test e	quipment
SETUP	→	➔ INVITE	
		← 407 Proxy A	Authentication Required
		→ ACK	
		→ INVITE	
PROGRESS	÷	← 183 Session	n Progress
	CCBS	activation in-band	
RELEASE	÷	← 486 Busy H	lere
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			
Session Progress and in su	ccession an announceme	r is not responding and CCNR is p nt with the prompt to activate the c minated by receiving a 199 Early I	all completion procedure. After
SIP header values			
DSS1 Parameter values			
Message flow			
Test ec	quipment	Test e	quipment
SETUP	→	→ INVITE	
		← 407 Proxy A	Authentication Required
		→ ACK	
		→ INVITE	
PROGRESS	+	← 183 Session	Progress
	CCNI	R activation in-band	
RELEASE	+	← 199 Early D	vialogue Terminated
RELEASE COMPLETE	→		

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