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Technical Specification

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

Reference
RTS/TISPAN-06046-1-NGN-R2

Keywords
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Contents

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Foreword

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

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

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

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

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

1 Scope

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

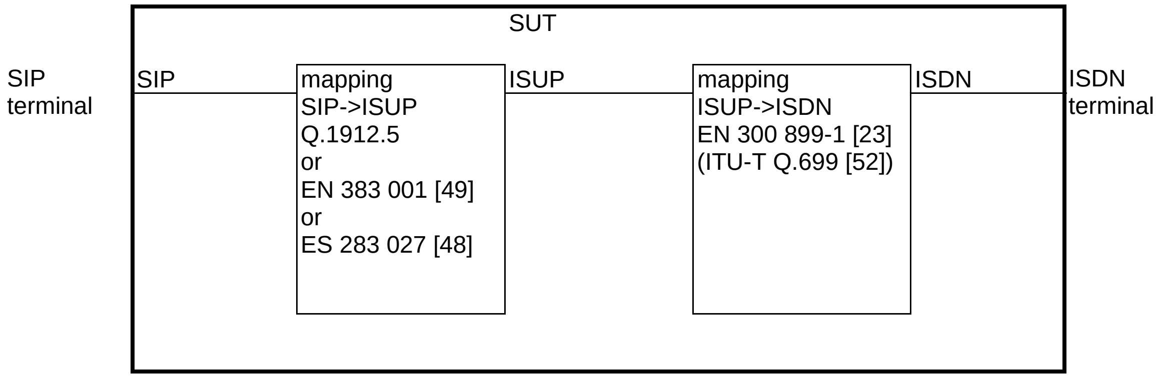

Figure 1: SIP-ISDN inter-working testing architecture

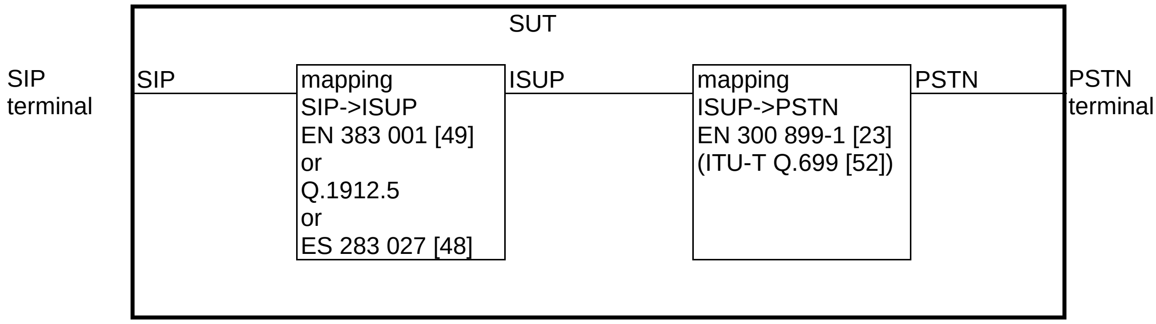

Figure 2a: SIP-PSTN inter-working testing architecture

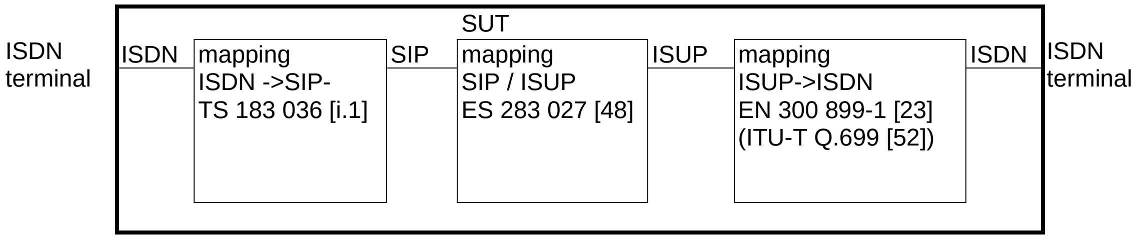

Figure 2b: ISDN-ISDN inter-working testing architecture with ISUP

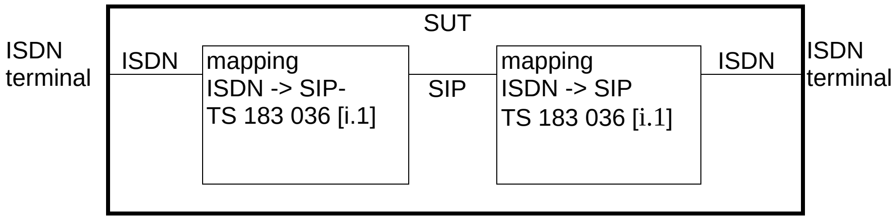

Figure 2c: ISDN-ISDN inter-working testing architecture without ISUP

2 References

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the reference document (including any amendments) applies.

Referenced documents which are not found to be publicly available in the expected location might be found at http://docbox.etsi.org/Reference.

NOTE: While any hyperlinks included in this clause were valid at the time of publication ETSI cannot guarantee their long term validity.

2.1 Normative references

The following referenced documents are necessary for the application of the present document.

[1] Void.

[2] ITU-T Recommendations Q.1902.2 (2001): "Bearer Independent Call Control protocol (Capability Set 2) and Signalling System No.7 ISDN User Part: General functions of messages and parameters".

[3] Void.

[4] Void.

[5] Void.

[6] Void.

[7] Void.

[8] Void.

[9] Void.

[10] Void.

[11] Void.

[12] Void.

[13] Void.

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

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

[16] Void.

[17] Void.

[18] Void.

[19] Void.

[20] Void.

[21] Void.

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

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

[24] Void.

[25] IETF RFC 4566 (2006): "SDP: Session Description Protocol".

[26] IETF RFC 3966 (2004): "The tel URI for Telephone Numbers".

[27] Void.

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

[29] Void.

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

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

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

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

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

[35] Void.

[36] Void.

[37] Void.

[38] Void.

[39] Void.

[40] Void.

[41] Void.

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

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

[44] ETSI TS 183 008: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); PSTN/ISDN simulation services Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR); Protocol specification".

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

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

[47] Void.

[48] ETSI ES 283 027: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN);Endorsement of the SIP-ISUP Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks [3GPP TS 29.163 (Release 7), modified]".

[49] ETSI EN 383 001: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control (BICC) Protocol or ISDN User Part (ISUP) [ITU-T Recommendation Q.1912.5, modified]".

[50] Void.

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

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

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

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

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

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

[57] ETSI EG 201 299-1: "Integrated Services Digital Network (ISDN); Network Integration Testing (NIT); ISDN/PSTN end-to-end testing; Part 1: Test Suite Structure and Test Purposes (TSS&TP) specification".

2.2 Informative references

The following referenced documents are not necessary for the application of the present document but they assist the user with regard to a particular subject area.

[i.1] ETSI TS 183 036: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); ISDN/SIP interworking; Protocol specification".

[i.2] ETSI EG 201 018: "Integrated Services Digital Network (ISDN); Application of the Bearer Capability (BC), High Layer Compatibility (HLC) and Low Layer Compatibility (LLC) information elements by terminals supporting ISDN services".

[i.3] ETSI EN 300 403-1: "Integrated Services Digital Network (ISDN); Digital Subscriber Signalling System No. one (DSS1) protocol; Signalling network layer for circuit-mode basic call control; Part 1: Protocol specification [ITU-T Recommendation Q.931 (1993), modified]".

[i.4] ETSI EN 300 093-1: "Integrated Services Digital Network (ISDN); Calling Line Identification Restriction (CLIR) supplementary service; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".

[i.5] ETSI EN 300 207-1: "Integrated Services Digital Network (ISDN); Diversion supplementary services; Digital Subscriber Signalling System No. One (DSS1); Part 1: Protocol specification".

[i.6] ETSI EN 300 188-1: "Integrated Services Digital Network (ISDN); Three-Party (3PTY) supplementary service; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".

[i.7] ETSI EN 300 141-1: "Integrated Services Digital Network (ISDN); Call Hold (HOLD) supplementary service; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".

[i.8] ETSI EN 300 185-1: "Integrated Services Digital Network (ISDN); Conference call, add-on (CONF) supplementary service; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".

[i.9] ETSI EN 300 196-1: "Integrated Services Digital Network (ISDN); Generic functional protocol for the support of supplementary services; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".

[i.10] ETSI EN 300 138-1: "Integrated Services Digital Network (ISDN); Closed User Group (CUG) supplementary service; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".

[i.11] ETSI TS 124 147: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Conferencing using the IP Multimedia (IM) Core Network (CN) subsystem; Stage 3 (3GPP TS 24.147 version 9.1.0 Release 9)".

[i.12] ETSI EN 300 001: "Attachments to the Public Switched Telephone Network (PSTN); General technical requirements for equipment connected to an analogue subscriber interface in the PSTN".

[i.13] ETSI ETS 300 648: "Public Switched Telephone Network (PSTN); Calling Line Identification Presentation (CLIP) supplementary service; Service description".

[i.14] ETSI EN 300 092-1: "Integrated Services Digital Network (ISDN); Calling Line Identification Presentation (CLIP) supplementary service; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".

[i.15] ETSI EN 300 659: "Access and Terminals (AT); Analogue access to the Public Switched Telephone Network (PSTN); Subscriber line protocol over the local loop for display (and related) services".

[i.16] ETSI TBR 008: "Integrated Services Digital Network (ISDN); Telephony 3,1 kHz teleservice; Attachment requirements for handset terminals".

[i.17] ITU-T Recommendation Q.951: "Stage 3 description for number identification supplementary services using DSS 1".

[i.18] ITU-T Recommendation Q.939: "Typical DSS 1 service indicator codings for ISDN telecommunications services".

[i.19] ETSI TS 183 028: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Common Basic Communication procedures; Protocol specification".

[i.20] ETSI TS 129 163: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (3GPP TS 29.163 version 9.1.0 Release 9)".

[i.21] ISO/IEC 9646 (1994): "Information technology - Open Systems Interconnection -Conformance testing methodology and framework".

3 Definitions and abbreviations

3.1 Definitions

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

For BICC or ISUP specific terminology, reference shall be made to ITU-T Recommendation Q.1902.2 [2]. For SIP and SDP specific terminology, reference shall be made to RFC 3261 [28] and RFC 4566 [25] respectively. Definitions for additional terminology used in this interworking Recommendation are as follows:

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

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

Basic Call Control (BCC): signalling protocol associated with the DSS1 - ISDN Basic Call control procedures of ITU-T recommendation Q.931 [53] (EN 300 403-1 [i.3])

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

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

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

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

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

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

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

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

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

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

NOTE: TSIs test description can be used as the basis for a formal test specification (e.g. Abstract Test Suite in TTCN). See ISO/IEC 9646 [i.21].

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

3.1.1 Conventions for representation of SIP/SDP information

1) All letters of SIP method names are capitalized.

EXAMPLE 1: INVITE, INFO.

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

EXAMPLE 2: To, From, Call-ID.

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

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

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

EXAMPLE 4: sip: URI, tel: URI.

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

EXAMPLE 5: 100 Trying, 484 Address Incomplete.

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

EXAMPLE 6: 200 OK INVITE, 200 OK PRACK.

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

EXAMPLE 7: m=line, a=line.

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

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

3.2 Abbreviations

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

ATS	Abstract tests suite
CFNR	Call forwarding no reply
CFU	Call Forwarding Unconditional
COLR	Called Line Identification Restriction
GW	GateWay
I	Inopportune
ISDN	Integrated Services Digital Network
IUT	Implementation Under Test
NIT	Network integration testing
OIR	Originating Identification presentation Restriction
PER	Packed Encoding Rules
PICS	Protocol Implementation Conformance Statement
PIXIT	Protocol Implementation eXtra Information for Testing
PSTN	Public Switched Telephone Network
QoS	Quality of service
S	Syntactically invalid
SIP	Session Initiation Protocol
TP	Test Purpose
TSS	Test Suite Structure
V	Valid

4 Test Suite Structure (TSS)

4.1 Test Suite Structure (TSS)

4.1.1 ISDN-SIP

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

4.1.2 SIP-ISDN

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

4.1.3 PSTN-SIP

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

4.1.4 SIP-PSTN

C - Plane / U - Plane			
Basic_Call	Successful	3,1 kHz audio	SP_AU_xx
	Unsuccessful		SP_XX_Uxx
Supplementary Services		CLIP	SP_XXSSCLIPxx
		CLIR	SP_XXSSCLIRxx
		CFU	SPS_XXSSCFUxx
			SPP_XXSSCFUxx
		CFB	SP_XXSSCFBxx
			SPP_XXSSCFBxx
		CFNR	SPS_XXSSCFNRxx
			SPP_XXSSCFNRxx

4.1.5 ISDN-ISDN

ISDN-ISDN	Basic_Call (1)	Successful (1)	Speech	1101xx
			UDI	1102xx
			Audio	1103xx
			UDI-TA	1104xx
		Unsuccessful (2)	Speech	1201xx
			UDI	1202xx
			Audio	1203xx
			UDI-TA	1204xx
	Supplementary Services (2)		CLIP	2101xx
			CLIR	2102xx
			COLP	2103xx
			COLR	2104xx
			CUG	2105xx
			SUB	2106xx
			TP	2107xx
			UUS	2108xx
			CONF	2109xx
			CFU	2111xx
			CFB	2112xx
			CFNR	2113xx
			CD	2114xx
			FPH	2115xx
			MCID	2116xx
			3PTY	2117xx
			HOLD	2118xx
			CW	2119xx
			ECT	2120xx
			CCBS	2121xx
			CCNR	2122xx
			Comb	2123xx
			DDI	2124xx
	B-channel (3)	(0)	Speech	3001xx
			UDI	3002xx
			Audio	3003xx
			UDI-TA	3004xx

5 Numbering Scheme

5.1 General description

Pos 1: Network of the A-Subscriber.

Pos. 2: Network of the B-Subscriber.

Pos. 3: Network of the C-Subscriber.

Pos. 4: Network of the D-Subscriber.

Pos. 5: Network of the E-Subscriber.

The following Network Codes apply:

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

(underscore makes it easier to read the name)

P: PSTN

I: ISDN

S: SIP

(Extensions will be added when needed)

Pos. 6 and 7: Bearer- or Teleservice involved

XX: defined per PIXIT value

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

SP: Speech

AU: 3,1 kHz Audio

UD: UDI

UT: UDI/TA

CN: Codec negotiation

DT: DTMF

UP: UPDATE Method

Pos. 8 and 9:

_: No Supplementary Services Involved / Successful

_U: No Supplementary Services Involved / Unsuccessful

SS: Supplementary Services Involved

SI: Supplementary Services interaction

SN: Nonsymmetrical Supplementary Services Involved

ST: Supplementary Services transparent

5.2 Basic Call

Speech	IS_XX_XX

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

5.3 Supplementary Services

CLIP	IS_XXSSCLIP XX

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

6 Test purposes

The registration and application usage procedures in the ATS shall be compliant to RFC 3261 [28] and ES 283 003 [42] (modified 3GPP TS 24.229). The validation of the registration procedure is out of scope of the present document and is part of the Preamble used in the test cases.

The registration conformance tests based on 3GPP TS 24.229 [55**] described in 3GPP TS 34.229-1 [**54**].**

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

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

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

Dial string parameters options

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

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

6.1 ISDN - SIP

6.1.1 Basic Call

6.1.1.1 Test purposes for ISDN-SIP Basic call Successful - Speech or 3,1 kHz audio

Successful Speech or 3,1 kHz audio calls

IS_XX_01	**ISDN reference to: EN 300 403-1 [i.3], clause 5.1.5.1 EN 300 899-1 [23], clause 2.1.1**				**NGN reference to: Q.1912.5 [51], clause 7.1.1 EN 383 001 [49], clause 7.1.1 ES 283 027 [48], clause 7.2.3.2**
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice				
Selection criteria:	Basic_call				
Test purpose:	Ensure that the call establishment using **en-bloc sending** is performed correctly. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters). The call is released from the **calling** user. At the call establishment the SDP parameters in table 1 can be used.				
ISDN Parameter values:	BC=PIXIT, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition				
Comments:					
	ISDN		SUT		SIP
	Case a)				
	SETUP	→		→	INVITE
	CALL PROC	←			
	ALERTING	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	Case b) IMS with 100 rel				
	SETUP	→		→	INVITE
	CALL PROC	←			
	ALERTING	←		←	180 Ringing
				→	PRACK
				←	200 OK
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	Case c) IMS				
	SETUP	→		→	INVITE
	CALL PROC	←			
				←	183 Session Progress
				→	PRACK
				←	200 OK
				→	UPDATE
				←	200 OK
	ALERTING	←		←	180 Ringing
				→	PRACK

				←	200 OK
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE

IS_XX_02	**ISDN reference to:** **EN 300 403-1 [i.3],clause 5.1.5.1** **EN 300 899-1 [23], clause 2.1.1**				**NGN reference to:** **Q.1912.5 [51], clause 7.3.1** **EN 383 001 [49], clause 7.3.1** **ES 283 027 [48], clause 7.2.3.2.5**
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice				
Selection criteria:	Basic_call, SIP Profile A or EN 383 001 [49] Profile B or ES 283 027 [48];				
Test purpose:	Ensure that call establishment using en-bloc sending is performed correctly. Ensure that the ISDN user in the state U3 receives an ALERTING message with the progress indicator information element "call is not end-to-end ISDN (#1)" when the SIP user answers with 180 Ringing message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters). The call is released from the **called** user.				
ISDN Parameter values:	BC=PIXIT, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition				
Comments:					
	ISDN		SUT		SIP
	Case a)				
	SETUP	→		→	INVITE
	CALL PROC	←			
	ALERTING PI #1	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	←		←	BYE
	REL	→		→	200 OK BYE
	Case c)				
	SETUP	→		→	INVITE
	CALL PROC	←			
				←	183 Session Progress
				→	PRACK
				←	200 OK
				→	UPDATE
				←	200 OK
	ALERTING PI #1	←		←	180 Ringing
				→	PRACK
				←	200 OK
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	←		←	BYE
	REL	→		→	200 OK BYE

<table>
<tr><td colspan="2">IS_XX_02A</td><td colspan="2">ISDN reference to:
EN 300 403-1 [i.3],clause 5.1.5.1
EN 300 899-1 [23], clause 2.1.1</td><td colspan="3">NGN reference to:
Q.1912.5 [51], clause 7.3.1
EN 383 001 [49], clause 7.3.1
ES 283 027 [48], clause 7.2.3.2.5</td></tr>
<tr><td>TSS reference:</td><td colspan="6">ISDN-SIP/Basic_call/Successful/Voice</td></tr>
<tr><td>Selection criteria:</td><td colspan="6">Basic_call, ES 283 027 [48]</td></tr>
<tr><td>Test purpose:</td><td colspan="6">Ensure that call establishment using en-bloc sending is performed correctly.
P-Early-Media header not supported, 183 is not interworked sending complete indication received.
Ensure that the ISDN user in the state U3 does not receive a Progress message when the SIP user answers with 183 Session Progress.
Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).
The call is released from the called user.</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="6">BC=PIXIT, no HLC</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="6">Dial string parameters options=PIXIT

TYPE_SDP= PIXIT;

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition</td></tr>
<tr><td>Comments:</td><td colspan="6"></td></tr>
<tr><td rowspan="30"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td colspan="2">SIP</td></tr>
<tr><td>Case a)</td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">183 Session Progress</td></tr>
<tr><td>ALERTING
PI #1</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td colspan="2"></td></tr>
<tr><td>DISC</td><td>←</td><td></td><td>←</td><td colspan="2">BYE</td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td colspan="2">200 OK BYE</td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>Case c)</td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">183 Session Progress</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK</td></tr>
<tr><td>ALERTING PI #1</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td colspan="2"></td></tr>
<tr><td>DISC</td><td>←</td><td></td><td>←</td><td colspan="2">BYE</td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td colspan="2">200 OK BYE</td></tr>
</table>

<table>
<tr><th>IS_XX_02B</th><th colspan="3">ISDN reference to:
EN 300 403-1 [i.3], clause 5.1.5.1
EN 300 899-1 [23], clause 2.1.1</th><th colspan="2">NGN reference to:
ES 283 027 [48], clause 7.2.3.2.5</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/Basic_call/Successful/Voice</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">Basic_call, ES 283 027 [48]; P-Early-Media header supported and inserted by the network</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that call establishment using overlap sending is performed correctly.
P-Early-Media header supported.
Ensure that the ISDN user in the state U2 receive a Call Proceeding message when the SIP user answers with 183 Session Progress and P-Early-Media header supported and inserted by the network.
Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).
The call is released from the called user.</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">BC=PIXIT, no HLC</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

TYPE_SDP= PIXIT;

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="11"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>Case a)</td><td></td><td></td><td></td><td></td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>CALL PROC
PI #8</td><td>←</td><td></td><td>←</td><td>183 Session Progress</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>DISC</td><td>←</td><td></td><td>←</td><td>BYE</td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>200 OK BYE</td></tr>
</table>

<table>
<tr><th>IS_XX_02C</th><th colspan="2">ISDN reference to:
EN 300 403-1 [i.3],clause 5.1.5.1
EN 300 899-1 [23], clause 2.1.1</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 7.3.1
EN 383 001 [49], clause 7.3.1
ES 283 027 [48], clause 7.2.3.2.5</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/Basic_call/Successful/Voice</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">Basic_call, ES 283 027 [48]; P-Early-Media header supported and inserted by the network</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that call establishment using en-bloc sending is performed correctly.
P-Early-Media header supported and inserted by the network.
Ensure that the ISDN user in the state U3 receives a Progress message when the SIP user answers with 183 Session Progress.
Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).
The call is released from the called user.</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">BC=PIXIT, no HLC</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

TYPE_SDP= PIXIT;

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="11"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>Case a)</td><td></td><td></td><td></td><td></td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PROGRESS
PI#8</td><td>←</td><td></td><td>←</td><td>183 Session Progress</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>DISC</td><td>←</td><td></td><td>←</td><td>BYE</td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>200 OK BYE</td></tr>
</table>

IS_XX_03	**ISDN reference to:** **EN 300 403-1 [i.3],clause 5.1.5.1** **EN 300 899-1 [23], clause 2.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 7.3**		
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice				
Selection criteria:	Basic_call; Q.1912.5 [51] Profile B with PI				
Test purpose:	Ensure that call establishment using **en-bloc sending** is performed correctly. Ensure that the ISDN user in the state U3 receives an ALERTING message with the progress indicator information element #1 or #2 or both location "Network beyond Interworking point" when the SIP user answers with 180 Ringing message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters). NOTE: According to Q.699 [52] every message sent to the ISDN access **may** contain 2 Progress indicator information elements.				
ISDN Parameter values:	BC=PIXIT , no HLC PI_VA (PIXIT)				
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition				
Comments:					
	ISDN		SUT		SIP
	Case a)				
	SETUP	→		→	INVITE
	CALL PROC	←			
	ALERTING PI	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	Case c)				
	SETUP	→		→	INVITE
	CALL PROC	←			
				←	183 Session Progress
				→	PRACK
				←	200 OK
				→	UPDATE
				←	200 OK
	ALERTING PI# VA	←		←	180 Ringing
				→	PRACK
				←	200 OK
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE

IS_XX_03A	**ISDN reference to:** **EN 300 403-1 [i.3],clause 5.1.5.1** **EN 300 899-1 [23], clause 2.1.1**	**NGN reference to:** **ES 283 027 [48], clause 7.2.3.2.5**
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice	
Selection criteria:	Basic_call; ES 283 027 [48] *P-Early-Media header not supported*	
Test purpose:	Ensure that call establishment using **en-bloc sending** is performed correctly. Ensure that the ISDN user in the state U3 receives an ALERTING message **including** a progress indicator I.E. with the descriptions "call is not end-to-end ISDN (#1)" or #2 or both, location "Network beyond Interworking point" when the SIP user answers with 180 Ringing message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters). NOTE: According to Q.699 [52] every message sent to the ISDN access **may** contain 2 Progress indicator information elements.	
ISDN Parameter values:	BC=PIXIT , no HLC PI_VA (PIXIT)	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition	
Comments:		

ISDN		SUT		SIP
Case a)				
SETUP	→		→	INVITE
CALL PROC	←			
ALERTING PI	←		←	180 Ringing
CONN	←		←	200 OK INVITE
			→	ACK
		Conversation		
DISC	→		→	BYE
REL	←		←	200 OK BYE
Case c)				
SETUP	→		→	INVITE
CALL PROC	←			
			←	183 Session Progress
			→	PRACK
			←	200 OK
			→	UPDATE
			←	200 OK
ALERTING PI# 1	←		←	180 Ringing
			→	PRACK
			←	200 OK
CONN	←		←	200 OK INVITE
			→	ACK
		Conversation		
DISC	→		→	BYE
REL	←		←	200 OK BYE

IS_XX_03B	**ISDN reference to:** **EN 300 403-1 [i.3],clause 5.1.5.1** **EN 300 899-1 [23], clause 2.1.1**				**NGN reference to:** **Q.1912.5 [51], clause 7.3.**
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice				
Selection criteria:	Basic_call; ES 283 027 [48] *P-Early-Media header supported*				
Test purpose:	Ensure that call establishment using **en-bloc sending** is performed correctly. Ensure that the ISDN user in the state U3 receives an ALERTING message **including** a PI#8 when the SIP user answers with 180 Ringing message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters). NOTE: According to Q.699 [52] every message sent to the ISDN access **may** contain 2 Progress indicator information elements.				
ISDN Parameter values:	BC=PIXIT , no HLC PI_VA (PIXIT)				
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition				
Comments:					
	ISDN		SUT		SIP
	Case a)				
	SETUP	→		→	INVITE
	CALL PROC	←			
	ALERTING PI#8	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	Case c)				
	SETUP	→		→	INVITE
	CALL PROC	←			
				←	183 Session Progress
				→	PRACK
				←	200 OK
				→	UPDATE
				←	200 OK
	ALERTING PI# 1, PI#8	←		←	180 Ringing
				→	PRACK
				←	200 OK
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE

IS_XX_04	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.1.5.1** **EN 300 899-1 [23], clause 2.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 7.3.1** **EN 383 001 [49], clause 7.1.1**		
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice				
Selection criteria:	Basic_call; Q.1912.5 [51] Profile B without PI				
Test purpose:	Ensure that call establishment using **en-bloc sending** is performed correctly. Ensure that the ISDN user in the state U3 receives an ALERTING message **without** the progress indicator information element "call is not end-to-end ISDN (#1)" when the SIP user answers with 180 Ringing message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).				
ISDN Parameter values:	BC=PIXIT, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition				
Comments:					
	ISDN		SUT		SIP
	Case a)				
	SETUP	→		→	INVITE
	CALL PROC	←			
	ALERTING	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	Case c)				
	SETUP	→		→	INVITE
	CALL PROC	←			
				←	183 Session Progress
				→	PRACK
				←	200 OK
				→	UPDATE
				←	200 OK
	ALERTING	←		←	180 Ringing
				→	PRACK
				←	200 OK
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE

IS_XX_05	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.1.5.1** **EN 300 899-1 [23], clause 2.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 7.3** **EN 383 001 [49], clause 7.3**
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice	
Selection criteria:	Basic_call; SIP Profile A or EN 383 001 [49] Profile B or ES 283 027 [48]	
Test purpose:	Ensure that the call establishment using **en-bloc sending** is performed correctly. Ensure that the ISDN user in the state U3 receives a CONNECT message **including** a progress indicator I.E. with the descriptions "call is not end-to-end ISDN (#1)" location "Network beyond Interworking point" when the SIP user answers with a 200 OK message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC=PIXIT, no HLC	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition	
Comments:		

	ISDN		SUT		SIP
	Case a)				
	SETUP	→		→	INVITE
	CALL PROC	←			
	CONN PI# 1	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	Case c)				
	SETUP	→		→	INVITE
	CALL PROC	←			
				←	183 Session Progress
				→	PRACK
				←	200 OK
				→	UPDATE
				←	200 OK
	CONN PI# 2	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE

<table>
<tr><th>IS_XX_06</th><th colspan="3">ISDN reference to:
EN 300 403-1 [i.3], clause 5.1.5.1
EN 300 899-1 [23], clause 2.1.1</th><th colspan="2">NGN reference to:
Q.1912.5 [51], clause 7.3</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/Basic_call/Successful/Voice</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">Basic_call; Q.1912.5 [51] Profile B with PI</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that call establishment using en-bloc sending is performed correctly.
Ensure that the ISDN user in the state U3 receives a CONNECT message with the progress indicator information element #1 or #2 or both location "Network beyond Interworking point" when the SIP user answers with 180 Ringing message.
Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).
NOTE: According to Q.699 [52] every message sent to the ISDN access may contain 2 Progress indicator information elements.</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">BC=PIXIT , no HLC
PI_VA (PIXIT)</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

TYPE_SDP= PIXIT;

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="25"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>Case a)</td><td></td><td></td><td></td><td></td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CONN PI# VA</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>Case c) Supported: 100 rel and precondition</td><td></td><td></td><td></td><td></td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>183 Session Progress</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK</td></tr>
<tr><td>CONN PI# VA</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>

<table>
<tr><th>IS_XX_07</th><th colspan="2">ISDN reference to:
EN 300 403-1 [i.3], clause 5.1.5.1
EN 300 899-1 [23], clause 2.1.1</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 7.3.
EN 383 001 [49], clause 7.3</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/Basic_call/Successful/Voice</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">Basic_call; Q.1912.5 [51] without PI</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the call establishment using en-bloc sending is performed correctly.
Ensure that the ISDN user in the state U3 receives a CONNECT message without a progress indicator I.E. with the descriptions "call is not end-to-end ISDN (#1)" location "Network beyond Interworking point" when the SIP user answers with a 200 OK message.
Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">BC=PIXIT, no HLC</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

TYPE_SDP= PIXIT;

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition</td></tr>
<tr><td rowspan="25">Comments:</td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>Case a)</td><td></td><td></td><td></td><td></td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>Case c)</td><td></td><td></td><td></td><td></td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>183 Session Progress</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>

IS_SP_08	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.1.5.2** **EN 300 899-1 [23], clause 2.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 7.3** **EN 383 001 [49], clause 7.3** **ES 283 027 [48], clause 7.2.3.5.1**
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice	
Selection criteria:	Basic_call; SIP Profile A or EN 383 001 [49] Profile B or ES 283 027 [48]	
Test purpose:	Ensure that call establishment using **overlap sending** is performed correctly. Ensure that the ISDN user in the state U2 receives an ALERTING message **with** a progress indicators "call is not end-to-end ISDN (#1)" location Network beyond Interworking point" when the SIP user answers with a 180 Ringing message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC=speech, no HLC	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition	
Comments:		

ISDN		SUT		SIP
Case a)				
SETUP	→			
SETUP ACK	←			
INFO	→		→	INVITE
ALERTING PI #1	←		←	180 Ringing
CONN	←		←	200 OK INVITE
			→	ACK
		Conversation		
DISC	→		→	BYE
REL	←		←	200 OK BYE
Case c)				
SETUP	→			
SETUP ACK	←			
INFO	→		→	INVITE
			←	183 Session Progress
			→	PRACK
			←	200 OK
			→	UPDATE
			←	200 OK
ALERTING PI #1	←		←	180 Ringing
			→	PRACK
			←	200 OK
CONN	←		←	200 OK INVITE
			→	ACK
		Conversation		
DISC	→		→	BYE
REL	←		←	200 OK BYE

<table>
<tr><th>IS_SP_08A</th><th colspan="3">ISDN reference to:
EN 300 403-1 [i.3], clause 5.1.5.2
EN 300 899-1 [23], clause 2.1.1</th><th colspan="2">NGN reference to:
ES 283 027 [48], clause 7.2.3.5.1</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/Basic_call/Successful/Voice</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">Basic_call; 283 027 overlap receiving supported; In-Dialog Method
SIP PBX</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that call establishment using overlap sending is performed correctly.
Ensure that the ISDN user in the state U2 receives an ALERTING message when the SIP user answers with a 180 Ringing message.
Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">BC=speech, no HLC</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

TYPE_SDP= PIXIT;

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="19"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>Case a)</td><td></td><td></td><td></td><td></td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>SETUP ACK</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>INFO</td><td>→</td><td></td><td>→</td><td>INVITE cseg1</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>183 Session Progress cseg1</td></tr>
<tr><td>INFO</td><td>→</td><td></td><td>→</td><td>INFO cseg2</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK cseg2</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>Call proceeding</td><td>←</td><td></td><td>←</td><td>183 cseg1</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>

<table>
<tr><th>IS_SP_08B</th><th colspan="3">ISDN reference to:
EN 300 403-1 [i.3], clause 5.1.5.2
EN 300 899-1 [23], clause 2.1.1</th><th colspan="2">NGN reference to:
ES 283 027 [48], clause 7.2.3.5.1</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/Basic_call/Successful/Voice</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">Basic_call; ES 283 027 [48] overlap receiving supported; multiple INVITE Overlap Dialling Procedures; one dialog open</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that call establishment using overlap sending is performed correctly.
Ensure that the ISDN user in the state U2 receives an ALERTING message when the SIP user answers with a 180 Ringing message.
Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">BC=speech, no HLC</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

TYPE_SDP= PIXIT;

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="17"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>Case a)</td><td></td><td></td><td></td><td></td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>SETUP ACK</td><td>←</td><td></td><td>←</td><td>484</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>INFO</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>484</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>Call proceeding</td><td>←</td><td></td><td>←</td><td>183 Session Progress</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>

IS_SP_08C	ISDN reference to: EN 300 403-1 [i.3], clause 5.1.5.2 EN 300 899-1 [23], clause 2.1.1	NGN reference to: ES 283 027 [48], clause 7.2.3.5.1
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice	
Selection criteria:	Basic_call; ES 283 027 [48] overlap receiving supported; multiple INVITE Overlap Dialling Procedures; two dialogs open	
Test purpose:	Ensure that call establishment using **overlap sending** is performed correctly. Ensure that the ISDN user in the state U2 receives an ALERTING message when the SIP user answers with a 180 Ringing message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC=speech, no HLC	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition	
Comments:		

	ISDN		SUT		SIP
	Case a)				
	SETUP	→		→	INVITE csq 1
	SETUP ACK	←			
	INFO	→		→	INVITE csq 2
				←	484 csq 1
				→	ACK
	Call proceeding	←		←	183 Session csq 2 Progress
	ALERTING	←		←	180 Ringing csq2
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE

IS_XX_09	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.1.5.2** **EN 300 899-1 [23], clause 2.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 7.3** **EN 383 001 [49], clause 7.3**		
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice				
Selection criteria:	Basic_call;Q.1912.5 [51] Profile B with PI				
Test purpose:	Ensure that call establishment using **overlap sending** is performed correctly. Ensure that the ISDN user in the state U2 receives a ALERTING message with a progress indicators #1 or #2 or both location "Network beyond Interworking point" when the SIP user answers with a 180 Ringing message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters) NOTE: According to Q.699 [52] every message sent to the ISDN access **may** contain 2 Progress indicator information elements.				
ISDN Parameter values:	BC=PIXIT, no HLC PI_VA (PIXIT)				
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition				
Comments:					
	ISDN		SUT		SIP
	Case a)				
	SETUP	→			
	SETUP ACK	←			
	INFO	→		→	INVITE
	ALERTING PI #VA	←		←	180 Ringing
				→	INVITE
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	Case c)				
	SETUP	→			
	SETUP ACK	←			
	INFO	→		→	INVITE
				←	183 Session Progress
				→	PRACK
				←	200 OK
				→	UPDATE
				←	200 OK
	ALERTING PI #VA	←		←	180 Ringing
				→	PRACK
				←	200 OK
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE

IS_XX_10	**ISDN reference to: EN 300 403-1 [i.3], clause 5.1.5.1 EN 300 899-1 [23], clause 2.1.1**				**NGN reference to: Q.1912.5 [51], clause 7.3.1 EN 383 001 [49], clause 7.1.1**
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice				
Selection criteria:	Basic_call; Q.1912.5 [51] Profile B without PI				
Test purpose:	Ensure that call establishment using **overlap sending** is performed correctly. Ensure that the ISDN user in the state U2 receives an ALERTING message **without** the progress indicator information element "call is not end-to-end ISDN (#1)" when the SIP user answers with 180 Ringing message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).				
ISDN Parameter values:	BC=PIXIT, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition				
Comments:					
	ISDN		SUT		SIP
	Case a)				
	SETUP	→			
	SETUP ACK	←			
	INFO	→		→	INVITE
	ALERTING	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	Case c)				
	SETUP	→			
	SETUP ACK	←			
	INFO	→		→	INVITE
				←	183 Session Progress
				→	PRACK
				←	200 OK
				→	UPDATE
				←	200 OK
	ALERTING	←		←	180 Ringing
				→	PRACK
				←	200 OK
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE

IS_XX_11	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.1.5.2** **EN 300 899-1 [23], clause 2.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 7.3** **EN 383 001 [49], clause 7.3** **ES 283 027 [48], clause 7.2.3.5.1**
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice	
Selection criteria:	Basic_call; SIP Profile A or EN 383 001 [49] Profile B or ES 283 027 [48];	
Test purpose:	Ensure that call establishment using **overlap sending** is performed correctly. Ensure that the ISDN user in the state U2 receives a CONNECT message **with** a progress indicators "call is not end-to-end ISDN (#1)" location "Network beyond Interworking point" when the SIP user answers with a 200 OK message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC=PIXIT, no HLC	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition	
Comments:		

ISDN		SUT		SIP
Case a)				
SETUP	→			
SETUP ACK	←			
INFO	→		→	INVITE
CONN PI #1	←		←	200 OK INVITE
			→	ACK
		Conversation		
DISC	→		→	BYE
REL	←		←	200 OK BYE
Case c)				
SETUP	→		→	INVITE
CALL PROC	←			
			←	183 Session Progress
			→	PRACK
			←	200 OK
			→	UPDATE
			←	200 OK
CONN PI #1	←		←	200 OK INVITE
			→	ACK
		Conversation		
DISC	→		→	BYE
REL	←		←	200 OK BYE

IS_XX_12	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.1.5.2** **EN 300 899-1 [23], clause 2.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 7.3** **EN 383 001 [49], clause 7.3**
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice	
Selection criteria:	Basic_call; Q.1912.5 [51] Profile B with PI	
Test purpose:	Ensure that call establishment using **overlap sending** is performed correctly. Ensure that the ISDN user in the state U2 receives a CONNECT message **with** a progress indicators set to PI_VA location "Network beyond Interworking point" when the SIP user answers with a 200 OK message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters)	
ISDN Parameter values:	BC= PIXIT, no HLC PI_VA (PIXIT)	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition	
Comments:		

ISDN		SUT		SIP
Case a)				
SETUP	→			
SETUP ACK	←			
INFO	→		→	INVITE
CONN PI #VA	←		←	200 OK INVITE
			→	ACK
		Conversation		
DISC	→		→	BYE
REL	←		←	200 OK BYE
Case c) Supported: 100 rel and precondition				
SETUP	→			
SETUP ACK	←			
INFO			→	INVITE
			←	183 Session Progress
			→	PRACK
			←	200 OK
			→	UPDATE
			←	200 OK
CONN PI #VA	←		←	200 OK INVITE
			→	ACK
		Conversation		
DISC	→		→	BYE
REL	←		←	200 OK BYE

Values for test purposes IS_XX_12	
VA	PI information element
VA_1	# 1 (call is not end-to-end ISDN)
VA_2	# 2 (destination address is non-ISDN)
VA_3	PI # 1 and PI # 2

<table>
<tr><td>IS_XX_13</td><td>ISDN reference to:
EN 300 403-1 [i.3], clause 5.1.5.2
EN 300 899-1 [23], clause 2.1.1</td><td colspan="3">NGN reference to:
Q.1912.5 [51], clause 7.3
EN 383 001 [49], clause 7.3
ES 283 027 [48], clause 7.2.3.5.1</td></tr>
<tr><td>TSS reference:</td><td colspan="4">ISDN-SIP/Basic_call/Successful/Voice</td></tr>
<tr><td>Selection criteria:</td><td colspan="4">Basic_call; Q.1912.5 [51] Profile B without PI</td></tr>
<tr><td>Test purpose:</td><td colspan="4">Ensure that call establishment using overlap sending is performed correctly.
Ensure that the ISDN user in the state U3 receives an ALERTING message without a progress indicators "call is not end-to-end ISDN (#1)" location Network beyond Interworking point" when the SIP user answers with a 180 Ringing message .
Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="4">BC=PIXIT, no HLC</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="4">Dial string parameters options=PIXIT

TYPE_SDP= PIXIT;

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition</td></tr>
<tr><td>Comments:</td><td colspan="4"></td></tr>
</table>

ISDN		SUT		SIP
Case a)				
SETUP	→			
SETUP ACK	←			
INFO	→		→	INVITE
ALERTING	←		←	180 Ringing
CONN	←		←	200 OK INVITE
			→	ACK
		Conversation		
DISC	→		→	BYE
REL	←		←	200 OK BYE
Case c)				
SETUP	→			
SETUP ACK	←			
INFO	→		→	INVITE
			←	183 Session Progress
			→	PRACK
			←	200 OK
			→	UPDATE
			←	200 OK
ALERTING	←		←	180 Ringing
			→	PRACK
			←	200 OK
CONN	←		←	200 OK INVITE
			→	ACK
		Conversation		
DISC	→		→	BYE
ISDN	←		←	200 OK BYE

IS_XX_14	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.3.3** **EN 300 899-1 [23], clause 2.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 7.7** **EN 383 001 [49], clause 7.7** **ES 283 027 [48], clause 7.2.3.2.13**
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice/	
Selection criteria:	Basic_call	
Test purpose:	Ensure that the call establishment and the call clearing procedure is performed correctly when the **calling user** clears after answering with a DISCONNECT message indicating the Cause value # 16 "normal call clearing". The called user shall receive a BYE message. According to ES 283 027 [48] and EN 383 001 [49] the Reason Header field shall be included with Cause Value #16. Ensure that in the Call Delivered call state U4 the transfer of tone or announcement on the B- channel is performed correctly.	
ISDN Parameter values:	BC=PIXIT, no HLC	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition	
Comments:		

ISDN		SUT		SIP
Case a)				
SETUP	→		→	INVITE
CALL PROC	←			
ALERTING	←		←	180 Ringing
CONN	←		←	200 OK INVITE
			→	ACK
		Conversation		
DISC	→		→	BYE
REL	←		←	200 OK BYE
Case c)				
SETUP	→		→	INVITE
CALL PROC	←			
			←	183 Session Progress
			→	PRACK
			←	200 OK
			→	UPDATE
			←	200 OK
ALERTING	←		←	180 Ringing
			→	PRACK
			←	200 OK
CONN	←		←	200 OK INVITE
			→	ACK
		Conversation		
DISC	→		→	BYE
ISDN	←		←	200 OK BYE

<table>
<tr><td>IS_XX_15</td><td colspan="2">ISDN reference to:
EN 300 403-1 [i.3], clause 5.3.3
EN 300 899-1 [23], clause 2.1.1</td><td colspan="3">NGN reference to:
Q.1912.5 [51], clause 7.7
EN 383 001 [49], clause 7.7
ES 283 027 [48], clause 7.2.3.2.13</td></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/Basic_call/Successful/Voice/</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">Basic_call</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the call clearing procedure is performed correctly when the called user clears after answering with a BYE message.
The calling user shall receive a DISCONNECT message with the Cause value # 16 "normal call clearing".
Ensure that in the Call Delivered call state U4 and disconnect indication state (N12) the transfer of tone or announcement on the B- channel is performed correctly.</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">BC=PIXIT, no HLC</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

TYPE_SDP= PIXIT;

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="30"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>Case a)</td><td></td><td></td><td></td><td></td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>DISC</td><td>←</td><td></td><td>←</td><td>BYE</td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>200 OK BYE</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>Case c) Supported: 100 rel and precondition</td><td></td><td></td><td></td><td></td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>183 Session Progress</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>DISC</td><td>←</td><td></td><td>←</td><td>BYE</td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>200 OK BYE</td></tr>
</table>

<table>
<tr><td>IS_SP_16</td><td colspan="3">ISDN reference to:
EN 300 403-1 [i.3], clause 4.5.16
EN 300 899-1 [23], clause 2.1.1
TBR 008 [i.16], clause 5.1.3,
EG 201 018 [i.2], clause 6.3.1</td><td colspan="2">NGN reference to:
Q.1912.5 [51], clause 7.3
EN 383 001 [49], clause 7.3
ES 283 027 [48], clause 7.2.3.2.2.2</td></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/Basic_call/Successful/Voice</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">Basic_call;</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that call establishment supporting the telephony 3,1 kHz teleservice is performed correctly.
Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">BC=speech, HLC=telephony</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

TYPE_SDP= PIXIT;

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td></td><td>Case a)</td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td></td><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td></td><td>DISC</td><td>←</td><td></td><td>←</td><td>BYE</td></tr>
<tr><td></td><td>REL</td><td>→</td><td></td><td>→</td><td>200 OK BYE</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>Case c) Supported: 100 rel and precondition</td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td>←</td><td>183 Session Progress</td></tr>
<tr><td></td><td></td><td></td><td></td><td>→</td><td>PRACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td>←</td><td>200 OK</td></tr>
<tr><td></td><td></td><td></td><td></td><td>→</td><td>UPDATE</td></tr>
<tr><td></td><td></td><td></td><td></td><td>←</td><td>200 OK</td></tr>
<tr><td></td><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td></td><td></td><td></td><td></td><td>→</td><td>PRACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td>←</td><td>200 OK</td></tr>
<tr><td></td><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td></td><td>DISC</td><td>←</td><td></td><td>←</td><td>BYE</td></tr>
<tr><td></td><td>REL</td><td>→</td><td></td><td>→</td><td>200 OK BYE</td></tr>
</table>

Table 1: PIXIT Values for test purposes IS_XX_01 to IS_XX_16

Variable	m= line			b= line	a= line
	<media>	<transport>	<fmt-list>	<modifier>:<bandwidth-value>	rtpmap:<dynamic-PT> <encoding name>/<clock rate> [/encoding parameters>
VA_01	audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)
VA_02	audio	RTP/AVP	Dynamic PT (and possibly a second Dynamic PT)	AS:64	rtpmap:<dynamic-PT> PCMU/8000 (and possibly rtpmap:<dynamic-PT> PCMA/8000)
VA_03	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000
VA_04	audio	RTP/AVP	Dynamic PT	AS:64	rtpmap:<dynamic-PT> PCMA/8000
VA_05	audio	RTP/AVP	0 and/or 8	AS:64	rtpmap:0 PCMU/8000 and/or rtpmap:8 PCMA/8000
VA_06	audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)
VA_07	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000

IS_AU_17	**ISDN reference to: EN 300 403-1 [i.3], clause 4.5.17 EN 300 899-1 [23], clause 2.1.1**			**NGN reference to: Q.1912.5 [51], clause 7.1**	
TSSreference:	ISDN-SIP/Basic_call/Successful/Voice				
Selection criteria:	Basic_call; Telefax G3 terminals; Telefax G3 terminals with T.38				
Test purpose:	Support of Telefax G3. Ensure that in the active call state (N10) the Fax transfer on the media and B-channels is performed correctly.				
ISDN Parameter values:	BC=3,1 kHz audio, HLC = Facsimile G2/G3				
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line Based on T.38. b = line AS: 64 m = line: udptl; T38				
Comments:					
	ISDN		SUT		SIP
	Case a)				
	SETUP	→			
				→	INVITE
	ALERTING	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	←		←	BYE
	REL	→		→	200 OK BYE
	Case c)				
	SETUP	→			
	SETUP ACK	←			
	INFO			→	INVITE
				←	183 Session Progress
				→	PRACK
				←	200 OK
				→	UPDATE
				←	200 OK
	ALERTING	←		←	180 Ringing
				→	PRACK
				←	200 OK
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	←		←	BYE
	REL	→		→	200 OK BYE

<table>
<tr><th>IS_AU_18</th><th colspan="3">ISDN reference to:
EN 300 403-1 [i.3], clause 4.5.17
EN 300 899-1 [23], clause 2.1.1</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 7.1
EN 383 001 [49], clause 7.1
ES 283 027 [48], clause 7.2.3.2</th></tr>
<tr><td>TSSreference:</td><td colspan="6">ISDN-SIP/Basic_call/Successful/Voice</td></tr>
<tr><td>Selection criteria:</td><td colspan="6">Basic_call; Telefax G3 terminals-inband</td></tr>
<tr><td>Test purpose:</td><td colspan="6">Support of Telefax G3.
Ensure that in the active call state (N10) the Fax transfer on the media and B-channels is performed correctly and the echo cancellers in the GW are not activated. No transcoding in the gateway takes place.</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="6">BC=3,1 kHz audio, HLC = Facsimile G2/G3</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="6">Dial string parameters options=PIXIT

TYPE_SDP= PIXIT;

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line RTP/AVP
b = line 64 kbit/s
m = line: 8</td></tr>
<tr><td>Comments:</td><td colspan="6"></td></tr>
<tr><td></td><td>ISDN</td><td></td><td>SUT</td><td></td><td colspan="2">SIP</td></tr>
<tr><td rowspan="30"></td><td>Case a)</td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">INVITE</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td colspan="2"></td></tr>
<tr><td>DISC</td><td>←</td><td></td><td>←</td><td colspan="2">BYE</td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td colspan="2">200 OK BYE</td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>Case c)</td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>SETUP ACK</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>INFO</td><td></td><td></td><td>→</td><td colspan="2">INVITE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">183 Session Progress</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td colspan="2"></td></tr>
<tr><td>DISC</td><td>←</td><td></td><td>←</td><td colspan="2">BYE</td></tr>
<tr><td></td><td>→</td><td></td><td>→</td><td colspan="2">200 OK BYE</td></tr>
</table>

IS_AU_19	**ISDN reference to: EN 300 403-1 [i.3], clause 4.5.5 EN 300 899-1 [23], clause 2.1.1**		**NGN reference to: Q.1912.5 [51], clause 7.1 ES 283 027 [48], clause 7.2.3.2**		
TSSreference:	ISDN-SIP/Basic_call/Successful/Voice				
Selection criteria:	Basic_call; Bearer service 3,1 kHz audio				
Test purpose:	Ensure that the ISDN SETUP with the BC parameter value information transfer capability 3,1 kHz audio, voice band data via modem, synchronous/ asynchronous mode is set to MODE is mapped to the SIP INVITE Message. Ensure that in the active call state (N10) the data transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters) and the echo cancellers in the GW are not activated.				
ISDN Parameter values:	BC= 3,1 kHz audio, voice band data via modem, synchronous/ asynchronous mode: MODE user rate: USER_RATE (table 2)				
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	Case a)				
	SETUP	→			
				→	INVITE
	ALERTING	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	←		←	BYE
	REL	→		→	200 OK BYE
	Case c)				
	SETUP	→			
				→	INVITE
				←	183 Session Progress
				→	PRACK
				←	200 OK
				→	UPDATE
				←	200 OK
	ALERTING	←		←	180 Ringing
				→	PRACK
				←	200 OK
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	←		←	BYE
	REL	→		→	200 OK BYE

IS_AU_20	**ISDN reference to:** **EN 300 403-1 [i.3], clause 4.5.18** **EN 300 899-1 [23], clause 2.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 7.1** **EN 383 001 [49], clause 7.1** **ES 283 027 [48], clause 7.2.3.2**		
TSSreference:	ISDN-SIP/Basic_call/Successful/Voice				
Selection criteria:	Basic_call; Bearer service 3,1 kHz audio				
Test purpose:	Ensure that the ISDN SETUP with the BC parameter value information transfer capability 3,1 kHz audio and the LLC ISDN Parameter values: 3,1 kHz audio, voice band data via modem, synchronous/ asynchronous mode is set to MODE, user rate set to USER_RATE is mapped to the SIP INVITE Message. Ensure that in the active call state (N10) the data transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters) and the echo cancellers in the GW are not activated.				
ISDN Parameter values:	BC = 3,1 kHz audio, LLC = 3,1 kHz audio, voice band data via modem, synchronous/ asynchronous mode: MODE user rate: USER_RATE (table 2)				
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	Case a)				
	SETUP	→			
				→	INVITE
	ALERTING	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	←		←	BYE
	REL	→		→	200 OK BYE
	Case c)				
	SETUP	→			
				→	INVITE
				←	183 Session Progress
				→	PRACK
				←	200 OK
				→	UPDATE
				←	200 OK
	ALERTING	←		←	180 Ringing
				→	PRACK
				←	200 OK
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	←		←	BYE
	REL	→		→	200 OK BYE

IS_AU_21	**ISDN reference to:** **EN 300 403-1 [i.3], clause 4.5.18** **EN 300 899-1 [23], clause 2.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 7.1** **EN 383 001 [49], clause 7.1** **ES 283 027 [48], clause 7.2.3.2**
TSSreference:	ISDN-SIP/Basic_call/Successful/Voice	
Selection criteria:	Basic_call; Bearer service 3,1 kHz audio	
SIP selection criteria:	Audio	
Test purpose:	Ensure that the ISDN SETUP with the BC parameter value information transfer capability 3,1 kHz audio voice band data via modem, synchronous/ asynchronous mode is set to MODE, user rate set to USER_RATE and the LLC ISDN Parameter values: 3,1 kHz audio, voice band data via modem, synchronous/ asynchronous mode is set to MODE, user rate set to USER_RATE is mapped to the SIP INVITE. In the active call state (N10) ensure that the data transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters) and the echo cancellers in the GW are not activated.	
ISDN Parameter values:	BC=LLC=3,1 kHz audio, voice band data via modem, synchronous/ asynchronous mode: MODE user rate: USER_RATE (table 2)	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		SUT		SIP
Case a)				
SETUP	→			
			→	INVITE
ALERTING	←		←	180 Ringing
CONN	←		←	200 OK INVITE
			→	ACK
		Conversation		
DISC	←		←	BYE
REL	→		→	200 OK BYE
Case c)				
SETUP	→			
			→	INVITE
			←	183 Session Progress
			→	PRACK
			←	200 OK
			→	UPDATE
			←	200 OK
ALERTING	←		←	180 Ringing
			→	PRACK
			←	200 OK
CONN	←		←	200 OK INVITE
			→	ACK
		Conversation		
DISC	←		←	BYE
REL	→		→	200 OK BYE

Table 2: Values for test purposes IS_AU_19 to IS_AU_21

VA_01	MODE: synchronous USER_RATE: 1,2 kbit/s
VA_02	MODE: synchronous USER_RATE: 2,4 kbit/s
VA_03	MODE: synchronous USER_RATE: 3,6 kbit/s
VA_04	MODE: synchronous USER_RATE: 4,8 kbit/s
VA_05	MODE: synchronous USER_RATE: 7,2 kbit/s
VA_06	MODE: synchronous USER_RATE: 8 kbit/s
VA_07	MODE: synchronous USER_RATE: 9,6 kbit/s
VA_08	MODE: synchronous USER_RATE: 14,4 kbit/s
VA_09	MODE: synchronous USER_RATE: 16 kbit/s
VA_10	MODE: synchronous USER_RATE: 19,2 kbit/s
VA_11	MODE: synchronous USER_RATE: 32 kbit/s
VA_12	MODE: synchronous USER_RATE: 48 kbit/s
VA_13	MODE: synchronous USER_RATE: 56,0 kbit/s
VA_14	MODE: synchronous USER_RATE: 64 kbit/s
VA_15	MODE: asynchronous USER_RATE: 1,2 kbit/s
VA_16	MODE: asynchronous USER_RATE: 2,4 kbit/s
VA_17	MODE: asynchronous USER_RATE: 3,6 kbit/s
VA_18	MODE: asynchronous USER_RATE: 4,8 kbit/s
VA_19	MODE: asynchronous USER_RATE: 7,2 kbit/s
VA_20	MODE: asynchronous USER_RATE: 8 kbit/s
VA_21	MODE: asynchronous USER_RATE: 9,6 kbit/s
VA_22	MODE: asynchronous USER_RATE: 14,4 kbit/s
VA_23	MODE: synchronous USER_RATE: 16 kbit/s
VA_24	MODE: asynchronous USER_RATE: 19,2 kbit/s
VA_25	MODE: asynchronous USER_RATE: 32 kbit/s
VA_26	MODE: asynchronous USER_RATE: 48 kbit/s
VA_27	MODE: asynchronous USER_RATE: 56 kbit/s
VA_28	MODE: asynchronous USER_RATE: 64 kbit/s

Table 3: PIXIT Values for the test purpose IS_AU_19 to IS_AU_21

m= line				b= line	a= line
VA	**<media>**	**<transport>**	**<fmt-list>**	**<modifier>:<bandwidth-value> (see note)**	**rtpmap:<dynamic-PT> <encoding name>/ <clock rate>[/encoding parameters>**
VA_01	Audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A
VA_02	Audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMU/8000
VA_03	Audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A
VA_04	Audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMA/8000
VA_05	Image	Udptl	t38	N/A or up to 64 kbit/s	Based on T.38
VA_06	Image	Tcptl	t38	N/A or up to 64 kbit/s	Based on T.38
NOTE: <bandwidth value> for <modifier> of AS is evaluated to be B kbit/s.					

IS_XX_22	**ISDN reference to: EN 300 403-1 [i.3], clause 4.5.18 EN 300 899-1 [23], clause 2.1.1**	**NGN reference to: ES 283 027 [48] clause A.1.4**
TSSreference:	ISDN-SIP/Basic_call/Successful/Voice	
Selection criteria:	Basic_call; announcement towards a PSTN/ISDN Providing announcements to a user during the establishment of a communication session	
SIP selection criteria:	Audio	
Test purpose:	Ensure that an announcement towards a PSTN/ISDN can be provided. During the establishment an AS in the IP network provides an announcement e.g. "The communication is forwarded" or "The user is not reachable". The announcement should be received after the CALL PROCEEDINGmessage with PI#8.	
ISDN Parameter values:	BC= PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

IS_AU_23	**ISDN reference to:** **EN 300 403-1 [i.3], clause 4.5.17** **EN 300 899-1 [23], clause 2.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 7.1** **EN 383 001 [49], clause 7.1** **ES 283 027 [48], clause 7.2.3.2**
TSSreference:	ISDN-SIP/Basic_call/Successful/Voice	
Selection criteria:	Basic_call; Telefax G3 terminals-inband	
Test purpose:	Support of Telefax G3. Ensure that in the active call state (N10) the Fax transfer on the media and B-channels is performed correctly and the echo cancellers in the GW are not activated. In the active call state **the callee** sends a re-INVITE with the T38.	
ISDN Parameter values:	BC=3,1 kHz audio, HLC = Facsimile G2/G3	
SIP Parameter values:	Dial string parameters options=PIXIT a = line RTP/AVP b = line 64 kbit/s m = line: 8 re-INVITE a = line Based on T.38. b = line AS: 64 m = line: udptl; T38	
Comments:		

ISDN		SUT		SIP
Case a)				
SETUP	→		→	INVITE
ALERTING	←		←	180 Ringing
CONN	←		←	200 OK INVITE
			→	ACK
		Conversation		
			←	INVITE
			→	200 OK INVITE
			←	ACK
DISC	←		←	BYE
REL	→		→	200 OK BYE
Case c) Supported: 100 rel and precondition				
SETUP	→			
SETUP ACK	←			
INFO			→	INVITE
			←	183 Session Progress
			→	PRACK
			←	200 OK
			→	UPDATE
			←	200 OK
ALERTING	←		←	180 Ringing
			→	PRACK
			←	200 OK
CONN	←		←	200 OK INVITE
			→	ACK
			←	INVITE
			→	200 OK INVITE
			←	ACK
DISC	←		←	BYE
REL	→		→	200 OK BYE

6.1.1.2 Codec negotiation

IS_CN_01	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.1.5.1** **EN 300 899-1 [23], clause 2.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 7.1.1** **EN 383 001 [49], clause 7.1.1** **ES 283 027 [48], clause 7.2.3.2.1** **RFC 3264 [30]**
TSS reference:	ISDN-SIP/Basic_call/Codec negotiation	
Selection criteria:	Basic_call	
Test purpose:	Ensure that the call establishment is performed correctly. The answer related to the SDP offer is contained in the 183 Session Progress message Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC=speech, no HLC	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		SUT		SIP
A) SDP pre-condition not requested				
SETUP	→		→	INVITE offer1
CALL PROC	←		←	183 Session Progress answer 1
ALERTING	←		←	180 Ringing
CONNECT	←		←	200 OK INVITE
	→		→	ACK
DISC	←		←	BYE
REL	→		→	200 OK BYE
b) pre-condition and 100 rel				
SETUP	→		→	INVITE offer 1
CALL PROC	←			
			←	183 Session Progress answer 1
			→	PRACK
			←	200 OK PRACK
			→	UPDATE
			←	200 OK
ALERTING	←		←	180 Ringing
			→	PRACK
			←	200 OK PRACK
CONNECT	←		←	200 OK INVITE
DISC	→		→	ACK
REL	←		←	BYE
	→		→	200 OK BYE

<table>
<tr><td>IS_CN_02</td><td colspan="3">ISDN reference to:
EN 300 403-1 [i.3], clause 5.1.5.1
EN 300 899-1 [23], clause 2.1.1</td><td colspan="3">NGN reference to:
Q.1912.5 [51], clause 7.1.1
EN 383 001 [49], clause 7.1.1
ES 283 027 [48], clause 7.2.3.2.1
RFC 3264 [30]</td></tr>
<tr><td>TSS reference:</td><td colspan="6">ISDN-SIP/Basic_call/Codec negotiation</td></tr>
<tr><td>Selection criteria:</td><td colspan="6">Basic_call</td></tr>
<tr><td>Test purpose:</td><td colspan="6">Ensure that the call establishment is performed correctly.
The answer related to the SDP offer is contained in the 180 Ringing message
Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="6">BC=speech, no HLC</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="6">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="6"></td></tr>
<tr><td rowspan="22"></td><td>ISDN</td><td colspan="3">SUT</td><td colspan="2">SIP</td></tr>
<tr><td colspan="6">a) SDP pre-condition not requested
ISDN 1 UE 1</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE offer1</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing answer 1</td></tr>
<tr><td>CONNECT</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE</td></tr>
<tr><td></td><td>→</td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td>DISC</td><td>←</td><td></td><td>←</td><td colspan="2">BYE</td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td colspan="2">200 OK BYE</td></tr>
<tr><td colspan="6">b) pre-condition and 100 rel</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE offer1</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">183 Session Progress</td></tr>
<tr><td></td><td>→</td><td></td><td>→</td><td colspan="2">PRACK</td></tr>
<tr><td></td><td>←</td><td></td><td>←</td><td colspan="2">200 OK PRACK</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing Answer 1</td></tr>
<tr><td></td><td>→</td><td></td><td>→</td><td colspan="2">PRACK</td></tr>
<tr><td></td><td>←</td><td></td><td>←</td><td colspan="2">200 OK PRACK</td></tr>
<tr><td>CONNECT</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE</td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td colspan="2">BYE</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td colspan="2">200 OK BYE</td></tr>
</table>

IS_CN_03	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.1.5.1** **EN 300 899-1 [23], clause 2.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 7.1.1** **EN 383 001 [49], clause 7.1.1** **ES 283 027 [48], clause 7.2.3.2.1** **RFC 3264 [30]**		
TSS reference:	ISDN-SIP/Basic_call/Codec negotiation				
Selection criteria:	Basic_call				
Test purpose:	Ensure that the call establishment is performed correctly. The answer related to the SDP offer is contained in the 200 OK INVITE message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).				
ISDN Parameter values:	BC=speech, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	a) SDP pre-condition not requested				
	SETUP	→		→	INVITE offer1
	CALL PROC	←			
	ALERTING	←		←	180 Ringing
	CONNECT	←		←	200 OK INVITE answer 1
		→		→	ACK
	DISC	←		←	BYE
	REL	→		→	200 OK BYE
	b) pre-condition and 100 rel				
	SETUP	→		→	INVITE offer 1
	CALL PROC	←			
				←	183 Session Progress
				→	PRACK
				←	200 OK PRACK
	ALERTING	←		←	180 Ringing
				→	PRACK
				←	200 OK PRACK
	CONNECT	←		←	200 OK INVITE answer 1
	DISC	→		→	ACK
	REL	←		←	BYE
		→		→	200 OK BYE

IS_CN_04	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.1.5.1** **EN 300 899-1 [23], clause 2.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 7.1.1** **EN 383 001 [49], clause 7.1.1** **ES 283 027 [48], clause 7.2.3.2.1** **RFC 3264 [30]**
TSS reference:	ISDN-SIP/Basic_call/Codec negotiation	
Selection criteria:	Basic_call	
Test purpose:	Ensure that the call establishment is performed correctly. Ensure that answer related to the SDP offer is contained in the 183 Session Progress message. A new offer (codec) is sent in the 180 Ringing. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC=speech, no HLC	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	ISDN		SUT		SIP
	a) SDP pre-condition not requested				
	SETUP	→		→	INVITE **Offer 1**
	CALL PROC	←			
				←	183 Session Progress **Answer 1**
	ALERTING	←		←	180 Ringing offer 2
	CONNECT	←		←	200 OK INVITE
				→	ACK answer 2
	DISC	←		←	BYE
	REL	→		→	200 OK BYE
	b) pre-condition and 100 rel				
	Option a)				
	SETUP	→		→	INVITE **Offer 1**
	CALL PROC	←			
				←	183 Session Progress **Answer 1**
				→	PRACK
				←	200 OK PRACK
	ALERTING	←		←	180 Ringing **offer 2**
				→	PRACK
				←	200 OK PRACK
	CONNECT	←		←	200 OK INVITE
		→		→	ACK **answer 2**
	DISC	←		←	BYE
	REL	→		→	200 OK BYE
	Option c)				
	SETUP	→		→	INVITE **Offer 1**
	CALL PROC	←			
				←	183 Session Progress **Answer 1**
				→	PRACK
				←	200 OK PRACK
	ALERTING	←		←	180 Ringing **offer 2**
				→	PRACK
				→	UPDATE **answer 2**
				←	200 OK
				←	200 OK PRACK
	CONNECT	←		←	200 OK INVITE
		→		→	ACK
	DISC	←		←	BYE
	REL	→		→	200 OK BYE

<table>
<tr><th>IS_CN_05</th><th colspan="4">ISDN reference to:
EN 300 403-1 [i.3], clause 5.1.5.1
EN 300 899-1 [23], clause 2.1.1</th><th colspan="2">NGN reference to:
Q.1912.5 [51], clause 7.1.1
EN 383 001 [49], clause 7.1.1
ES 283 027 [48], clause 7.2.3.2.1
RFC 3264 [30]</th></tr>
<tr><td>TSS reference:</td><td colspan="6">ISDN-SIP/Basic_call/Codec negotiation</td></tr>
<tr><td>Selection criteria:</td><td colspan="6">Basic_call; RE-INVITE</td></tr>
<tr><td>Test purpose:</td><td colspan="6">During the session, the called user decides to change the characteristics of the media session. This is accomplished by sending a re-INVITE containing a new media description. This re-INVITE references the existing dialog so that the other party knows that it is to modify an existing session instead of establishing a new session. The other party sends a 200 (OK) to accept the change. The requestor responds to the 200 (OK) with an ACK.</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="6">BC=speech, no HLC</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="6">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="6"></td></tr>
<tr><td rowspan="33"></td><td>ISDN</td><td colspan="3">SUT</td><td colspan="2">SIP</td></tr>
<tr><td colspan="6">a) SDP pre-condition not requested</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE Offer 1</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">183 Session Progress</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing</td></tr>
<tr><td>CONNECT</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">RE-INVITE offer 2</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">200 OK answer 2</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">ACK</td></tr>
<tr><td>DISC</td><td>←</td><td></td><td>←</td><td colspan="2">BYE</td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td colspan="2">200 OK BYE</td></tr>
<tr><td colspan="6">b) TS 124 229 [55] / ES 283 003 [42] (pre-condition and 100 rel)</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td colspan="2">INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">183 Session Progress</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK PRACK</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK</td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td colspan="2">180 Ringing</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">200 OK PRACK</td></tr>
<tr><td>CONNECT</td><td>←</td><td></td><td>←</td><td colspan="2">200 OK INVITE</td></tr>
<tr><td></td><td>→</td><td></td><td>→</td><td colspan="2">ACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">RE-INVITE offer 2</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td colspan="2">200 OK answer 2</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td colspan="2">ACK</td></tr>
<tr><td>DISC</td><td>←</td><td></td><td>←</td><td colspan="2">BYE</td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td colspan="2">200 OK BYE</td></tr>
</table>

6.1.1.3 Test purposes for ISDN-SIP Basic call Successful - UPDATE

IS_XX_UP_01			**NGN reference to: EN 383 001 [49], clause 7.1.1 ES 283 027 [48], clause 7.2.3.2.1 RFC 3311 [31], clause 5.1**		
TSS reference:	ISDN -SIP/Basic_call/Successful				
Selection criteria:	UPDATE procedure for the callee after INVITE transaction				
Test purpose:	Ensure that the callee can send UPDATE after completion of the initial INVITE transaction.				
ISDN Parameter values:	BC=PIXIT, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	SETUP	→		→	INVITE Offer 1
	CALL PROC	←			
				←	183 Session Progress answer 1
				→	PRACK
				←	200 OK PRACK
	ALERTING	←		←	180 Ringing
	CONNECT	←		←	200 OK INVITE
				→	ACK
			Conversation		
				←	UPDATE offer
				→	200 OK UPDATE
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE

IS_XX_UP_02					**NGN reference to: EN 383 001 [49], clause 7.1.1 ES 283 027 [48], clause 7.2.3.2.1 RFC 3311 [31], clause 5.1**
TSS reference:	ISDN - SIP/Basic_call/Successful				
Selection criteria:	Subsequent UPDATE is rejected if a pending offer (PRACK) is not answered				
Test purpose:	Ensure that a subsequent UPDATE following an UPDATE is rejected (500 Server Internal Error) as long as the pending offer (2) is not answered.				
ISDN Parameter values:	BC=speech, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	SETUP	→		→	INVITE with Offer 1
	CALL PROC	←			
				←	183 Session Progress answer 1
				→	PRACK
				←	200 OK PRACK
				←	UPDATE offer 2
				←	UPDATE offer 3
				→	500 Server Internal Error
				→	200 OK UPDATE answer 2
	ALERTING	←		←	180 Ringing
	CONNECT	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE

IS_XX_UP_03					**NGN reference to:** **EN 383 001 [49], clause 7.1.1** **ES 283 027 [48], clause 7.2.3.2.1** **RFC 3311 [31], clause 5.1**
TSS reference:	ISDN - SIP/Basic_call/Successful				
Selection criteria:	Received UPDATE is rejected if a pending offer (INVITE) is not answered				
Test purpose:	Ensure that a subsequent UPDATE following an INVITE with SDP offer is rejected (500 Server Internal Error) as long as the first offer is not answered.				
ISDN Parameter values:	BC=speech, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	SETUP	→		→	INVITE offer 1
	CALL PROC	←			
				←	UPDATE offer 2
				→	500 Server Internal Error
				←	183 Session Progress answer 1
				→	PRACK
				←	200 OK PRACK
	ALERTING	←		←	180 Ringing
	CONNECT	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE

IS_XX_UP_04		**NGN reference to:** **EN 383 001 [49], clause 7.1.1** **ES 283 027 [48], clause 7.2.3.2.1** **RFC 3311 [31], clause 5.1**		
TSS reference:	ISDN - SIP/Basic_call/Successful			
Selection criteria:	Execution of UPDATE procedure, SDP version identifier exists			
Test purpose:	Ensure that an UPDATE procedure is executed (e.g. bandwidth parameter changed), if the SDP version identifier is different.			
ISDN Parameter values:	BC=speech, no HLC			
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)			
Comments:				

	ISDN		SUT		SIP
	SETUP	→		→	INVITE with offer 1
					with SDP version identifier
	CALL PROC	←			
				←	183 Session Progress answer 1
				→	PRACK
				←	200 OK (PRACK)
				←	UPDATE with offer 2
					different SDP version identifier
				→	200 OK UPDATE answer 2
			Session parameter changed		
	ALERTING	←		←	180 Ringing
	CONNECT	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE

IS_XX_UP_05		**NGN reference to:** **EN 383 001 [49], clause 7.1.1** **ES 283 027 [48], clause 7.2.3.2.1** **RFC 3311 [31], clause 5.1**
TSS reference:	ISDN - SIP/Basic_call/Successful	
Selection criteria:	Execution of UPDATE procedure, SDP version identifier exists	
Test purpose:	Ensure that an UPDATE procedure is executed but bandwidth parameters are not changed, if the SDP version identifier and SDP content have not changed.	
ISDN Parameter values:	BC=speech, no HLC	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		SUT		SIP
SETUP	→		→	INVITE with offer 1
				with SDP version identifier
CALL PROC	←			
			←	183 Session Progress answer 1
			→	PRACK
			←	200 OK (PRACK)
			←	UPDATE with offer 2
				identical SDP as offer 1
			→	200 OK UPDATE answer 2
		Session parameter are not changed		
ALERTING	←		←	180 Ringing
CONNECT	←		←	200 OK INVITE
			→	ACK
		Conversation		
DISC	→		→	BYE
REL	←		←	200 OK BYE

6.1.1.4 Test purposes for ISDN-SIP Basic call Successful - DTMF Tests

IS_XX_DT_01	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.1.5.1** **EN 300 899-1 [23], clause 2.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 7.1.1** **EN 383 001 [49], clause 7.1.1** **ES 283 027 [48], clause 7.2.3.2.1**		
TSS reference:	ISDN-SIP/Basic_call/Successful/DTMF				
Selection criteria:	Basic call; DTMF -Inband				
Test purpose:	Ensure that the call establishment using **en-bloc sending** is performed correctly. Ensure that in the active call state (N10) the DTMF Digits (events 0 through 15) can be transmitted inband to the called user.				
ISDN Parameter values:	BC=speech, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	Case a)				
	SETUP	→		→	INVITE with offer 1
	CALL PROC	←			
	ALERTING	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	←		←	BYE
	REL	→		→	200 OK BYE
	Case c) Supported: 100 rel and precondition				
	SETUP	→		→	INVITE
	CALL PROC	←			
				←	183 Session Progress
				→	PRACK
				←	200 OK
				→	UPDATE
				←	200 OK
	ALERTING	←		←	180 Ringing
				→	PRACK
				←	200 OK
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	←		←	BYE
	REL	→		→	200 OK BYE

Values of codecs for test purposes IS_XX_DT_01					
VARIABLE	PT	Encoding	media type	clock rate	channels
VA_01	0	PCMU	A	8,000	1
VA_02	3	GSM	A	8,000	1
VA_03	8	PCMA	A	8,000	1

IS_XX_DT_02	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.1.5.1** **EN 300 899-1 [23], clause 2.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 7.1.1** **EN 383 001 [49], clause 7.1.1** **ES 283 027 [48], clause 7.2.3.2.1**		
TSS reference:	ISDN - SIP/Basic_call/Successful/DTMF				
Selection criteria:	Basic call; DTMF-RFC 2833 [56]				
Test purpose:	Ensure that the call establishment using **en-bloc sending** is performed correctly. Ensure that in the active call state (N10) the DTMF Digits (events 0 through 15) can be transmitted as payload for DTMF Digits (RFC 2833 [56]) to the called user.				
ISDN Parameter values:	BC=speech, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	Case a)				
	SETUP	→		→	INVITE with offer 1
	CALL PROC	←			
	ALERTING	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	←		←	BYE
	REL	→		→	200 OK BYE
	Case c) Supported: 100 rel and precondition				
	SETUP	→		→	INVITE
	CALL PROC	←			
				←	183 Session Progress
				→	PRACK
				←	200 OK
				→	UPDATE
				←	200 OK
	ALERTING	←		←	180 Ringing
				→	PRACK
				←	200 OK
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	←		←	BYE
	REL	→		→	200 OK BYE

6.1.1.5 Test purposes for ISDN-SIP Basic call Successful -UDI

Successful
UDI

IS_UD_01	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.1.5.1** **EN 300 899-1 [23], clause 2.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 7.1** **EN 383 001 [49], clause 7.1** **ES 283 027 [48], clause 7.2.3.2**
TSS reference:	ISDN - SIP/Basic_call/Successful/UDI	
Selection criteria:	Basic call; UDI	
Test purpose:	Ensure that the call establishment using **en-bloc sending** is performed correctly. Ensure that the mapping of the SETUP parameters and the INVITE message parameters is performed correctly. Ensure that in the active call state (N10) the data transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC= UDI, no HLC	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line : rtpmap:<dynamic-PT> CLEARMODE/8000 b = line AS: 64 m = RTP/AVP	
Comments:		

ISDN		SUT		SIP
Case a)				
SETUP	→		→	INVITE
CALL PROC	←			
ALERTING	←		←	180 Ringing
CONN	←		←	200 OK INVITE
			→	ACK
		Conversation		
DISC	→		→	BYE
REL	←		←	200 OK BYE
Case c)				
SETUP	→		→	INVITE
CALL PROC	←			
			←	183 Session Progress
			→	PRACK
			←	200 OK
			→	UPDATE
			←	200 OK
ALERTING	←		←	180 Ringing
			→	PRACK
			←	200 OK
CONN	←		←	200 OK INVITE
			→	ACK
		Conversation		
DISC	→		→	BYE
REL	←		←	200 OK BYE

IS_UD_02	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.1.5.1** **EN 300 899-1 [23], clause 2.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 7.3** **EN 383 001 [49], clause 7.1** **ES 283 027 [48], clause 7.2.3.2**
TSS reference:	ISDN-SIP/Basic_call/Successful/UDI	
Selection criteria:	Basic call; UDI; Q.1912.5 [51] Profile A	
Test purpose:	Ensure that call establishment using **en-bloc sending** is performed correctly. Ensure that the ISDN user in the state U3 receives an ALERTING message **with** the progress indicator information element "call is not end-to-end ISDN (#1)" location "Network beyond Interworking point" when the SIP user answers with 180 Ringing message. Ensure that in the active call state (N10) the data transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC= UDI, no HLC	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line : rtpmap:<dynamic-PT> CLEARMODE/8000 b = line AS: 64 m = RTP/AVP	
Comments:		

ISDN		SUT		SIP
Case a)				
SETUP	→		→	INVITE
CALL PROC	←			
ALERTING PI #1	←		←	180 Ringing
CONN	←		←	200 OK INVITE
			→	ACK
		Conversation		
DISC	→		→	BYE
REL	←		←	200 OK BYE
Case c)				
SETUP	→		→	INVITE
CALL PROC	←			
			←	183 Session Progress
			→	PRACK
			←	200 OK
			→	UPDATE
			←	200 OK
ALERTING PI #1	←		←	180 Ringing
			→	PRACK
			←	200 OK
CONN	←		←	200 OK INVITE
			→	ACK
		Conversation		
DISC	→		→	BYE
REL	←		←	200 OK BYE

IS_UD _03	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.1.5.1** **EN 300 899-1 [23], clause 2.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 7.3**		
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice				
Selection criteria:	Basic call; UDI; Q.1912.5 [51] Profile B with PI				
Test purpose:	Ensure that call establishment using **en-bloc sending** is performed correctly. Ensure that the ISDN user in the state U3 receives an ALERTING message **with** the progress indicator information element PI#1 or PI#2 or both location "Network beyond Interworking point" when the SIP user answers with 180 Ringing message. Ensure that in the active call state (N10) the data transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).				
ISDN Parameter values:	BC= UDI, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line : rtpmap:<dynamic-PT> CLEARMODE/8000 b = line AS: 64 m = RTP/AVP				
Comments:					
	ISDN		SUT		SIP
	Case a)				
	SETUP	→		→	INVITE
	CALL PROC	←			
	ALERTING PI	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	Case c)				
	SETUP	→		→	INVITE
	CALL PROC	←			
				←	183 Session Progress
				→	PRACK
				←	200 OK
				→	UPDATE
				←	200 OK
	ALERTING PI# VA	←		←	180 Ringing
				→	PRACK
				←	200 OK
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE

IS_UD_04	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.1.5.1** **EN 300 899-1 [23], clause 2.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 7.3** **EN 383 001 [49], clause 7.3** **ES 283 027 [48], clause 7.2.3.2**		
TSS reference:	ISDN-SIP/Basic_call/Successful/UDI				
Selection criteria:	Basic call; UDI; EN 383 001 [49] or ES 283 027 [48]				
Test purpose:	Ensure that call establishment using **en-bloc sending** is performed correctly. Ensure that the ISDN user in the state U3 receives an ALERTING message **without** the progress indicator information element when the SIP user answers with 180 Ringing message. Ensure that in the active call state (N10) the data transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).				
ISDN Parameter values:	BC= UDI, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line : rtpmap:<dynamic-PT> CLEARMODE/8000 b = line AS: 64 m = RTP/AVP				
Comments:					
	ISDN		SUT		SIP
	Case a)				
	SETUP	→		→	INVITE
	CALL PROC	←			
	ALERTING	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	Case c)				
	SETUP	→		→	INVITE
	CALL PROC	←			
				←	183 Session Progress
				→	PRACK
				←	200 OK
				→	UPDATE
				←	200 OK
	ALERTING	←		←	180 Ringing
				→	PRACK
				←	200 OK
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE

IS_UD_05	**ISDN reference to: EN 300 403-1 [i.3], clause 5.1.5.1 EN 300 899-1 [23], clause 2.1.1**				**NGN reference to: Q.1912.5 [51], clause 7.3**
TSS reference:	ISDN-SIP/Basic_call/Successful/UDI				
Selection criteria:	Basic call; UDI; SIP Profile A or Profile B optional				
Test purpose:	Ensure that call establishment using **en-bloc sending** is performed correctly. Ensure that the ISDN user in the state U3 receives a CONNECT message **with** the progress indicator information element "call is not end-to-end ISDN (#1)"location "Network beyond Interworking point" when the SIP user answers with 200 OK message. Ensure that in the active call state (N10) the data transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).				
ISDN Parameter values:	BC= UDI, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line : rtpmap:<dynamic-PT> CLEARMODE/8000 b = line AS: 64 m = RTP/AVP				
Comments:					
	ISDN		SUT		SIP
	Case a)				
	SETUP	→		→	INVITE
	CALL PROC	←			
	CONN PI# 1	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	Case c) Supported: 100 rel and precondition				
	SETUP	→		→	INVITE
	CALL PROC	←			
				←	183 Session Progress
				→	PRACK
				←	200 OK
				→	UPDATE
				←	200 OK
	CONN PI# 2	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE

<table>
<tr><th>IS_UD_06</th><th colspan="2">ISDN reference to:
EN 300 403-1 [i.3], clause 5.1.5.1
EN 300 899-1 [23], clause 2.1.1</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 7.3</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/Basic_call/Successful/Voice</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">Basic call; UDI; Q.1912.5 [51] Profile B with PI</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that call establishment using en-bloc sending is performed correctly.
Ensure that the ISDN user in the state U3 receives an CONNECT message with the progress indicator information element PI#1 or PI#2 or both location "Network beyond Interworking point" when the SIP user answers with 180 Ringing message.
Ensure that in the active call state (N10) the data transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">BC= UDI, no HLC</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line : rtpmap:<dynamic-PT> CLEARMODE/8000
b = line AS: 64
m = RTP/AVP</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="25"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>Case a)</td><td></td><td></td><td></td><td></td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CONN PI</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>Case c) Supported: 100 rel and precondition</td><td></td><td></td><td></td><td></td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>183 Session Progress</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK</td></tr>
<tr><td>CONN PI# VA</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>

IS_UD_07	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.1.5.1** **EN 300 899-1 [23], clause 2.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 7.3** **EN 383 001 [49], clause 7.1** **ES 283 027 [48], clause 7.2.3.2**		
TSS reference:	ISDN-SIP/Basic_call/Successful/UDI				
Selection criteria:	Basic call; UDI; EN 383 001 [49] or ES 283 027 [48];				
Test purpose:	Ensure that call establishment using **en-bloc sending** is performed correctly. Ensure that the ISDN user in the state U3 receives a CONNECT message **without** a progress indicator information element when the SIP user answers with 200 OK message. Ensure that in the active call state (N10) the data transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).				
ISDN Parameter values:	BC= UDI, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line : rtpmap:<dynamic-PT> CLEARMODE/8000 b = line AS: 64 m = RTP/AVP				
Comments:					
	ISDN		SUT		SIP
	Case a)				
	SETUP	→		→	INVITE
	CALL PROC	→			
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	Case c)				
	SETUP	→		→	INVITE
	CALL PROC	←			
				←	183 Session Progress
				→	PRACK
				←	200 OK
				→	UPDATE
				←	200 OK
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE

IS_UD_08	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.3.3** **EN 300 899-1 [23], clause 2.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 7.7** **EN 383 001 [49], clause 7.7** **ES 283 027 [48], clause 7.2.3.2**		
TSS reference:	ISDN-SIP/Basic_call/Successful/UDI				
Selection criteria:	Basic call; UDI				
Test purpose:	Ensure that the call establishment and the call clearing procedure is performed correctly when the **calling user** clears after answering with a DISCONNECT message indicating the Cause value # 16 "normal call clearing". The called user shall receive a BYE message. The Reason header shall contain Cause Value #16 in the case of EN 383 001 [49] and ES 283 027 [48].				
ISDN Parameter values:	BC= UDI, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line : rtpmap:<dynamic-PT> CLEARMODE/8000 b = line AS: 64 m = RTP/AVP				
Comments:					
	ISDN		SUT		SIP
	Case a)				
	SETUP	→		→	INVITE
	CALL PROC	←			
	ALERTING	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	Case c)				
	SETUP	→		→	INVITE
	CALL PROC	←			
				←	183 Session Progress
				→	PRACK
				←	200 OK
				→	UPDATE
				←	200 OK
	ALERTING	←		←	180 Ringing
				→	PRACK
				←	200 OK
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE

IS_UD_09	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.3.3** **EN 300 899-1 [23], clause 2.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 7.7** **EN 383 001 [49], clause 7.7** **ES 283 027 [48], clause 7.2.3.2**		
TSS reference:	ISDN-SIP/Basic_call/Successful/UDI				
Selection criteria:	Basic call; UDI; EN 383 001 [49] or ES 283 027 [48] and ITU optional				
Test purpose:	Ensure that the call clearing procedure is performed correctly when the **called user** clears after answering with a BYE message. The calling user shall receive a DISCONNECT message with the Cause value # 16 "normal call clearing".				
ISDN Parameter values:	BC= UDI, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line : rtpmap:<dynamic-PT> CLEARMODE/8000 b = line AS: 64 m = RTP/AVP				
Comments:					
	ISDN		SUT		SIP
	Case a)				
	SETUP	→		→	INVITE
	CALL PROC	←			
	ALERTING	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
				→	ACK
			Conversation		
	DISC	←		←	BYE
	REL	→		→	200 OK BYE
	Case c)				
	SETUP	→		→	INVITE
	CALL PROC	←			
				←	183 Session Progress
				→	PRACK
				←	200 OK
				→	UPDATE
				←	200 OK
	ALERTING	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
				→	PRACK
				←	200 OK
				→	ACK
			Conversation		
	DISC	←		←	BYE
	REL	→		→	200 OK BYE

6.1.1.6 Test purposes for ISDN-SIP Basic call Unsuccessful

Unsuccessful

<table>
<tr><td>IS_XX_U01</td><td colspan="2">ISDN reference to:
EN 300 403-1 [i.3],
clause 5.2.5.1, G.1.7
EN 300 899-1 [23], clause 2.1.1</td><td colspan="3">NGN reference to:
Q.1912.5 [51], clause 7.7.6
EN 383 001 [49], clause 7.7.6
ES 283 027 [48], clause 7.2.3.2</td></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/Basic_call/Unsuccessful</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">Basic call;</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that, when the called user is busy and responds with a 486 Busy Here message the circuit switched side is initiating call clearing with a DISCONNECT or RELEASE message indicating cause value #17 "user busy".</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">BC = PIXIT</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5">The originating exchange sends a DISCONNECT message to the calling user with progress indicator #8 thus indicating that in-band information is available. Normal release procedure applies after the in-band information has been connected. The calling user shall receive in the disconnect indication state (N12) the in-band tone/announcement on the B-channel.</td></tr>
<tr><td rowspan="5"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>DISC</td><td>←</td><td></td><td>←</td><td>486 Busy Here</td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td></td><td></td></tr>
</table>

IS_XX_U02	**ISDN reference to: EN 300 403-1 [i.3], clauses 5.2.5.1, G.1.7 EN 300 899-1 [23], clause 2.1.1**				**NGN reference to: Q.1912.5 [51], clause 7.7.6**
TSS reference:	ISDN-SIP/Basic_call/Unsuccessful				
Selection criteria:	Basic call;				
Test purpose:	Ensure that, when the called user is busy and the PROXY responds with a 486 Busy Here (NDUB). The circuit switched side is initiating call clearing with a DISCONNECT or RELEASE message indicating cause value #17 "user busy".				
ISDN Parameter values:	BC = PIXIT				
SIP Parameter values:	PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:	The originating exchange sends a DISCONNECT message to the calling user with progress indicator #8 thus indicating that in-band information is available. Normal release procedure applies after the in-band information has been connected. The calling user shall receive in the disconnect indication state (N12) the in-band tone/announcement on the B-channel.				
	ISDN		SUT		SIP
	SETUP	→			
	DISC	←			
	REL	→			
	RLC	←			

IS_XX_U03	**ISDN reference to:** **EN 300 403-1 [i.3],** **clauses 5.2.5.4, G.1.9** **EN 300 899-1 [23], clause 2.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 7.7.6** **EN 383 001 [49], clause 7.7.6** **ES 283 027 [48], clause 7.2.3.2**
TSS reference:	ISDN-SIP/Basic_call/Unsuccessful/	
Selection criteria:	Basic call; Reason Header field is supported	
Test purpose:	Ensure that when there is no answer from the called user (but user alerted), the ISDN network initiate call clearing to the calling user with a DISCONNECT message indicating cause value #19 "no answer from user (user alerted)" and sends to the called user a CANCEL or BYE message indicating cause # 102 "recovery on timer expire" in the Reason Header field.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROCEEDING	←			.
					…
					.
	ALERTING	←		←	180 Ringing
	DISC#19	←		→	CANCEL
				←	200 OK
				←	487 Request terminated
				→	ACK

IS_XX_U04	**ISDN reference to: EN 300 403-1 [i.3], clauses 5.2.5.4, G.1.9 EN 300 899-1 [23], clause 2.1.1**				**NGN reference to: Q.1912.5 [51], clause 7.7.6 EN 383 001 [49], clause 7.7.6 ES 283 027 [48], clause 7.2.3.2**
TSS reference:	ISDN-SIP/Basic_call/Unsuccessful/				
Selection criteria:	Basic call; Reason Header field is supported				
Test purpose:	Ensure that when there is no answer from the called user (but user alerted) and if the SIP network initiate call clearing before the SCN release the call, the SIP network shall send to the calling user a 480 Temporarily unavailable message and the SCN network initiate call clearing to the calling user with a DISCONNECT message indicating cause value # 20 Subscriber absent.				
ISDN Parameter values:	BC = PIXIT				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROCEEDING	←			.
					.
					.
	ALERTING	←		←	180 Ringing
	DISC#20	←		←	480 temp. Unavailable
	REL	→		→	ACK

IS_XX_U05	**ISDN reference to: EN 300 403-1 [i.3], clauses 5.2.5.4, G.1.9 EN 300 899-1 [23], clause 2.1.1**				**NGN reference to: Q.1912.5 [51], clause 7.7.6 EN 383 001 [49], clause 7.7.6 ES 283 027 [48], clause 7.2.3.2**
TSS reference:	ISDN-SIP/Basic_call/Unsuccessful/				
Selection criteria:	Basic call; Reason Header field is not supported				
Test purpose:	Ensure that when there is no answer from the called user (but user alerted), the ISDN network initiate call clearing to the calling user with a DISCONNECT message indicating cause value #19 "no answer from user (user alerted)" and sends to the called user a CANCEL or BYE.				
ISDN Parameter values:	BC = PIXIT				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROCEEDING	←			
					...
	ALERTING	←		←	180 Ringing
	DISC#19	←		→	CANCEL
				←	200 OK

IS_XX_U06	**ISDN reference to: EN 300 403-1 [i.3], clauses 5.2.5.4, G.1.9 EN 300 899-1 [23], clause 2.1.1**				**NGN reference to: Q.1912.5 [51], clause 7.7.6 EN 383 001 [49], clause 7.7.6 ES 283 027 [48], clause 7.2.3.2**
TSS reference:	ISDN-SIP/Basic_call/Unsuccessful/				
Selection criteria:	Basic call; Reason Header field is not supported				
Test purpose:	Ensure that when there is no answer from the called user (but user alerted) and if the SIP network initiate call clearing before the SCN release the call, the SIP network shall send to the calling user a 480 Temporarily unavailable message and the SCN network initiate call clearing to the calling user with a DISCONNECT message indicating cause value # 20 Subscriber absent.				
ISDN Parameter values:	BC = PIXIT				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROCEEDING	←			.
					.
					.
	ALERTING	←		←	180 Ringing
	DISC#20	←		←	480 temp. Unavailable
	REL	→		→	ACK

IS_XX_U07	**ISDN reference to: EN 300 403-1 [i.3], clauses 5.1.9, 5.3.2, G.1.10 EN 300 899-1 [23], clause 2.1.1**				**NGN reference to: Q.1912.5 [51], clause 7.7.6 EN 383 001 [49], clause 7.7.6 ES 283 027 [48], clause 7.2.3.2**
TSS reference:	ISDN-SIP/Basic_call/Unsuccessful/				
Selection criteria:	Basic call				
Test purpose:	Ensure that when the called side rejects the call and responds with a 603 Decline message containing the Cause information element indicating the cause value #21 "call reject". The circuit switched network initiates call clearing to the calling user with a DISCONNECT or RELEASE message indicating cause value # 21"call reject".				
ISDN Parameter values:	BC = PIXIT				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROCEEDING	←			.
	DISC#21	←		←	603 Decline Unavailable
				→	ACK

<table>
<tr><th>IS_XX_U08</th><th colspan="3">ISDN reference to:
EN 300 403-1 [i.3],
clauses 5.1.9, 5.3.2, G.1.10
EN 300 899-1 [23], clause 2.1.1</th><th colspan="2">NGN reference to:
Q.1912.5 [51], clause 7.7.6
EN 383 001 [49], clause 7.7.6
ES 283 027 [48], clause 7.2.3.2</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/Basic_call/Unsuccessful</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">Basic call;</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the call will be released when the called number is incomplete.
The circuit switched network initiates call clearing to the calling user with a DISCONNECT or RELEASE COMPLETE message with a cause value # 28.</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">BC = PIXIT</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5">In some networks tones or announcement can be generated in the destination exchange (or intermediate exchange) during call establishment.
The originating exchange sends a DISCONNECT message to the calling user with progress indicator #8 thus indicating that in-band information is available. Normal release procedure applies after the in-band information has been connected.
The calling user shall receive in the disconnect indication state (N12)
the in-band tone/announcement.</td></tr>
<tr><td rowspan="5"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP-S-CSCF</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROCEEDING</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>DISC#28</td><td>←</td><td></td><td>←</td><td>484 Address Incomplete</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
</table>

<table>
<tr><th>IS_XX_U09</th><th colspan="3">ISDN reference to:
EN 300 403-1 [i.3], clause G.1.13
EN 300 899-1 [23], clause 2.1.1</th><th colspan="2">NGN reference to:
Q.1912.5 [51], clause 7.7.6
EN 383 001 [49], clause 7.7.6
ES 283 027 [48], clause 7.2.3.2.12</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/Basic_call/Unsuccessful</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">Basic call;</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that when the called party is not registered.
The circuit switched network initiates call clearing to the calling user with a DISCONNECT or RELEASE COMPLETE message with a cause:
20 "subscriber absent"</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">BC = PIXIT</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="4"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td></td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>CALL PROCEEDING</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>DISC#20</td><td>←</td><td></td><td></td><td></td></tr>
</table>

IS_XX_U10	**ISDN reference to: EN 300 403-1 [i.3], clauses 5.2.2, G.5.7 EN 300 899-1 [23], clause 2.1.1**				**NGN reference to: Q.1912.5 [51], clause 7.7.6 EN 383 001 [49], clause 7.7.6 ES 283 027 [48], clause 7.2.3.2.12**
TSS reference:	ISDN-SIP/Basic_call/Unsuccessful				
Selection criteria:	Basic call				
Test purpose:	Ensure that when the called user is not compatible and responds with a 503 Service Unavailable the circuit switched network initiates call clearing to the calling user with a DISCONNECT with Cause value # 127.				
ISDN Parameter values:	BC = PIXIT				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROCEEDING	←			.
	DISC#127	←		←	503 Service Unavailable
				→	ACK

IS_XX_U11	**ISDN reference to: EN 300 403-1 [i.3], clause G.1.6 EN 300 899-1 [23], clause 2.1.1**				**NGN reference to: Q.1912.5 [51], clause 7.7.6 EN 383 001 [49], clause 7.7.6 ES 283 027 [48], clause 7.2.3.2.12**
TSS reference:	ISDN-SIP/Basic_call/Unsuccessful				
Selection criteria:	Basic call;				
Test purpose:	Ensure that when the calling user clears with cause value #16 "normal call clearing" before answer from called user, the network initiates call clearing to the called user with a CANCEL or BYE message.				
ISDN Parameter values:	BC=PIXIT				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROCEEDING	←			.
	ALERTING	←		←	180 Ringing
	DISC#16	→		→	CANCEL
	REL	←		←	200 OK CANCEL
	RLC	→		←	487 Request terminated
				→	ACK

IS_XX_U012	**ISDN reference to:** **EN 300 403-1 [i.3], G.1.6** **EN 300 899-1 [23], clause 2.1.1**				**NGN reference to:** **Q.1912.5 [51], clause 7.7.6** **EN 383 001 [49], clause 7.7.6** **ES 283 027 [48], clause 7.2.3.2.12**
TSS reference:	ISDN-SIP/Basic_call/Unsuccessful				
Selection criteria:	Basic call;				
Test purpose:	Ensure that the call will be released when the number is changed the circuit switched network initiates call clearing to the calling user with a DISCONNECT with Cause value # 22 number changed.				
ISDN Parameter values:	BC=PIXIT				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		
	SETUP	→			
	CALL PROCEEDING	←			
	DISC#22	←			
	REL	→			

IS_XX_U13	**ISDN reference to:** **EN 300 403-1 [i.3], clause G.1.6** **EN 300 899-1 [23], clause 2.1.1**				**NGN reference to:** **Q.1912.5 [51], clause 7.7.6,** **EN 383 001 [49], clause 7.7.6** **ES 283 027 [48], clause 7.2.3.2.12** **RFC 3261 [28] and RFC 4566 [25]**
TSS reference:	ISDN-SIP/Basic_call/Unsuccessful				
Selection criteria:	Basic call				
Test purpose:	Ensure that when there is no answer from the called user (there is no response from INVITE messages), the network initiate call clearing to the calling user with a DISCONNECT message indicating cause value # 20 "Subscriber absent"				
ISDN Parameter values:	BC=PIXIT				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROCEEDING	←			
				→	INVITE
				→	INVITE
				→	INVITE
				→	INVITE
				→	INVITE
	DISC#20	←		→	INVITE
	REL	→			
	RLC	←			

IS_XX_U14	**ISDN reference to:** **EN 300 403-1 [i.3], clause G.1.6** **EN 300 899-1 [23], clause 2.1.1**				**NGN reference to:** **Q.1912.5 [51], clause 7.7.6,** **EN 383 001 [49], clause 7.7.6** **ES 283 027 [48], clause 7.2.3.2.12** **RFC 3261 [28] and RFC 4566 [25]**
TSS reference:	ISDN-SIP/Basic_call/Unsuccessful				
Selection criteria:	Basic call; Reason Header field is supported				
Test purpose:	Ensure that the SUT if the SIP Failure response is interworked on receipt of a Failure message 4XX defined as SIP_Failure_VA. Sends a DISC or RELEASE message. The Cause Value in the header field set to CV_SIP is mapped to the ISDNCause Value field in the ISDNREL message with the Cause value set to CV_ ISDN.				
ISDN Parameter values:	BC=PIXIT				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROCEEDING	←			
	DISC#CV_ISDN	←		←	SIP_Failure_VA
	REL	→		→	ACK
	RLC	←			

Values for test purposes IS_XX_U14		
	←REL (Cause Value) CV_ ISDN	←4XX/5XX/6XX SIP message SIP_Failure_VA
VA_01	CV_ ISDN	415 Unsupported Media type CV_SIP (PIXIT)
VA_02	CV_ ISDN	420 Bad Extension CV_SIP (PIXIT)
VA_03	CV_ ISDN	421 Extension required CV_SIP (PIXIT)

CV_SIP = CV_ISDN

IS_XX_U15	**ISDN reference to: EN 300 403-1 [i.3], clause G.1.6 EN 300 899-1 [23], clause 2.1.1**				**NGN reference to: Q.1912.5 [51], clause 7.7.6, EN 383 001 [49], clause 7.7.6 ES 283 027 [48], clause 7.2.3.2.12 RFC 3261 [28] and RFC 4566 [25]**
TSS reference:	ISDN-SIP/Basic_call/Unsuccessful				
Selection criteria:	Basic call; Reason Header field is **not** supported				
Test purpose:	Ensure that the SUT if the SIP Failure response is interworked on receipt of a Failure message 4XX defined as SIP_Failure_VA sends a DISC with Cause Value 127.				
ISDN Parameter values:	BC=PIXIT				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		
	SETUP	→		→	INVITE
	CALL PROCEEDING	←			
	DISC#127	←		←	SIP_Failure_VA
	REL	→		→	ACK
	RLC	←			

Values for test purposes IS_XX_U15		
	←REL (Cause Value)	←4XX/5XX/6XX SIP message SIP_Failure_VA
VA_01	127	415 Unsupported Media type
VA_02	127	420 Bad Extension
VA_03	127	421 Extension required

IS_UD_U16	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.3.3** **EN 300 899-1 [23], clause 2.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 7.7** **EN 383 001 [49], clause 7.7** **ES 283 027 [48], clause 7.2.3.2.12**			
TSS reference:	ISDN-SIP/Basic_call/Unsuccessful				
Selection criteria:	Basic call; SIP Network does not support UDI				
Test purpose:	Ensure that when the SIP Network is not supporting UDI, the network initiate call clearing to the calling user with a DISCONNECT message indicating cause value # 65 "Bearer capability not implemented"				
ISDN Parameter values:	BC= UDI, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		
	SETUP	→		→	INVITE
	CALL PROCEEDING	←			
	DISC#65	←			
	REL	→			
	RLC	←			

IS_AU_U17	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.3.3** **EN 300 899-1 [23], clause 2.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 7.7** **EN 383 001 [49], clause 7.7** **ES 283 027 [48], clause 7.2.3.2.12**			
TSS reference:	ISDN-SIP/Basic_call/Unsuccessful				
Selection criteria:	Basic call; SIP Network does not support Teleservice FAX G3				
Test purpose:	Ensure that when the SIP Network is not supporting the Teleservice Fax G3, the network initiate call clearing to the calling user with a DISCONNECT message indicating cause value # 79 "Service or option not implemented"				
ISDN Parameter values:	BC=3,1 kHz audio, HLC= Facsimile G2/G3				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		
	SETUP	→		→	INVITE
	CALL PROCEEDING	←			
	DISC#79	←			
	REL	→			
	RLC	←			

IS_XX_U18	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.3.3** **EN 300 899-1 [23], clause 2.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 7.7** **EN 383 001 [49], clause 7.7** **ES 283 027 [48], clause 7.2.3.2.12**
TSS reference:	ISDN-SIP /Basic_call/Unsuccessful	
Selection criteria:	Basic call	
Test purpose:	During the session, the called user decide to change the characteristics of the media session. This is accomplished by sending a re-INVITE containing a new media description. This re-INVITE references the existing dialog so that the other party knows that it is to modify an existing session instead of establishing a new session. Ensure that if the other party does not accept the change, he sends an error response such as 488 (Not Acceptable Here), which also receives an ACK. On the ISDN side the user is still in the active state.	
ISDN Parameter values:	BC=PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

IS_XX_U19	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.1.5.1** **EN 300 899-1 [23], clause 2.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 7.1.1** **EN 383 001 [49], clause 7.1.1** **ES 283 027 [48], clause 7.2.3.2** **RFC 3264 [30], clause 6**
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice	
Selection criteria:	Basic call	
Test purpose:	Ensure that answer related to the SDP offer is contained in the 180 Ringing message. The media stream is rejected (port number is set to zero). Ensure that the call is rejected by sending a CANCEL or BYE.	
ISDN Parameter values:	BC=PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROCEEDING	←			.
	ALERTING	←		←	180 Ringing answer 1
	Case a)				
	DISC	←		→	CANCEL
	REL	→		←	200 OK CANCEL
	RLC	←		←	487 Request terminated
				→	ACK
	Case b)				
	DISC	←		→	BYE
	REL	→		←	200 OK
	RLC	←		←	487 Request terminated
				→	ACK

IS_XX_U20	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.1.5.1** **EN 300 899-1 [23], clause 2.1.1**			**NGN reference to:** **Q.1912.5 [51], clause 7.1.1** **RFC 3264 [30], clause 6**
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice				
Selection criteria:	Basic call				
Test purpose:	Ensure that answer related to the SDP offer is contained in the 200 OK INVITE message. The media stream is rejected (port number is set to zero). Ensure that the call is rejected by sending a BYE.				
ISDN Parameter values:	BC=PIXIT				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	SETUP	→		→	INVITE offer 1
	CALL PROCEEDING	←			.
				←	200 OK INVITE answer 1
				→	ACK
	DISC	←		→	BYE
	REL	→		←	200 OK
	RLC	←		←	487 Request terminated
				→	ACK

6.1.2 Test purposes for ISDN-SIP Supplementary services

6.1.2.1 CLIP/OIP

IS_XXSSCLIP01	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3], clauses 4.5.10, 4.5.11**	**NGN reference to:** **Q.1912.5 [51], clause 7.1.3** **EN 383 001 [49], clause 7.1.3** **ES 283 027 [48], clause 7.2.3.2.2.3** **TS 183 007 [43]**
TSS reference:	ISDN-SIP/Supplementary_services/CLIP	
Selection criteria:	The called user is provided with CLIP	
Test purpose:	Ensure that when the Calling party number is provided by the calling user (verified and passed) with the APRI "presentation allowed". The Type of number is defined as : TYPE_NUMBER, the Calling party number information elements is correctly delivered to the called (served) with following mapping rules: The SIP P-Asserted-Identity header is Derived from Calling party information element Address Signal. The Calling Party Number is mapped into the SIP From header.	
ISDN Parameter values:	BC= PIXIT,	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROCEEDING	←			
					…
	ALERTING	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
			Conversation		
	DISC	→		→	BYE
	REL	←			200 OK BYE
	RLC	→			

Values for the test purpose IS_XXSSCLIP01

<table>
<tr><th colspan="2">SETUP→</th><th colspan="2">INVITE→</th></tr>
<tr><th colspan="2">Calling party number i.e.</th><th rowspan="2">From Header Field</th><th rowspan="2">P-Asserted -Identity</th></tr>
<tr><th>TYPE_NUMBER</th><th>Numbering plan identification</th></tr>
<tr><td>National number</td><td rowspan="4">ISDN/telephony numbering plan
or
Unknown</td><td>The user is derived from the address string of the calling party number IE
sip: „+"CC+ NDC + SN @hostportion; user=phone</td><td>The user is derived from the address string of the calling party number IE
„+"CC+ NDC + SN @hostportion; user=phone</td></tr>
<tr><td>International number</td><td>The user is derived from the address string of the calling party number IE
sip: „+"CC+ NDC+ SN @hostportion; user=phone</td><td>The user is derived from the address string of the calling party number IE
sip: „+"CC+ NDC + SN @hostportion; user=phone</td></tr>
<tr><td>Unknown</td><td>The user is derived from the address string of the calling party number IE
sip: „+"CC+ NDC + SN @hostportion; user=phone</td><td>The user is derived from the address string of the calling party number IE
sip: „+"CC+ NDC + SN @hostportion; user=phone</td></tr>
<tr><td>Subscriber number</td><td>The user is derived from the address string of the calling party number IE
sip: „+"CC+ NDC + SN @hostportion; user=phone</td><td>The user is derived from the address string of the calling party number IE
sip: „+"CC+ NDC + SN @hostportion; user=phone</td></tr>
</table>

<table>
<tr><th>IS_XXSSCLIP02</th><th colspan="2">ISDN reference to:
EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 7.1.3
EN 383 001 [49], clause 7.1.3
ES 283 027 [48], clause 7.2.3.2.2.3
TS 183 007, [43]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/Supplementary_services/CLIP</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The called user is provided with CLIP</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that when no Calling party number information element is provided by the calling user, (and no Calling party subaddress), the Calling party number information element is network provided and correctly delivered to the called (served) user with following mapping rules:
The SIP P-Asserted-Identity header is Derived from Calling party information element Address Signal.
The Calling Party Number is mapped into the SIP From header.</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">BC = PIXIT</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="11"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROCEEDING</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td>…</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td></td><td></td></tr>
</table>

Values for the test purpose IS_XXSSCLIP02

SETUP→		INVITE→	
Calling party number i.e.		**From Header Field**	**P-Asserted -Identity**
TYPE_NUMBER	**Numbering plan identification**		
No or invalid calling party number information element (see note)		Default Public user identity sip: „+"CC+ NDC + SN @hostportion; user=phone	Default Public user identity sip: „+"CC+ NDC + SN @hostportion; user=phone
NOTE: Validity conditions of the calling party number information element are defined in clause 3.5.2.2.1 of ITU-T Q.951 [i.17].			

IS_XXSSCLIP03	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3], clauses 4.5.10, 4.5.11**	**NGN reference to:** **Q.1912.5 [51], clause 7.1.3** **EN 383 001 [49], clause 7.1.3** **ES 283 027 [48], clause 7.2.3.2.2.3** **TS 183 007 [43]**
TSS reference:	ISDN-SIP/Supplementary_services/CLIP	
Selection criteria:	The called user is provided with CLIP Special arrangement applies	
Test purpose:	Ensure that when a **special arrangement applies** and a Calling party number information element and a valid calling number is provided by the calling user, the user provided, not screened number (generic number) delivered to the called (served) user. In the SIP From header field the addr-spec is derived from unscreened Calling party number. The SIP P-Asserted-Identity header is Derived from Calling party information element Address Signal.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROCEEDING	←			
				...
ALERTING	←		←	180 Ringing
CONN	←		←	200 OK INVITE
		Conversation		
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

Values for the test purpose IS_XXSSCLIP03

SETUP→		INVITE→	
Calling party number i.e.		From Header Field	P-Asserted -Identity
TYPE_NUMBER	Numbering plan identification		
National number	*ISDN/telephony numbering plan* or *Unknown*	The user is derived from the address string of the calling party number IE sip: „+"CC+ NDC + SN @hostportion; user=phone	Default Public user identity sip: „+"CC+ NDC + SN @hostportion; user=phone
International number		The user is derived from the address string of the calling party number IE sip: „+"CC+ NDC + SN @hostportion; user=phone	Default Public user identity sip: „+"CC+ NDC + SN @hostportion; user=phone

IS_XXSSCLIP04	ISDN reference to: EN 300 092-1 [i.14], clause 9.3 EN 300 403-1 [i.3], clauses 4.5.10, 4.5.11	NGN reference to: Q.1912.5 [51], clause 7.1.3 EN 383 001 [49], clause 7.1.3 ES 283 027 [48], clause 7.2.3.2.2.3 TS 183 007 [43]
TSS reference:	ISDN-SIP/Supplementary_services/CLIP	
Selection criteria:	The called user is provided with CLIP Special arrangement applies.	
Test purpose:	Ensure that when a **special arrangement applies** and no Calling party number information element is provided by the calling user, the default number of the access of the calling user is correctly delivered to the called (served) user. The mapping rules are described in the attached table.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROCEEDING	←			
				…
ALERTING	←		←	180 Ringing
CONN	←		←	200 OK INVITE
		Conversation		
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

Mapping rules for the test purpose IS_XXSSCLIP04

SETUP→		**INVITE→**	
Calling party number i.e.		**From Header Field**	**P-Asserted -Identity**
TYPE_NUMBER	**Numbering plan identification**		
No or invalid calling party number information element		Default Public user identity sip: „+"CC+ NDC + SN @hostportion; user=phone	Default Public user identity sip: „+"CC+ NDC + SN @hostportion; user=phone

6.1.2.2 CLIR/OIR

IS_XXSSCLIR01	**ISDN reference to:** **EN 300 093-1 [i.4], clause 9.4.1,** **EN 300 092-1 [i.14], clause A.2 Figure 2**	**NGN reference to:** **Q.1912.5 [51], clause 7.1.3** **EN 383 001 [49], clause 7.1.3** **ES 283 027 [48], clause 7.2.3.2.2.3** **TS 183 007 [43]** **RFC 3323 [33], RFC 3325 [34]**
TSS reference:	ISDN-SIP/Supplementary_services/CLIR	
Selection criteria:	The calling user is provided with CLIR permanent mode subscription	
Test purpose:	Ensure that when the Calling party number is provided by the calling user, with Calling party subaddress: Sends a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROCEEDING	←			
				…
ALERTING	←		←	180 Ringing
CONN	←		←	200 OK INVITE
		Conversation		
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

IS_XXSSCLIR02	**ISDN reference to:** **EN 300 093-1 [i.4], clause 9.4.1,** **EN 300 092-1 [i.14], clause A.2 Figure 2**		**NGN reference to:** **Q.1912.5 [51], clause 7.1.3** **EN 383 001 [49], clause 7.1.3** **ES 283 027 [48], clause 7.2.3.2.2.3** **TS 183 007 [43]** **RFC 3323 [33], RFC 3325 [34]**	
TSS reference:	ISDN-SIP/Supplementary_services/CLIR			
Selection criteria:	The calling user is provided with CLIR temporary mode subscription			
Test purpose:	Ensure that when the Calling party number is provided by the calling user with the APRI "presentation restricted" and with Calling party subaddress: Sends a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received.			
ISDN Parameter values:	BC=PIXIT			
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)			
Comments:				

	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROCEEDING	←			
					…
	ALERTING	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	RLC	→			

IS_XXSSCLIR03	**ISDN reference to:** **EN 300 093-1 [i.4], clause 9.4.1** **EN 300 092-1 [i.14], clause A.2 Figure 2**				**NGN reference to:** **Q.1912.5 [51], clause 7.1.3** **EN 383 001 [49], clause 7.1.3** **ES 283 027 [48], clause 7.2.3.2.2.3** **TS 183 007 [43]** **RFC 3323 [33], RFC 3325 [34]**
TSS reference:	ISDN-SIP/Supplementary_services/CLIR				
Selection criteria:	The calling user is provided with CLIR permanent mode subscription				
Test purpose:	Ensure that when no Calling party number is provided by the calling user (and no Calling party subaddress): Sends a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received.				
ISDN Parameter values:	BC=PIXIT				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROCEEDING	←			
					...
	ALERTING	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	RLC	→			

IS_XXSSCLIR04	**ISDN reference to:** **EN 300 093-1 [i.4], clause 9.4.1** **EN 300 092-1 [i.14], clause A.2** **Figure 2**				**NGN reference to:** **Q.1912.5 [51], clause 7.1.3** **EN 383 001 [49], clause 7.1.3** **ES 283 027 [48], clause 7.2.3.2.2.3** **TS 183 007 [43]** **RFC 3323 [33], RFC 3325 [34]**
TSS reference:	ISDN-SIP/Supplementary_services/CLIR/				
Selection criteria:	The calling user is provided with CLIR temporary mode subscription; The called user is provided with CLIP; Special arrangement applies.				
Test purpose:	Ensure that when a **special arrangement applies** and a Calling party number information element and a valid calling number with presentation in not allowed is provided by the calling user: Sends a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received.				
ISDN Parameter values:	BC = PIXIT				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROCEEDING	←			
					…
	ALERTING	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
			Conversation		
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	RLC	→			

6.1.2.3 COLP/COLR (TIP/TIR)

<table>
<tr><th>IS_XXSSCOLP01</th><th>ISDN reference to:
EN 300 092-1 [i.14]
EN 300 403-1 [i.3]</th><th colspan="4">NGN reference to:
EN 383 001 [49]
ES 283 027 [48]
TS 183 008 [44]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">Temporary mode presentation not restricted</td></tr>
<tr><td>ISDN selection criteria:</td><td colspan="5">COLP service has been requested by the calling party</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT on receipt of a provisional 1XX response defined as SIP_MESSAGE_VA with priv-value component is set to "none" has been received from the terminating user:
• no P-Preferred-Identity header field is provided within the 1xx response
• sends a CONNECT message with the Connected Party Number information element coded:
- Address signals = default public user identity
- Numbering plan indicator = ISDN numbering plan
- Type of number = PIXIT
- Screening indicator = network provided
- Address presentation restriction indicator = presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">CONNECT;
Connected number
User provided, not screened Connected Party Number not present</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="11"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>XXXX</td><td>←</td><td></td><td>←</td><td>SIP_MESSAGE_VA</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>Conversation</td><td></td><td>Conversation</td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td></td><td></td></tr>
</table>

Values for test purpose IS_XXSSCOLP01	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	183 Session Progress

<table>
<tr><th>IS_XXSSCOLP02</th><th colspan="3">ISDN reference to:
EN 300 092-1 [i.14]
EN 300 403-1 [i.3]</th><th colspan="2">NGN reference to:
EN 383 001 [49]
ES 283 027 [48]
TS 183 008 [44]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">Temporary mode presentation not restricted</td></tr>
<tr><td>ISDN selection criteria:</td><td colspan="5">COLP service has been requested by the calling party</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT on receipt of a provisional 1XX response defined as SIP_MESSAGE_VA with priv-value component is set to "id" has been received from the terminating user:
• no P-Preferred-Identity header field is provided within the 1xx response
• sends a CONNECT message with the Connected Party Number information element coded:
- Address signals = not available
- Numbering plan indicator = ISDN numbering plan
- Type of number = (PIXIT)
- Screening indicator = network provided
- Address presentation restriction indicator = presentation restricted</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">CONNECT;
Connected number
User provided, not screened Connected Party Number not present</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="11"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>XXXX</td><td>←</td><td></td><td>←</td><td>SIP_MESSAGE_VA</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>Conversation</td><td></td><td>Conversation</td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td></td><td></td></tr>
</table>

Values for test purpose IS_XXSSCOLP02	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	183 Session Progress

<table>
<tr><th>IS_XXSSCOLP03</th><th colspan="4">ISDN reference to:
EN 300 092-1 [i.14]
EN 300 403-1 [i.3]</th><th>NGN reference to:
EN 383 001 [49]
ES 283 027 [48]
TS 183 008 [44]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">Temporary mode presentation not restricted</td></tr>
<tr><td>ISDN selection criteria:</td><td colspan="5">COLP service has been requested by the calling party</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT on receipt of a provisional 1XX response defined as SIP_MESSAGE_VA with priv-value component is set to "user" has been received from the terminating user:
• no P-Preferred-Identity header field is provided within the 1xx response
• sends a CONNECT message with the Connected Party Number information element coded:
- Address signals = not available
- Numbering plan indicator = ISDN numbering plan
- Type of number = PIXIT
- Screening indicator = network provided
- Address presentation restriction indicator = presentation restricted</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">CONNECT;
Connected number
User provided, not screened Connected Party Number not present</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="11"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>XXXX</td><td>←</td><td></td><td>←</td><td>SIP_MESSAGE_VA</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>Conversation</td><td></td><td>Conversation</td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td></td><td></td></tr>
</table>

Values for test purpose IS_XXSSCOLP03	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	183 Session Progress

IS_XXSSCOLP04	**ISDN reference to:** **EN 300 092-1 [i.14]** **EN 300 403-1 [i.3]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]** **TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	Temporary mode presentation not restricted	
ISDN selection criteria:	COLP service has been requested by the calling party	
Test purpose:	Ensure that the SUT on receipt of a provisional 1XX response defined as **SIP_MESSAGE_VA** with priv-value component is set to "header" has been received from the terminating user: • no P-Preferred-Identity header field is provided within the 1xx response • sends a CONNECT message with the **Connected Party Number information element** coded: - Address signals = not available - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = **presentation restricted**	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	CONNECT; **Connected number** **User provided, not screened Connected Party Number** not present	
Comments:		

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROC	←			
XXXX	←		←	**SIP_MESSAGE_VA**
CONN	←		←	200 OK INVITE
			→	ACK
	Conversation		Conversation	
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

Values for test purpose IS_XXSSCOLP04	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	183 Session Progress

<table>
<tr><td>IS_XXSSCOLP05</td><td>ISDN reference to:
EN 300 092-1 [i.14]
EN 300 403-1 [i.3]</td><td>NGN reference to:
EN 383 001 [49]
ES 283 027 [48]
TS 183 008 [44]</td></tr>
<tr><td>TSS reference:</td><td colspan="2">ISDN-SIP/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">Temporary mode presentation not restricted</td></tr>
<tr><td>ISDN selection criteria:</td><td colspan="2">COLP service has been requested by the calling party</td></tr>
<tr><td>Test purpose:</td><td colspan="2">Ensure that the SUT having not received a provisional 1XX response, on receipt of a 200 OK INVITE for this call without a Privacy header field:
• no P-Preferred-Identity header field is provided within the 200 OK response
• sends a CONNECT message with the Connected Party Number information element coded:
- Address signals = default public user identity
- Numbering plan indicator = ISDN numbering plan
- Type of number = PIXIT
- Screening indicator = network provided
- Address presentation restriction indicator = presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">Dial string parameters options=PIXIT
PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition
a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="2">ANM;
Connected number
User provided, not screened Connected Party Number not present</td></tr>
<tr><td>Comments:</td><td colspan="2"></td></tr>
</table>

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROC	←			
CONN	←		←	200 OK INVITE
			→	ACK
	Conversation		Conversation	
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

IS_XXSSCOLP06	**ISDN reference to:** **EN 300 092-1 [i.14]** **EN 300 403-1 [i.3]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]** **TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	Temporary mode presentation not restricted	
ISDN selection criteria:	COLP service has been requested	
Test purpose:	Ensure that the SUT on receipt of a 200 OK INVITE for this call with a Privacy header field was received with the value "none" has been received: • no P-Preferred-Identity header field is provided within the 200 OK response • sends a CONNECT message with the Connected Party Number information element coded: - Address signals = default public user identity - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = presentation allowed	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	CONNECT; Connected Party Number information element User provided, not screened Connected Party Number not present NoAS: NoA_VALUE	
Comments:		

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROC	←			
ALERTING	←		←	180 Ringing
CONN	←		←	200 OK INVITE
			→	ACK
	Conversation		Conversation	
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

<table>
<tr><th>IS_XXSSCOLP07</th><th colspan="3">ISDN reference to:
EN 300 092-1 [i.14],
EN 300 403-1 [i.3]</th><th colspan="2">NGN reference to:
EN 383 001 [49]
ES 283 027 [48]
TS 183 008 [44]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">Temporary mode presentation not restricted</td></tr>
<tr><td>ISDN selection criteria:</td><td colspan="5">COLP service has been requested</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT on receipt of a 200 OK INVITE for this call with a Privacy header field was received with the value "PRIV_VALUE" has been received:
• no P-Preferred-Identity header field is provided within the 200 OK response
• sends a CONNECT message with the Connected Party Number information element coded:
- Address signals = not available
- Numbering plan indicator = ISDN numbering plan
- Type of number = PIXIT
- Screening indicator = network provided
- Address presentation restriction indicator = presentation restricted</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">CONNECT;
Connected Party Number information element
User provided, not screened Connected Party Number not present</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="12"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>Conversation</td><td></td><td>Conversation</td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td></td><td></td></tr>
</table>

Values for test purpose IS_XXSSCOLP07	
VA	PRIV_VALUE
VA_1	Id
VA_2	User
VA_3	Header

IS_XXSSCOLP08	**ISDN reference to:** **EN 300 092-1 [i.14]** **EN 300 403-1 [i.3]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]** **TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	Temporary mode presentation restricted	
ISDN selection criteria:	COLP service has been requested by the calling party	
Test purpose:	Ensure that the SUT on receipt of a provisional 1XX response defined as SIP_MESSAGE_VA with priv-value component is set to "none" has been received from the terminating user: • no P-Preferred-Identity header field is provided within the 1xx response • sends a CONNECT message with the Connected Party Number information element coded: - Address signals = default public user identity - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = presentation allowed	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	CONNECT; Connected number User provided, not screened Connected Party Number not present	
Comments:		

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROC	←			
XXXX	←		←	SIP_MESSAGE_VA
CONN	←		←	200 OK INVITE
			→	ACK
	Conversation		Conversation	
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

Values for test purpose IS_XXSSCOLP08	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	183 Session Progress

<table>
<tr><td>IS_XXSSCOLP09</td><td>ISDN reference to:
EN 300 092-1 [i.14]
EN 300 403-1 [i.3]</td><td>NGN reference to:
EN 383 001 [49]
ES 283 027 [48]
TS 183 008 [44]</td></tr>
<tr><td>TSS reference:</td><td colspan="2">ISDN-SIP/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">Temporary mode presentation restricted</td></tr>
<tr><td>ISDN selection criteria:</td><td colspan="2">COLP service has been requested by the calling party</td></tr>
<tr><td>Test purpose:</td><td colspan="2">Ensure that the SUT on receipt of a provisional 1XX response defined as SIP_MESSAGE_VA with priv-value component is set to "id" has been received from the terminating user:
• no P-Preferred-Identity header field is provided within the 1xx response
• sends a CONNECT message with the Connected Party Number information element coded:
- Address signals = not available
- Numbering plan indicator = ISDN numbering plan
- Type of number = (PIXIT)
- Screening indicator = network provided
- Address presentation restriction indicator = presentation restricted</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="2">CONNECT;
Connected number
User provided, not screened; Connected Party Number not present</td></tr>
<tr><td>Comments:</td><td colspan="2"></td></tr>
</table>

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROC	←			
XXXX	←		←	SIP_MESSAGE_VA
CONN	←		←	200 OK INVITE
			→	ACK
	Conversation		Conversation	
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

Values for test purpose IS_XXSSCOLP09	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	183 Session Progress

<table>
<tr><td>IS_XXSSCOLP10</td><td>ISDN reference to:
EN 300 092-1 [i.14]
EN 300 403-1 [i.3]</td><td colspan="4">NGN reference to:
EN 383 001 [49]
ES 283 027 [48]
TS 183 008 [44]</td></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">Temporary mode presentation restricted</td></tr>
<tr><td>ISDN selection criteria:</td><td colspan="5">COLP service has been requested by the calling party</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT on receipt of a provisional 1XX response defined as SIP_MESSAGE_VA with priv-value component is set to "user" has been received from the terminating user.
• no P-Preferred-Identity header field is provided within the 1xx response
• sends a CONNECT message with the Connected Party Number information element coded:
- Address signals = not available
- Numbering plan indicator = ISDN numbering plan
- Type of number = PIXIT
- Screening indicator = network provided
- Address presentation restriction indicator = presentation restricted</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">CONNECT;
Connected number
User provided, not screened; Connected Party Number not present</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="11"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>XXXX</td><td>←</td><td></td><td>←</td><td>SIP_MESSAGE_VA</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>Conversation</td><td></td><td>Conversation</td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td></td><td></td></tr>
</table>

Values for test purpose IS_XXSSCOLP10	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	183 Session Progress

IS_XXSSCOLP11	**ISDN reference to:** **EN 300 092-1 [i.14]** **EN 300 403-1 [i.3]**			**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]** **TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP			
SIP selection criteria:	Temporary mode presentation restricted			
ISDN selection criteria:	COLP service has been requested by the calling party			
Test purpose:	Ensure that the SUT on receipt of a provisional 1XX response defined as SIP_MESSAGE_VA with priv-value component is set to "header" has been received from the terminating user: • no P-Preferred-Identity header field is provided within the 1xx response • sends a CONNECT message with the Connected Party Number information element coded: - Address signals = not available - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = presentation restricted			
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)			
ISDN Parameter values:	CONNECT; Connected number User provided, not screened; Connected Party Number not present			
Comments:				

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROC	←			
XXXX	←		←	SIP_MESSAGE_VA
CONN	←		←	200 OK INVITE
			→	ACK
	Conversation		Conversation	
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

Values for test purpose IS_XXSSCOLP11	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	183 Session Progress

IS_XXSSCOLP12	**ISDN reference to:** **EN 300 092-1 [i.14]** **EN 300 403-1 [i.3]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]** **TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	Temporary mode presentation restricted	
ISDN selection criteria:	COLP service has been requested by the calling party	
Test purpose:	Ensure that the SUT having **not** received a provisional 1XX response, on receipt of a 200 OK INVITE for this call **without** a Privacy header field: • no P-Preferred-Identity header field is provided within the 1xx response • sends a CONNECT message with the Connected Party Number information element coded: - Address signals = not available - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = presentation restricted	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	CONNECT; Connected number User provided, not screened Connected Party Number not present	
Comments:		

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROC	←			
CONN	←		←	200 OK INVITE
			→	ACK
	Conversation		Conversation	
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

<table>
<tr><td>IS_XXSSCOLP13</td><td colspan="3">ISDN reference to:
EN 300 092-1 [i.14]
EN 300 403-1 [i.3]</td><td colspan="2">NGN reference to:
EN 383 001 [49]
ES 283 027 [48]
TS 183 008 [44]</td></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">Temporary mode presentation restricted</td></tr>
<tr><td>ISDN selection criteria:</td><td colspan="5">COLP service has been requested by the calling party</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT having not received a provisional 1XX response, on receipt of a 200 OK INVITE for this call, a Privacy header field with the value "none" has been received:
• no P-Preferred-Identity header field is provided within the 200 OK response
• sends a CONNECT message with the Connected Party Number information element coded:
- Address signals = default public user identity
- Numbering plan indicator = ISDN numbering plan
- Type of number = PIXIT
- Screening indicator = network provided
- Address presentation restriction indicator = presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">CONNECT;
Connected number Address signals = default public user identity
User provided, not screened; Connected Party Number not present</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="10"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>Conversation</td><td></td><td>Conversation</td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td></td><td></td></tr>
</table>

IS_XXSSCOLP14	**ISDN reference to:** **EN 300 092-1 [i.14],** **EN 300 403-1 [i.3]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]** **TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	Temporary mode presentation restricted	
ISDN selection criteria:	COLP service has been requested by the calling party	
Test purpose:	Ensure that the SUT having **not** received a provisional 1XX response, on receipt of a 200 OK INVITE for this call, a Privacy header field with the value "PRIV_VALUE" has been received: • no P-Preferred-Identity header field is provided within the 200 OK response • sends a CONNECT message with the Connected Party Number information element coded: - Address signals = not available - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = presentation restricted	
SIP Parameter values:	PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	CONNECT; Connected number Address signals = not available User provided, not screened Connected Party Number not present	
Comments:		

	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROC	←			
	CONN	←		←	200 OK INVITE
				→	ACK
		Conversation		Conversation	
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	RLC	→			

IS_XXSSCOLP15	**ISDN reference to:** **EN 300 092-1 [i.14]** **EN 300 403-1 [i.3]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]** **TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	Temporary mode presentation restricted	
ISDN selection criteria:	COLP service has been requested	
Test purpose:	Ensure that the SUT on receipt of a 200 OK INVITE for this call with a Privacy header field was received with the value "none" has been received: • no P-Preferred-Identity header field is provided within the 200 OK response • sends a CONNECT message with the Connected Party Number information element coded: - Address signals = default public user identity - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = presentation allowed	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	COONECT; Connected Party Number information element User provided, not screened Connected Party Number not present NoAS: NoA_VALUE	
Comments:		

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROC	←			
ALERTING	←		←	180 Ringing
CONN	←		←	200 OK INVITE
			→	ACK
	Conversation		Conversation	
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

IS_XXSSCOLP16	**ISDN reference to:** **EN 300 092-1 [i.14],** **EN 300 403-1 [i.3]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]** **TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	Temporary mode presentation not restricted	
ISDN selection criteria:	COLP service has been requested	
Test purpose:	Ensure that the SUT on receipt of a 200 OK INVITE for this call with a Privacy header field was received with the value "PRIV_VALUE" has been received: • no P-Preferred-Identity header field is provided within the 200 OK response • sends a CONNECT message with the Connected Party Number information element coded: - Address signals = not available - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = presentation restricted	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	CONNECT; Connected Party Number information element User provided, not screened Connected Party Number not present NoAS: NoA_VALUE	
Comments:		

	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROC	←			
	ALERTING	←		←	180 Ringing
	CONN	←		←	200 OK INVITE
				→	ACK
		Conversation		Conversation	
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	RLC	→			

Values for test purpose IS_XXSSCOLP16	
VA	PRIV_VALUE
VA_1	Id
VA_2	User
VA_3	Header

IS_XXSSCOLP17	**ISDN reference to:** **EN 300 092-1 [i.14]** **EN 300 403-1 [i.3]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]** **TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	permanent mode	
ISDN selection criteria:	COLP service has been requested by the calling party	
Test purpose:	Ensure that the SUT on receipt of a provisional 1XX response defined as SIP_MESSAGE_VA with priv-value component is set to "none" has been received from the terminating user: • no P-Preferred-Identity header field is provided within the 200 OK response • sends a CONNECT message with the Connected Party Number information element coded: - Address signals = default public user identity - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = presentation allowed	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	CONNECT; Connected number User provided, not screened Connected Party Number not present	
Comments:		

	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROC	←			
	XXXX	←		←	SIP_MESSAGE_VA
	CONN	←		←	200 OK INVITE
				→	ACK
		Conversation		Conversation	
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	RLC	→			

Values for test purpose IS_XXSSCOLP17	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	183 Session Progress

<table>
<tr><td>IS_XXSSCOLP18</td><td colspan="3">ISDN reference to:
EN 300 092-1 [i.14]
EN 300 403-1 [i.3]</td><td colspan="2">NGN reference to:
EN 383 001 [49]
ES 283 027 [48]
TS 183 008 [44]</td></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">Permanent mode</td></tr>
<tr><td>ISDN selection criteria:</td><td colspan="5">COLP service has been requested by the calling party</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT having not received a provisional 1XX response, on receipt of a 200 OK INVITE for this call without a Privacy header field:
• no P-Preferred-Identity header field is provided within the 200 OK response
• sends a CONNECT message with the Connected Party Number information element coded:
- Address signals = not available
- Numbering plan indicator = ISDN numbering plan
- Type of number = PIXIT
- Screening indicator = network provided
- Address presentation restriction indicator = presentation restricted</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">CONNECT;
Connected number
User provided, not screened Connected Party Number not present</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="10"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>Conversation</td><td></td><td>Conversation</td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td></td><td></td></tr>
</table>

IS_XXSSCOLP19	**ISDN reference to:** **EN 300 092-1 [i.14]** **EN 300 403-1 [i.3]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]** **TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	Permanent mode	
ISDN selection criteria:	COLP service has been requested by the calling party	
Test purpose:	Ensure that the SUT on receipt of a 200 OK INVITE for this call with a Privacy header field was received with the value "none" has been received: • no P-Preferred-Identity header field is provided within the 200 OK response • sends a CONNECT message with the Connected Party Number information element coded: - Address signals = not available - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = presentation allowed	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	CONNECT; Connected Party Number information element User provided, not screened Connected Party Number not present NoAS: NoA_VALUE	
Comments:		

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROC	←			
ALERTING	←		←	180 Ringing
CONN	←		←	200 OK INVITE
			→	ACK
	Conversation		Conversation	
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

IS_XXSSCOLP20	**ISDN reference to: EN 300 092-1 [i.14], EN 300 403-1 [i.3]**	**NGN reference to: EN 383 001 [49] ES 283 027 [48] TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	Temporary mode presentation not restricted	
ISDN selection criteria:	COLP service has been requested by the calling party	
Test purpose:	Ensure that the SUT on receipt of a provisional 1XX response defined as SIP_MESSAGE_VA with priv-value component is set to "none" has been received from the terminating user: • P-Preferred-Identity header field is provided within the 1xx response • sends a CONNECT message with the Connected Party Number information element coded: - Address signals = default public user identity - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = presentation allowed	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	CONNECT; Connected number User provided, not screened Connected Party Number not present	
Comments:		

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROC	←			
XXXX	←		←	SIP_MESSAGE_VA
CONN	←		←	200 OK INVITE
			→	ACK
	Conversation		Conversation	
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

Values for test purpose IS_XXSSCOLP20	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	183 Session Progress

<table>
<tr><th>IS_XXSSCOLP21</th><th colspan="4">ISDN reference to:
EN 300 092-1 [i.14]
EN 300 403-1 [i.3]</th><th>NGN reference to:
EN 383 001 [49]
ES 283 027 [48]
TS 183 008 [44]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">Temporary mode presentation not restricted</td></tr>
<tr><td>ISDN selection criteria:</td><td colspan="5">COLP service has been requested by the calling party</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT on receipt of a provisional 1XX response defined as SIP_MESSAGE_VA with priv-value component is set to "id" has been received from the terminating user:
• P-Preferred-Identity header field is provided within the 1xx response
• Sends a CONNECT message with the Connected Party Number information element coded:
- Address signals = not available
- Numbering plan indicator = ISDN numbering plan
- Type of number = (PIXIT)
- Screening indicator = network provided
- Address presentation restriction indicator = presentation restricted</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">CONNECT;
Connected number
User provided, not screened Connected Party Number not present</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="11"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>XXXX</td><td>←</td><td></td><td>←</td><td>SIP_MESSAGE_VA</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>Conversation</td><td></td><td>Conversation</td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td></td><td></td></tr>
</table>

Values for test purpose IS_XXSSCOLP21	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	183 Session Progress

<table>
<tr><th>IS_XXSSCOLP22</th><th colspan="4">ISDN reference to:
EN 300 092-1 [i.14]
EN 300 403-1 [i.3]</th><th>NGN reference to:
EN 383 001 [49]
ES 283 027 [48]
TS 183 008 [44]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">Temporary mode presentation not restricted</td></tr>
<tr><td>ISDN selection criteria:</td><td colspan="5">COLP service has been requested by the calling party</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT on receipt of a provisional 1XX response defined as SIP_MESSAGE_VA with priv-value component is set to "user" has been received from the terminating user:
• P-Preferred-Identity header field is provided within the 1xx response
• Sends a CONNECT message with the Connected Party Number information element coded:
- Address signals = not available
- Numbering plan indicator = ISDN numbering plan
- Type of number = PIXIT
- Screening indicator = network provided
- Address presentation restriction indicator = presentation restricted</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">CONNECT;
Connected number
User provided, not screened Connected Party Number not present</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td></td><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td>XXXX</td><td>←</td><td></td><td>←</td><td>SIP_MESSAGE_VA</td></tr>
<tr><td></td><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td>Conversation</td><td></td></tr>
<tr><td></td><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td></td><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td></td><td>RLC</td><td>→</td><td></td><td></td><td></td></tr>
</table>

Values for test purpose IS_XXSSCOLP22	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	183 Session Progress

IS_XXSSCOLP23	**ISDN reference to:** **EN 300 092-1 [i.14]** **EN 300 403-1 [i.3]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]** **TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	Temporary mode presentation not restricted	
ISDN selection criteria:	COLP service has been requested by the calling party	
Test purpose:	Ensure that the SUT on receipt of a provisional 1XX response defined as SIP_MESSAGE_VA with priv-value component is set to "header" has been received from the terminating user: • P-Preferred-Identity header field is provided within the 1xx response • Sends a CONNECT message with the Connected Party Number information element coded: - Address signals = not available - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = presentation restricted	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	CONNECT; Connected number User provided, not screened Connected Party Number not present	
Comments:		

	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROC	←			
	XXXX	←		←	SIP_MESSAGE_VA
	CONN	←		←	200 OK INVITE
				→	ACK
		Conversation		Conversation	
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	RLC	→			

Values for test purpose IS_XXSSCOLP23	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	183 Session Progress

IS_XXSSCOLP24	**ISDN reference to:** **EN 300 092-1 [i.14]** **EN 300 403-1 [i.3]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]** **TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	Temporary mode presentation not restricted	
ISDN selection criteria:	COLP service has been requested by the calling party	
Test purpose:	Ensure that the SUT having **not** received a provisional 1XX response, on receipt of a 200 OK INVITE for this call **without** a Privacy header field: • P-Preferred-Identity header field is provided within the 200 OK response • Sends a CONNECT message with the Connected Party Number information element coded: - Address signals = default public user identity - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = presentation allowed	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	ANM; Connected number User provided, not screened Connected Party Number not present	
Comments:		

	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROC	←			
	CONN	←		←	200 OK INVITE
				→	ACK
		Conversation		Conversation	
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	RLC	→			

IS_XXSSCOLP25	**ISDN reference to:** **EN 300 092-1 [i.14]** **EN 300 403-1 [i.3]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]** **TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	Temporary mode presentation not restricted	
ISDN selection criteria:	COLP service has been requested	
Test purpose:	Ensure that the SUT on receipt of a 200 OK INVITE for this call with a Privacy header field was received with the value "none" has been received: • P-Preferred-Identity header field is provided within the 200 OK response • Sends a CONNECT message with the Connected Party Number information element coded: - Address signals = default public user identity - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = presentation allowed	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	CONNECT; Connected Party Number information element User provided, not screened Connected Party Number not present NoAS: NoA_VALUE	
Comments:		

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROC	←			
ALERTING	←		←	180 Ringing
CONN	←		←	200 OK INVITE
			→	ACK
	Conversation		Conversation	
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

IS_XXSSCOLP26	ISDN reference to: EN 300 092-1 [i.14] EN 300 403-1 [i.3]	NGN reference to: EN 383 001 [49] ES 283 027 [48] TS 183 008 [44]
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	Temporary mode presentation not restricted	
ISDN selection criteria:	COLP service has been requested	
Test purpose:	Ensure that the SUT on receipt of a 200 OK INVITE for this call with a Privacy header field was received with the value "PRIV_VALUE" has been received: • P-Preferred-Identity header field is provided within the 200 OK response • Sends a CONNECT message with the Connected Party Number information element coded: - Address signals = not available - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = presentation restricted	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	CONNECT; Connected Party Number information element User provided, not screened Connected Party Number not present	
Comments:		

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROC	←			
ALERTING	←		←	180 Ringing
CONN	←		←	200 OK INVITE
			→	ACK
	Conversation		Conversation	
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

Values for test purpose IS_XXSSCOLP26	
VA	PRIV_VALUE
VA_1	Id
VA_2	User
VA_3	Header

<table>
<tr><th>IS_XXSSCOLP27</th><th colspan="3">ISDN reference to:
EN 300 092-1 [i.14]
EN 300 403-1 [i.3]</th><th colspan="2">NGN reference to:
EN 383 001 [49]
ES 283 027 [48]
TS 183 008 [44]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">Temporary mode presentation restricted</td></tr>
<tr><td>ISDN selection criteria:</td><td colspan="5">COLP service has been requested by the calling party</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT on receipt of a provisional 1XX response defined as SIP_MESSAGE_VA with priv-value component is set to "none" has been received from the terminating user:
• P-Preferred-Identity header field is provided within the 1xx response
• Sends a CONNECT message with the Connected Party Number information element coded:
- Address signals = default public user identity
- Numbering plan indicator = ISDN numbering plan
- Type of number = PIXIT
- Screening indicator = network provided
- Address presentation restriction indicator = presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">CONNECT;
Connected number
User provided, not screened Connected Party Number not present</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="11"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>XXXX</td><td>←</td><td></td><td>←</td><td>SIP_MESSAGE_VA</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>Conversation</td><td></td><td>Conversation</td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td></td><td></td></tr>
</table>

Values for test purpose IS_XXSSCOLP27	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	183 Session Progress

IS_XXSSCOLP28	**ISDN reference to:** **EN 300 092-1 [i.14]** **EN 300 403-1 [i.3]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]** **TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	Temporary mode presentation restricted	
ISDN selection criteria:	COLP service has been requested by the calling party	
Test purpose:	Ensure that the SUT on receipt of a provisional 1XX response defined as SIP_MESSAGE_VA with priv-value component is set to "id" has been received from the terminating user: • P-Preferred-Identity header field is provided within the 1xx response • Sends a CONNECT message with the Connected Party Number information element coded: - Address signals = not available - Numbering plan indicator = ISDN numbering plan - Type of number = (PIXIT) - Screening indicator = network provided - Address presentation restriction indicator = presentation restricted	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	CONNECT; Connected number User provided, not screened; Connected Party Number not present	
Comments:		

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROC	←			
XXXX	←		←	SIP_MESSAGE_VA
CONN	←		←	200 OK INVITE
			→	ACK
	Conversation		Conversation	
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

Values for test purpose IS_XXSSCOLP28	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	183 Session Progress

IS_XXSSCOLP29	**ISDN reference to:** **EN 300 092-1 [i.14]** **EN 300 403-1 [i.3]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]** **TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	Temporary mode presentation restricted	
ISDN selection criteria:	COLP service has been requested by the calling party	
Test purpose:	Ensure that the SUT on receipt of a provisional 1XX response defined as SIP_MESSAGE_VA with priv-value component is set to "user" has been received from the terminating user: • P-Preferred-Identity header field is provided within the 1xx response • sends a CONNECT message with the Connected Party Number information element coded: - Address signals = not available - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = presentation restricted	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	CONNECT; Connected number User provided, not screened; Connected Party Number not present	
Comments:		

	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROC	←			
	XXXX	←		←	SIP_MESSAGE_VA
	CONN	←		←	200 OK INVITE
				→	ACK
		Conversation		Conversation	
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	RLC	→			

Values for test purpose IS_XXSSCOLP29	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	183 Session Progress

IS_XXSSCOLP30	**ISDN reference to:** **EN 300 092-1 [i.14]** **EN 300 403-1 [i.3]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]** **TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	Temporary mode presentation restricted	
ISDN selection criteria:	COLP service has been requested by the calling party	
Test purpose:	Ensure that the SUT on receipt of a provisional 1XX response defined as SIP_MESSAGE_VA with priv-value component is set to "header" has been received from the terminating user: • P-Preferred-Identity header field is provided within the 1xx response • sends a CONNECT message with the Connected Party Number information element coded: - Address signals = not available - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = presentation restricted	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	CONNECT; Connected number User provided, not screened Connected Party Number, number not present	
Comments:		

	ISDN		SUT		SIP
	SETUP	→		→	INVITE
	CALL PROC	←			
	XXXX	←		←	SIP_MESSAGE_VA
	CONN	←		←	200 OK INVITE
				→	ACK
		Conversation		Conversation	
	DISC	→		→	BYE
	REL	←		←	200 OK BYE
	RLC	→			

Values for test purpose IS_XXSSCOLP30	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	183 Session Progress

<table>
<tr><td>IS_XXSSCOLP31</td><td>ISDN reference to:
EN 300 092-1 [i.14]
EN 300 403-1 [i.3]</td><td colspan="3">NGN reference to:
EN 383 001 [49]
ES 283 027 [48]
TS 183 008 [44]</td></tr>
<tr><td>TSS reference:</td><td colspan="4">ISDN-SIP/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="4">Temporary mode presentation restricted</td></tr>
<tr><td>ISDN selection criteria:</td><td colspan="4">COLP service has been requested by the calling party</td></tr>
<tr><td>Test purpose:</td><td colspan="4">Ensure that the SUT having not received a provisional 1XX response, on receipt of a 200 OK INVITE for this call without a Privacy header field:
• P-Preferred-Identity header field is provided within the 1xx response
• sends a CONNECT message with the Connected Party Number information element coded:
- Address signals = not available
- Numbering plan indicator = ISDN numbering plan
- Type of number = PIXIT
- Screening indicator = network provided
- Address presentation restriction indicator = presentation restricted</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="4">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="4">CONNECT;
Connected number
User provided, not screened Connected Party Number not present</td></tr>
<tr><td>Comments:</td><td colspan="4"></td></tr>
</table>

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROC	←			
CONN	←		←	200 OK INVITE
			→	ACK
	Conversation		Conversation	
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

IS_XXSSCOLP32	**ISDN reference to:** **EN 300 092-1 [i.14]** **EN 300 403-1 [i.3]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]** **TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	Temporary mode presentation restricted	
ISDN selection criteria:	COLP service has been requested by the calling party	
Test purpose:	Ensure that the SUT having not received a provisional 1XX response, on receipt of a 200 OK INVITE for this call, a Privacy header field with the value "none" has been received: • P-Preferred-Identity header field is provided within the 200 OK response • sends a CONNECT message with the Connected Party Number information element coded: - Address signals = default public user identity - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = presentation allowed	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	CONNECT; Connected number Address signals = **default public user identity** User provided, not screened; Connected Party Number not present	
Comments:		

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROC	←			
CONN	←		←	200 OK INVITE
			→	ACK
	Conversation		Conversation	
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

<table>
<tr><th>IS_XXSSCOLP33</th><th colspan="4">ISDN reference to:
EN 300 092-1 [i.14]
EN 300 403-1 [i.3]</th><th>NGN reference to:
EN 383 001 [49]
ES 283 027 [48]
TS 183 008 [44]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">Temporary mode presentation restricted</td></tr>
<tr><td>ISDN selection criteria:</td><td colspan="5">COLP service has been requested by the calling party</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT having not received a provisional 1XX response, on receipt of a 200 OK INVITE for this call, a Privacy header field with the value "PRIV_VALUE" has been received:
• P-Preferred-Identity header field is provided within the 200 OK response
• sends a CONNECT message with the Connected Party Number information element coded:
- Address signals = not available
- Numbering plan indicator = ISDN numbering plan
- Type of number = PIXIT
- Screening indicator = network provided
- Address presentation restriction indicator = presentation restricted</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

Dial string parameters options=PIXIT
PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">CONNECT;
Connected number Address signals = not available
User provided, not screened Connected Party Number not present</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="11"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>Conversation</td><td></td><td>Conversation</td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td></td><td></td></tr>
</table>

Values for test purpose IS_XXSSCOLP33	
VA	PRIV_VALUE
VA_1	Id
VA_2	User
VA_3	Header

<table>
<tr><td>IS_XXSSCOLP34</td><td colspan="3">ISDN reference to:
EN 300 092-1 [i.14]
EN 300 403-1 [i.3]</td><td colspan="2">NGN reference to:
EN 383 001 [49]
ES 283 027 [48]
TS 183 008 [44]</td></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">Temporary mode presentation restricted</td></tr>
<tr><td>ISDN selection criteria:</td><td colspan="5">COLP service has been requested</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT on receipt of a 200 OK INVITE for this call with a Privacy header field was received with the value "none" has been received:
• P-Preferred-Identity header field is provided within the 200 OK response
• sends a CONNECT message with the Connected Party Number information element coded:
- Address signals = default public user identity
- Numbering plan indicator = ISDN numbering plan
- Type of number = PIXIT
- Screening indicator = network provided
- Address presentation restriction indicator = presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">COONECT;
Connected Party Number information element
User provided, not screened Connected Party Number not present
NoAS: NoA_VALUE</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="11"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>Conversation</td><td></td><td>Conversation</td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td></td><td></td></tr>
</table>

IS_XXSSCOLP35	**ISDN reference to:** **EN 300 092-1 [i.14]** **EN 300 403-1 [i.3]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]** **TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	Temporary mode presentation not restricted	
ISDN selection criteria:	COLP service has been requested	
Test purpose:	Ensure that the SUT on receipt of a 200 OK INVITE for this call with a Privacy header field was received with the value "PRIV_VALUE" has been received: • P-Preferred-Identity header field is provided within the 200 OK response • sends a CONNECT message with the Connected Party Number information element coded: - Address signals = not available - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = presentation restricted	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	CONNECT; Connected Party Number information element User provided, not screened Connected Party Number not present NoAS: NoA_VALUE	
Comments:		

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROC	←			
ALERTING	←		←	180 Ringing
CONN	←		←	200 OK INVITE
			→	ACK
	Conversation		Conversation	
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

Values for test purpose IS_XXSSCOLP35	
VA	PRIV_VALUE
VA_1	Id
VA_2	User
VA_3	Header

IS_XXSSCOLP36	**ISDN reference to:** **EN 300 092-1 [i.14]** **EN 300 403-1 [i.3]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]** **TS 183 008 [44]**
TSS reference:	ISDN-SIP/SS/COLP	
SIP selection criteria:	permanent mode	
ISDN selection criteria:	COLP service has been requested by the calling party	
Test purpose:	Ensure that the SUT on receipt of a provisional 1XX response defined as SIP_MESSAGE_VA with priv-value component is set to "none" has been received from the terminating user: • P-Preferred-Identity header field is provided within the 200 OK response • sends a CONNECT message with the Connected Party Number information element coded: - Address signals = default public user identity - Numbering plan indicator = ISDN numbering plan - Type of number = PIXIT - Screening indicator = network provided - Address presentation restriction indicator = presentation allowed	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	CONNECT; Connected number User provided, not screened Connected Party Number not present	
Comments:		

ISDN		SUT		SIP
SETUP	→		→	INVITE
CALL PROC	←			
XXXX	←		←	SIP_MESSAGE_VA
CONN	←		←	200 OK INVITE
			→	ACK
	Conversation		Conversation	
DISC	→		→	BYE
REL	←		←	200 OK BYE
RLC	→			

Values for test purpose IS_XXSSCOLP36	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	183 Session Progress

<table>
<tr><th>IS_XXSSCOLP37</th><th colspan="3">ISDN reference to:
EN 300 092-1 [i.14]
EN 300 403-1 [i.3]</th><th colspan="2">NGN reference to:
EN 383 001 [49]
ES 283 027 [48]
TS 183 008 [44]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">Permanent mode</td></tr>
<tr><td>ISDN selection criteria:</td><td colspan="5">COLP service has been requested by the calling party</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT having not received a provisional 1XX response, on receipt of a 200 OK INVITE for this call without a Privacy header field:
• P-Preferred-Identity header field is provided within the 200 OK response
• sends a CONNECT message with the Connected Party Number information element coded:
- Address signals = not available
- Numbering plan indicator = ISDN numbering plan
- Type of number = PIXIT
- Screening indicator = network provided
- Address presentation restriction indicator = presentation restricted</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">CONNECT;
Connected number
User provided, not screened Connected Party Number not present</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="10"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>Conversation</td><td></td><td>Conversation</td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><th>IS_XXSSCOLP38</th><th colspan="3">ISDN reference to:
EN 300 092-1 [i.14]
EN 300 403-1 [i.3]</th><th colspan="2">NGN reference to:
EN 383 001 [49]
ES 283 027 [48]
TS 183 008 [44]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">Permanent mode</td></tr>
<tr><td>ISDN selection criteria:</td><td colspan="5">COLP service has been requested by the calling party</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT on receipt of a 200 OK INVITE for this call with a Privacy header field was received with the value "none" has been received:
• P-Preferred-Identity header field is provided within the 200 OK response
• sends a CONNECT message with the Connected Party Number information element coded:
- Address signals = not available
- Numbering plan indicator = ISDN numbering plan
- Type of number = PIXIT
- Screening indicator = network provided
- Address presentation restriction indicator = presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">CONNECT;
Connected Party Number information element
User provided, not screened Connected Party Number not present
NoAS: NoA_VALUE</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="11"></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CALL PROC</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>CONN</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>Conversation</td><td></td><td>Conversation</td><td></td></tr>
<tr><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><td>IS_XXSSCOLP39</td><td colspan="3">ISDN reference to:
EN 300 092-1 [i.14],
EN 300 403-1 [i.3]</td><td colspan="2">NGN reference to:
EN 383 001 [49],
ES 283 027 [48],
TS 183 008 [44], TIP/TIR reference clause 4.5.2.12</td></tr>
<tr><td>TSS reference:</td><td colspan="5">ISDN-SIP/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">Temporary mode presentation not restricted
Terminated user has special arrangement</td></tr>
<tr><td>ISDN selection criteria:</td><td colspan="5">COLP service has been requested by the calling party</td></tr>
<tr><td>Test purpose:</td><td colspan="5">The Terminating UE sends an UPDATE request with an updated From and To header.

Ensure that if the UE receives a "from-change" tag in a Supported header in an initial INVITE, and the user equipment sends an UPDATE request after the ACK for the 200 OK INVITE containing a connected identity in the From header

• sends a CONNECT message with the Connected Party Number information element coded:
- Address signals = unscreened number
- Numbering plan indicator = ISDN numbering plan
- Type of number = PIXIT
- Screening indicator = network provided
- Address presentation restriction indicator = presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)

SIP messages: INVITE: Supported: from-change
18x/200: Supported: from-change
UPDATE: From <identity user equipment>)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">CONNECT;
Connected number
User provided, not screened Connected Party Number not present</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td></td><td>ISDN</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td></td><td>SETUP</td><td>→</td><td></td><td>→</td><td>INVITE with "from-change" tag</td></tr>
<tr><td></td><td>ALERTING</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td></td><td></td><td></td><td></td><td>→</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td>←</td><td>UPDATE
with updated From and To header</td></tr>
<tr><td></td><td></td><td></td><td></td><td>→</td><td>200 OK UPDATE</td></tr>
<tr><td></td><td>CONN</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td>Conversation</td><td></td></tr>
<tr><td></td><td>DISC</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td></td><td>REL</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
<tr><td></td><td>RLC</td><td>→</td><td></td><td></td><td></td></tr>
</table>

6.1.2.4 CFU

6.1.2.4.1 CFU - ISI

ISI_XXSSCFU 01	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU. User C is point-to-multipoint	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←											
					INVITE	←									
			INVITE	←											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→	BYE	→	BYE	→							
DISC (UE2)	←		BYE	←	BYE	←	BYE	←							
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFU 02	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is provided with CFU ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = Yes). User C is point-to-multipoint	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
							181	←	MESSAGE	→					
					181	←					MESSAGE	→			
NOTIFY (UE 1)	←		181	←											
SETUP (UE 2)	←		INVITE	←											
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFU 03	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no). User C is point-to-multipoint	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed).User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←							
					INVITE	←									
SETUP (UE 2)	←		INVITE	←											
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFU 04	**ISDN reference to:** **EN 300 207-1 [i.5]** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFU The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no). User C is point-to-multipoint	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is not notified of call diversion and not informed of the diverted-to number (user C has COLP) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
					INVITE	←									
			INVITE	←											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFU 05	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFU	
Configuration:	The user B is in network N2 and is provided with CFU. User C is point-to-multipoint	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported user C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFU was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←							
					INVITE	←									
			INVITE	←											
SETUP (UE 2)	←														
RLC (UE 2)	→		486 Busy here	→	486 Busy here	→	486 Busy here	→							
DISC (UE 1)	←		486 Busy here	←	486 Busy here	←	486 Busy here	←							
REL	→		ACK	→	ACK	→	ACK	→							
RLC	←		ACK	←	ACK	←	ACK	←							

ISI_XXSSCFU 06	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFU	
Configuration:	The user B is in network N2 and is provided with CFU. User C is point-to-multipoint	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported user C is network determined user busy	
Test purpose:	To verify that a call is released correctly if CFU was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←							
					INVITE	←									
			INVITE	←											
			486 Busy here	→	486 Busy here	→	486 Busy here	→							
DISC (UE 1)	←		486 Busy here	←	486 Busy here	←	486 Busy here	←							
REL	→		ACK	→	ACK	→	ACK	→							
RLC	←		ACK	←	ACK	←	ACK	←							

ISII_XXSSCFU 07	**ISDN reference to: EN 300 207-1 [i.5], clause 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFU. User D forwards the call to back to user B. User C is point-to-multipoint	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C and D. User D forwards the call to back to user B. Ensure that the call is released.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SII_XXSSCFU 08	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFU. User D forwards the call. User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported Network option: maximal number of diversions for a single call N= 3	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, C to D. User D has call forwarding activated. Ensure that the call is released.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

6.1.2.4.2 CFU - ISS

ISS_XXSSCFU 01	**ISDN reference to: EN 300 207-1 [i.5], clause 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFU	
Configuration:	The user B is provided with CFU	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
							181	←							
					181	←									
NOTIFY	←		181	←					INVITE	→			INVITE	→	
													100 Trying	←	
									180	←			180	←	
							180	→							
					180	←	180	←							
ALERTING	←		180	←											
									200 OK	←			200 OK	←	
							200 OK	→							
							200 OK	←							
					200 OK	←									
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→									
							BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISS_XXSSCFU 02	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFU	
Configuration:	The user B is provided with CFU ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = Yes).	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no TIR) and user C is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←	MESSAGE	→					
							100 Trying	→			MESSAGE	→			
							181	←							
					181	←									
NOTIFY	←		181	←					INVITE	→			INVITE	→	
													100 Trying	←	
									100 Trying	←					
									180	←			180	←	
							180	→							
					180	←	180	←							
ALERTING	←		180	←											
									200 OK	←			200 OK	←	
							200 OK	→							
							200 OK	←							
					200 OK	←									
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→									
							BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISS_XXSSCFU 03	**ISDN reference to:** **EN 300 207-1 [i.5]** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFU	
Configuration:	User B is provided with CFU ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = No).	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - TIR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
							181	←							
					181	←									
NOTIFY	←		181	←					INVITE	→			INVITE	→	
													100 Trying	←	
									100 Trying	←					
									180	←			180	←	
							180	→							
					180	←	180	←							
ALERTING	←		180	←											
									200 OK	←			200 OK	←	
							200 OK	→							
							200 OK	←							
					200 OK	←									
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→									
							BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISS_XXSSCFU 04	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFU	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFU The user A and the user C are in network N1 and user C is provided with COLP. The user B is in network N2 and is provided with CFU ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no).	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→			INVITE	→	
													100 Trying	←	
									100 Trying	←					
									180	←			180	←	
							180	→							
					180	←	180	←							
ALERTING	←		180	←											
									200 OK	←			200 OK	←	
							200 OK	→							
							200 OK	←							
					200 OK	←									
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→									
							BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISS_XXSSCFU 05	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFU	
Configuration:	The user B is in network N2 and is provided with CFU	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	To verify that a call is released correctly if **CFU was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
							181	←							
					181	←	INVITE			→					
NOTIFY	←		181	←									INVITE	→	
									486	←			486	←	
							486	→					ACK	→	
					486	←	486	←							
DISC # 17	←		486	←											
REL	→		ACK	→											
RLC	←				ACK	→									
							ACK	→							
							ACK	←							
									ACK	→					

ISS_XXSSCFU 06	ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5	NGN reference to: TS 183 004 [45]
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFU	
Configuration:	The user B is in network N2 and is provided with CFU	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	To verify that a call is released correctly if CFU **was not successful**. User A calls user B, the call is forwarded to user C who is **network determined user busy.**	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
							INVITE			→					
									486	←					
							486	→							
					486	←	486	←							
DISC # 17	←		486	←											
REL	→		ACK	→											
RLC	←				ACK	→									
							ACK	→							
							ACK	←							
									ACK	→					

ISSI_XXSSCFU 09	ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5	NGN reference to: TS 183 004 [45]
TSS reference:	ISDN-SIP-SIP-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU. User D in network N2 forwards the call to back to user B.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C and then forwarded to user D. User D forwards the call to back to user B. Ensure that the call is released.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISSI_XXSSCFU 10	ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5	NGN reference to: TS 183 004 [45]
TSS reference:	ISDN-SIP-SIP-ISDN/Supplementary_services/CFU	
Configuration:	The user is A in network N1. The user B and the user C are in network N2. User B is provided with CFU. User D forwards the call.	
Selection criteria:	Network option: maximal number of diversions for a single call N=3	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, C to D. User D has call forwarding activated Ensure that the call is released.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

6.1.2.5 CFB

6.1.2.5.1 CFB - ISI

ISI_XXSSCFB 01	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFB (network determined). User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network Call forwarding busy supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→					
									486	←					
							486	→							
							ACK	←							
									ACK	→					
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←							
							100 Trying	→							
					INVITE	←									
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFB 02	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFB (user determined). User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network Call forwarding busy supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
									100 Trying	←					
											INVITE	→			
									486	←	486	←			
							486	→							
							ACK	←							
									ACK	→	ACK	→			
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←							
							100 Trying	→							
					INVITE	←									
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFB 03	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is provided with CFB ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = Yes). User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network Call forwarding busy supported (user determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
									100 Trying	←					
											INVITE	→			
									486	←	486	←			
							486	→							
							ACK	←							
									ACK	→	ACK	→			
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←	MESSAGE	→					
							100 Trying	→			MESSAGE	→			
					INVITE	←									
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFB 04	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is provided with CFB ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = No). User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network Call forwarding busy supported (user determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number (user B has presentation allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF			I-CSCF		S-CSCF		AS		P-CSCF		UE-B	UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
									100 Trying	←					
											INVITE	→			
									486	←	486	←			
							486	→							
							ACK	←							
									ACK	→	ACK	→			
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←							
							100 Trying	→							
					INVITE	←									
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFB 05	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFB ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = No). User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network Call forwarding Busy supported (user determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
									100 Trying	←					
											INVITE	→			
									486	←	486	←			
							486	→							
							ACK	←							
									ACK	→	ACK	→			
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←							
							100 Trying	→							
					INVITE	←									
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFB 06	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFB The user A and the user C are in network N1 and user C is provided with COLP. The user B is in network N2 and is provided with CFB ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no). User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network Call forwarding busy supported (user determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B	UE-C
SETUP (UE 1)	→	INVITE	→	INVITE	→								
				100 Trying	←								
						INVITE	→						
						100 Trying	←						
						INVITE	←						
						100 Trying	→						
								INVITE	→				
								100 Trying	←				
										INVITE	→		
								486	←	486	←		
						486	→						
						ACK	←						
								ACK	→	ACK	→		
						INVITE	←						
						100 Trying	→						
				INVITE	←								
				100 Trying	→								
		INVITE	←										
		100 Trying	→										
SETUP (UE 2)	←												
ALERTING (UE 2)	→	180	→										
				180	→	180	→						
				180	←	180	←						
ALERTING (UE 1)	←	180	←										
CONNECT (UE 2)	→	200 OK	→										
				200 OK	→	200 OK	→						
				200 OK	←	200 OK	←						
CONNECT (UE 1)	←	200 OK	←										
		ACK	→	ACK	→	ACK	→						
		ACK	←	ACK	←	ACK	←						
DISC (UE1)	→	BYE	→										
				BYE	→	BYE	→						
				BYE	←	BYE	←						
DISC (UE2)	←	BYE	←										
REL (UE2)	→	200 OK BYE	→										
RLC (UE2)	←			200 OK BYE	→	200 OK BYE	→						
				200 OK BYE	←	200 OK BYE	←						
REL (UE1)	←	200 OK BYE	←										
RLC (UE1)	→												

ISI_XXSSCFB 07	ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5	NGN reference to: TS 183 004 [45]
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is provided with CFB ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = Yes). User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network Call forwarding busy supported (network determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B	UE-C
SETUP (UE 1)	→	INVITE	→	INVITE	→								
				100 Trying	←								
						INVITE	→						
						100 Trying	←						
						INVITE	←						
						100 Trying	→						
								INVITE	→				
								486	←				
						486	→						
						ACK	←						
								ACK	→				
						181	←						
				181	←								
NOTIFY (UE 1)	←	181	←			INVITE	←	MESSAGE	→				
						100 Trying	→			MESSAGE	→		
				INVITE	←								
				100 Trying	→								
		INVITE	←										
		100 Trying	→										
SETUP (UE 2)	←												
ALERTING (UE 2)	→	180	→										
				180	→	180	→						
				180	←	180	←						
ALERTING (UE 1)	←	180	←										
CONNECT (UE 2)	→	200 OK	→										
				200 OK	→	200 OK	→						
				200 OK	←	200 OK	←						
CONNECT (UE 1)	←	200 OK	←										
		ACK	→	ACK	→	ACK	→						
		ACK	←	ACK	←	ACK	←						
DISC (UE1)	→	BYE	→										
				BYE	→	BYE	→						
				BYE	←	BYE	←						
DISC (UE2)	←	BYE	←										
REL (UE2)	→	200 OK BYE	→										
RLC (UE2)	←			200 OK BYE	→	200 OK BYE	→						
				200 OK BYE	←	200 OK BYE	←						
REL (UE1)	←	200 OK BYE	←										
RLC (UE1)	→												

ISI_XXSSCFB 08	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is provided with CFB ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = No). User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network Call forwarding busy supported (network determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
									486	←					
							486	→							
							ACK	←							
									ACK	→					
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←							
							100 Trying	→							
					INVITE	←									
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFB 09	**ISDN reference to: EN 300 207-1 [i.5], clause 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFB ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = No). User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network Call forwarding Busy supported (network determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
									486	←					
							486	→							
							ACK	←							
									ACK	→					
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←							
							100 Trying	→							
					INVITE	←									
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

SI_XXSSCFB 10	ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5	NGN reference to: TS 183 004 [45]
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFB The user A and the user C are in network N1 and user C is provided with COLP. The user B is in network N2 and is provided with CFB ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no). User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network Call forwarding busy supported (network determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
									100 Trying	←					
									486	←					
							486	→							
							ACK	←							
									ACK	→					
							INVITE	←							
							100 Trying	→							
					INVITE	←									
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFB 12	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFB	
Configuration:	The user B is in network N2 and is provided with CFB. User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network Call forwarding busy supported user C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFB was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN			MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
									100 Trying	←					
											INVITE	→			
											486	←			
									486	←					
							486	→							
							ACK	←							
									ACK	→	ACK	→			
							INVITE	←							
							100 Trying	→							
					INVITE	←									
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
RLC (UE 2)	→		486 Busy here	→											
					486 Busy here	→	486 Busy here	→							
					ACK	←	ACK	←							
			ACK	←											
					486 Busy here	←	486 Busy here	←							
DISC (UE 1)	←		486 Busy here	←											
REL	→		ACK	→											
RLC	←				ACK	→	ACK	→							

ISI_XXSSCFB 13	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFB	
Configuration:	The user B is in network N2 and is provided with CFB. User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network Call forwarding busy supported user C is network determined user busy	
Test purpose:	To verify that a call is released correctly if CFB was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
									100 Trying	←					
											INVITE	→			
											486	←			
									486	←					
							486	→							
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←							
							100 Trying	→							
					INVITE	←									
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
			486 Busy here	→											
					486 Busy here	→									
					ACK	←									
			ACK	←											
					486 Busy here	←									
DISC (UE 1)	←		486 Busy here	←											
REL	→		ACK	→											
RLC	←				ACK	→									

6.1.2.5.1 CFB - ISS

ISS_XXSSCFB 01	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFB	
Configuration:	The user B is provided with CFB	
Selection criteria:	Call forwarding by the network Call forwarding busy supported (user determined)	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→	INVITE	→			
									486	←	486	←			
							486	→			ACK	→			
							ACK	←							
									ACK	→					
NOTIFY (UE 1)	←		181	←	181	←	181	←							
							INVITE	←							
							INVITE	→					INVITE	→	
									180	←			180	←	
							180	→							
ALERTING	←		180	←	180	←	180	←							
													200 OK	←	
									200 OK	←					
							200 OK	→							
CONNECT	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
							ACK	←							
DISC (UE1)	→		BYE	→					ACK	→			ACK	→	
					BYE	→	BYE	→							
							BYE	←							
									BYE	→					
													BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISS_XXSSCFB 02	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFB	
Configuration:	The user B is provided with CFB ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = Yes).	
Selection criteria:	Call forwarding by the network Call forwarding busy supported (user determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no TIR) and user C is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→	INVITE	→			
									486	←	486	←			
							486	→			ACK	→			
							ACK	←							
									ACK	→					
NOTIFY (UE 1)	←		181	←	181	←	181	←							
							INVITE	←							
							INVITE	→	MESSAGE	→					
											MESSAGE	→			
													INVITE	→	
							180	→	180	←			180	←	
ALERTING	←		180	←	180	←	180	←					200 OK	←	
									200 OK	←					
							200 OK	→							
CONNECT	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
							ACK	←							
DISC (UE1)	→		BYE	→					ACK	→			ACK	→	
					BYE	→	BYE	→							
							BYE	←							
									BYE	→					
													BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISS_XXSSCFB 03	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFB	
Configuration:	User B is provided with CFB ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no).	
Selection criteria:	Call forwarding by the network Call forwarding busy supported (user determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - TIR) and user C is not informed of the forwarding number (user B has presentation not allowed).User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→	INVITE	→			
									486	←	486	←			
							486	→			ACK	→			
							ACK	←							
									ACK	→					
NOTIFY (UE 1)	←		181	←	181	←	181	←							
							INVITE	←							
							INVITE	→					INVITE	→	
									180	←			180	←	
							180	→							
ALERTING	←		180	←	180	←	180	←							
													200 OK	←	
									200 OK	←					
							200 OK	→							
CONNECT	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
							ACK	←							
DISC (UE1)	→		BYE	→					ACK	→			ACK	→	
					BYE	→	BYE	→							
							BYE	←							
									BYE	→					
													BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISS_XXSSCFB 04	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFB. The user A and the user C are in network N1. ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no).	
Selection criteria:	Call forwarding by the network Call forwarding busy supported (user determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number (user B has presentation not allowed).User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→	INVITE	→			
									486	←	486	←			
							486	→			ACK	→			
							ACK	←							
									ACK	→					
NOTIFY (UE 1)	←		181	←	181	←	181	←							
							INVITE	←							
							INVITE	→					INVITE	→	
									180	←			180	←	
							180	→							
ALERTING	←		180	←	180	←	180	←							
													200 OK	←	
									200 OK	←					
							200 OK	→							
CONNECT	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
							ACK	←							
DISC (UE1)	→		BYE	→					ACK	→			ACK	→	
					BYE	→	BYE	→							
							BYE	←							
									BYE	→					
													BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISS_XXSSCFB 05	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFB	
Configuration:	The user B is provided with CFB	
Selection criteria:	Call forwarding by the network Call forwarding busy supported (network determined)	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN			MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C
SETUP	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→					
									486	←					
							486	→							
							ACK	←							
									ACK	→					
NOTIFY (UE 1)	←		181	←	181	←	181	←							
							INVITE	←							
							INVITE	→					INVITE	→	
									180	←			180	←	
							180	→							
ALERTING	←		180	←	180	←	180	←							
													200 OK	←	
									200 OK	←					
							200 OK	→							
CONNECT	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
							ACK	←							
DISC (UE1)	→		BYE	→					ACK	→			ACK	→	
					BYE	→	BYE	→							
							BYE	←							
									BYE	→					
													BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISS_XXSSCFB 06	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFB	
Configuration:	The user B is provided with CFB ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = Yes).	
Selection criteria:	Call forwarding by the network Call forwarding busy supported (network determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no TIR) and user C is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→					
									486	←					
							486	→							
							ACK	←							
									ACK	→					
NOTIFY (UE 1)	←		181	←	181	←	181	←							
							INVITE	←	MESSAGE	→					
							INVITE	→			MESSAGE	→			
							180	→					INVITE	→	
ALERTING	←		180	←	180	←	180	←	180	←			180	←	
													200 OK	←	
									200 OK	←					
							200 OK	→							
CONNECT	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
							ACK	←							
DISC (UE1)	→		BYE	→					ACK	→			ACK	→	
					BYE	→	BYE	→							
							BYE	←							
									BYE	→					
													BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISS_XXSSCFB 07	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFB	
Configuration:	User B is provided with CFB ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no).	
Selection criteria:	Call forwarding by the network Call forwarding busy supported (network determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - TIR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→	INVITE	→			
									486	←	486	←			
							486	→			ACK	→			
							ACK	←							
									ACK	→					
NOTIFY (UE 1)	←		181	←	181	←	181	←							
							INVITE	←							
							INVITE	→					INVITE	→	
									180	←			180	←	
							180	→							
ALERTING	←		180	←	180	←	180	←							
													200 OK	←	
									200 OK	←					
							200 OK	→							
CONNECT	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
							ACK	←							
DISC (UE1)	→		BYE	→					ACK	→			ACK	→	
					BYE	→	BYE	→							
							BYE	←							
									BYE	→					
													BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISS_XXSSCFB 08	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFB. The user A and the user C are in network N1. ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no).	
Selection criteria:	Call forwarding by the network Call forwarding busy supported (network determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→					
									486	←					
							486	→							
							ACK	←							
									ACK	→					
							INVITE	←							
							INVITE	→					INVITE	→	
									180	←			180	←	
							180	→							
ALERTING	←		180	←	180	←	180	←							
													200 OK	←	
									200 OK	←					
							200 OK	→							
CONNECT	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
							ACK	←							
DISC (UE1)	→		BYE	→					ACK	→			ACK	→	
					BYE	→	BYE	→							
							BYE	←							
									BYE	→					
													BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISS_XXSSCFB 09	**ISDN reference to: EN 300 207-1 [i.5] clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFB. The user A and the user C are in network N1. The user B is in network N2 and is provided with CFB ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No, "diverting number is released to the diverted-to user" = No).	
Selection criteria:	User B has activated the CALL DEFLECTION service Call forwarding busy supported performed by the UE CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number (user B has presentation not allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
									100 Trying	←					
											INVITE	→			
											100 Trying	←			
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
							INVITE	←							
							100 Trying	→							
									INVITE	→			INVITE	→	
													100 Trying	←	
													180	←	
									180	←					
							180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
													200 OK	←	
									200 OK	←					
							200 OK	→							
							200 OK	←							
					200 OK	←									
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC	→		BYE	→											
					BYE	→									
							BYE	→							
							BYE	←							
									BYE	→			BYE	→	
									200 OK BYE	←			200 OK BYE	←	
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL	←		200 OK BYE	←											
RLC	→														

ISS_XXSSCFB 10	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFB	
Configuration:	The user B is in network N2 and is provided with CFB	
Selection criteria:	Call forwarding by the network Call forwarding busy supported	
Test purpose:	To verify that a call is released correctly if **CFB was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→					
									486	←					
							486	→							
							ACK	←							
									ACK	→					
							INVITE	←							
									INVITE	→					
													INVITE	→	
													486	←	
									486	←			ACK	→	
							486	→							
DISC # 17	←		486	←	486	←	486	←							
REL	→		ACK	→	ACK	→	ACK	→							
RLC	←						ACK	←							
									ACK	→					

ISS_XXSSCFB 11	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFB	
Configuration:	The user B is in network N2 and is provided with CFB	
Selection criteria:	Call forwarding by the network Call forwarding busy supported	
Test purpose:	To verify that a call is released correctly if CFB **was not successful**. User A calls user B, the call is forwarded to user C who is **network determined user busy**.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→					
									486	←					
							486	→							
							ACK	←							
									ACK	→					
							INVITE	←							
									INVITE	→					
									486	←					
							486	→							
DISC # 17	←		486	←	486	←	486	←							
REL	→		ACK	→	ACK	→	ACK	→							
RLC	←						ACK	←							
									ACK	→					

ISSI_XXSSCFB 14	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU. User D forwards the call to back to user B.	
Selection criteria:	Call forwarding busy by the network Call forwarding busy supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C and D. User D forwards the call to back to user B. Ensure that the call is released.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISSI_XXSSCFB 15	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: ES 283 027 [48]**
TSS reference:	ISDN-SIP-SIP-ISDN/Supplementary_services/CFU	
Configuration:	The user is A in network N1. The user B and the user C are in network N2. User B is provided with CFU. User E forwards the call to back to user B.	
Selection criteria:	Network option: hop counter supported N<5	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, C to D. User D forwards the call to back to user B. User D forwards the call to back to user B. Ensure that the call is released.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

6.1.2.6 CFNR

6.1.2.6.1 CFNR - ISI

ISI_XXSSCFNR01	**ISDN reference to:** **EN 300 403-1 [i.3],** **clauses 9.2.2, 9.2.4.4, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFNR/	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR. User C is point-to-multipoint. Served user communication retention on invocation of diversion (forwarding or deflection) = No [Clear call to the served user on invocation of call diversion]	
Selection criteria:	CFNR supported	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
							180	←							
					180	←									
			180	←											
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←											
							CANCEL	←							
							CANCEL	→							
											CANCEL	→			
											487 Request terminated	←			
									487 Request terminated	←					
							487 Request terminated	→							
							ACK	←							
									ACK	→					
											ACK	→			
							INVITE	←							
					INVITE	←									
					100 Trying	→									
SETUP (UE 2)	←		INVITE	←											
			100 Trying	→											
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→	200 OK	→	200 OK	→							
CONNECT (UE 1)	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→	BYE	→	BYE	→							
DISC (UE2)	←		BYE	←	BYE	←	BYE	←							
REL (UE2)	→		200 OK BYE	→											

RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFNR02	**ISDN reference to: EN 300 403-1 [i.3], clauses 9.2.2, 9.2.4.4, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFNR/	
Configuration:	The user A and the user C are in network N1. The user B is provided with CFNR ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = Yes, Served user communication retention on invocation of diversion (forwarding or deflection) = No [Clear call to the served user on invocation of call diversion]. User C is point-to-multipoint.	
Selection criteria:	CFNR supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C. User A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
							180	←							
					180	←									
			180	←											
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←											
							CANCEL	←							
							CANCEL	→							
											CANCEL	→			
											487 Request terminated	←			
									487 Request terminated	←					
							487 Request terminated	→							
							ACK	←							
									ACK	→					
											ACK	→			
							INVITE	←	MESSAGE	→					
					INVITE	←					MESSAGE	→			
					100 Trying	→									
SETUP (UE 2)	←		INVITE	←											
			100 Trying	→											
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→	200 OK	→	200 OK	→							

CONNECT (UE 1)	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→	BYE	→	BYE	→							
DISC (UE2)	←		BYE	←	BYE	←	BYE	←							
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFNR 03	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no) Served user communication retention on invocation of diversion (forwarding or deflection) = No. User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network Call forwarding not reply supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF			I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
							180	←							
					180	←									
			180	←											
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←											
							CANCEL	←							
							CANCEL	→							
											CANCEL	→			
											487 Request terminated	←			
									487 Request terminated	←					
							487 Request terminated	→							
							ACK	←							
									ACK	→					
											ACK	→			
							INVITE	←							

					INVITE	←									
					100 Trying	→									
SETUP (UE 2)	←		INVITE	←											
			100 Trying	→											
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→	200 OK	→	200 OK	→							
CONNECT (UE 1)	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→	BYE	→	BYE	→							
DISC (UE2)	←		BYE	←	BYE	←	BYE	←							
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFNR 04	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFNR	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFNR. The user A and the user C are in network N1 and user C is provided with COLP. The user B is in network N2 and is provided with CFNR ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no); Served user communication retention on invocation of diversion (forwarding or deflection) = No. User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network Call forwarding not reply supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF			I-CSCF		S-CSCF		AS		P-CSCF		UE-B	UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
							180	←							
					180	←									
			180	←											
ALERTING (UE1)	←						CANCEL	←							
							CANCEL	→							
											CANCEL	→			
											487 Request terminated	←			
									487 Request terminated	←					
							487 Request terminated	→							
							ACK	←							
									ACK	→					
											ACK	→			
							INVITE	←							
					INVITE	←									
					100 Trying	→									
SETUP (UE 2)	←		INVITE	←											
			100 Trying	→											
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
			180	←											
CONNECT (UE 2)	→		200 OK	→	200 OK	→	200 OK	→							
CONNECT (UE 1)	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→	BYE	→	BYE	→							
DISC (UE2)	←		BYE	←	BYE	←	BYE	←							
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFNR 05	ISDN reference to: EN 300 403-1 [i.3], clauses 9.2.2, 9.2.4.4, 9.2.5	NGN reference to: TS 183 004 [45]
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFNR/	
Configuration:	The user A and the user C are in network N1. The user B is provided with CFNR Served user communication retention on invocation of diversion (forwarding or deflection) = Yes. User C is point-to-multipoint.	
Selection criteria:	CFNR supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C The diverting user accepts the communication after sending the INVITE request, the communication path towards the diverted to user shall be released according to the rules and procedures in RFC 3261 [28].	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF			I-CSCF		S-CSCF		AS		P-CSCF		UE-B	UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
											180	←			
									180	←					
							180	→							
							180	←							
					180	←									
ALERTING (UE1)		←	180	←											
NOTIFY (UE 1)	←		181	←	181	←	181	←							
					INVITE	←	INVITE	←							
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→	180	→	180	→							
							CANCEL	←							
							CANCEL	→							
											CANCEL	→			
											487 Request terminated	←			
									487 Request terminated	←					
							487 Request terminated	→							
							ACK	←							
									ACK	→					
											ACK	→			
			180	←	180	←	180	←							
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFNR 06	**ISDN reference to: EN 300 403-1 [i.3], clauses 9.2.2, 9.2.4.4, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFNR/	
Configuration:	The user A and the user C are in network N1. The user B is provided with CFNR ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = Yes, Served user communication retention on invocation of diversion (forwarding or deflection) = Yes. User C is point-to-multipoint.	
Selection criteria:	CFNR supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C The diverting user accepts the communication after sending the INVITE request, the communication path towards the diverted to user shall be released according to the rules and procedures in RFC 3261 [28]. User A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion	
ISDN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
											180	←			
									180	←					
							180	→							
							180	←							
					180	←									
ALERTING (UE1)		←	180	←											
NOTIFY (UE 1)	←		181	←	181	←	181	←							
					INVITE	←	INVITE	←	MESSAGE	→					
					100 Trying	→					MESSAGE	→			
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→	180	→	180	→							
							CANCEL	←							
							CANCEL	→			CANCEL	→			
											487 Request terminated	←			
									487 Request terminated	←					
							487 Request terminated	→							
							ACK	←							
									ACK	→					
											ACK	→			
			180	←	180	←	180	←							
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFNR 07	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no) Served user communication retention on invocation of diversion (forwarding or deflection) = Yes. User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network Call forwarding not reply supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C The diverting user accepts the communication after sending the INVITE request, the communication path towards the diverted to user shall be released according to the rules and procedures in RFC 3261 [28]. User A is notified of call diversion and informed of the diverted-to number and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
							180	←							
ALERTING (UE1)		←	180	←	180	←									
NOTIFY (UE 1)	←		181	←	181	←	181	←							
					INVITE	←	INVITE	←							
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→	180	→	180	→							
							CANCEL	←							
							CANCEL	→							
											CANCEL	→			
											487 Request terminated	←			
									487 Request terminated	←					
							487 Request terminated	→							
							ACK	←							
									ACK	→	ACK	→			
			180	←	180	←	180	←							
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFNR 08	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFNR	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFNR The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no); Served user communication retention on invocation of diversion (forwarding or deflection) = Yes. User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network Call forwarding not reply supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C The diverting user accepts the communication after sending the INVITE request, the communication path towards the diverted to user shall be released according to the rules and procedures in RFC 3261 [28].User A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
							180	←							
ALERTING (UE1)		←	180	←	180	←									
NOTIFY (UE 1)	←		181	←	181	←	181	←							
					INVITE	←	INVITE	←							
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→	180	→	180	→							
							CANCEL	←							
							CANCEL	→							
											CANCEL	→			
											487 Request terminated	←			
									487 Request terminated	←					
							487 Request terminated	→							
							ACK	←							
									ACK	→	ACK	→			
			180	←	180	←	180	←							
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											

REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFNR 09	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR, Served user communication retention on invocation of diversion (forwarding or deflection) = No [Clear call to the served user on invocation of call diversion]. User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported user C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFNR was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
ALERTING		←	180	←	180	←	180	←							
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←											
							CANCEL	←							
							CANCEL	→							
											CANCEL	→			
											487 Request terminated	←			
									487 Request terminated	←					
							487 Request terminated	→							
							ACK	←							
									ACK	→					
											ACK	→			
					INVITE	←	INVITE	←							
					100 Trying	→	100 Trying	→							
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
RLC (UE 2)	→		486 Busy here	→	486 Busy here	→	486 Busy here	→							
DISC (UE 1)	←		486 Busy here	←	486 Busy here	←	486 Busy here	←							
REL (UE1)	→		ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							

ISI_XXSSCFNR 10	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR, Served user communication retention on invocation of diversion (forwarding or deflection) = No. User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network CFNR supported user C is network determined user busy	
Test purpose:	To verify that a call is released correctly if CFNR was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
ALERTING		←	180	←	180	←	180	←							
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←											
							CANCEL	←							
							CANCEL	→							
											CANCEL	→			
											487 Request terminated	←			
									487 Request terminated	←					
							487 Request terminated	→							
							ACK	←							
									ACK	→					
											ACK	→			
					INVITE	←	INVITE	←							
					100 Trying	→	100 Trying	→							
			INVITE	←											
			100 Trying	→											
			486 Busy here	→	486 Busy here	→	486 Busy here	→							
DISC (UE 1)	←		486 Busy here	←	486 Busy here	←	486 Busy here	←							
REL (UE1)	→		ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							

ISI_XXSSCFNR 11	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR, Served user communication retention on invocation of diversion (forwarding or deflection) = Yes. User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported user C is user determined user busy	
Test purpose:	User A calls user B, the call is forwarded to user C who is **user** determined user busy. The forwarding user User B continues to alert.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→	INVITE	→			
									180	←	180	←			
							180	→							
ALERTING		←	180	←	180	←	180	←							
					INVITE	←	INVITE	←							
SETUP (UE 2)	←		INVITE	←											
RLC (UE 2)	→		486 Busy here	→	486 Busy here	→	486 Busy here	→							
			ACK	←	ACK	←	ACK	←							
									200 OK	←	200 OK	←			
							200 OK	→							
CONNECT (UE 1)	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→	ACK	→			
DISC (UE1)	→		BYE	→	BYE	→	BYE	→							
							BYE	←							
									BYE	→	BYE	→			
									200 OK BYE	←	200 OK BYE	←			
							200 OK BYE	→							
REL (UE1)	←		200 OK BYE	←	200 OK BYE	←	200 OK BYE	←							
RLC (UE1)	→														

ISI_XXSSCFNR 12	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR, Served user communication retention on invocation of diversion (forwarding or deflection) = Yes. User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network CFNR supported user C is network determined user busy	
Test purpose:	User A calls user B, the call is forwarded to user C who is **network** determined user busy. The forwarding user User B continues to alert.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→	INVITE	→			
									180	←	180	←			
							180	→							
ALERTING		←	180	←	180	←	180	←							
					INVITE	←	INVITE	←							
			INVITE	←											
			486 Busy here	→	486 Busy here	→	486 Busy here	→							
			ACK	←	ACK	←	ACK	←							
									200 OK	←	200 OK	←			
							200 OK	→							
CONNECT (UE 1)	←		200 OK	←	200 OK	←	200 OK	←							
DISC (UE1)	→		BYE	→	BYE	→	BYE	→							
							BYE	←							
									BYE	→	BYE	→			
									200 OK BYE	←	200 OK BYE	←			
							200 OK BYE	→							
REL (UE1)	←		200 OK BYE	←	200 OK BYE	←	200 OK BYE	←							
RLC (UE1)	→														

ISII_XXSSCFNR 13	ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5	NGN reference to: TS 183 004 [45]
TSS reference:	ISDN-SIP-ISDN-ISDN/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR. User D forwards the call to back to user B. User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network CFNR supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C and D. User D forwards the call to back to user B. Ensure that the call is released.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

6.1.2.6.2 CFNR - ISS

ISS_XXSSCFNR 01	**ISDN reference to:** **EN 300 207-1 [i.5]** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	The user B is provided with CFNR Served user communication retention on invocation of diversion (forwarding or deflection) = No.	
Selection criteria:	Call forwarding by the network CFNR supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C according CFNR procedures. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
							180	←							
ALERTING		←	180	←	180	←									
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←											
							CANCEL	←							
							CANCEL	→			CANCEL	→			
									487 Request terminated	←	487 Request terminated	←			
							487 Request terminated	→							
							ACK	←							
									ACK	→					
											ACK	→			
							INVITE	←							
									INVITE	→					
									100 Trying	←			INVITE	→	
													100 Trying	←	
									180	←			180	←	
							180	→							
ALERTING	←		180	←	180	←	180	←							
													200 OK	←	
									200 OK	←					
							200 OK	→							
					200 OK	←	200 OK	←							
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
REL (UE1)	←		200 OK BYE	←	200 OK BYE	←	200 OK BYE	←							
RLC (UE1)	→														

ISS_XXSSCFNR 02	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	The user B is provided with CFNR ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = Yes, Served user communication retention on invocation of diversion (forwarding or deflection) = No.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no TIR) and user C is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
							180	←							
ALERTING		←	180	←	180	←									
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←											
							CANCEL	←							
							CANCEL	→			CANCEL	→			
									487 Request terminated	←	487 Request terminated	←			
							487 Request terminated	→							
							ACK	←							
									ACK	→					
											ACK	→			
							INVITE	←	MESSAGE	→					
											MESSAGE	→			
									INVITE	→					
									100 Trying	←			INVITE	→	
													100 Trying	←	
									180	←			180	←	
							180	→							
ALERTING	←		180	←	180	←	180	←							
													200 OK	←	
									200 OK	←					
							200 OK	→							
					200 OK	←	200 OK	←							
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
REL (UE1)	←		200 OK BYE	←	200 OK BYE	←	200 OK BYE	←							
RLC (UE1)	→														

ISS_XXSSCFNR 03	**ISDN reference to: EN 300 207-1 [i.5] clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	The user B is provided with CFNR ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = No, Served user communication retention on invocation of diversion (forwarding or deflection) = No.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no TIR) and user C is not informed of the forwarding number (user B has presentation allowed). User B is not notified of call diversion.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
							180	←							
ALERTING		←	180	←	180	←									
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←											
							CANCEL	←							
							CANCEL	→			CANCEL	→			
									487 Request terminated	←	487 Request terminated	←			
							487 Request terminated	→							
							ACK	←							
									ACK	→					
											ACK	→			
							INVITE	←							
									INVITE	→					
									100 Trying	←			INVITE	→	
													100 Trying	←	
									180	←			180	←	
							180	→							
ALERTING	←		180	←	180	←	180	←							
													200 OK	←	
									200 OK	←					
							200 OK	→							
					200 OK	←	200 OK	←							
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
REL (UE1)	←		200 OK BYE	←	200 OK BYE	←	200 OK BYE	←							
RLC (UE1)	→														

ISS_XXSSCFNR 04	**ISDN reference to: EN 300 207-1 [i.5] clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	User B is provided with CFNR ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no) Served user communication retention on invocation of diversion (forwarding or deflection) = No.	
Selection criteria:	Call forwarding by the network CFNR supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
							180	←							
ALERTING		←	180	←	180	←									
							CANCEL	←							
							CANCEL	→			CANCEL	→			
									487 Request terminated	←	487 Request terminated	←			
							487 Request terminated	→							
							ACK	←							
									ACK	→					
											ACK	→			
							INVITE	←							
									INVITE	→					
									100 Trying	←			INVITE	→	
													100 Trying	←	
									180	←			180	←	
							180	→							
ALERTING	←		180	←	180	←	180	←							
													200 OK	←	
									200 OK	←					
							200 OK	→							
					200 OK	←	200 OK	←							
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
REL (UE1)	←		200 OK BYE	←	200 OK BYE	←	200 OK BYE	←							
RLC (UE1)	→														

ISS_XXSSCFNR 05	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	The user B is provided with CFNR Served user communication retention on invocation of diversion (forwarding or deflection) = Yes).	
Selection criteria:	Call forwarding by the network CFNR supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C The diverting user accepts the communication after sending the INVITE request, the communication path towards the diverted to user shall be released according to the rules and procedures in RFC 3261 [28].	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
							180	←							
ALERTING		←	180	←	180	←									
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←											
									INVITE	→					
									100 Trying	←			INVITE	→	
													100 Trying	←	
									180	←			180	←	
							180	→							
ALERTING	←		180	←	180	←	180	←							
							CANCEL	←							
							CANCEL	→			CANCEL	→			
									487 Request terminated	←	487 Request terminated	←			
							487 Request terminated	→							
							ACK	←							
									ACK	→					
											ACK	→			
													200 OK	←	
									200 OK	←					
							200 OK	→							
					200 OK	←	200 OK	←							
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
REL (UE1)	←		200 OK BYE	←	200 OK BYE	←	200 OK BYE	←							
RLC (UE1)	→														

ISS_XXSSCFNR 06	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	The user B is provided with CFNR ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = Yes, Served user communication retention on invocation of diversion (forwarding or deflection) = Yes).	
Selection criteria:	Call forwarding by the network CFNR supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C The diverting user accepts the communication after sending the INVITE request, the communication path towards the diverted to user shall be released according to the rules and procedures in RFC 3261 [28].User A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
							180	←							
ALERTING		←	180	←	180	←									
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←											
									INVITE	→					
									100 Trying	←			INVITE	→	
									MESSAGE	→					
											MESSAGE	→			
													100 Trying	←	
									180	←			180	←	
							180	→							
			180	←	180	←	180	←							
							CANCEL	←							
							CANCEL	→			CANCEL	→			
									487 Request terminated	←	487 Request terminated	←			
							487 Request terminated	→							
							ACK	←							
									ACK	→					
											ACK	→			
													200 OK	←	
									200 OK	←					
							200 OK	→							
					200 OK	←	200 OK	←							
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
REL (UE1)	←		200 OK BYE	←	200 OK BYE	←	200 OK BYE	←							
RLC (UE1)	→														

ISS_XXSSCFNR 07	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	The user B is provided with CFNR ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = No, Served user communication retention on invocation of diversion (forwarding or deflection) = Yes).	
Selection criteria:	Call forwarding by the network CFNR supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C The diverting user accepts the communication after sending the INVITE request, the communication path towards the diverted to user shall be released according to the rules and procedures in RFC 3261 [28].User A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number (user B has presentation allowed). User B is not notified of call diversion.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
							180	←							
ALERTING		←	180	←	180	←									
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←											
									INVITE	→					
									100 Trying	←			INVITE	→	
									MESSAGE	→					
											MESSAGE	→			
													100 Trying	←	
									180	←			180	←	
							180	→							
			180	←	180	←	180	←							
							CANCEL	←							
							CANCEL	→			CANCEL	→			
									487 Request terminated	←	487 Request terminated	←			
							487 Request terminated	→							
							ACK	←							
									ACK	→					
											ACK	→			
													200 OK	←	
									200 OK	←					
							200 OK	→							
					200 OK	←	200 OK	←							
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
REL (UE1)	←		200 OK BYE	←	200 OK BYE	←	200 OK BYE	←							
RLC (UE1)	→														

ISS_XXSSCFNR 08	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	The user B is provided with CFNR ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = No, Served user communication retention on invocation of diversion (forwarding or deflection) = Yes).	
Selection criteria:	Call forwarding by the network CFNR supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C The diverting user accepts the communication after sending the INVITE request, the communication path towards the diverted to user shall be released according to the rules and procedures in RFC 3261 [28]. User A is notified of call diversion and informed of the diverted-to number and user C is not informed of the forwarding number (user B has presentation allowed). User B is not notified of call diversion.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
							180	←							
ALERTING		←	180	←	180	←									
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←											
									INVITE	→					
									100 Trying	←			INVITE	→	
													100 Trying	←	
									180	←			180	←	
							180	→							
ALERTING	←		180	←	180	←	180	←							
							CANCEL	←							
							CANCEL	→			CANCEL	→			
									487 Request terminated	←	487 Request terminated	←			
							487 Request terminated	→							
							ACK	←							
									ACK	→					
											ACK	→			
													200 OK	←	
									200 OK	←					
							200 OK	→							
					200 OK	←	200 OK	←							
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
REL (UE1)	←		200 OK BYE	←	200 OK BYE	←	200 OK BYE	←							
RLC (UE1)	→														

ISS_XXSSCFNR 09	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	User B is provided with CFNR ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no) Served user communication retention on invocation of diversion (forwarding or deflection) = Yes).	
Selection criteria:	Call forwarding by the network CFNR supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C The diverting user accepts the communication after sending the INVITE request, the communication path towards the diverted to user shall be released according to the rules and procedures in RFC 3261 [28]. User A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
							180	←							
ALERTING		←	180	←	180	←									
									INVITE	→					
									100 Trying	←			INVITE	→	
													100 Trying	←	
									180	←			180	←	
							180	→							
ALERTING	←		180	←	180	←	180	←							
							CANCEL	←							
							CANCEL	→			CANCEL	→			
									487 Request terminated	←	487 Request terminated	←			
							487 Request terminated	→							
							ACK	←							
									ACK	→					
											ACK	→			
													200 OK	←	
									200 OK	←					
							200 OK	→							
					200 OK	←	200 OK	←							
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
REL (UE1)	←		200 OK BYE	←	200 OK BYE	←	200 OK BYE	←							
RLC (UE1)	→														

ISS_XXSSCFNR 10	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR Served user communication retention on invocation of diversion (forwarding or deflection) = No [Clear call to the served user on invocation of call diversion], Served user communication retention when forwarding is rejected at forwarded-to user = No action at the forwarding user).	
Selection criteria:	Call forwarding by the network Call forwarding on no reply supported	
Test purpose:	To verify that a call is released correctly if **CFNR was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
							180	←							
ALERTING		←	180	←	180	←									
							CANCEL	←							
							CANCEL	→							
											CANCEL	→			
											487 Request terminated	←			
									487 Request terminated	←					
							487 Request terminated	→							
							ACK	←							
									ACK	→					
											ACK	→			
							INVITE	←							
									INVITE	→					
													INVITE	→	
									486	←			486	←	
							486	→					ACK	→	
DISC # 17	←		486	←	486	←	486	←							
REL	→		ACK	→	ACK	→	ACK	→							
RLC	←						ACK	←							
									ACK	→					

ISS_XXSSCFNR 11	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR Served user communication retention on invocation of diversion (forwarding or deflection) = No [Clear call to the served user on invocation of call diversion.	
Selection criteria:	Call forwarding by the network Call forwarding on no reply supported	
Test purpose:	To verify that a call is released correctly if CFNR **was not successful**. User A calls user B, the call is forwarded to user C who is **network determined user busy**.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
							180	←							
ALERTING		←	180	←	180	←									
							CANCEL	←							
							CANCEL	→							
											CANCEL	→			
											487 Request terminated	←			
							487 Request terminated	→							
							ACK	←							
											ACK	→			
							INVITE	←							
									INVITE	→					
													INVITE	→	
									486	←			486	←	
							486	→							
DISC # 17	←		486	←	486	←	486	←							
REL	→		ACK	→	ACK	→	ACK	→							
RLC	←						ACK	←							
									ACK	→					
													ACK	→	

ISS_XXSSCFNR 12	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR. Served user communication retention on invocation of diversion (forwarding or deflection = Yes)	
Selection criteria:	Call forwarding by the network Call forwarding on no reply supported	
Test purpose:	User A calls user B, the call is forwarded to user C who is **user** determined user busy. The forwarding user User B continues to alert.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN			MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C
SETUP (UE 1)	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→					
											INVITE	→			
											180	←			
									180	←					
							180	→							
							180	←							
					180	←									
ALERTING		←	180	←											
					181	←	181	←							
NOTIFY (UE 1)	←		181	←											
							INVITE	←							
									INVITE	→					
													INVITE	→	
									486	←			486	←	
							486	→			ACK	→			
							ACK	←							
									ACK	→			ACK	→	
									200 OK	←	200 OK	←			
							200 OK	→			ACK	→			
					200 OK	←	200 OK	←							
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→					
DISC (UE1)	→		BYE	→											
					BYE	→									
							BYE	→							
							BYE	←							
									BYE	→	BYE	→			
											200 OK BYE	←			
									200 OK BYE	←					
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISS_XXSSCFNR 13	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR. Served user communication retention on invocation of diversion (forwarding or deflection) = Yes.	
Selection criteria:	Call forwarding by the network Call forwarding on no reply supported	
Test purpose:	User A calls user B, the call is forwarded to user C who is **network** determined user busy. The forwarding user User B continues to alert.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→					
											INVITE	→			
											180	←			
									180	←					
							180	→							
							180	←							
					180	←									
ALERTING		←	180	←											
					181	←	181	←							
NOTIFY (UE 1)	←		181	←											
							INVITE	←							
									INVITE	→					
									486	←					
							486	→	ACK	→					
							ACK	←							
									200 OK	←	200 OK	←			
							200 OK	→			ACK	→			
					200 OK	←	200 OK	←							
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→					
DISC (UE1)	→		BYE	→											
					BYE	→									
							BYE	→							
							BYE	←							
									BYE	→	BYE	→			
											200 OK BYE	←			
									200 OK BYE	←					
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISSI_XXSSCFNR 14	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN-ISDN/Supplementary_services/CFNR	
Configuration:	The user A and the user D are in network N1. The user B and C are in network N2, and user B is provided with CFNR. User D forwards the call to back to user B.	
Selection criteria:	Call forwarding by the network Call forwarding on no reply supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C and D. User D forwards the call to back to user B. Ensure that the call is released.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

6.1.2.7 CFNL

ISI_XXSSCFNL 01	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/ CFNL	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNL. User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network CFNL supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:		
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←							
							100 Trying	→							
					INVITE	←									
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→	BYE	→	BYE	→							
DISC (UE2)	←		BYE	←	BYE	←	BYE	←							
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFNL 02	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFNL	
Configuration:	The user A and the user C are in network N1. The user B is provided with CFNL ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes). User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network CFNL supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:		
Comments:		

ISDN		MGCF			I-CSCF		S-CSCF		AS	P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←							
							100 Trying	→							
					INVITE	←									
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFNL 03	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFNL	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNL ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No). User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network CFNL supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:		
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←							
							100 Trying	→							
					INVITE	←									
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCFNL 04	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFNL	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNL. The user B is in network N2 and is provided with CFNL ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No, "diverting number is released to the diverted-to user" = No). User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network CFNL supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number (user B has presentation not allowed).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:		
Comments:		

ISDN		MGCF			I-CSCF		S-CSCF		AS	P-CSCF	UE-B	UE-C
SETUP (UE 1)	→		INVITE	→	INVITE	→						
					100 Trying	←						
							INVITE	→				
							100 Trying	←				
							INVITE	←				
							100 Trying	→				
							INVITE	←				
							100 Trying	→				
					INVITE	←						
					100 Trying	→						
			INVITE	←								
			100 Trying	→								
SETUP (UE 2)	←											
ALERTING (UE 2)	→		180	→								
					180	→	180	→				
					180	←	180	←				
ALERTING (UE 1)	←		180	←								
CONNECT (UE 2)	→		200 OK	→								
					200 OK	→	200 OK	→				
					200 OK	←	200 OK	←				
CONNECT (UE 1)	←		200 OK	←								
			ACK	→	ACK	→	ACK	→				
			ACK	←	ACK	←	ACK	←				
DISC (UE1)	→		BYE	→								
					BYE	→	BYE	→				
					BYE	←	BYE	←				
DISC (UE2)	←		BYE	←								
REL (UE2)	→		200 OK BYE	→								
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→				
					200 OK BYE	←	200 OK BYE	←				
REL (UE1)	←		200 OK BYE	←								
RLC (UE1)	→											

ISI_XXSSCFNL 05	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFNL	
Configuration:	The user B is in network N2 and is provided with CFNL. User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network CFNL supported user C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFNL was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:		
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←							
							100 Trying	→							
					INVITE	←									
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
RLC (UE 2)	→		486 Busy here	→	486 Busy here	→	486 Busy here	→							
DISC (UE 1)	←		486 Busy here	←	486 Busy here	←	486 Busy here	←							
REL	→		ACK	→	ACK	→	ACK	→							
RLC	←		ACK	←	ACK	←	ACK	←							

ISI_XXSSCFNL 06	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CFNL	
Configuration:	The user B is in network N2 and is provided with CFNL. User C is point-to-multipoint.	
Selection criteria:	Call forwarding by the network CFNL supported user C is network determined user busy	
Test purpose:	To verify that a call is released correctly if CFNL was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:		
Comments		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←							
							100 Trying	→							
					INVITE	←									
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
			486 Busy here	→	486 Busy here	→	486 Busy here	→							
DISC (UE 1)	←		486 Busy here	←	486 Busy here	←	486 Busy here	←							
REL	→		ACK	→	ACK	→	ACK	→							
RLC	←		ACK	←	ACK	←	ACK	←							

ISS_XXSSCFNL 07	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFNL	
Configuration:	The user B is provided with CFNL.	
Selection criteria:	Call forwarding by the network CFNL supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:		
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
							181	←							
					181	←									
NOTIFY	←		181	←					INVITE	→			INVITE	→	
													100 Trying	←	
									100 Trying	←					
									180	←			180	←	
							180	→							
					180	←	180	←							
ALERTING	←		180	←											
									200 OK	←			200 OK	←	
							200 OK	→							
							200 OK	←							
					200 OK	←									
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→									
							BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISS_XXSSCFNL 08	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFNL	
Configuration:	The user B is provided with CFNL ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes).	
Selection criteria:	Call forwarding by the network CFNL supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no TIR) and user C is informed of the forwarding number (user B has presentation allowed).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:		
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
							181	←							
					181	←									
NOTIFY	←		181	←					INVITE	→			INVITE	→	
													100 Trying	←	
									100 Trying	←					
									180	←			180	←	
							180	→							
					180	←	180	←							
ALERTING	←		180	←											
									200 OK	←			200 OK	←	
							200 OK	→							
							200 OK	←							
					200 OK	←									
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→									
							BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISS_XXSSCFNL 09	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFNL	
Configuration:	User B is provided with CFNL ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No).	
Selection criteria:	Call forwarding by the network CFNL supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - TIR) and user C is not informed of the forwarding number (user B has presentation not allowed).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:		
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
							181	←							
					181	←									
NOTIFY	←		181	←					INVITE	→			INVITE	→	
													100 Trying	←	
									100 Trying	←					
									180	←			180	←	
							180	→							
					180	←	180	←							
ALERTING	←		180	←											
									200 OK	←			200 OK	←	
							200 OK	→							
							200 OK	←							
					200 OK	←									
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→									
							BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISS_XXSSCFNL 10	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFNL	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNL The user B is in network N2 and is provided with CFNL ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No, "diverting number is released to the diverted-to user" = No)	
Selection criteria:	Call forwarding by the network CFNL supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number (user B has presentation not allowed)	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:		
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→			INVITE	→	
													100 Trying	←	
									100 Trying	←					
									180	←			180	←	
							180	→							
					180	←	180	←							
ALERTING	←		180	←											
									200 OK	←			200 OK	←	
							200 OK	→							
							200 OK	←							
					200 OK	←									
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→									
							BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISS_XXSSCFNL 11	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFNL	
Configuration:	The user B is in network N2 and is provided with CFNL	
Selection criteria:	Call forwarding by the network CFNL supported	
Test purpose:	To verify that a call is released correctly if **CFNL was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:		
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
							181	←							
					181	←	INVITE			→					
NOTIFY	←		181	←									INVITE	→	
									486	←			486	←	
							486	→					ACK	→	
					486	←	486	←							
DISC # 17	←		486	←											
REL	→		ACK	→											
RLC	←				ACK	→									
							ACK	→							
							ACK	←							
									ACK	→					

ISS_XXSSCFNL 12	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/CFNL	
Configuration:	The user B is in network N2 and is provided with CFNL	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	To verify that a call is released correctly if CFNL **was not successful**. User A calls user B, the call is forwarded to user C who is **network determined user busy.**	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:		
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
							INVITE			→					
									486	←					
							486	→							
					486	←	486	←							
DISC # 17	←		486	←											
REL	→		ACK	→											
RLC	←				ACK	→									
							ACK	→							
							ACK	←							
									ACK	→					

6.1.2.8 CD

6.1.2.8.1 Call Deflection-ISI

ISI_XXSSCD 01	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CD- Immediate response	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CD ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = **No**). User C is point-to-multipoint.	
Selection criteria:	User B has activated the CALL DEFLECTION service	
Test purpose:	Ensure that when user A calls user B, the call is deflected to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→					
											INVITE	→			
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←							
					INVITE	←									
			INVITE	←											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCD 02	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/CD- Immediate response	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CD ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no). User C is point-to-multipoint.	
Selection criteria:	User B has activated the CALL DEFLECTION service	
Test purpose:	Ensure that when user A calls user B, the call is deflected to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→					
											INVITE	→			
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←							
					INVITE	←									
			INVITE	←											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCD 03	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/ CD- Immediate response	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CD The user B is in network N2 and is provided with CD ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no). User C is point-to-multipoint.	
Selection criteria:	User B has activated the CALL DEFLECTION service	
Test purpose:	Ensure that when user A calls user B, the call is deflected to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number (user B has presentation not allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→					
											INVITE	→			
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
						181	←								
					INVITE	←									
			INVITE	←											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
CONNECT (UE 1)	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCD 04	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/ CD- Immediate response	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CD	
Selection criteria:	User B has activated the CALL DEFLECTION service	
Test purpose:	Ensure that when user A calls user B, the call is deflected to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters). User C is point-to-multipoint.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→					
											INVITE	→			
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←							
					INVITE	←									
			INVITE	←											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→											
					180	→	180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
CONNECT (UE 2)	→		200 OK	→											
					200 OK	→	200 OK	→							
					200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
CONNECT (UE 1)	←		200 OK	←											
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
					BYE	←	BYE	←							
DISC (UE2)	←		BYE	←											
REL (UE2)	→		200 OK BYE	→											
RLC (UE2)	←				200 OK BYE	→	200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL (UE1)	←		200 OK BYE	←											
RLC (UE1)	→														

ISI_XXSSCD 05	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/ CD- Immediate response	
Configuration:	The user B is in network N2 and is provided with CD. User C is point-to-multipoint.	
Selection criteria:	user C is network determined user busy User B has activated the CALL DEFLECTION service	
Test purpose:	To verify that a call is released correctly if CD was not successful. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
						INVITE	→								
						INVITE	←			INVITE	→				
										100 Trying	←				
										302	←				
								302	←						
						302	→								
						ACK	←								
								ACK	→	ACK	→				
						181	←								
				181	←										
NOTIFY (UE 1)	←		181	←			INVITE	←							
					INVITE	←									
			INVITE	←											
SETUP (UE 2)	←														
RLC (UE 2)	→		486 Busy here	→	486 Busy here	→	486 Busy here	→							
DISC (UE 1)	←		486 Busy here	←	486 Busy here	←	486 Busy here	←							
REL	→		ACK	→	ACK	→	ACK	→							
RLC	←		ACK	←	ACK	←	ACK	←							

ISI_XXSSCD 06	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/ CD- Immediate response	
Configuration:	The user B is in network N2 and is provided with CD. User C is point-to-multipoint.	
Selection criteria:	user C is network determined network busy User B has activated the CALL DEFLECTION service	
Test purpose:	To verify that a call is released correctly if CD was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→					
											INVITE	→			
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←			INVITE	←							
					INVITE	←									
			INVITE	←											
			486 Busy here	→	486 Busy here	→	486 Busy here	→							
DISC (UE 1)	←		486 Busy here	←	486 Busy here	←	486 Busy here	←							
REL	→		ACK	→	ACK	→	ACK	→							
RLC	←		ACK	←	ACK	←	ACK	←							

ISI_XXSSCD 07	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/ CD- during alerting	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CD- during alerting. User C is point-to-multipoint.	
Selection criteria:	User B has activated the CALL DEFLECTION service	
Test purpose:	Ensure that when user A calls user B, the call is deflected during alerting to user C.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
									100 Trying	←					
									INVITE	→					
											INVITE	→			
											180	←			
									180	←					
							180	→							
							180	←							
ALERTING		←	180	←	180	←									
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
					181	←	181	←							
NOTIFY (UE 1)	←		181	←											
					INVITE	←	INVITE	←							
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→	180	→	180	→							
ALERTING (UE 1)	←		180	←	180	←	180	←							
CONNECT (UE 2)	→		200 OK	→	200 OK	→	200 OK	→							
CONNECT (UE 1)	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→	BYE	→	BYE	→							
DISC (UE2)	←		BYE	←	BYE	←	BYE	←							
REL (UE2)	→		200 OK BYE	→	200 OK BYE	→	200 OK BYE	→							
RLC (UE2)	←														
REL (UE1)	←		200 OK BYE	←	200 OK BYE	←	200 OK BYE	←							
RLC (UE1)	→														

ISI_XXSSCD 08	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/ CD- during alerting	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CD ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = NO). User C is point-to-multipoint.	
Selection criteria:	User B has activated the CALL DEFLECTION service	
Test purpose:	Ensure that when user A calls user B, the call is deflected during alerting to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
									100 Trying	←					
									INVITE	→					
											INVITE	→			
											180	←			
									180	←					
							180	→							
							180	←							
ALERTING		←	180	←	180	←									
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
					181	←	181	←							
NOTIFY (UE 1)	←		181	←											
					INVITE	←	INVITE	←							
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→	180	→	180	→							
ALERTING (UE 1)	←		180	←	180	←	180	←							
CONNECT (UE 2)	→		200 OK	→	200 OK	→	200 OK	→							
CONNECT (UE 1)	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→	BYE	→	BYE	→							
DISC (UE2)	←		BYE	←	BYE	←	BYE	←							
REL (UE2)	→		200 OK BYE	→	200 OK BYE	→	200 OK BYE	→							
RLC (UE2)	←														
REL (UE1)	←		200 OK BYE	←	200 OK BYE	←	200 OK BYE	←							
RLC (UE1)	→														

ISI_XXSSCD 09	**ISDN reference to:** **EN 300 207-1 [i.5]** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/ CD- during alerting	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no). User C is point-to-multipoint.	
Selection criteria:	User B has activated the CALL DEFLECTION service	
Test purpose:	Ensure that when user A calls user B, the call is deflected during alerting to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
									100 Trying	←					
									INVITE	→					
											INVITE	→			
											180	←			
									180	←					
							180	→							
							180	←							
ALERTING		←	180	←	180	←									
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
					181	←	181	←							
NOTIFY (UE 1)	←		181	←											
					INVITE	←	INVITE	←							
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→	180	→	180	→							
ALERTING (UE 1)	←		180	←	180	←	180	←							
CONNECT (UE 2)	→		200 OK	→	200 OK	→	200 OK	→							
CONNECT (UE 1)	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→	BYE	→	BYE	→							
DISC (UE2)	←		BYE	←	BYE	←	BYE	←							
REL (UE2)	→		200 OK BYE	→	200 OK BYE	→	200 OK BYE	→							
RLC (UE2)	←														
REL (UE1)	←		200 OK BYE	←	200 OK BYE	←	200 OK BYE	←							
RLC (UE1)	→														

ISI_XXSSCD 10	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/ CD- during alerting	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CD- during alerting The user A and the user C are in network N1. The user B is in network N2 and is provided with CD ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no). User C is point-to-multipoint.	
Selection criteria:	User B has activated the CALL DEFLECTION service	
Test purpose:	Ensure that when user A calls user B, the call is deflected during alerting to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number (user B has presentation not allowed).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF			I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
									100 Trying	←					
									INVITE	→					
											INVITE	→			
											180	←			
									180	←					
							180	→							
							180	←							
ALERTING		←	180	←	180	←									
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
					181	←	181	←							
NOTIFY (UE 1)	←		181	←											
					INVITE	←	INVITE	←							
					100 Trying	→									
			INVITE	←											
			100 Trying	→											
SETUP (UE 2)	←														
ALERTING (UE 2)	→		180	→	180	→	180	→							
ALERTING (UE 1)	←		180	←	180	←	180	←							
CONNECT (UE 2)	→		200 OK	→	200 OK	→	200 OK	→							
CONNECT (UE 1)	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
			ACK	←	ACK	←	ACK	←							
DISC (UE1)	→		BYE	→	BYE	→	BYE	→							
DISC (UE2)	←		BYE	←	BYE	←	BYE	←							
REL (UE2)	→		200 OK BYE	→	200 OK BYE	→	200 OK BYE	→							
RLC (UE2)	←														
REL (UE1)	←		200 OK BYE	←	200 OK BYE	←	200 OK BYE	←							
RLC (UE1)	→														

6.1.2.8.2 CD-ISS

ISS_XXSSCD 01	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/ CD- Immediate response	
Configuration:	The user B is provided with CD- Immediate response	
Selection criteria:	User B has activated the CALL DEFLECTION service	
Test purpose:	Ensure that when user A calls user B, the call is deflected immediate response to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
									100 Trying	←					
											INVITE	→			
											100 Trying	←			
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←											
							INVITE	←							
							100 Trying	→							
									INVITE	→			INVITE	→	
													100 Trying	←	
													180	←	
									180	←					
							180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
													200 OK	←	
									200 OK	←					
							200 OK	→							
							200 OK	←							
					200 OK	←									
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC	→		BYE	→											
					BYE	→									
							BYE	→							
							BYE	←							
									BYE	→			BYE	→	
									200 OK BYE	←			200 OK BYE	←	
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL	←		200 OK BYE	←											
RLC	→														

ISS_XXSSCD 02	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/ CD- Immediate response	
Configuration:	The user B is provided with CD- Immediate response ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes).	
Selection criteria:	User B has activated the CALL DEFLECTION service	
Test purpose:	Ensure that when user A calls user B, the call is deflected immediate response to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF			I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
									100 Trying	←					
											INVITE	→			
											100 Trying	←			
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←											
							INVITE	←	MESSAGE	→					
							100 Trying	→			MESSAGE	→			
									INVITE	→			INVITE	→	
													100 Trying	←	
													180	←	
									180	←					
							180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
													200 OK	←	
									200 OK	←					
							200 OK	→							
							200 OK	←							
					200 OK	←									
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC	→		BYE	→											
					BYE	→									
							BYE	→							
							BYE	←							
									BYE	→			BYE	→	
									200 OK BYE	←			200 OK BYE	←	
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL	←		200 OK BYE	←											
RLC	→														

ISS_XXSSCD 03	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/ CD- Immediate response	
Configuration:	The user B is provided with CD- Immediate response ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No).	
Selection criteria:	User B has activated the CALL DEFLECTION service	
Test purpose:	Ensure that when user A calls user B, the call is deflected immediate response to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
									100 Trying	←					
											INVITE	→			
											100 Trying	←			
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←											
							INVITE	←							
							100 Trying	→							
									INVITE	→			INVITE	→	
													100 Trying	←	
													180	←	
									180	←					
							180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
													200 OK	←	
									200 OK	←					
							200 OK	→							
							200 OK	←							
					200 OK	←									
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC	→		BYE	→											
					BYE	→									
							BYE	→							
							BYE	←							
									BYE	→			BYE	→	
									200 OK BYE	←			200 OK BYE	←	
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL	←		200 OK BYE	←											
RLC	→														

ISS_XXSSCD 04	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/ CD- Immediate response	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CD- Immediate response. The user B is in network N2 and is provided with CD- Immediate response ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No, "diverting number is released to the diverted-to user" = No).	
Selection criteria:	User B has activated the CALL DEFLECTION service CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is deflected immediate response to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number (user B has presentation not allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF			I-CSCF	S-CSCF			AS		P-CSCF		UE-B		UE-C
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							100 Trying	←							
							INVITE	←							
							100 Trying	→							
									INVITE	→					
									100 Trying	←					
											INVITE	→			
											100 Trying	←			
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
							181	←							
					181	←									
NOTIFY (UE 1)	←		181	←											
							INVITE	←							
							100 Trying	→							
									INVITE	→			INVITE	→	
													100 Trying	←	
													180	←	
									180	←					
							180	→							
					180	←	180	←							
ALERTING (UE 1)	←		180	←											
													200 OK	←	
									200 OK	←					
							200 OK	→							
							200 OK	←							
					200 OK	←									
CONNECT	←		200 OK	←											
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC	→		BYE	→											
					BYE	→									
							BYE	→							
							BYE	←							
									BYE	→			BYE	→	
									200 OK BYE	←			200 OK BYE	←	
							200 OK BYE	→							
					200 OK BYE	←	200 OK BYE	←							
REL	←		200 OK BYE	←											
RLC	→														

ISS_XXSSCD 05	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/ CD- Immediate response	
Configuration:	The user B is in network N2 and is provided with CD- Immediate response	
Selection criteria:	Call forwarding by the network User B has activated the CALL DEFLECTION service	
Test purpose:	To verify that a call is released correctly if CD- Immediate response **was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
ALERTING		←	180	←	180	←	180	←							
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
							181	←							
					181	←									
NOTIFY. (UE 1)	←		181	←											
							INVITE	←							
									INVITE	→			INVITE	→	
									486	←			486	←	
							486	→					ACK	→	
					486	←	486	←							
DISC # 17	←		486	←											
REL	→		ACK	→	ACK	→	ACK	→							
RLC	←						ACK	←							
									ACK	→					

ISS_XXSSCD 06	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/ CD- during alerting	
Configuration:	The user B is provided with CFNR - CALL DEFLECTION	
Selection criteria:	User B has activated the CALL DEFLECTION service	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							INVITE	←							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
ALERTING		←	180	←	180	←	180	←							
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
					181	←	181	←							
NOTIFY (UE 1)	←		181	←											
							INVITE	←							
									INVITE	→					
													INVITE	→	
													100 Trying	←	
									180	←			180	←	
							180	→							
ALERTING	←		180	←	180	←	180	←							
									200 OK	←			200 OK	←	
							200 OK	→							
CONNECT	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
REL (UE1)	←		200 OK BYE	←	200 OK BYE	←	200 OK BYE	←							
RLC (UE1)	→														

ISS_XXSSCD 07	**ISDN reference to:** **EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/ CD- during alerting	
Configuration:	The user B is provided with CD- during alerting ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = **No**).	
Selection criteria:	User B has activated the CALL DEFLECTION service	
Test purpose:	Ensure that when user A calls user B, the call is deflected during alerting to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF			I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							INVITE	←							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
ALERTING		←	180	←	180	←	180	←							
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
					181	←	181	←							
NOTIFY (UE 1)	←		181	←											
							INVITE	←							
									INVITE	→					
									MESSAGE	→					
											MESSAGE	→			
													INVITE	→	
													100 Trying	←	
									180	←			180	←	
							180	→							
ALERTING	←		180	←	180	←	180	←							
									200 OK	←			200 OK	←	
							200 OK	→							
CONNECT	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
REL (UE1)	←		200 OK BYE	←	200 OK BYE	←	200 OK BYE	←							
RLC (UE1)	→														

ISS_XXSSCD 08	ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5	NGN reference to: TS 183 004 [45]
TSS reference:	ISDN-SIP-SIP/Supplementary_services/ CD- during alerting	
Configuration:	The user B is provided with CD- during alerting ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no).	
Selection criteria:	User B has activated the CALL DEFLECTION service	
Test purpose:	Ensure that when user A calls user B, the call is deflected during alerting to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has TIR) and user C is not informed of the forwarding number (user B has presentation not allowed).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							INVITE	←							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
ALERTING		←	180	←	180	←	180	←							
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
					181	←	181	←							
NOTIFY (UE 1)	←		181	←											
							INVITE	←							
									INVITE	→					
													INVITE	→	
													100 Trying	←	
									180	←			180	←	
							180	→							
ALERTING	←		180	←	180	←	180	←							
									200 OK	←			200 OK	←	
							200 OK	→							
CONNECT	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
REL (UE1)	←		200 OK BYE	←	200 OK BYE	←	200 OK BYE	←							
RLC (UE1)	→														

ISS_XXSSCD 09	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/ CD- during alerting	
Configuration:	The user B is provided with CD- during alerting ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no).	
Selection criteria:	User B has activated the CALL DEFLECTION service	
Test purpose:	Ensure that when user A calls user B, the call is deflected during alerting to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number (user B has presentation not allowed).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
					100 Trying	←									
							INVITE	→							
							INVITE	←							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
ALERTING		←	180	←	180	←	180	←							
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
							INVITE	←							
									INVITE	→					
													INVITE	→	
													100 Trying	←	
									180	←			180	←	
							180	→							
ALERTING	←		180	←	180	←	180	←							
									200 OK	←			200 OK	←	
							200 OK	→							
CONNECT	←		200 OK	←	200 OK	←	200 OK	←							
			ACK	→	ACK	→	ACK	→							
							ACK	←							
									ACK	→			ACK	→	
DISC (UE1)	→		BYE	→											
					BYE	→	BYE	→							
							BYE	←							
									BYE	→			BYE	→	
													200 OK BYE	←	
									200 OK BYE	←					
							200 OK BYE	→							
REL (UE1)	←		200 OK BYE	←	200 OK BYE	←	200 OK BYE	←							
RLC (UE1)	→														

ISS_XXSSCD 10	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/ CD- during alerting	
Configuration:	The user B is in network N2 and is provided with CD- during alerting	
Selection criteria:	User B has activated the CALL DEFLECTION service	
Test purpose:	To verify that a call is released correctly if CD- during alerting **was not successful**. User A calls user B, the call is forwarded to user C who is **network determined user busy**.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN		MGCF		I-CSCF		S-CSCF		AS		P-CSCF		UE-B		UE-C	
SETUP (UE 1)	→		INVITE	→	INVITE	→									
							INVITE	→							
							INVITE	←							
									INVITE	→					
											INVITE	→			
									180	←	180	←			
							180	→							
ALERTING		←	180	←	180	←	180	←							
											302	←			
									302	←					
							302	→							
							ACK	←							
									ACK	→	ACK	→			
							181	←							
					181	←									
NOTIFY . (UE 1)	←		181	←											
							INVITE	←							
									INVITE	→					
									486	←					
							486	→							
					486	←	486	←							
DISC # 17	←		486	←											
REL	→		ACK	→	ACK	→	ACK	→							
RLC	←						ACK	←							
									ACK	→					

6.1.2.9 3PTY

ISI_XXSS3PTY01	**ISDN reference to: EN 300 188-1 [i.6], clause 9.2**	**NGN reference to: EN 383 001 [49]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/3PTY	
Selection criteria:	The user A is in network N1 and is provided with 3PTY. The user B is in network N2 user C in the network N1.	
Test purpose:	Ensure that user A can establish a three-way conversation call with user B and user C and release the Active-Idle connection (A-C). After the completion of the Retrieve function, the call clearing procedure is performed from user A.	
ISDN Parameter values:	BC = PIXIT	
Comments:		

UE A (ISDN)				UE B (SIP)			UE C (ISDN)
SETUP(CRx)	→		→	INVITE			
ALERTING	←		←	180 Ringing			
CONNECT	←		←	200 OK			
			→	ACK			
HOLD(CRx)	→		→	INVITE(sendonly)			
			←	200 OK(recvonly)			
			→	ACK			
SETUP(CRy)	→					→	SETUP
ALERTING	←					←	ALERTING
CONNECT	←					←	CONNECT
FAC(3PTY_begin_invoke, CRx)	→		→	INVITE(sendrecv)			
FAC(3PTY_begin_ret_res, CRx)	←		←	200 OK(sendrecv)			
			→	ACK		→	NOTIFY(conf est)
				3 Party conversation			
DISC(CRy)	→		→	INVITE(sendonly)		→	DISC
RELEASE	←		←	200 OK(recvonly)		←	RELEASE
REL COMP	→		→	ACK		→	REL COMP
RETRIEVE (CRx)	→		→	INVITE(sendrecv)			
			←	200 OK(sendrecv)			
			→	ACK			
		Conversation					
DISC(CRx)	→		→	BYE			
RELEASE	←		←	200 OK			
REL COMP	→						

ISI_XXSS3PTY02	**ISDN reference to: EN 300 188-1 [i.6], clause 9.2, figure A.2**	**NGN reference to: EN 383 001 [49]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/3PTY	
Selection criteria:	The user A is in network N1 and is provided with 3PTY. The user B is in network N2 user C in the network N1.	
Test purpose:	Ensure that user A can establish a three-way conversation call with user B and user C and release the Active-Held connection (A-B). The call clearing procedure is performed from user A.	
ISDN Parameter values:	BC = PIXIT	
Comments:	User A calls user B (with CRx). After initiating of call hold, the call A-B has an Active-Held connection. User A is calling user C (with the CRy). The call (A-C) has an Active-Idle connection. When user A sends a FACILITY message for CRx containing a facility IE with a Begin3PTY invoke component the network shall respond with a FACILITY message containing a facility IE with a Begin3PTY return result component for CRx. User C shall receive a NOTIFY message containing a Notification Indicator IE with a notification description of "Conference established". The three-way bridge is established. On receipt of a DISCONNECT message from the user A relating to the Active-Held connection (CRx) the network shall clear the call to user B. After the release of the three-way bridge the network is sending to the remote user C a NOTIFY message containing a Notification indicator IE with a notification description of "Conference disconnected". The call A-C has an Active-Idle connection. The call clearing procedure is performed from user A with a DISCONNECT message.	

ISI_XXSS3PTY03	**ISDN reference to: EN 300 188-1 [i.6], clause 9.2**	**NGN reference to: EN 383 001 [49]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/3PTY	
Selection criteria:	The user A is in network N1 and is provided with 3PTY. The user B is in network N2 user C in the network N1.	
Test purpose:	Ensure that user A can establish a three-way conversation call with user B and user C and user B sends disconnect during the Three-Party communication.	
ISDN Parameter values:	BC = PIXIT	
Comments:		

ISI_XXSS3PTY04	**ISDN reference to: EN 300 188-1 [i.6], clause 9.2**	**NGN reference to: Figure 2-9 of Q.734.2 [15] - User C disconnects**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/3PTY	
Selection criteria:	The user A is in network N1 and is provided with 3PTY. The user B is in network N2 user C in the network N1.	
Test purpose:	Ensure that user A can establish a three-way conversation call with user B and user C and user C sends disconnect during the Three-Party communication.	
ISDN Parameter values:	BC = PIXIT	
Comments:		

ISI_XXSS3PTY05	**ISDN reference to: EN 300 188-1 [i.6], clause 9.2**	**NGN reference to: EN 383 001 [49]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/3PTY	
Selection criteria:	The user A is in network N1 and is provided with 3PTY. The user B is in network N2 user C in the network N1.	
Test purpose:	Ensure that user A can establish a three-way conversation call with user B and user C and release of both remote users, user C is released first.	
ISDN Parameter values:	BC = speech	
Comments:	User A calls user B (with CRx). After initiating of call hold, the call A-B has an Active-Held connection. User A is calling user C (with the CRy). The call (A-C) has an Active-Idle connection. When user A sends a FACILITY message for CRx containing a facility IE with a Begin3PTY invoke component the network shall respond with a FACILITY message containing a facility IE with a Begin3PTY return result component for CRx. User C receives a NOTIFY message containing a Notification Indicator IE with a notification description of "Conference established". The three-way bridge is established. On receipt of a DISCONNECT message from the user A relating to the Active-Idle connection (CRy) the network shall clear the call to user C with a DISCONNECT message. On receipt of a DISCONNECT message from the user A relating to the Active-Held connection (CRx) the network shall clear the call to user B.	

ISI_XXSS3PTY06	**ISDN reference to: EN 300 188-1 [i.6], clause 9.2**	**NGN reference to: EN 383 001 [49]**
TSS reference:	The user A is in network N1 and is provided with 3PTY. The user B is in network N2 user C in the network N1.	
Selection criteria:	The user A is in network N1 and is provided with 3PTY. The user B is in network N2 user C in the network N1.	
Test purpose:	Ensure that user A can establish a three-way conversation call with user B and user C and create a private communication with user B. The call clearing procedure is performed from user A	
ISDN Parameter values:	BC = speech	
Comments:	User A calls user B (with CRx). After initiating of call hold, the call A-B has an Active-Held connection. User A is calling user C (with the CRy). The call (A-C) has an Active-Idle connection. When user A sends a FACILITY message for CRx containing a facility IE with a Begin3PTY invoke component the network shall respond with a FACILITY message containing a facility IE with a Begin3PTY return result component for CRx. User B an C receive a NOTIFY message containing a Notification Indicator IE with a notification description of "Conference established". The three-way bridge is established. The served user shall send an End3PTY invoke component to the network in a FACILITY message with that CRx. On receiving such an invoke component in a FACILITY message, the network shall: i) remove the three-way bridge from both the Active-Idle connection and the Active-Held connection; ii) release the three-way bridge; iii) return to the served user an End3PTY return result component, within a FACILITY message using the CRx of the Active-Held connection; iv) send a NOTIFY message to the remote user with which private communication is required containing a Notification indicator information element with a notification description of "Remote hold"; and, v) send a NOTIFY message to the other remote user containing a Notification indicator information element with a notification description of "Conference disconnected". When the served user receives a correctly encoded End3PTY return result component, within a FACILITY message, the user shall accept the provided information and shall: i) use the CR relating to the Active-Idle connection, perform the Hold function ii) use the CR relating to the Active-Held connection, perform the Retrieve function The network shall complete the Hold and Retrieve functions. On successful completion of the Hold function (i.e. the HOLD ACKNOWLEDGE message is sent) the network shall send a NOTIFY message, to the remote user that is not to be included in the private communication, containing a Notification indicator information element with a notification description of "Remote hold". On successful completion of the Retrieve function (i.e. RETRIEVE ACKNOWLEDGE message is sent) the network shall send a NOTIFY message, to the remote user for whom private communication is desired, containing a Notification indicator information element with a notification description of "Conference disconnected". (A Notification indicator information element with a notification description of "Remote retrieval" is not sent to the remote user under these circumstances.) As a result of the procedures of this item of this clause, the call state of the connections, at both the network and the served user, is unchanged. The auxiliary state of the connection of the private communication changes from Call Held to Idle. The auxiliary state of the other connection changes from Idle to Call Held. The call clearing procedure is performed from user A with a DISCONNECT message.	

ISI_XXSS3PTY07	**ISDN reference to: EN 300 188-1 [i.6], clause 9.2**	**NGN reference to: EN 383 001 [49]**
TSS reference:	ISDN-SIP-ISDN/Supplementary_services/3PTY	
Selection criteria:	The user A is in network N1 and is provided with 3PTY. The user B is in network N2 user C in the network N1.	
Test purpose:	Ensure that user A can establish a three-way conversation call with user B and user C and create a private communication with user C. The call clearing procedure is performed from user A	
ISDN Parameter values:	BC = speech	
Comments:	User A calls user B (with CRx). After initiating of call hold, the call A-B has an Active-Held connection. User A is calling user C (with the CRy). The call (A-C) has an Active-Idle connection. When user A sends a FACILITY message for CRx containing a facility IE with a Begin3PTY invoke component the network shall respond with a FACILITY message containing a facility IE with a Begin3PTY return result component for CRx. User C receives a NOTIFY message containing a Notification Indicator IE with a notification description of "Conference established". The three-way bridge is established. If the remote user, for which a private communication is required, is identified at the served user by the CRy relating to the Active-Idle connection, the served user shall send an End3PTY invoke component to the network in a FACILITY message with that CRy. On receiving such an invoke component in a FACILITY message, the network shall: i) remove the three-way bridge from both the Active-Idle connection and the Active-Held connection; ii) release the three-way bridge; iii) return to the served user an End3PTY return result component, within a FACILITY message, using the CRy of the Active-Idle connection; iv) send a NOTIFY message to both remote users containing a Notification indicator information element with a notification description of "Conference disconnected"; and, v) send to the remote user for which private communication is not required, either in the same NOTIFY message as (iv), or in a subsequent NOTIFY message, a Notification indicator information element with a notification description of "Remote hold". If any intervening protocol between the network of the served user and the network of the remote user does not support transmission of two notification descriptions in the same message, then this should be mapped at that point to a message containing a single notification description of "Conference disconnected", and a subsequent message containing a notification description of "Remote hold". When the served user receives a correctly encoded End3PTY return result component, within a FACILITY message, the user shall accept the provided information and take no further action. As a result of the procedures of this item of this clause, the call state and the auxiliary state of the connections, at both the network and the served user, are unchanged. The call clearing procedure is performed from user A with a DISCONNECT message	

ISS_XXSS3PTY08	**ISDN reference to: EN 300 188-1 [i.6], clause 9.2**	**NGN reference to: EN 383 001 [49]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/3PTY	
Selection criteria:	The user A is in network N1 and is provided with 3PTY. The user B is in network N2 user C in the network N2.	
Test purpose:	Ensure that user A can establish a three-way conversation call with user B and user C and release the Active-Idle connection (A-C). After the completion of the Retrieve function, the call clearing procedure is performed from user A.	
ISDN Parameter values:	BC = PIXIT	
Comments:		

UE A (ISDN)				UE B (SIP)			UE C (SIP)
SETUP(CRx)	→		→	INVITE			
ALERTING	←		←	180 Ringing			
CONNECT	←		←	200 OK			
HOLD(CRx)	→		→	INVITE(sendonly)			
			←	200 OK(recvonly)			
SETUP(CRy)	→					→	INVITE
ALERTING	←					←	180 Ringing
CONNECT	←					←	200 OK
FAC(3PTY_begin_invoke, CRx)	→		→	INVITE(sendrecv)			
DISC(3PTY_begin_ret_res, CRx)	←		←	200 OK(sendrecv)			
3 Party conversation							
DISC(CRy)	→		→	INVITE(sendonly)		→	BYE
RELEASE	←		←	200 OK(recvonly)		←	200 OK
REL COMP	→						
RETRIVE	→		→	INVITE(sendrecv)			
			←	200 OK(sendrecv)			
DISC(CRx)	→		→	BYE			
RELEASE	←		←	200 OK			
REL COMP	→						

ISS_XXSS3PTY09	**ISDN reference to: EN 300 188-1 [i.6], clause 9.2, figure A.2**	**NGN reference to: EN 383 001 [49]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/3PTY	
Selection criteria:	The user A is in network N1 and is provided with 3PTY. The user B is in network N2 user C in the network N2.	
Test purpose:	Ensure that user A can establish a three-way conversation call with user B and user C and release the Active-Held connection (A-B). The call clearing procedure is performed from user A.	
ISDN Parameter values:	BC = PIXIT	
Comments:		

ISS_XXSS3PTY10	**ISDN reference to: EN 300 188-1 [i.6], clause 9.2**	**NGN reference to: EN 383 001 [49]**
TSS reference:	ISDN-SIP-SIP/Supplementary_services/3PTY	
Selection criteria:	The user A is in network N1 and is provided with 3PTY. The user B is in network N2 user C in the network N2.	
Test purpose:	Ensure that user A can establish a three-way conversation call with user B and user C and user B sends disconnect during the Three-Party communication.	
ISDN Parameter values:	BC = PIXIT	
Comments:		

ISS_XXSS3PTY11	ISDN reference to: EN 300 188-1 [i.6], clause 9.2	NGN reference to: EN 383 001 [49]
TSS reference:	ISDN-SIP-SIP/Supplementary_services/3PTY	
Selection criteria:	The user A is in network N1 and is provided with 3PTY. The user B and user C are network N2.	
Test purpose:	Ensure that user A can establish a three-way conversation call with user B and user C and user C sends disconnect during the Three-Party communication.	
ISDN Parameter values:	BC = PIXIT	
Comments:		

ISS_XXSS3PTY12	ISDN reference to: EN 300 188-1 [i.6], clause 9.2	NGN reference to: EN 383 001 [49]
TSS reference:	ISDN-SIP-SIP/Supplementary_services/3PTY	
Selection criteria:	The user A is in network N1 and is provided with 3PTY. The user B is in network N2 user C in the network N2.	
Test purpose:	Ensure that user A can establish a three-way conversation call with user B and user C and release of both remote users, user C is released first.	
ISDN Parameter values:	BC = speech	
Comments:		

ISS_XXSS3PTY13	ISDN reference to: EN 300 188-1 [i.6], clause 9.2	NGN reference to: EN 383 001 [49]
TSS reference:	ISDN-SIP-SIP/Supplementary_services/3PTY	
Selection criteria:	The user A is in network N1 and is provided with 3PTY. The user B is in network N2 user C in the network N2	
Test purpose:	Ensure that user A can establish a three-way conversation call with user B and user C and create a private communication with user B. The call clearing procedure is performed from user A	
ISDN Parameter values:	BC = speech	
Comments:		

ISS_XXSS3PTY14	ISDN reference to: EN 300 188-1 [i.6], clause 9.2	NGN reference to: EN 383 001 [49]
TSS reference:	ISDN-SIP-SIP/Supplementary_services/3PTY	
Selection criteria:	The user A is in network N1 and is provided with 3PTY. The user B and user C are network N2.	
Test purpose:	Ensure that user A can establish a three-way conversation call with user B and user C and create a private communication with user C. The call clearing procedure is performed from user A	
ISDN Parameter values:	BC = speech	
Comments:		

6.1.2.10 HOLD

IS_SPSSHOLD 01	**ISDN reference to:** **EN 300 403-1 [i.3],** **EN 300 141-1 [i.7]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]**
TSS reference:	ISDN-SIP/SS/HOLD/	
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams	
ISDN selection criteria:	Support the generic notification procedure for HOLD supplementary service	
Test purpose:	Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold The calling party should be able to retrieve the other party The called party should be able to put the other party on hold The called party should be able to retrieve the other party	
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>	
ISDN Parameter values:		
Comments:		

ISDN		MGCF		SIP
SETUP	→		→	INVITE
ALERT	←		←	180 Ringing
CONNECT	←		←	200 OK INVITE
HOLD	→		→	INVITE(sendonly)
			←	200 OK INVITE(recvonly)
RETRIEVE	→		→	INVITE(sendrecv)
			←	200 OK INVITE(sendrecv)
NOTIFY(HOLD)	←		←	INVITE(sendonly)
			→	200 OK INVITE(recvonly)
NOTIFY(RETRIEVE)	←		←	INVITE(sendrecv)
			→	200 OK INVITE(sendrecv)

IS_SPSSHOLD 02	**ISDN reference to:** **EN 300 403-1 [i.3],** **EN 300 141-1 [i.7]**	**NGN reference to:** **EN 383 001 [49],** **ES 283 027 [48]**
TSS reference:	ISDN-SIP/SS/HOLD/	
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams Support the invocation of the service in the alerting state	
ISDN selection criteria:	Support the generic notification procedure for HOLD supplementary service	
Test purpose:	Ensure that a party can put the other party on hold in the alerting state. Ensure that the party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold The calling party should be able to retrieve the other party	
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>	
ISDN Parameter values:		
Comments:		

ISDN		MGCF		SIP
SETUP	→		→	INVITE
ALERTING	←		←	180 Ringing
HOLD	→		→	UPDATE(sendonly)
			←	200 OK UPDATE(recevonly)
RETRIEVE	→		→	UPDATE(sendrecv)
			←	200 OK UPDATE(sendrecv)

IS_SPSSHOLD 03	**ISDN reference to:** **EN 300 403-1 [i.3]** **EN 300 141-1 [i.7]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]**
TSS reference:	ISDN-SIP/SS/HOLD/	
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams	
ISDN selection criteria:	Support the generic notification procedure for HOLD supplementary service	
Test purpose:	Ensure that a party can put the other party on hold after the calling user has provided all of the information necessary for processing the call. Ensure that the party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold The calling party should be able to retrieve the other party	
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>	
ISDN Parameter values:		
Comments:		

ISDN		MGCF		SIP
SETUP	→		→	INVITE
HOLD	→		→	UPDATE(sendonly)
			←	200 OK UPDATE(recevonly)
RETRIEVE	→		→	UPDATE(sendrecv)
			←	200 OK UPDATE(sendrecv)
ALERTING	←		←	180 Ringing
CONNECT	←		←	200 OK INVITE

IS_SPSSHOLD 04	**ISDN reference to:** **EN 300 403-1 [i.3]** **EN 300 141-1 [i.7]**	**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]**
TSS reference:	ISDN-SIP/SS/HOLD/	
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams The MGCF sends the update of the media stream in an UPDATE message	
ISDN selection criteria:	Support the generic notification procedure for HOLD supplementary service	
Test purpose:	Ensure that a party can put the other party on hold in the confirmed state using an UPDATE request. Ensure that the party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold The calling party should be able to retrieve the other party	
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>	
ISDN Parameter values:	FACILITY: Generic notification: remote hold Event indicator PROGRESS (put on hold) Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)	
Comments:		

ISDN		MGCF		SIP
SETUP	→		→	INVITE
ALERTING	←		←	180 Ringing
CONNECT	←		←	200 OK INVITE
HOLD	→		→	UPDATE(sendonly)
			←	200 OK UPDATE(recevonly)
RETRIEVE	→		→	UPDATE(sendrecv)
			←	200 OK UPDATE(sendrecv)

IS_SPSSHOLD 05	**ISDN reference to:** **EN 300 403-1 [i.3]** **EN 300 141-1 [i.7]**			**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]**	
TSS reference:	ISDN-SIP/SS/HOLD/				
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams				
ISDN selection criteria:	Support the generic notification procedure for HOLD supplementary service				
Test purpose:	Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold The called party should be able to put the other party on hold The calling party should be able to retrieve the other party The called party should be able to retrieve the other party				
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>				
ISDN Parameter values:					
Comments:					
	ISDN		MGCF		SIP
	SETUP	→		→	INVITE
	ALERTING	←		←	180 Ringing
	CONNECT	←		←	200 OK INVITE
	HOLD	→		→	INVITE(sendonly)
				←	200 OK INVITE(recvonly)
	NOTIFY Remote HOLD	←		←	INVITE(inactive)
				→	200 OK INVITE(inactive)
	RETRIEVE	→		→	INVITE(recvonly)
				←	200 OK INVITE(sendonly)
	NOTIFY Remote RETRIEVAL	←		←	INVITE(sendrecv)
				→	200 OK INVITE(sendrecv)

IS_SPSSHOLD 06	**ISDN reference to:** **EN 300 403-1 [i.3]** **EN 300 141-1 [i.7]**			**NGN reference to:** **EN 383 001 [49]** **ES 283 027 [48]**	
TSS reference:	ISDN-SIP/SS/HOLD/				
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams				
ISDN selection criteria:	Support the generic notification procedure for HOLD supplementary service				
Test purpose:	Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold The called party should be able to put the other party on hold The called party should be able to retrieve the other party The calling party should be able to retrieve the other party				
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>				
ISDN Parameter values:					
Comments:					
	ISDN		MGCF		SIP
	SETUP	→		→	INVITE
	ALERTING	←		←	180 Ringing
	CONNECT	←		←	200 OK INVITE
	HOLD	→		→	INVITE(sendonly)
				←	200 OK INVITE(recvonly)
	NOTIFY- Remote HOLD	←		←	INVITE(inactive)
				→	200 OK INVITE(inactive)
	NOTIFY - Remote RETRIEVAL	←		←	INVITE(recvonly)
				→	200 OK INVITE(sendonly)
	RETRIEVE	→		→	INVITE(sendrecv)
				←	200 OK INVITE(sendrecv)

6.1.2.11 CONF (Outgoing Call)

IS_SPSSCONF01	**ISDN reference to:** **EN 300 403-1 [i.3]** **EN 300 185-1 [i.8]**			**NGN reference:** **TS 183 005 [46]** **ES 283 027 [48], clause 7.4.14**	
TSS reference:	ISDN-SIP/SS/CONF/				
SIP selection criteria:					
ISDN selection criteria:	SUPPORT OF SERVICE CONFERENCE CALL, ADD-ON (CONF)				
Test purpose:	Ensure that the SUT in the confirmed dialogue can establish a conference				
SIP Parameter values:					
ISDN Parameter values:					
Comments:					
	ISDN		MGCF		SIP
	SETUP	→		→	INVITE
	ALERTING	←		←	180 Ringing
	CONNECT	←		←	200 OK INVITE
	BeginCONF	→			
	DISC	→		→	BYE
	RELEASE	←		←	200 OK BYE
	REL_COMP	→			

IS_SPSSCONF03	**ISDN reference to:** **EN 300 403-1 [i.3]** **EN 300 185-1 [i.8]**		**NGN reference:** **TS 183 005 [46]** **ES 283 027 [48], clause 7.4.14**		
TSS reference:	ISDN-SIP/SS/CONF/				
SIP selection criteria:					
ISDN selection criteria:	SUPPORT OF SERVICE CONFERENCE CALL, ADD-ON (CONF)				
Test purpose:	Ensure that the SUT in the confirmed dialogue can establish a conference, can isolate a party and reattach this party				
SIP Parameter values:	SDP: a= a_LINE_VA **(Table 5) or a line is omitted**				
ISDN Parameter values:					
Comments:					
	ISDN		MGCF		SIP
	SETUP	→		→	INVITE
	ALERTING	←		←	180 Ringing
	CONNECT	←		←	200 OK INVITE
	BeginCONF	→			
	IsolatedCONF	→		→	INVITE(sendonly)
				←	200 OK INVITE(recvonly
				→	ACK
	ReattachCONF	→		→	INVITE(sendrecv)
				←	200 OK INVITE(sendrecv)
				→	ACK
	DISC	→		→	BYE
	RELEASE	←		←	200 OK BYE
	REL_COMP	→			

IS_SPSSCONF05	**ISDN reference to:** **EN 300 403-1 [i.3]** **EN 300 185-1 [i.8]**		**NGN reference:** **TS 183 005 [46]** **TS 183 005 [46], clause 7.4.1.1.1**		
TSS reference:	ISDN-SIP/SS/CONF/				
SIP selection criteria:	Conference event package supported				
ISDN selection criteria:					
Test purpose:	*Conference notification information is mapped into "conference established"* Upon the receipt of a conference information document with the <conference-state-type> element *active* is set to 'true', the ISDN Network shall send a NOTIFY message '*conference established*'.				
SIP Parameter values:	NOTIFY 1: *<conference-state>* <active>true</active> if present				
ISDN Parameter values:	AddCONF				
Comments:					
	ISDN		MGCF		SIP
	SETUP	→		→	INVITE
	ALERTING	←		←	180 Ringing
	CONNECT	←		←	200 OK INVITE
	NOTIFY(conference established)	←		←	NOTIFY 1
				→	200 OK NOTIFY
	REL	→		→	BYE
	RLC	←		←	200 OK BYE

IS_SPSSCONF06	**ISDN reference to:** **EN 300 403-1 [i.3]** **EN 300 185-1 [i.8]**			**NGN reference:** **TS 183 005 [46]** **TS 183 005 [46], clause 7.4.1.1.1**	
TSS reference:	ISDN-SIP/SS/CONF/				
SIP selection criteria:	Conference event package supported				
ISDN selection criteria:					
Test purpose:	*Conference notification information is mapped into "other party added"* Upon the receipt of a conference information document with the <endpoint-type> and the element *status of endpoint-status-type* is set to 'connected' and it was not set to 'on-hold' before and the Contact URI in the element *entity* is not the address of the served PSTN/ISDN participant, the ISDN Network shall send a NOTIFY message '*other party added*'.				
SIP Parameter values:	NOTIFY 1: *<conference-state>* <active>true</active> if present NOTIFY 2: <endpoint entity=endpoint SIPx URI <status>connected</status>				
ISDNParameter values:	FACILITY(other party added)				
Comments:					
	ISDN		MGCF		SIP
	SETUP	→		→	INVITE
	ALERTING	←		←	180 Ringing
	CONNECT	←		←	200 OK INVITE
	NOTIFY(conference established)	←		←	NOTIFY 1
				→	200 OK NOTIFY
	NOTIFY(other party added)	←		←	NOTIFY 2
				→	200 OK NOTIFY
	REL	→		→	BYE
	RLC	←		←	200 OK BYE
	The connection to SIP2 is not shown in the message flow.				

IS_SPSSCONF11	**ISDN reference to: EN 300 403-1 [i.3] EN 300 185-1 [i.8]**		**NGN reference: TS 183 005 [46], clause 7.4.1.1.1**		
TSS reference:	ISDN-SIP/SS/CONF/				
SIP selection criteria:	Conference event package supported				
ISDN selection criteria:					
Test purpose:	*Conference notification information is mapped into "other party disconnected"* Upon the receipt of a conference information document with the <endpoint-type> and the element *status of endpoint-status-type* is set to 'disconnected' and the element *joining-method of joining-type* is not set to 'focus-owner, the ISDN network shall send a NOTIFY message '*other party disconnected*'.				
SIP Parameter values:	NOTIFY 3: <endpoint entity=endpoint SIPx URI <status>disconnected</status>				
Comments:					
	ISDN		MGCF		SIP
	SETUP	→		→	INVITE
	ALERTING	←		←	180 Ringing
	CONNECT	←		←	200 OK INVITE
	NOTIFY(conference established)	←		←	NOTIFY 1
				→	200 OK NOTIFY
	NOTIFY(other party added)	←		←	NOTIFY 2
				→	200 OK NOTIFY
	NOTIFY(other party disconnected)	←		←	NOTIFY 3
				→	200 OK NOTIFY
	REL	→		→	BYE
	RLC	←		←	200 OK BYE
	The connection to SIP2 is not shown in the message flow.				

IS_SPSSCONF12	**ISDN reference to: EN 300 403-1 [i.3] EN 300 185-1 [i.8]**		**NGN reference: TS 183 005 [46], clause 7.4.14**		
TSS reference:	ISDN-SIP/SS/CONF/				
SIP selection criteria:	The temporarily stops sending one or more unicast media streams is not supported				
ISDN selection criteria:	SUPPORT OF SERVICE CONFERENCE CALL, ADD-ON (CONF)				
Test purpose:	Ensure that the SUT on receipt of FACILITY messages due to the CONF supplementary service, **no mapping, no disrupting of the SIP procedure.**				
SIP Parameter values:					
ISDN Parameter values:					
Comments:					
	ISDN		MGCF		SIP
	SETUP	→		→	INVITE
	ALERTING	←		←	180 Ringing
	CONNECT	←		←	200 OK INVITE
	HOLD	→			
	BeginCONF	→			
	IsolatedCONF	→			
	ReattachCONF				
	REL	→		→	BYE
	RLC	←		←	200 OK BYE

IS_SPSSCONF13	**ISDN reference to:** **EN 300 403-1 [i.3]** **EN 300 185-1 [i.8]**				**NGN reference:** **TS 183 005 [46], clause 7.4.1.1.1**
TSS reference:	ISDN-SIP/SS/CONF/				
SIP selection criteria:	Conference event package not supported				
ISDN selection criteria:					
Test purpose:	*Conference notification information is not mapped to ISDN* Upon the receipt of a conference information document the conference notification information is not mapped to the PSTN side. No NOTIFY is sent to the ISDN user.				
SIP Parameter values:	NOTIFY 1: *<conference-state>* <active>**true**</active> if present NOTIFY 2: <endpoint entity=endpoint SIPx URI <status>**connected**</status>				
ISDNParameter values:					
Comments:					
	ISDN		MGCF		SIP
	SETUP	→		→	INVITE
	ALERTING	←		←	180 Ringing
	CONNECT	←		←	200 OK INVITE
				←	NOTIFY 1
				→	200 OK NOTIFY
				←	NOTIFY 2
				→	200 OK NOTIFY
	REL	→		→	BYE
	RLC	←		←	200 OK BYE
	The connection to SIP2 is not shown in the message flow.				

IS_SPSSCONF14	**ISDN reference to:** **EN 300 403-1 [i.3]** **EN 300 185-1 [i.8]**				**NGN reference:** **ETSI TS 183 005 [46], clause 7.4.1.1.1**
TSS reference:	ISDN-SIP/SS/CONF/				
SIP selection criteria:					
ISDN selection criteria:					
Test purpose:	*The referring of MGCF is not possible a call is established* Ensure that a REFER request received by the MGCF is not successful. The request is rejected with . 403 Forbidden. The CS -site is not affected.				
SIP Parameter values:	REFER: Request URI contained the conference URI Refer-To contains the URI of ISDNx, method=invite Referred-By contains SIP or tel URI of SIPx				
ISDN Parameter values:	AddCONF				
Comments:					
	ISDN		MGCF		SIP
	SETUP	→		→	INVITE
	ALERTING	←		←	180 Ringing
	CONNECT	←		←	200 OK INVITE
				←	REFER
				→	403 Forbidden

6.2 Test purposes for SIP-ISDN

6.2.1 Basic Call

6.2.1.1 Test purposes for SIP-ISDN, Basic call, Successful 3,1 kHz audio

SI_AU_01	**ISDN reference to:** **EN 300 403-1 [i.3],clause 5.2.1** **EN 300 899-1 [23], clause 3.1.1**		**NGN reference to:** **Q.1912.5 [51] clause 6.1.1, 6.1.3** **EN 383 001 [49], clause 6.1.1, 6.1.3** **ES 283 027 [48]**		
TSS reference:	SIP-ISDN/Basic_call/Successful/3,1 kHz audio				
Selection criteria:	INVITE received without an SDP offer; not ES 283 027 [48]				
Test purpose:	Ensure that call establishment, upon receipt of the first INVITE with sufficient digits, without an SDP offer and reliable provisional responses is supported, is performed correctly.				
ISDN Parameter values:	SETUP = 3,1 kHz audio				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	INVITE	→			
	183 Session Progress offer 1	←			
	PRACK answer 1	→			
	200 OK PRACK	←			
				→	SETUP
	180 Ringing	←		←	ALERT
	PRACK	→			
	200 OK PRACK	←			
	200 OK INVITE	←		←	CON
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_AU_02	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.2.1** **EN 300 899-1 [23], clause 3.1.1**			**NGN reference to:** **Q.1912.5 [51], clauses 6.1.1, 6.1.3** **EN 383 001 [49], clauses 6.1.1, 6.1.3**
TSS reference:	SIP-ISDN/Basic_call/Successful/3,1 kHz audio			
Selection criteria:	INVITE received without an SDP offer; not 283 027			
Test purpose:	Ensure that call establishment with an INVITE which does not contain an SDP offer and reliable provisional responses is not supported, is performed correctly.			
ISDN Parameter values:	SETUP = 3,1 kHz audio;			
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)			
Comments:				

	ISDN		SUT		SIP
	INVITE	→			
	200 OK INVITE offer 1	←		→	SETUP
	PRACK answer 1	→		←	CON
	ACK answer 1	←			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_AU_03	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.2.1** **EN 300 899-1 [23], clause 3.1.1**		**NGN reference to:** **Q.1912.5 [51], clauses 6.1.2, 6.1.3** **EN 383 001 [49], clauses 6.1.1, 6.1.3** **ES 283 027 [48], clause 7.2.3.1**		
TSS reference:	SIP-ISDN/Basic_call/Successful/3,1 kHz audio				
Selection criteria:	INVITE received with an SDP offer.				
Test purpose:	Ensure that call establishment with an INVITE which contains an SDP offer is performed correctly.				
ISDN Parameter values:	SETUP = 3,1 kHz audio;				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	A) Without SDP pre-condition				
	INVITE SDP	→		→	SETUP
	100 Trying	←			
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CON
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL
	B) ES 283 027 [48] (pre-condition and 100 rel)				
	INVITE SDP	→			
	100 Trying	←			
	183 Session Progress SDP	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→			
	200 OK (UPDATE)	←			
				→	SETUP
	180 Ringing	←		←	ALERT
	PRACK	→			
	200 OK PRACK	←			
	200 OK INVITE	←		←	CON
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_AU_04	**ISDN reference to: EN 300 403-1 [i.3], clause 5.2.1 EN 300 899-1 [23], clause 3.1.1**		**NGN reference to: Q.1912.5 [51], clause 6.1.3 EN 383 001 [49], clause 6.1.3 ES 283 027 [48], clause 7.2.3.1**		
TSS reference:	SIP-ISDN/Basic_call/Successful/3,1 kHz audio				
Selection criteria:					
Test purpose:	Ensure that call establishment and the mapping of the a = line b=line and m =line parameters defined in table 5 between INVITE message and the SETUP message is performed correctly. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).				
ISDN Parameter values:	SETUP = 3,1 kHz audio;				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT- Table 5) b = line (PIXIT - Table 5) m = line (PIXIT - Table 5)				
Comments:					
	ISDN		SUT		SIP
	A) Without SDP pre-condition				
	INVITE	→		→	SETUP
	100 Trying	←			
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CON
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL
	b) ES 283 027 [48] (pre-condition met)				
	INVITE SDP	→			
	100 Trying	←			
	183 Session Progress SDP	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→			
	200 OK (UPDATE)	←			
				→	SETUP
	180 Ringing	←		←	ALERT
	PRACK	→			
	200 OK PRACK	←			
	200 OK INVITE	←		←	CON
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_AU_04A	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.2.1** **EN 300 899-1 [23], clause 3.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 6.5** **EN 383 001 [49], clause 6.5** **ES 283 027 [48], clause 7.2.3.1**		
TSS reference:	SIP-ISDN/Basic_call/Successful/3,1 kHz audio				
Selection criteria:	ISDN = point to point Configuration: with DDI; P-Early-Media Header is supported				
Test purpose:	Ensure that the SIP user receives a 183 Session Progress message when the ISDN User in call state U03 is sending a Call Proceedingmessage. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters). Ensure that in the Call Delivered call state U4 the transfer of tone or announcement on the media channel is performed correctly. In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in table 6 applies.				
ISDN Parameter values:	SETUP = 3,1 kHz audio;				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT- Table 6) b = line (PIXIT - Table 6) m = line (PIXIT - Table 6)				
Comments:					
	ISDN		SUT		SIP
	a) Without SDP pre-condition				
	INVITE	→		→	SETUP
	183 Session Progress Including the P-Early-Media Header	←		←	CALL PROCEEDING PI#8
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CON
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL
	b) ES 283 027 [48] (pre-condition met)				
	INVITE SDP	→			
	100 Trying	←			
	183 Session Progress SDP	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→			
	200 OK (UPDATE)	←			
				→	SETUP
	180 Ringing	←		←	ALERT
	PRACK	→			
	200 OK PRACK	←			
	200 OK INVITE	←		←	CON
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_AU_05	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.2.1** **EN 300 899-1 [23], clause 3.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 6.5** **EN 383 001 [49], clause 6.5** **ES 283 027 [48], clause 7.2.3.1**		
TSS reference:	SIP-ISDN/Basic_call/Successful/3,1 kHz audio				
Selection criteria:	ISDN = point to point Configuration: with DDI				
Test purpose:	Ensure that the SIP user receives a 180 Ringing message when the ISDN User in call state U07 is sending an ALERTING message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters). Ensure that in the Call Delivered call state U4 the transfer of tone or announcement on the media channel is performed correctly. In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in table 6 applies.				
ISDN Parameter values:	SETUP = 3,1 kHz audio				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT - Table 6) b = line (PIXIT - Table 6) m = line (PIXIT - Table 6)				
Comments:					
	ISDN		SUT		SIP
	a) Without SDP pre-condition				
	INVITE	→		→	SETUP
	100 Trying	←			
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CON
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL
	b) ES 283 027 [48] (pre-condition met)				
	INVITE SDP	→			
	100 Trying	←			
	183 Session Progress SDP	←			
	PRACK	→			
	200 OK PRACK	←			
	UPDATE	→			
	200 OK (UPDATE)	←			
				→	SETUP
	180 Ringing	←		←	ALERT
	PRACK	→			
	200 OK PRACK	←			
	200 OK INVITE	←		←	CON
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_AU_05A	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.2.1** **EN 300 899-1 [23], clause 3.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 6.5** **EN 383 001 [49], clause 6.5** **ES 283 027 [48], clause 7.2.3.1**
TSS reference:	SIP-ISDN/Basic_call/Successful/3,1 kHz audio	
Selection criteria:	ISDN = point to point Configuration: with DDI P-Early-Media Header is supported	
Test purpose:	Ensure that the SIP user receives a 180 Ringing message when the ISDN User in call state U07 is sending an ALERTING message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters). Ensure that in the Call Delivered call state U4 the transfer of tone or announcement on the media channel is performed correctly. In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in table 6 applies.	
ISDN Parameter values:	SETUP = 3,1 kHz audio	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT- Table 6) b = line (PIXIT - Table 6) m = line (PIXIT - Table 6)	
Comments:		

ISDN		SUT		SIP
a) Without SDP pre-condition				
INVITE	→		→	SETUP
100 Trying	←			
180 Ringing Including the P-Early-Media Header	←		←	ALERT
200 OK INVITE	←		←	CON
ACK	→			
BYE	←		←	DISC
200 OK BYE	→		→	REL
b) ES 283 027 [48] (pre-condition met)				
INVITE SDP	→			
100 Trying	←			
183 Session Progress SDP	←			
PRACK	→			
200 OK PRACK	←			
UPDATE	→			
200 OK (UPDATE)	←			
			→	SETUP
180 Ringing Including the P-Early-Media Header	←		←	ALERT PI#8
PRACK	→			
200 OK PRACK	←			
200 OK INVITE	←		←	CON
ACK	→			
BYE	←		←	DISC
200 OK BYE	→		→	REL

SI_AU_07	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.2.1** **EN 300 899-1 [23], clause 3.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 6.5** **EN 383 001 [49], clause 6.5** **ES 283 027 [48], clause 7.2.3.1**		
TSS reference:	SIP-ISDN/Basic_call/Successful/3,1 kHz audio				
Selection criteria:					
Test purpose:	Ensure that SIP user receives a 180 Ringing message when the ISDN User in call state N09 is sending an ALERTING message. Ensure that the ringing tone can be heard in the early dialogue Ensure that in the active call state (N10) the transfer of tone or announcement on the media and B-channels is performed correctly (e.g. testing QoS parameters).				
ISDN Parameter values:	SETUP = 3,1 kHz audio;				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	a) Without SDP pre-condition				
	INVITE	→		→	SETUP
	100 Trying	←			
				←	CALL PROC
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CON
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL
	b) SDP pre-condition met				
	INVITE SDP	→			
	100 Trying	←			
	183 Session Progress SDP	←			
	PRACK	→			
	200 OK (PRACK)	←			
	UPDATE	→			
	200 OK (UPDATE)	←			
				→	SETUP
				←	CALL PROC
	180 Ringing	←		←	ALERT
	PRACK	→			
	200 OK (PRACK)	←			
	200 OK INVITE	←		←	CON
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_AU_08	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.2.1** **EN 300 899-1 [23], clause 3.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 6.5** **EN 383 001 [49], clause 6.5** **ES 283 027 [48], clause 7.2.3.1**	
TSS reference:	SIP-ISDN/Basic_call/Successful/3,1 kHz audio			
Selection criteria:	FAX G3			
Test purpose:	Ensure that call establishment and the mapping of the defined SDP parameters for T.38 between INVITE message and the SETUP message is performed correctly. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).			
ISDN Parameter values:	SETUP = 3,1 kHz audio HLC = "Facsimile Group 2/3"			
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line Based on T.38. b = line AS: 64 m = line: **VA_Transport**; T38 (Table 4)			
Comments:				

	SIP		SUT		ISDN
	a) Without SDP pre-condition				
	INVITE	→		→	SETUP
	100 Trying	←			
				←	CALL PROC
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CON
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL
	b) SDP pre-condition met				
	INVITE SDP	→			
	100 Trying	←			
	183 Session Progress SDP	←			
	PRACK	→			
	200 OK (PRACK)	←			
	UPDATE	→			
	200 OK (UPDATE)	←			
				→	SETUP
				←	CALL PROC
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CON
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

Table 4

Parameter transport protocol VA_Transport	
VA_Transport_1	udptl
VA_Transport_2	tcptl

SI_AU_09	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.2.1** **EN 300 899-1 [23], clause 3.1.1**				**NGN reference to:** **Q.1912.5 [51], clause 6.5** **EN 383 001 [49], clause 6.5** **ES 283 027 [48], clause 7.2.3.1**
TSS reference:	SIP-ISDN/Basic_call/Successful/3,1 kHz audio				
Selection criteria:	FAX G3-T.30				
Test purpose:	Ensure that call establishment and the mapping of the defined SDP parameters INVITE message and the SETUP message is performed correctly. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).				
ISDN Parameter values:	SETUP = 3,1 kHz audio;				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	a) Without SDP pre-condition				
	INVITE	→		→	SETUP
	100 Trying	←			
				←	CALL PROC
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CON
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL
	b) SDP pre-condition met				
	INVITE SDP	→			
	100 Trying	←			
	183 Session Progress SDP	←			
	PRACK	→			
	200 OK (PRACK)	←			
	UPDATE	→			
	200 OK (UPDATE)	←			
				→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CON
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_AU_10	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.3.3** **EN 300 899-1 [23], clause 3.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 6.11** **EN 383 001 [49], clause 6.11** **ES 283 027 [48], clause 7.2.3.1**		
TSS reference:	SIP-ISDN/Basic_call/Successful/3,1 kHz audio				
Selection criteria:					
Test purpose:	Ensure that the call establishment and the call clearing procedure are performed correctly when the **calling user** clears after answering with a BYE message. The called user shall receive a DISCONNECT message indicating the Cause value # 16 "normal call clearing" with the progress indicator #8 or a Progress message with the progress indicator #8.				
ISDN Parameter values:	SETUP = 3,1 kHz audio;				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	a) Without SDP pre-condition				
	INVITE	→		→	SETUP
	100 Trying	←			
				←	CALL PROC
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CON
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL
	b) SDP pre-condition met				
	INVITE SDP	→			
	100 Trying	←			
	183 Session Progress SDP	←			
	PRACK	→			
	200 OK (PRACK)	←			
	UPDATE	→			
	200 OK (UPDATE)	←			
				→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CON
	ACK	→			
	BYE	→		→	DISC
	200 OK BYE	←		←	REL

SI_AU_11	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.3.3** **EN 300 899-1 [23], clause 3.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 6.11** **EN 383 001 [49], clause 6.11** **ES 283 027 [48], clause 7.2.3.1**		
TSS reference:	SIP-ISDN/Basic_call/Successful/3,1 kHz audio				
Selection criteria:					
Test purpose:	Ensure that the call clearing procedure is performed correctly when the **called user** clears after answering with a DISCONNECT message indicating the Cause value # 16 "normal call clearing". The calling user shall receive a BYE message. A reason header field with value 16 is sent in case of EN 383 001 [49] and ES 283 027 [48], optional in Q.1912.5 [51].				
ISDN Parameter values:	SETUP = 3,1 kHz audio;				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	a) Without SDP pre-condition				
	INVITE	→		→	SETUP
	100 Trying	←			
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CON
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL
	b) SDP pre-condition met				
	INVITE SDP	→			
	183 Session Progress SDP	←			
	PRACK	→			
	200 OK (PRACK)	←			
	UPDATE	→			
	200 OK (UPDATE)	←			
				→	SETUP
	180 Ringing	←		←	ALERT
	PRACK	→			
	200 OK (PRACK)	←			
	200 OK INVITE	←		←	CON
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

Table 5

m= line				B= line	A= line	BC parameter (see note 2)		HLC parameter
	<media>	**<transport>**	**<fmt-list>**	**<modifier>:<bandwidth-value> (see note 3)**	**rtpmap:<dynamic-PT> <encoding name>/ <clock rate>[/encoding parameters>**	**Information Transport Capability**	**User Information Layer 1 Protocol Indicator**	**High Layer Characteristics Identification**
VA_01	audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A	*"3,1 kHz audio"*	*"G.711 A-law "*	Note 2
VA_02	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMU/8000	*"3,1 kHz audio"*	*"G.711 A-law "*	Note 2
VA_03	audio audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A	*"3,1 kHz audio"*	*"G.711 A-law "*	Note 2
VA_04	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMA/8000	*"3,1 kHz audio"*	*"G.711 A-law"*	Note 2

NOTE 1: In this table the codec G.711 is used only as an example. Other codec is possible.
NOTE 2: HLC normally absent in this case. It is possible for HLC to be present with the value "Telephony", although clause 6.3.1/Q.939 [i.18] indicates that this would normally be accompanied by a value of "Speech" for the Information Transfer Capability element.
NOTE 3: <bandwidth value> for <modifier> of AS is evaluated to be B kbit/s.

Table 6: Values for test purposes SI_AU_04 and SI_AU_11

VARIABLE	PT	Encoding	media type	clock rate	channels
VA_01	0	PCMU	A	8,000	1
VA_02	3	GSM	A	8,000	1
VA_03	4	G723	A	8,000	1
VA_04	5	DVI4	A	8,000	1
VA_05	7	LPC	A	8,000	1
VA_06	8	PCMA	A	8,000	1
VA_07	9	G722	A	8,000	1
VA_08	12	QCELP	A	8,000	1
VA_09	13	CN	A	8,000	1
VA_10	18	G729	A	8,000	1
VA_11	Dyn	G726-40	A	8,000	1
VA_12	Dyn	G726-32	A	8,000	1
VA_13	Dyn	G726-24	A	8,000	1
VA_14	Dyn	G726-16	A	8,000	1
VA_15	Dyn	G729D	A	8,000	1
VA_16	Dyn	G729E	A	8,000	1
VA_17	Dyn	GSM-EFR	A	8,000	1

<table>
<tr><th>SI_AU_12</th><th colspan="4">ISDN reference to:
EN 300 403-1 [i.3], clause 5.2.1
EN 300 899-1 [23,], clause 3.1.1</th><th>NGN reference to:
Q.1912.5 [51], clause 6.1.3
EN 383 001 [49], clause 6.1.3
ES 283 027 [48], clause 7.2.3.1</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISDN/Basic_call/Successful/3,1 kHz audio</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">FAX - inband</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that call establishment and the mapping of the a = line: PCMA/8000; b=line - 64 kbit/s and m = RTP/AVP 8 for FAX - inband between INVITE message and the SETUP message is performed correctly. Ensure that in the active call state (N10) the data transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">SETUP = 3,1 kHz audio;</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line: PCMA/8000 or PCMU/8000 (PIXIT)
b = line: 64 kbit/s
m = line: = RTP/AVP 8 or 0 (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="27"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td colspan="5">a) Without SDP pre-condition</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>100 Trying</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CON</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td colspan="5">b) SDP pre-condition met</td></tr>
<tr><td>INVITE SDP</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>100 Trying</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>183 Session Progress SDP</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK (PRACK)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK (UPDATE)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK (PRACK)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CON</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

SI_AU_13	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.2.1** **EN 300 899-1 [23], clause 3.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.1, 6.1.3** **EN 383 001 [49], clause 6.1.1, 6.1.3** **ES 283 027 [48], clause 7.2.3.1**
TSS reference:	SIP-ISDN/Basic_call/Successful/3,1 kHz audio	
Selection criteria:	SIP Profile A or Q.1912.5 [51] Profile B with PI or EN 383 001 [49] or ES 283 027 [48]	
Test purpose:	Ensure that call establishment upon receipt of the first INVITE with sufficient digits, is performed correctly. During call establishment a Progress indicator information element shall be included in the SETUP message sent to the called user with progress description value #1 "call is not end-to-end ISDN"	
ISDN Parameter values:	SETUP = 3,1 kHz audio;	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SIP		SUT		ISDN
a) Without SDP pre-condition				
INVITE	→		→	SETUP PI #1
100 Trying	←			
180 Ringing	←		←	ALERT
200 OK INVITE	←		←	CON
ACK	→			
BYE	←		←	DISC
200 OK BYE	→		→	REL
b) SDP pre-condition met				
INVITE SDP	→			
100 Trying	←			
183 Session Progress SDP	←			
PRACK	→			
200 OK (PRACK)	←			
UPDATE	→			
200 OK (UPDATE)	←			
			→	SETUP PI #1
180 Ringing	←		←	ALERT
PRACK	→			
200 OK (PRACK)	←			
200 OK INVITE	←		←	CON
ACK	→			
BYE	←		←	DISC
200 OK BYE	→		→	REL
NOTE: The 183 Session Progress message with SDP answer should be sent only when the OBCI in the ALERTING is set to: inband info or appropriate pattern is now available.				

SI_AU_14	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.2.1** **EN 300 899-1 [23], clause 3.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 6.1.1, 6.1.3** **EN 383 001 [49], clause 6.1.1, 6.1.3** **ES 283 027 [48], clause 7.2.3.1**		
TSS reference:	SIP-ISDN/Basic_call/Successful/3,1 kHz audio				
Selection criteria:					
Test purpose:	During the session, the **calling user** decides to change the characteristics of the media session. This is accomplished by sending a re-INVITE containing a new media description. This re-INVITE references the existing dialog so that the IWU knows that it is to modify an existing session instead of establishing a new session. The IWU sends a 200 (OK) to accept the change. The requestor responds to the 200 (OK) with an ACK. In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in table 16 applies.				
ISDN Parameter values:	SETUP = 3,1 kHz audio;				
ISDN Parameter values:	SETUP = 3,1 kHz audio;				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	a) Without SDP pre-condition				
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERTING
	200 OK INVITE	←		←	CON
	ACK	→			
	RE-INVITE	→			
	200 OK	←			
	ACK	→			
			Communication		
	BYE	←		←	DISC
	200 OK BYE	→		→	REL
	b) SDP pre-condition met				
	INVITE	→			
	183 Session Progress	←			
	PRACK	→			
	200 OK (PRACK)	←			
	UPDATE	→			
	200 OK (UPDATE)	←			
				→	SETUP
	180 Ringing	←		←	ALERTING
	PRACK	→			
	200 OK (PRACK)	←			
	200 OK INVITE	←		←	CON
	ACK	→			
			Communication		
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_AU_15	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.3.3** **EN 300 899-1 [23], clause 3.1.1**	**NGN reference to:** **ES 283 027 [48], clause 7.2.3.1** **TS 183 028 [i.19]**
TSS reference:	SIP-ISDN/Basic_call/Successful/3,1 kHz audio	
Selection criteria:	Announcements controlled provided by the PSTN/ISDN Providing announcements to a user during the establishment of a communication session	
Test purpose:	During the establishment of the communication the PSTN/ISDN provides an announcement e.g. "The communication is forwarded" or "The user is not reachable". Ensure that the transfer of tone or announcement on the media is performed correctly. The flow assume the use of the option-tag "100rel.	
ISDN Parameter values:	SETUP = 3,1 kHz audio;	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SI_AU_16	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.3.3** **EN 300 899-1 [23], clause 3.1.1**	**NGN reference to:** **ES 283 027 [48], clause 7.2.3.1** **TS 183 028 [i.19]**
TSS reference:	SIP-ISDN/Basic_call/Successful/3,1 kHz audio	
Selection criteria:	Announcements controlled provided by the PSTN/ISDN Providing announcements to a user during the establishment of a communication session	
Test purpose:	During the establishment of the communication the PSTN/ISDN provides an announcement e.g. "The communication is forwarded" or "The user is not reachable". Ensure that the transfer of tone or announcement on the media is performed correctly. The flow assumes the use of the P-Ealry media header.	
ISDN Parameter values:	SETUP = 3,1 kHz audio;	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

IS_SP_08A	**ISDN reference to: EN 300 403-1 [i.3] EN 300 899-1 [23]**		**NGN reference to: TS 129 163 [i.20], clause 7.2.3.1.3A.1**		
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice				
Selection criteria:	Basic_call; 283 027 overlap receiving supported; In-Dialog Method				
Test purpose:	Ensure that call establishment using **overlap sending** is performed correctly. Ensure that the SIP user receives an 180 Ringing message when the ISDN user answers with an ALERTING message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).				
ISDN Parameter values:	BC=speech, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition				
Comments:					
	INVITE	→		→	SETUP
				←	SETUP ACK
	183 Session progress	←			
	INFO	→		→	INFO
	200 OK	←			
	183 Session progress	←		←	Call proceeding
	180 Ringing	←		←	ALERTING
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	→		→	DISC
	200 OK BYE	←		←	REL

IS_SP_08B	**ISDN reference to: EN 300 403-1 [i.3] EN 300 899-1 [23]**			**NGN reference to: TS 129 163 [i.20], clause 7.2.3.1.3A.1**	
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice				
Selection criteria:	Basic_call; 283 027 overlap receiving supported; Multiple INVITE Method; one dialog used				
Test purpose:	Ensure that call establishment using **overlap sending** is performed correctly. Ensure that the SIP user receives an 180 Ringing message when the ISDN user answers with an ALERTING message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).				
ISDN Parameter values:	BC=speech, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition				
Comments:					
	INVITE	→		→	SETUP
	484	←		←	SETUP ACK
	INVITE	→			
				→	INFO
	484	←			
	INVITE	→		→	INFO
	183 Session progress	←		←	Call proceeding
	180 Ringing	←		←	ALERTING
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	→		→	DISC
	200 OK BYE	←		←	REL

IS_SP_08C	**ISDN reference to: EN 300 403-1 [i.3], EN 300 899-1 [23]**				**NGN reference to: ES 129163 clause 7.2.3.1.3A.1**
TSS reference:	ISDN-SIP/Basic_call/Successful/Voice				
Selection criteria:	Basic_call; 283 027 overlap receiving supported; Multiple INVITE Method; two dialogs used				
Test purpose:	Ensure that call establishment using **overlap sending** is performed correctly. Ensure that the SIP user receives a 180 Ringing message when the ISDN user answers with an ALERTING message. The sending of the 183 Session Progress is optional. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).				
ISDN Parameter values:	BC=speech, no HLC				
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition				
Comments:					
	INVITE csq 1	→		→	SETUP
				←	SETUP ACK
	INVITE csq2	→			
				→	INFO
	484 csq 1	←			
	183 Session progress csq 2	←		←	Call proceeding
	180 Ringing csq2	←		←	ALERTING
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	→		→	DISC
	200 OK BYE	←		←	REL

6.2.1.2 Codec negotiation

SI_XX_CN_01	**ISDN reference to: EN 300 403-1 [i.3], clause 5.3.3 EN 300 899-1 [23], clause 3.1.1**	**NGN reference to: Q.1912.5 [51], clause 6.11 EN 383 001 [49], clause 6.11 ES 283 027 [48], clause 7.2.3.1**
TSS reference:	SIP-SIP/Basic_call/Codec negotiation	
Selection criteria:		
Test purpose:	During the session, the **calling** user decides to change the characteristics of the media session in the confirmed state. This is accomplished by sending a re-INVITE containing a new media description. This re-INVITE references the existing dialog so that the other party knows that it is to modify an existing session instead of establishing a new session. The other party sends a 200 (OK) to accept the change. The requestor responds to the 200 (OK) with an ACK.	
ISDN Parameter values:	SETUP = 3,1 kHz audio	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SI_XX_CN_02	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.3.3** **EN 300 899-1 [23], clause 3.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 6.11** **EN 383 001 [49], clause 6.11** **ES 283 027 [48], clause 7.2.3.1**
TSS reference:	SIP-ISDN /Basic_call/Codec negotiation	
Selection criteria:		
Test purpose:	During the session, the calling user decides to change the characteristics of the media session. Ensure that the calling user can send UPDATE after completion of the initial INVITE transaction. The other party sends a 200 (OK) to accept the change. The requestor responds to the 200 (OK) with an ACK.	
ISDN Parameter values:	SETUP = 3,1 kHz audio	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

6.2.1.3 Test purposes for SIP-ISDN, Basic call, DTMF

<table>
<tr><th>SI_XX_DT_01</th><th colspan="2">ISDN reference to:
EN 300 403-1 [i.3], clause 5.2.1
EN 300 899-1 [23], clause 3.1.1</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.1.3
EN 383 001 [49], clause 6.1.3
ES 283 027 [48], clause 7.2.3.1</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISDN/Basic_call/Successful/3,1 kHz audio</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">DTMF inband</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that call establishment is performed correctly. Ensure that in the active call state (N10) the DTMF digits (events 0 through 15) can be transmitted inband to the called user in case when the SDP parameter in attached table applies.</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">SETUP = 3,1 kHz audio;</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="23"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td colspan="5">a) without SDP pre-condition</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERTING</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CON</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>Communication</td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>BYE</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>200 OK BYE</td></tr>
<tr><td colspan="5">b) SDP pre-condition met</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK (PRACK)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK (UPDATE)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERTING</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>Communication</td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

Values for test purposes SI_XX_DT_01					
VARIABLE	PT	Encoding	media type	clock rate	channels
VA_01	0	PCMU	A	8,000	1
VA_02	3	GSM	A	8,000	1
VA_03	8	PCMA	A	8,000	1
VA_04	dyn	GSM-EFR	A	8,000	1

SI_XX_DT_02	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.2.1** **EN 300 899-1 [23], clause 3.1.1**			**NGN reference to:** **Q.1912.5 [51], clause 6.1.3** **EN 383 001 [49], clause 6.1.3** **ES 283 027 [48], clause 7.2.3.1**	
TSS reference:	SIP-ISDN/Basic_call/Successful/3,1 kHz audio				
Selection criteria:	DTMF with RFC 2833 [56]				
Test purpose:	Ensure that call establishment is performed correctly. Ensure that in the active call state (N10) the DTMF Digits (events 0 through 15) can be transmitted as payload for DTMF Digits (RFC 2833 [56]) to the called user in case when the SDP parameter in table 6 applies.				
ISDN Parameter values:	SETUP = 3,1 kHz audio;				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	a) Without SDP pre-condition				
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERTING
	200 OK INVITE	←		←	CON
	ACK	→			
			Communication		
	BYE	←		←	DISC
	200 OK BYE	→		→	REL
	b) SDP pre-condition met				
	INVITE	→			
	183 Session Progress	←			
	PRACK	→			
	200 OK (PRACK)	←			
	UPDATE	→			
	200 OK (UPDATE)	←			
				→	SETUP
	180 Ringing	←		←	ALERTING
	PRACK	→			
	200 OK (PRACK)	←			
	200 OK INVITE	←		←	CONN
	ACK	→			
			Communication		
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

6.2.1.4 Test purposes for SIP-ISDN, Basic call, UDI

SI_UD_01	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.2.1** **EN 300 899-1 [23], clause 3.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3** **EN 383 001 [49], clause 6.1.3** **ES 283 027 [48], clause 7.2.3.1**
TSS reference:	SIP-ISDN/Basic_call/Successful/UDI	
Selection criteria:		
Test purpose:	Ensure that call establishment and the mapping of the parameters is performed correctly. Ensure that in the active call state (N10) the data transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	SETUP = UDI;	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line : rtpmap:<dynamic-PT> CLEARMODE/8000 b = line AS: 64 m = RTP/AVP	
Comments:		

SIP		SUT		ISDN
a) Without SDP pre-condition				
INVITE	→		→	SETUP
180 Ringing	←		←	ALERTING
200 OK INVITE	←		←	CON
ACK	→			
		Communication		
BYE	←		←	BYE
200 OK BYE	→		→	200 OK BYE
b) SDP pre-condition met				
INVITE	→			
183 Session Progress	←			
PRACK	→			
200 OK (PRACK)	←			
UPDATE	→			
200 OK (UPDATE)	←			
			→	SETUP
180 Ringing	←		←	ALERTING
PRACK	→			
200 OK (PRACK)	←			
200 OK INVITE	←		←	CONN
ACK	→			
		Communication		
BYE	←		←	DISC
200 OK BYE	→		→	REL

SI_UD_03	**ISDN reference to: EN 300 403-1 [i.3], clause 5.3.3 EN 300 899-1 [23], clause 3.1.1**		**NGN reference to: Q.1912.5 [51], clause 6.11 EN 383 001 [49], clause 6.11 ES 283 027 [48], clause 7.2.3.1**		
TSS reference:	SIP-ISDN/Basic_call/Successful/UDI				
Selection criteria:					
Test purpose:	Ensure that the call establishment and the call clearing procedure are performed correctly when the **calling user** clears after answering with a BYE message. The called user shall receive a DISCONNECT message indicating the Cause value # 16 "normal call clearing". Ensure that in the Call Delivered call state U4 the transfer of tone or announcement on the media channel is performed correctly.				
ISDN Parameter values:	SETUP = UDI;				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line : rtpmap:<dynamic-PT> CLEARMODE/8000 b = line AS: 64 m = RTP/AVP				
Comments:					
	SIP		SUT		ISDN
	a) Without SDP pre-condition				
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERTING
	200 OK INVITE	←		←	CON
	ACK	→			
	RE-INVITE	→			
	200 OK	←			
	ACK	→			
			Communication		
	BYE	←		←	DISC
	200 OK BYE	→		→	REL
	b) SDP pre-condition met				
	INVITE	→			
	183 Session Progress	←			
	PRACK	→			
	200 OK (PRACK)	←			
	UPDATE	→			
	200 OK (UPDATE)	←			
				→	SETUP
	180 Ringing	←		←	ALERTING
	200 OK INVITE	←		←	CONN
	ACK	→			
			Communication		
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_UD_04	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.3.3** **EN 300 899-1 [23], clause 3.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 6.11** **EN 383 001 [49], clause 6.11** **ES 283 027 [48], clause 7.2.3.1**
TSS reference:	SIP-ISDN/Basic_call/Successful/UDI	
Selection criteria:		
Test purpose:	Ensure that the call clearing procedure is performed correctly when the **called user** clears after answering with a DISCONNECT message indicating the Cause value # 16 "normal call clearing". The calling user shall receive a BYE message. A reason header field with value 16 is sent in case of EN 383 001 [49] and ES 283 027 [48], optional in Q.1912.5 [51].	
ISDN Parameter values:	SETUP = UDI;	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line : rtpmap:<dynamic-PT> CLEARMODE/8000 b = line AS: 64 m = RTP/AVP	
Comments:		

SIP		SUT		ISDN
a) Without SDP pre-condition				
INVITE	→		→	SETUP
180 Ringing	←		←	ALERTING
200 OK INVITE	←		←	CON
ACK	→			
		Communication		
BYE	←		←	DISC
200 OK BYE	→		→	REL
b) SDP pre-condition met				
INVITE	→			
183 Session Progress	←			
PRACK	→			
200 OK (PRACK)	←			
UPDATE	→			
200 OK (UPDATE)	←			
			→	SETUP
180 Ringing	←		←	ALERTING
PRACK	→			
200 OK (PRACK)	←			
200 OK INVITE	←		←	CONN
ACK	→			
		Communication		
BYE	←		←	DISC
200 OK BYE	→		→	REL

SI_UD_05	**ISDN reference to:** **EN 300 403-1 [i.3],clause 5.2.1** **EN 300 899-1 [23], clause 3.1.1**		**NGN reference to:** **Q.1912.5 [51], clause 6.1.1, 6.1.3** **EN 383 001 [49], clause 6.11**		
TSS reference:	SIP-ISDN/Basic_call/Successful/UDI				
Selection criteria:	SIP Profile A or Q.1912.5 [51] Profile B with PI or EN 383 001 [49]				
Test purpose:	Ensure that call establishment upon receipt of the first INVITE with sufficient digits, is performed correctly. During call establishment a Progress indicator information element shall be included in the SETUP message sent to the called user with progress description value #1 "call is not end-to-end ISDN"				
ISDN Parameter values:	SETUP = UDI				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line : rtpmap:<dynamic-PT> CLEARMODE/8000 b = line AS: 64 m = RTP/AVP				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP PI #1
	100 Trying	←			
				←	CALL PROC
	180 Ringing	←		←	ALERT
	PRACK	→			
	200 OK (PRACK)	←			
	200 OK INVITE	←		←	CON
	ACK	→			
			Communication		
	BYE	←		←	DISC
	200 OK BYE	→		→	REL
	NOTE: EN 383 001 [49] is not conform to the ETSI ISDN basic standard regarding UDI (PI).				

SI_UD_06	**ISDN reference to:** **EN 300 403-1 [i.3], clause 5.2.1** **EN 300 899-1 [23], clause 3.1.1**	**NGN reference to:** **EN 383 001 [49], clause 6.1.1** **ES 283 027 [48], clause 7.2.3.1**
TSS reference:	SIP-ISDN/Basic_call/Successful/UDI	
Selection criteria:	In case of UDI, the FCI is sent with "No interworking encountered" and "Originating access is ISDN"	
Test purpose:	Ensure that call establishment upon receipt of the first INVITE with sufficient digits, is performed correctly. During call establishment a Progress indicator information element **shall not be included** in the SETUP message.	
ISDN Parameter values:	SETUP = UDI	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line : rtpmap:<dynamic-PT> CLEARMODE/8000 b = line AS: 64 m = RTP/AVP	
Comments:		

SIP		SUT		ISDN
a) Without SDP pre-condition				
INVITE	→		→	SETUP
100 Trying	←			
			←	CALL PROC
180 Ringing	←		←	ALERT
200 OK INVITE	←		←	CON
ACK	→			
		Communication		
BYE	←		←	DISC
200 OK BYE	→		→	REL
b) SDP pre-condition met				
INVITE	→			
183 Session Progress	←			
PRACK	→			
200 OK (PRACK)	←			
UPDATE	→			
200 OK (UPDATE)	←			
			→	SETUP
180 Ringing	←		←	ALERTING
PRACK	→			
200 OK (PRACK)	←			
200 OK INVITE	←		←	CONN
ACK	→			
		Communication		
BYE	←		←	DISC
200 OK BYE	→		→	REL

6.2.1.5 Test purposes for SIP-ISDN, Basic call, Unsuccessful

Unsuccessful

SI_XX_U01	**ISDN reference to:** **EN 300 403-1 [i.3],** **clauses 5.1.4, G.1.1**				**NGN reference to:** **Q.1912.5 [51], clause 6.11** **EN 383 001 [49], clause 6.11** **ES 283 027 [48], clause 7.2.3.1**
TSS reference:	SIP-ISDN/Basic_call/Unsuccessful				
Selection criteria:					
Test purpose:	Ensure that, when calling to **unallocated number**, the network initiate call clearing to the calling user with a Not Found message. If the Reason Header field is implemented the cause value #1 "unassigned number" should be mapped to the Reason Header field. (According ES 283 027 [48] and EN 383 001 [49] the Reason Header field shall be added to the SIP final response)				
ISDN Parameter values:	SETUP = PIXIT				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→			
	100 Trying	←			
	404 Not Found	←			
	ACK	→			

<table>
<tr><td>SI_XX_U02</td><td colspan="3">ISDN reference to:
EN 300 403-1 [i.3],
clauses 5.1.4, G.1.1</td><td colspan="2">NGN reference to:
Q.1912.5 [51], clause 6.11
EN 383 001 [49], clause 6.11
ES 283 027 [48], clause 7.2.3.1</td></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISDN/Basic_call/Unsuccessful</td></tr>
<tr><td>Selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the call will be released when there is no route to destination. The network initiates call clearing to the calling user with a 500 Server Internal Error. If the Reason Header field is implemented the cause value # 3 "no route to destination should be contained in the Reason Header field.
(According ES 283 027 [48] and EN 383 001 [49] the Reason Header field shall be added to the SIP final response)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">SETUP = PIXIT</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td></td><td>INVITE</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td>100 Trying</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td>500 Server Internal Error</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><td>SI_XX_U03</td><td colspan="3">ISDN reference to:
EN 300 403-1 [i.3],
clauses 5.2.5.1, G.1.7</td><td colspan="2">NGN reference to:
Q.1912.5 [51], clause 6.11
EN 383 001 [49], clause 6.11
ES 283 027 [48], clause 7.2.3.1</td></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISDN/Basic_call/Unsuccessful</td></tr>
<tr><td>Selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that, when the called user is user determined user busy the network initiate call clearing to the calling user with a 486 Busy Here message. If the Reason Header field is implemented the cause value # 17 should be contained in the Reason Header field.
(According ES 283 027 [48] and EN 383 001 [49] the Reason Header field shall be added to the SIP final response)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">SETUP = PIXIT</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5">In the Request-URI a sip: URI with the user=phone parameter, and the user info part of that URI is an E.164 number encoded as specified by the telephone-subscriber rule of RFC 3966 [26].</td></tr>
<tr><td></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td></td><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td></td><td>100 Trying</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td>486 Busy Here</td><td>←</td><td></td><td>←</td><td>REL_COMP #17</td></tr>
<tr><td></td><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><th>SI_XX_U04</th><th colspan="3">ISDN reference to:
EN 300 403-1 [i.3],
clauses 5.2.5.4, G.1.9</th><th colspan="2">NGN reference to:
Q.1912.5 [51], clause 6.11
EN 383 001 [49], clause 6.11
ES 283 027 [48], clause 7.2.3.1</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISDN/Basic_call/Unsuccessful</td></tr>
<tr><td>Selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that when there is no answer from the called user ("no user responding"), the network initiate call clearing to the calling user with a 480 Temporarily unavailable message.
If the Reason Header field is implemented the cause value # 18 should be contained in the Reason Header field.
(According ES 283 027 [48] and EN 383 001 [49] the Reason Header field shall be added to the SIP final response)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">SETUP = PIXIT</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5">In the Request-URI a sip: URI with the user=phone parameter, and the user info part of that URI is an E.164 number encoded as specified by the telephone-subscriber rule of RFC 3966 [26].</td></tr>
<tr><td rowspan="5"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>100 Trying</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>480 Temporarily unavailable</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
</table>

SI_XX_U05	**ISDN reference to:** **EN 300 403-1 [i.3],** **clauses 5.2.5.4, G.1.9**				**NGN reference to:** **Q.1912.5 [51], clause 6.11** **EN 383 001 [49], clause 6.11** **ES 283 027 [48], clause 7.2.3.1**
TSS reference:	SIP-ISDN/Basic_call/Unsuccessful				
Selection criteria:					
Test purpose:	Ensure that when there is **no answer from the called user** ("no answer from the user"), the ISDN network initiate call clearing to the calling user with a 480 Temporarily unavailable message. If the Reason Header field is implemented the cause **value # 102** should be contained in the Reason Header field. (According ES 283 027 [48] and EN 383 001 [49] the Reason Header field shall be added to the SIP final response)				
ISDN Parameter values:	SETUP = PIXIT				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	100 Trying	←			
	180 Ringing	←		←	ALERTING
			ISUP T9 expired		
	480 Temporarily unavailable	←		→	DISC #102
				←	RELEASE
	ACK	→		→	REL COMP

SI_XX_U06	**ISDN reference to: EN 300 403-1 [i.3], clauses 5.2.5.4, G.1.9**				**NGN reference to: Q.1912.5 [51], clause 6.11 EN 383 001 [49], clause 6.11 ES 283 027 [48], clause 7.2.3.1**
TSS reference:	SIP-ISDN/Basic_call/Unsuccessful/				
Selection criteria:	Basic call; Reason Header field is supported				
Test purpose:	Ensure that when there is no answer from the called user (but user alerted) and if the SIP network initiate call clearing before the SCN release the call, the SIP network shall send to the calling user a 480 Temporarily unavailable message and the SCN network initiate call clearing to the calling user with a DISCONNECT message indicating cause value # 20 Subscriber absent.				
ISDN Parameter values:	BC = PIXIT				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	ISDN		SUT		SIP
	SETUP	←		←	INVITE
	CALL PROCEEDING	→			.
					.
					.
	ALERTING	→		→	180 Ringing
	DISC#20	←		←	480 temp. Unavailable
	REL	←		←	ACK

<table>
<tr><th>SI_XX_U07</th><th colspan="2">ISDN reference to:
EN 300 403-1 [i.3],
clauses 5.1.9, 5.3.2, G.1.10</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.11
EN 383 001 [49], clause 6.11
ES 283 027 [48], clause 7.2.3.1</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISDN/Basic_call/Unsuccessful</td></tr>
<tr><td>Selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that when the called user rejects the call and responds with a RELEASE COMPLETE message indicating cause value # 21 "call rejected", the call will be released. The network initiates call clearing to the calling user with a 480 Temporarily unavailable message.
If the Reason Header field is implemented the cause value # 21 should be contained in the Reason Header field.
(According ES 283 027 [48] and EN 383 001 [49] the Reason Header field shall be added to the SIP final response)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">SETUP = PIXIT</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="5"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>100 Trying</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>480 Temporarily unavailable</td><td>←</td><td></td><td>←</td><td>REL COMP # 21</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><th>SI_XX_U08</th><th colspan="2">ISDN reference to:
EN 300 403-1 [i.3],
clauses 5.2.5.4, G.1.9</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.11
EN 383 001 [49], clause 6.11
ES 283 027 [48], clause 7.2.3.1</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISDN/Basic_call/Unsuccessful</td></tr>
<tr><td>Selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that when the number is changed, the network initiate call clearing to the calling user with a 410 Gone message.
If the Reason Header field is implemented the cause value # 22 ("number changed") should be contained in the Reason Header field.
(According ES 283 027 [48] and EN 383 001 [49] the Reason Header field shall be added to the SIP final response)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">SETUP = PIXIT</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="5"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>100 Trying</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>410 Gone</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><th>SI_XX_U09</th><th colspan="2">ISDN reference to:
EN 300 403-1 [i.3],
clauses 5.2.5.4, G.1.9</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.11
EN 383 001 [49], clause 6.11
ES 283 027 [48], clause 7.2.3.1</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISDN/Basic_call/Unsuccessful</td></tr>
<tr><td>Selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that when the destination is out of order, the network initiate call clearing to the calling user with a 502 Bad Gateway message.
If the Reason Header field is implemented the cause value Cause Value No. 27 ("destination out of order") should be contained in the Reason Header field.
(According ES 283 027 [48] and EN 383 001 [49] the Reason Header field shall be added to the SIP final response)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">SETUP = PIXIT</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5">In the Request-URI a sip: URI with the user=phone parameter, and the user info part of that URI is an E.164 number encoded as specified by the telephone-subscriber rule of RFC 3966 [26].</td></tr>
<tr><td rowspan="5"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>100 Trying</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>502 Bad Gateway</td><td>←</td><td></td><td>←</td><td>REL # 27</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>RLC</td></tr>
</table>

<table>
<tr><th>SI_XX_U10</th><th colspan="2">ISDN reference to:
EN 300 403-1 [i.3],
clauses 5.1.9, 5.3.2, G.1.10</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.11
EN 383 001 [49], clause 6.11
ES 283 027 [48], clause 7.2.3.1</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISDN/Basic_call/Unsuccessful</td></tr>
<tr><td>Selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the call will be released when the called number is incomplete. The network initiates call clearing to the calling user according the CAUSE_VA interworking to SIP_MESSAGE_VA.
If the Reason Header field is implemented the cause value should be contained in Reason Header field.
(According ES 283 027 [48] and EN 383 001 [49] the Reason Header field shall be added to the SIP final response)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">SETUP = PIXIT</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="5"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>100 Trying</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>SIP_MESSAGE_VA</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
</table>

SI_XX_U10		
	SIP_MESSAGE_VA	**CAUSE_VA** PSTN cause (Destination number PIXIT)
VA_1	404 Not Found	value # 1 "Unassigned (unallocated) number"
VA_2	500 server internal error	value # 3 "No route to destination"
VA_3	410 Gone	value # 22 "Number changed"
VA_4	484 Address Incomplete	value# 28 "Invalid number format (incomplete number")

<table>
<tr><th>SI_XX_U11</th><th colspan="2">ISDN reference to:
EN 300 403-1 [i.3],
clauses 5.2.5.4, G.1.9</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.11
EN 383 001 [49], clause 6.11
ES 283 027 [48], clause 7.2.3.1</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISDN/Basic_call/Unsuccessful</td></tr>
<tr><td>Selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that when the call is released with Cause Value No. 31 ("normal unspecified"), the network initiate call clearing to the calling user with a 480 Temporarily unavailable message.
If the Reason Header field is implemented the cause value # 31 should be contained in the Reason Header field.
(According ES 283 027 [48] and EN 383 001 [49] the Reason Header field shall be added to the SIP final response)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">SETUP = PIXIT</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="5"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>100 Trying</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>480 Temporarily unavailable</td><td>←</td><td></td><td>←</td><td>RELEASE #31</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>REL_COMP</td></tr>
</table>

SI_XX_U12	**ISDN reference to: EN 300 403-1 [i.3], clause G.1.6**			**NGN reference to: Q.1912.5 [51], clause 6.11 EN 383 001 [49], clause 6.11 ES 283 027 [48], clause 7.2.3.1**		
TSS reference:	SIP-ISDN/Basic_call/Unsuccessful					
Selection criteria:	Multipoint Configuration: for the called side					
Test purpose:	Ensure that when the calling user clears the call with a **SIP_MESSAGE_VA** before answer from called user.					
ISDN Parameter values:	SETUP = PIXIT					
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)					
Comments:						
	SIP		SUT		ISDN	
	INVITE	→		→	SETUP	
	100 Trying	←				
				←	CALL PROC	
	180 Ringing	←		←	ALERT	
	SIP_MESSAGE_VA	→		→	DISC	
	200 OK	←		←	REL	
	487 Request terminated	←				
	ACK	→				

SIP_MESSAGE_VA: SI_XX_U12	
VA_1	CANCEL
VA_2	BYE

SI_XX_U13	**ISDN reference to: EN 300 403-1 [i.3], clause G.1.6**			**NGN reference to: Q.1912.5 [51], clause 6.11 EN 383 001 [49], clause 6.11 ES 283 027 [48], clause 7.2.3.1**
TSS reference:	SIP-ISDN/Basic_call/Unsuccessful			
Selection criteria:	Point-to-point Configuration: for the called side			
Test purpose:	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out a SETUP message, having received a CALL PROCEEDING, on receipt of an RELEASE, where the cause value defined as CV_ISDN, the SUT immediately requests the disconnection of the internal bearer path. the SUT shall send the appropriate SIP status defined as **SIP_FAILURE_VA** The ISDN Cause Value field in the ISDN RELEASE message is mapped to the Reason header field if implemented According ES 283 027 [48] and EN 383 001 [49] the Reason Header field shall be added to the SIP final response.			
ISDN Parameter values:	SETUP = PIXIT			
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)			
Comments:				
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
				←	CALL PROC
	SIP_MESSAGE_VA	←		←	RELEASE
	ACK	→		←	REL COMP

Values for test purposes SI_XX_U13		
←SIP Message SIP_FAILURE_VA CV_SIP		←DISC Cause Indicators parameter CV_ISDN
VA_1	486 Busy Here Cause Value No. 17	Cause Value No. 17 ("user busy")
VA_2	480 Temporarily unavailable Cause Value No. 18	Cause Value No. 18 ("No user responding")
VA_3	480 Temporarily unavailable Cause Value No. 21	Cause Value No. 21 ("all rejected")
VA_4	410 Gone Cause Value No. 23	Cause Value No. 22 ("number changed")
VA_5	502 Bad Gateway Cause Value No. 27	Cause Value No. 27 ("destination out of order")
VA_6	484 Address Incomplete Cause Value No. 28	Cause Value No. 28 ("invalid number format (address incomplete")
VA_7	480 Temporarily unavailable Cause Value No. 31	Cause Value No. 31 ("normal unspecified") (Class default)
VA_8	500 Server Internal Error Cause Value No. 47	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38-47) (47 is class default)
VA_9	500 Server Internal Error Cause Value No. 63	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_10	500 Server Internal Error Cause Value No. 88	Cause Value No. 88 ("incompatible destination")
VA_11	500 Server Internal Error Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)

<table>
<tr><th>SI_XX_U14</th><th colspan="3">ISDN reference to:
EN 300 403-1 [i.3], clause G.1.6</th><th colspan="2">NGN reference to:
Q.1912.5 [51], clause 6.11
EN 383 001 [49], clause 6.11
ES 283 027 [48], clause 7.2.3.1</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISDN/Basic_call/Unsuccessful</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">Point-to-point Configuration: for the called side</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out a SETUP message, having received a ALERTING message, on receipt of an DISCONNECT where the cause value defined as CV_ISDN,
the SUT immediately requests the disconnection of the internal bearer path. the SUT shall send SIP_FAILURE_VA message
The Cause Value field in the DISCONNECT message is mapped to the Reason header field if implemented
(According ES 283 027 [48] and EN 383 001 [49] the Reason Header field shall be added to the SIP final response)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">SETUP = PIXIT</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="5"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td></td><td>ALERTING</td></tr>
<tr><td>SIP_MESSAGE_VA</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

<table>
<tr><th colspan="3">Values for test purposes SI_XX_U14</th></tr>
<tr><td colspan="2">←SIP Message
SIP_FAILURE_VA
CV_SIP</td><td>←DISC
Cause Indicators parameter
CV_ISDN</td></tr>
<tr><td>VA_1</td><td>480 Temporarily unavailable
Cause Value No. 21</td><td>Cause Value No. 21 ("all rejected")</td></tr>
<tr><td>VA_2</td><td>502 Bad Gateway
Cause Value No. 27</td><td>Cause Value No. 27 ("destination out of order")</td></tr>
<tr><td>VA_3</td><td>480 Temporarily unavailable
Cause Value No. 31</td><td>Cause Value No. 31 ("normal unspecified")
(Class default)</td></tr>
<tr><td>VA_4</td><td>500 Server Internal Error
Cause Value No. 38</td><td>Cause Value No. 38 ("Network out of order")</td></tr>
<tr><td>VA_5</td><td>500 Server Internal Error
Cause Value No. 41</td><td>Cause Value No. 41 ("Temporary failure ")</td></tr>
<tr><td>VA_6</td><td>500 Server Internal Error
Cause Value No. 111</td><td>Cause Value No. 111 ("protocol error, unspecified")
(Class default)</td></tr>
</table>

SI_XX_U15	**ISDN reference to: EN 300 403-1 [i.3], clause G.1.6**			**NGN reference to: Q.1912.5 [51], clause 6.11 EN 383 001 [49], clause 6.11 ES 283 027 [48], clause 7.2.3.1**
TSS reference:	SIP-ISDN/Basic_call/Unsuccessful			
Selection criteria:	Point-to-point Configuration: for the called side Cause Value is mapped to the Reason header field			
Test purpose:	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out a SETUP message, having received a ALERTING message, having received a CONNECT", a 200 OK message is sent, on receipt of an DISCONNECT where the cause value defined as CV_ISDN, the SUT immediately requests the disconnection of the internal bearer path. the SUT shall send BYE message The Cause Value field in the DISCONNECT message is mapped to the Reason header field if implemented (According ES 283 027 [48] and EN 383 001 [49] the Reason Header field shall be added to the SIP final response)			
ISDN Parameter values:	SETUP = PIXIT			
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)			
Comments:				
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERTING
	200 OK INVITE	←		←	CONNECT
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

Values for test purposes SI_XX_U15		
←SIP Message SIP_FAILURE_VA CV_SIP		←DISC Cause Indicators parameter CV_ISDN
VA_1	BYE Cause Value No. 16	Cause Value No. 16
VA_2	BYE Cause Value No. 27	Cause Value No. 27 ("destination out of order")
VA_3	BYE Cause Value No. 31	Cause Value No. 31 ("normal unspecified") (Class default)
VA_4	BYE Cause Value No. 38	Cause Value No. 38 ("Network out of order")
VA_5	BYE Cause Value No. 41	Cause Value No. 41 ("Temporary failure ")

<table>
<tr><th>SI_XX_U16</th><th colspan="3">ISDN reference to:
EN 300 403-1 [i.3], clause G.1.6</th><th colspan="2">NGN reference to:
Q.1912.5 [51], clause 6.11
EN 383 001 [49], clause 6.11
ES 283 027 [48], clause 7.2.3.1</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISDN/Basic_call/Unsuccessful</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">ISDN = point to point Configuration: with DDI</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the call will be released with cause 102 (Recovery on timer expiry) 484 Address Incomplete on the SIP side after the expire of timer T 304 when called user is in call state U02 when the called number is incomplete.</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">SETUP = PIXIT</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5">In the Request-URI a sip: URI with the user=phone parameter, and the user info part of that URI is an E.164 number encoded as specified by the telephone-subscriber rule of RFC 3966 [26].</td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>100 Trying</td><td>←</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>SETUP ACK</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>INFO SC (optional)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>RELEASE # 102</td></tr>
<tr><td>484 Address Incomplete</td><td>←</td><td></td><td>→</td><td>REL_COMP</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
</table>

SI_XX_U17	**ISDN reference to: EN 300 403-1 [i.3], clause G.1.6**	**NGN reference to: Q.1912.5 [51], clause 6.11 EN 383 001 [49], clause 6.11 ES 283 027 [48], clause 7.2.3.1**
TSS reference:	SIP-ISDN/Basic_call/Unsuccessful	
Selection criteria:		
Test purpose:	During the session, the calling user decides to change the characteristics of the media session. This is accomplished by sending a re-INVITE containing a new media description. This re-INVITE references the existing dialog so that the other party knows that it is to modify an existing session instead of establishing a new session. Ensure that if the other party (the GW) does not accept the change, he sends an error response such as 488 (Not Acceptable Here), which also receives an ACK. The session remains in the active state.	
ISDN Parameter values:	SETUP = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERTING
	200 OK INVITE	←		←	CONN
	ACK	→			
			Communication		
	RE-INVITE	→			
	484 Not Acceptable Here	←			
	ACK	→			
			Communication		
	BYE	←		←	Disconnect
	200 OK BYE	→		→	Release

SI_XX_U18	**ISDN reference to: EN 300 403-1 [i.3], clause G.1.6**	**NGN reference to: Q.1912.5 [51], clause6.11 RFC 3261 [28], clause 4**
TSS reference:	SIP-ISDN/Basic_call/Unsuccessful	
Selection criteria:		
Test purpose:	Ensure that 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, the originating UE shall send a new INVITE request using the precondition mechanism, if the originating UE supports the precondition mechanism.	
ISDN Parameter values:	SETUP = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

<table>
<tr><th>SI_XX_U19</th><th>ISDN reference to:
EN 300 403-1 [i.3], clause G.1.6</th><th>NGN reference to:
Q.1912.5 [51], clause6.11
RFC 3261 [28], clause 4</th></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-ISDN/Basic_call/Unsuccessful</td></tr>
<tr><td>Selection criteria:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2">Ensure that if a codec is required and the MGCF does not find an available matching codec at the MGW for the received initial INVITE request, the MGCF shall:
• send 503 (Service Unavailable) response if the type of codec was acceptable but none were available; or
• send 488 (Not Acceptable Here) response if the type of codec was not supported, and may include SDP in the message body to indicate the codecs supported by the MGCF/MGW.</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="2">SETUP = PIXIT</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="2"></td></tr>
</table>

6.2.2 Test purposes for SIP - ISDN Supplementary services

6.2.2.1 OIP/CLIP

SI_XXSSOIP01	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3],** **clauses 4.5.10, 4.5.11**				**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6** **TS 183 007 [43]**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks				
Selection criteria:					
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has not** been received • the priv-value component is set to "none" • the SIP From header field containing a URI with an identity in the format of a tel URI **has not been** received and • the Calling Party Number is correctly delivered to the called (served) user with the **Calling party information element** coded: - Address signals = default number derived from the P-Asserted-Identity - Screening indicator = network provided - Numbering plan indicator = ISDN numbering plan - Address Presentation Restricted Indicator = **Presentation allowed**				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

<table>
<tr><td>SI_XXSSOIP02</td><td colspan="3">ISDN reference to:
EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</td><td colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</td></tr>
<tr><td>TSS reference:</td><td colspan="6">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="6">The user subscribes OIR "temporary mode" default "not restricted"
No priv value is sent
Special arrangement applies</td></tr>
<tr><td>Test purpose:</td><td colspan="6">Ensure that the SUT in the Idle state, on receipt of a INVITE message where
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) with an identity in the format of a tel URI has not been received
• No priv value is received
• the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has been received
• the Calling Party Number is correctly delivered to the called (served) user with the 1stCalling party information element coded:
- Address signals = default number derived from the From header
- Screening indicator = user provided not verified
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed
2ndCalling party information element coded:
- Address signals = default number derived from the P-Asserted-Identity
- Screening indicator = network provided
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="6">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="6"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td colspan="2">SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td colspan="2"></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td colspan="2"></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td colspan="2"></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td colspan="2"></td><td></td><td></td></tr>
<tr><td></td><td></td><td colspan="2"></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td colspan="2"></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td colspan="2"></td><td>→</td><td>REL</td></tr>
</table>

SI_XXSSOIP03	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3],** **clauses 4.5.10, 4.5.11**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6** **TS 183 007 [43]**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" No priv value is sent No Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has not** been received • No priv value is received • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • the Calling Party Number is correctly delivered to the called (served) user with the **Calling party information element** coded: - Address signals = default number derived from the P-Asserted-Identity - Screening indicator = network provided - Numbering plan indicator = ISDN numbering plan - Address Presentation Restricted Indicator = **Presentation allowed**	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SIP		SUT		ISDN
INVITE	→		→	SETUP
180 Ringing	←		←	ALERT
200 OK INVITE	←		←	CONN
ACK	→			
BYE	←		←	DISC
200 OK BYE	→		→	REL

<table>
<tr><td>SI_XXSSOIP04</td><td>ISDN reference to:
EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</td><td colspan="4">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</td></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">SIP URI or SIPS URI are used in the P-Preferred-Identity
The user subscribes OIR "temporary mode" default "not restricted"
No priv value is sent</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) with an identity in the format of a tel URI has not been received
• no priv value is received
• the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has not been received
• the Calling Party Number is correctly delivered to the called (served) user with the Calling party information element coded:
- Address signals = default number derived from the P-Asserted-Identity
- Screening indicator = network provided
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

SI_XXSSOIP05	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3],** **clauses 4.5.10, 4.5.11**			**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6** **TS 183 007 [43]**	
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks				
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" Special arrangement applies				
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • the priv-value component is set to "none" • the Calling Party Number is correctly delivered to the called (served) user with the **1stCalling party information element** coded: - Address signals = default number derived from the From header - Screening indicator = *user provided not verified* - Numbering plan indicator = ISDN numbering plan - Address Presentation Restricted Indicator = **Presentation allowed** **2ndCalling party information element** coded: - Address signals = default number derived from the P-Asserted-Identity - Screening indicator = network provided - Numbering plan indicator = ISDN numbering plan - Address Presentation Restricted Indicator = **Presentation allowed**				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

<table>
<tr><th>SI_XXSSOIP06</th><th colspan="3">ISDN reference to:
EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</th></tr>
<tr><td>TSS reference:</td><td colspan="6">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="6">The user subscribes OIR "temporary mode" default "not restricted"
No Special arrangement applies</td></tr>
<tr><td>Test purpose:</td><td colspan="6">Ensure that the SUT in the Idle state, on receipt of a INVITE message where
• the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has been received
• the priv-value component is set to "none"
• the Calling Party Number is correctly delivered to the called (served) user with the Calling party information element coded:
- Address signals = default number derived from the P-Asserted-Identity
- Screening indicator = network provided
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="6">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="6"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td colspan="2">SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td colspan="2"></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td colspan="2"></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td colspan="2"></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td colspan="2"></td><td></td><td></td></tr>
<tr><td></td><td></td><td colspan="2"></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td colspan="2"></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td colspan="2"></td><td>→</td><td>REL</td></tr>
</table>

<table>
<tr><td>SI_XXSSOIP07</td><td>ISDN reference to:
EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</td><td colspan="4">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</td></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "not restricted"
Special arrangement applies</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where
• the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has been received
• No priv value is received
• the Calling Party Number is correctly delivered to the called (served) user with the 1stCalling party information element coded:
- Address signals = default number derived from the From header
- Screening indicator = user provided not verified
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed
2ndCalling party information element coded:
- Address signals = default number derived from the P-Asserted-Identity
- Screening indicator = network provided
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

<table>
<tr><td>SI_XXSSOIP08</td><td colspan="3">ISDN reference to:
EN 300 092-1 [i.14], clause 9.3;
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</td><td colspan="2">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</td></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "not restricted"
No Special arrangement applies</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where
• the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has been received
• no priv value is received
• the Calling Party Number is correctly delivered to the called (served) user with the Calling party information element coded:
- Address signals = default number derived from the P-Asserted-Identity
- Screening indicator = network provided
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

SI_XXSSOIP09	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3],** **clauses 4.5.10, 4.5.11**				**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6** **TS 183 007 [43]**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks				
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" Special arrangement applies				
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI **has not** been received • the priv-value component is set to "none" • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • the Calling Party Number is correctly delivered to the called (served) user with the **1stCalling party information element** coded: - Address signals = default number derived from the From header - Screening indicator = *user provided not verified* - Numbering plan indicator = ISDN numbering plan - Address Presentation Restricted Indicator = **Presentation allowed** **2ndCalling party information element** coded: - Address signals = default number derived from the P-Asserted-Identity - Screening indicator = network provided - Numbering plan indicator = ISDN numbering plan - Address Presentation Restricted Indicator = **Presentation allowed**				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_XXSSOIP10	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3],** **clauses 4.5.10, 4.5.11**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6** **TS 183 007 [43]**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI **has not** been received • the priv-value component is set to "none" • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • the Calling Party Number is correctly delivered to the called (served) user with the **Calling party information element** coded: - Address signals = default number derived from the P-Asserted-Identity - Screening indicator = network provided - Numbering plan indicator = ISDN numbering plan - Address Presentation Restricted Indicator = **Presentation allowed**	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_XXSSOIP11	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3],** **clauses 4.5.10, 4.5.11**				**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6** **TS 183 007 [43]**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks				
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" Special arrangement applies				
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI **has not** been received • no priv value is received • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • the Calling Party Number is correctly delivered to the called (served) user with the **1stCalling party information element** coded: - Address signals = default number derived from the From header - Screening indicator = *user provided not verified* - Numbering plan indicator = ISDN numbering plan - Address Presentation Restricted Indicator = **Presentation allowed** **2ndCalling party information element** coded: - Address signals = default number derived from the P-Asserted-Identity - Screening indicator = network provided - Numbering plan indicator = ISDN numbering plan - Address Presentation Restricted Indicator = **Presentation allowed**				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

<table>
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EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</th></tr>
<tr><td>TSS reference:</td><td colspan="6">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="6">The user subscribes OIR "temporary mode" default "not restricted"
No Special arrangement applies</td></tr>
<tr><td>Test purpose:</td><td colspan="6">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has not been received
• no priv value is received
• the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has been received
• the Calling Party Number is correctly delivered to the called (served) user with the Calling party information element coded:
- Address signals = default number derived from the P-Asserted-Identity
- Screening indicator = network provided
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="6">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="6"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td colspan="2">SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td colspan="2"></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td colspan="2"></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td colspan="2"></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td colspan="2"></td><td></td><td></td></tr>
<tr><td></td><td></td><td colspan="2"></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td colspan="2"></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td colspan="2"></td><td>→</td><td>REL</td></tr>
</table>

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EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</th><th colspan="2">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "not restricted"</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received
• the priv-value component is set to "none"
• the SIP From header field containing a SIP URI (PIXIT) in the format of a tel URI has not been received
• the Calling Party Number is correctly delivered to the called (served) user with the Calling party information element coded:
- Address signals = default number derived from the P-Asserted-Identity
- Screening indicator = network provided
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

SI_XXSSOIP14	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3],** **clauses 4.5.10, 4.5.11**		**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6** **TS 183 007 [43]**	
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks			
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted"			
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received • no priv value is received • the SIP From header field containing a SIP URI (PIXIT) in the format of a tel URI **has not been** received • the Calling Party Number is correctly delivered to the called (served) user with the **Calling party information element** coded: - Address signals = default number derived from the P-Asserted-Identity - Screening indicator = network provided - Numbering plan indicator = ISDN numbering plan - Address Presentation Restricted Indicator = **Presentation allowed**			
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)			
Comments:				
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

<table>
<tr><td>SI_XXSSOIP15</td><td colspan="3">ISDN reference to:
EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</td><td colspan="2">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</td></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "not restricted"
Special arrangement applies</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received
• the priv-value component is set to "none"
• the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has been received
• the Calling Party Number is correctly delivered to the called (served) user with the 1stCalling party information element coded:
- Address signals = default number derived from the From header
- Screening indicator = user provided not verified
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed
2ndCalling party information element coded:
- Address signals = default number derived from the P-Asserted-Identity
- Screening indicator = network provided
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

SI_XXSSOIP16	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3],** **clauses 4.5.10, 4.5.11**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6** **TS 183 007 [43]**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a URI (PIXIT) in the format of a tel URI has been received • no priv value is received • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has been received • the Calling Party Number is correctly delivered to the called (served) user with the **1stCalling party information element** coded: - Address signals = default number derived from the From header - Screening indicator = *user provided not verified* - Numbering plan indicator = ISDN numbering plan - Address Presentation Restricted Indicator = **Presentation allowed** **2ndCalling party information element** coded: - Address signals = default number derived from the P-Asserted-Identity - Screening indicator = network provided - Numbering plan indicator = ISDN numbering plan - Address Presentation Restricted Indicator = **Presentation allowed**	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SIP		SUT		ISDN
INVITE	→		→	SETUP
180 Ringing	←		←	ALERT
200 OK INVITE	←		←	CONN
ACK	→			
BYE	←		←	DISC
200 OK BYE	→		→	REL

<table>
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EN 300 092-1 [i.14], clause 9.3;
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</td><td colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</td></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "not restricted"
Special arrangement applies</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has been received
• the priv-value component is set to "none"
• the Calling Party Number is correctly delivered to the called (served) user with the 1stCalling party information element coded:
- Address signals = default number derived from the From header
- Screening indicator = user provided not verified
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed
2ndCalling party information element coded:
- Address signals = default number derived from the P-Asserted-Identity
- Screening indicator = network provided
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
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SI_XXSSOIP18	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3],** **clauses 4.5.10, 4.5.11**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6** **TS 183 007 [43]**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • no priv value is received • the Calling Party Number is correctly delivered to the called (served) user with the **1stCalling party information element** coded: - Address signals = default number derived from the From header - Screening indicator = *user provided not verified* - Numbering plan indicator = ISDN numbering plan - Address Presentation Restricted Indicator = **Presentation allowed** **2ndCalling party information element** coded: - Address signals = default number derived from the P-Asserted-Identity - Screening indicator = network provided - Numbering plan indicator = ISDN numbering plan - Address Presentation Restricted Indicator = **Presentation allowed**	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

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EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "not restricted"</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has not been received
• the priv-value component is set to "none"
• the SIP From header is set to anonymous
• the Calling Party Number is correctly delivered to the called (served) user with the Calling party information element coded:
- Address signals = default number derived from the P-Asserted-Identity
- Screening indicator = network provided
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

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EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</th></tr>
<tr><td>TSS reference:</td><td colspan="6">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="6">The user subscribes OIR "temporary mode" default "not restricted"</td></tr>
<tr><td>Test purpose:</td><td colspan="6">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has not been received
• no priv value is received
• the SIP From header is set to anonymous
• the Calling Party Number is correctly delivered to the called (served) user with the Calling party information element coded:
- Address signals = default number derived from the P-Asserted-Identity
- Screening indicator = network provided
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="6">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="6"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td colspan="2">SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td colspan="2"></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td colspan="2"></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td colspan="2"></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td colspan="2"></td><td></td><td></td></tr>
<tr><td></td><td></td><td colspan="2"></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td colspan="2"></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td colspan="2"></td><td>→</td><td>REL</td></tr>
</table>

<table>
<tr><th>SI_XXSSOIP21</th><th colspan="3">ISDN reference to:
EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</th><th colspan="2">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "restricted"
Special arrangement applies</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) with an identity in the format of a tel URI has not been received
• the priv-value component is set to "none"
• the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has been received
• the Calling Party Number is correctly delivered to the called (served) user with the 1stCalling party information element coded:
- Address signals = default number derived from the From header
- Screening indicator = user provided not verified
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed
2ndCalling party information element coded:
- Address signals = default number derived from the P-Asserted-Identity
- Screening indicator = network provided
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

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<tr><th>SI_XXSSOIP22</th><th colspan="3">ISDN reference to:
EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</th><th colspan="2">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "restricted"</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) with an identity in the format of a tel URI has not been received
• the priv-value component is set to "none"
• the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has not been received
• the Calling Party Number is correctly delivered to the called (served) user with the Calling party information element coded:
- Address signals = default number derived from the P-Asserted-Identity
- Screening indicator = network provided
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

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<tr><th>SI_XXSSOIP23</th><th colspan="2">ISDN reference to:
EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "restricted"
Special arrangement applies</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has been received
• the priv-value component is set to "none"
• the Calling Party Number is correctly delivered to the called (served) user with the 1stCalling party information element coded:
- Address signals = default number derived from the From header
- Screening indicator = user provided not verified
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed
2ndCalling party information element coded:
- Address signals = default number derived from the P-Asserted-Identity
- Screening indicator = network provided
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

SI_XXSSOIP24	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3],** **clauses 4.5.10, 4.5.11**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6** **TS 183 007 [43]**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "restricted" Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI **has not been** received • the priv-value component is set to "none" • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • the Calling Party Number is correctly delivered to the called (served) user with the **1stCalling party information element** coded: - Address signals = default number derived from the From header - Screening indicator = *user provided not verified* - Numbering plan indicator = ISDN numbering plan - Address Presentation Restricted Indicator = **Presentation allowed** **2ndCalling party information element** coded: - Address signals = default number derived from the P-Asserted-Identity - Screening indicator = network provided - Numbering plan indicator = ISDN numbering plan - Address Presentation Restricted Indicator = **Presentation allowed**	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_XXSSOIP25	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3],** **clauses 4.5.10, 4.5.11**				**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6** **TS 183 007 [43]**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks				
Selection criteria:	The user subscribes OIR "temporary mode" default "restricted"				
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received: • the priv-value component is set to "none" • the SIP From header field containing a SIP URI (PIXIT) not in the format of a tel URI **has not been** received • the Calling Party Number is correctly delivered to the called (served) user with the **Calling party information element** coded: - Address signals = default number derived from the P-Asserted-Identity - Screening indicator = network provided - Numbering plan indicator = ISDN numbering plan - Address Presentation Restricted Indicator = **Presentation allowed**				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

<table>
<tr><td>SI_XXSSOIP26</td><td colspan="2">ISDN reference to:
EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</td><td colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</td></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "restricted"
Special arrangement applies</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received
• the priv-value component is set to "none"
• the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has been received
• the Calling Party Number is correctly delivered to the called (served) user with the 1stCalling party information element coded:
- Address signals = default number derived from the From header
- Screening indicator = user provided not verified
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed
2ndCalling party information element coded:
- Address signals = default number derived from the P-Asserted-Identity
- Screening indicator = network provided
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

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<tr><th>SI_XXSSOIP27</th><th colspan="3">ISDN reference to:
EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</th><th colspan="2">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "restricted"</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has not been received
• the priv-value component is set to "none"
• the SIP From header is set to anonymous
• the Calling Party Number is correctly delivered to the called (served) user with the Calling party information element coded:
- Address signals = default number derived from the P-Asserted-Identity
- Screening indicator = network provided
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

SI_XXSSOIP28	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3],** **clauses 4.5.10, 4.5.11**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted"	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received • the priv-value component is set to "none" • the SIP From header is set to **anonymous** • the Calling Party Number is correctly delivered to the called (served) user with the **Calling party information element** coded: - Address signals = default number derived from the P-Asserted-Identity - Screening indicator = network provided - Numbering plan indicator = ISDN numbering plan - Address Presentation Restricted Indicator = **Presentation allowed**	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

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EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</th><th colspan="2">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "not restricted"</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received:
• no priv value is received
• the SIP From header is set to anonymous
• the Calling Party Number is correctly delivered to the called (served) user with the Calling party information element coded:
- Address signals = default number derived from the P-Asserted-Identity
- Screening indicator = network provided
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

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EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</td><td>NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</td></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "restricted"</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received
• the priv-value component is set to "none"
• the SIP From header is set to anonymous
• the Calling Party Number is correctly delivered to the called (served) user with the Calling party information element coded:
- Address signals = default number derived from the P-Asserted-Identity
- Screening indicator = network provided
- Numbering plan indicator = ISDN numbering plan
- Address Presentation Restricted Indicator = Presentation allowed</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

6.2.2.2 OIR/CLIR

<table>
<tr><td colspan="3">SI_XXSSOIR01</td><td colspan="3">ISDN reference to:
EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</td><td colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</td></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "restricted"</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
the SIP P-Preferred-Identity containing a SIP URI (PIXIT) has been received
the priv-value component is set to "id"
the SIP From header field containing a SIP URI (PIXIT) has been received
the Calling party number information element is delivered to the called user without any digit information</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

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EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "restricted"</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) has been received
• the priv-value not present
• the SIP From header field containing a SIP URI (PIXIT) has been received
• the Calling party number information element is delivered to the called user without any digit information</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
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EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</td><td colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</td></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "restricted"</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) has been received
• priv-value component is set to "id"
• the SIP From header is set to anonymous
• the Calling party number information element is delivered to the called user without any digit information</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

SI_XXSSOIR04	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3],** **clauses 4.5.10, 4.5.11**				**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6** **TS 183 007 [43]**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks				
Selection criteria:	The user subscribes OIR "temporary mode" default "restricted"				
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) has been received • priv-value component is not present • the SIP From header is set to **anonymous** • the Calling party number information element is delivered to the called user without any digit information				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_XXSSOIR05	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3],** **clauses 4.5.10, 4.5.11**				**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks				
Selection criteria:	The user subscribes OIR temporary mode" default "not restricted				
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) **has been** received • the priv-value component is set to "id" • the SIP From header field containing a SIP URI (PIXIT) **has been** received • the Calling party number information element is delivered to the called user without any digit information				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_XXSSOIR06	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3],** **clauses 4.5.10, 4.5.11**		**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**		
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks				
Selection criteria:	The user subscribes OIR temporary mode" default "not restricted				
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) has been received • priv-value component is set to "id" • the SIP From header is set to **anonymous** • the Calling party number information element is delivered to the called user without any digit information				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_XXSSOIR07	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3],** **clauses 4.5.10, 4.5.11**		**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**		
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks				
Selection criteria:	The user subscribes OIR permanent mode				
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) **has been** received • the priv-value component is set to "id" • the SIP From header field containing a SIP URI (PIXIT) **has been** received • the Calling party number information element is delivered to the called user without any digit information				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_XXSSOIR08	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3],** **clauses 4.5.10, 4.5.11**		**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**		
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks				
Selection criteria:	The user subscribes OIR permanent mode				
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) **has been** received • the priv-value component is not present • the SIP From header field containing a SIP URI (PIXIT) **has been** received • the Calling party number information element is delivered to the called user without any digit information				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_XXSSOIR09	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3],** **clauses 4.5.10, 4.5.11**		**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**		
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks				
Selection criteria:	The user subscribes OIR permanent mode				
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) has been received • priv-value component is set to "id" • the SIP From header is set to **anonymous** • the Calling party number information element is delivered to the called user without any digit information				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

<table>
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EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6</th></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR permanent mode</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) has been received
• priv-value component is not present
• the SIP From header is set to anonymous
• the Calling party number information element is delivered to the called user without any digit information</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

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EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "permanent mode"
Special arrangement applies</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) with an identity in the format of a tel URI has not been received
• the priv-value component is set to "none"
• the SIP From header field containing a SIP URI (PIXIT) not in the format of a tel URI has been received
• the Calling party number information element is delivered to the called user without any digit information</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
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SI_XXSSOIR12	**ISDN reference to:** **EN 300 092-1 [i.14], clause 9.3** **EN 300 403-1 [i.3],** **clauses 4.5.10, 4.5.11**				**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6** **TS 183 007 [43]**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks				
Selection criteria:	SIP URI or SIPS URI is used in the P-Preferred-Identity The user subscribes OIR "permanent mode"				
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has not** been received • SIP URI or SIPS URI are used in the in the P-Preferred -Identity • the priv-value component is set to "none" • the SIP From header field containing a SIP URI (PIXIT) not in the format of a tel URI **has not been** received • the Calling party number information element is delivered to the called user without any digit information				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_XXSSOIR13	**ISDN reference to: EN 300 092-1 [i.14], clause 9.3 EN 300 403-1 [i.3], clauses 4.5.10, 4.5.11**		**NGN reference to: Q.1912.5 [51], clause 6.1.3.6 EN 383 001 [49], clause 6.1.3.6 ES 283 027 [48], clause 7.2.3.1.2.6 TS 183 007 [43]**		
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks				
Selection criteria:	The user subscribes OIR "permanent mode"				
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • the priv-value component is set to "none" • the Calling party number information element is delivered to the called user without any digit information				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

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EN 300 092-1 [i.14], clause 9.3
EN 300 403-1 [i.3],
clauses 4.5.10, 4.5.11</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6
TS 183 007 [43]</th></tr>
<tr><td>TSS reference:</td><td colspan="4">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="4">The user subscribes OIR "permanent mode"</td></tr>
<tr><td>Test purpose:</td><td colspan="4">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received
the priv-value component is set to "none"
the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has been received
the Calling party number information element is delivered to the called user without any digit information</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="4">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="4"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

SI_XXSSOIR15	**ISDN reference to: EN 300 092-1 [i.14], clause 9.3 EN 300 403-1 [i.3], clauses 4.5.10, 4.5.11**			**NGN reference to: Q.1912.5 [51], clause 6.1.3.6 EN 383 001 [49], clause 6.1.3.6 ES 283 027 [48], clause 7.2.3.1.2.6 TS 183 007 [43]**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks			
Selection criteria:	The user subscribes OIR "permanent mode"			
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received • the priv-value component is set to "none" • the SIP From header is set to **anonymous** • the Calling party number information element is delivered to the called user without any digit information			
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)			
Comments:				

	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

6.2.2.3 TIP/COLP

<table>
<tr><th>SI_XXSSCOLP01</th><th colspan="3">ISDN reference to:
EN 300 092-1 [i.14]
EN 300 403-1 [i.3]</th><th colspan="2">NGN reference to:
TS 183 008 [44]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISDN/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5"></td></tr>
<tr><td>ISDN Parameter criteria:</td><td colspan="5">Temporary presentation allowed</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT on receipt of an CONNECT message with a Connected Party Number information element coded Address presentation restricted parameter = presentation allowed nature of address indicator = ISDN_NAI Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = ISDN_SI Address signals included

sends a 200 OK INVITE to the UAC with a P-Asserted-Identity header field containing a URI with an identity in the format of a tel URI has been received.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">CONNECT;
Connected number</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

<table>
<tr><th colspan="2">Values for test purposes SI_XXSSCOLP01</th></tr>
<tr><td>VA_01</td><td>ISDN_NAI = national number, '010'B
ISDN_SI = user provided verified and passed, '01'B</td></tr>
<tr><td>VA_02</td><td>ISDN_NAI = national number, '010'B B
ISDN_SI = user provided, not screened, '00'B</td></tr>
<tr><td>VA_03</td><td>ISDN_NAI = international number, '001'B
ISDN_SI = user provided, not screened '00'B</td></tr>
<tr><td>VA_04</td><td>ISDN_NAI = international number, '001'B
ISDN_SI = user provided, verified and passed '01'B</td></tr>
</table>

SI_XXSSCOLP02	**ISDN reference to:** **EN 300 092-1 [i.14]** **EN 300 403-1 [i.3]**			**NGN reference to:** **TS 183 008 [44]**
TSS reference:	SIP-ISDN/SS/COLP			
SIP selection criteria:				
ISDN Parameter criteria:	Temporary presentation allowed			
Test purpose:	Ensure that the SUT on receipt of an CONNECT message without a Connected Party Number information element sends a 200 OK INVITE to the UAC with a P-Asserted-Identity header field containing a URI with an identity in the format of a tel URI has been received.			
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)			
ISDN Parameter values:	CONNECT; Connected number			
Comments:				

	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_XXSSCOLP03	**ISDN reference to: EN 300 092-1 [i.14] EN 300 403-1 [i.3]**			**NGN reference to: TS 183 008 [44]**	
TSS reference:	SIP-ISDN/SS/COLP				
SIP selection criteria:					
ISDN Parameter criteria:	Temporary presentation allowed				
Test purpose:	Ensure that the SUT on receipt of an CONNECT message with a Unscreened Connected Party Number information element sends a 200 OK INVITE to the UAC with a P-Asserted-Identity header field containing a URI with an identity in the format of a tel URI has been received. The unscreened number is delivered in a From Header of the UPDATE request send after the 200 OK INVITE.				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
ISDN Parameter values:	CONNECT; Connected number , screening indicator user provided not verified				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	UPDATE	←			
	200 OK UPDATE	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

6.2.2.4 TIR/COLR

SI_XXSSCOLR01	**ISDN reference to:** **EN 300 092-1 [i.14]** **EN 300 403-1 [i.3]**			**NGN reference to:** **TS 183 008 [44]**
TSS reference:	SIP-ISDN/SS/COLP			
SIP selection criteria:				
ISDN Parameter criteria:	Temporary presentation allowed			
Test purpose:	Ensure that the SUT on receipt of an ANM message with a Connected Party Number information element coded: • Address presentation restricted parameter = presentation restricted, • Nature of address indicator = ISDN_NAI • Numbering plan indicator = ISDN/Telephony numbering plan • Screening indicator = ISDN_SI • Address signal included • sends a 200 OK INVITE to the UAC with • A P-Asserted-Identity header field containing a URI has **not** been received • APrivacy header field was received and the priv-value component is set to "id".			
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)			
ISDN Parameter values:	CONNECT; Connected number			
Comments:				
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

Values for test purposes SI_XXSSCOLR01	
VA_01	ISDN_NAI = national number, '010'B ISDN_SI = user provided verified and passed, '01'B
VA_02	ISDN_NAI = national number, '010'B B ISDN_SI = user provided, not screened, '00'B
VA_03	ISDN_NAI = international number, '001'B ISDN_SI = user provided, not screened '00'B
VA_04	ISDN_NAI = international number, '001'B ISDN_SI = user provided, verified and passed '01'B

<table>
<tr><th>SI_XXSSCOLR02</th><th colspan="3">ISDN reference to:
EN 300 092-1 [i.14]
EN 300 403-1 [i.3]</th><th colspan="2">NGN reference to:
TS 183 008 [44]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISDN/SS/COLP</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5"></td></tr>
<tr><td>ISDN Parameter criteria:</td><td colspan="5">Permanent presentation restricted</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT on receipt of an CONNECT message with a Connected Party Number information element coded:
• Address presentation restricted parameter = presentation allowed
• Nature of address indicator = ISDN_NAI
• Numbering plan indicator = ISDN/Telephony numbering plan
• Screening indicator = ISDN_SI
• Address signals included
• Sends a 200 OK INVITE to the UAC with
• A P-Asserted-Identity header field has not been received
• A Privacy header field was received and the priv-value component is set to "id".</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">CONNECT;
Connected number</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERT</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONN</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISC</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
</table>

Values for test purposes SI_XXSSCOLR02	
VA_01	ISDN_NAI = national number, '010'B ISDN_SI = user provided verified and passed, '01'B
VA_02	ISDN_NAI = national number, '010'B B ISDN_SI = user provided, not screened, '00'B
VA_03	ISDN_NAI = international number, '001'B ISDN_SI = user provided, not screened '00'B
VA_04	ISDN_NAI = international number, '001'B ISDN_SI = user provided, verified and passed '01'B

SI_XXSSCOLR03	**ISDN reference to: EN 300 092-1 [i.14] EN 300 403-1 [i.3]**			**NGN reference to: TS 183 008 [44]**	
TSS reference:	SIP-ISDN/SS/COLP				
SIP selection criteria:					
ISDN Parameter criteria:	Temporary presentation restricted				
Test purpose:	Ensure that the SUT receiving a CONNECT message without a Connected number: • Sends a 200 OK INVITE to the UAC with • A P-Asserted-Identity header field containing a URI with an identity in the format of a tel URI has **not** been received • A Privacy header field was received and the priv-value component is set to "id".				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
ISDN Parameter values:	CONNECT; Connected number				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

6.2.2.5 CFU

6.2.2.5.1 CFU - SIS

SIS_XXSSCFU 01	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU. User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFU 02	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B in network N2 is provided with CFU "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = Yes). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFU 03	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B in network N2 is provided with CFU "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = No). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFU 04	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFU 05	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFU	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFU. The user A and the user C are in network N1. The user B is provided with CFU "calling user is notified of call diversion with diverted to number" = No, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation allowed - no TIR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFU 06	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Partial Rerouting. User B has a point-to-point Configuration.	
Selection criteria:	User B has activated the Partial Rerouting service Call forwarding unconditional supported performed by the UE	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
			REL	←	→	INVITE	→	INVITE	→	INVITE			→	INVITE	
			RLC	→									←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFU 07	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Partial Rerouting "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes. User B has a point-to-point Configuration.	
Selection criteria:	User B has activated the Partial Rerouting service Call forwarding unconditional supported performed by the UE	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
			REL	←	→	INVITE	→	INVITE	→	INVITE			→	INVITE	
			RLC	→									←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFU 08	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Partial Rerouting "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = No. User B has a point-to-point Configuration.	
Selection criteria:	User B has activated the Partial Rerouting service Call forwarding unconditional supported performed by the UE	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
			REL	←	→	INVITE	→	INVITE	→	INVITE			→	INVITE	
			RLC	→									←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFU 09	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFU	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with Partial Rerouting User C is provided with TIR. The user B is provided with Partial Rerouting "calling user is notified of call diversion with diverted to number" = No, "diverting number is released to the diverted-to user" = No. User B has a point-to-point Configuration.	
Selection criteria:	User B has activated the Partial Rerouting service Call forwarding unconditional supported performed by the UE CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - TIR) and user C is not informed of the forwarding number (user B has presentation not allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→											
			REL	←	→	INVITE	→	INVITE	→	INVITE			→	INVITE	
			RLC	→									←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFU 10	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFU	
Configuration:	The user B is in network N2 and is provided with CFU User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported user C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFU was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
					←	486	←	486	←	486			←	486	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SIS_XXSSCFU 11	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFU	
Configuration:	The user B is in network N2 and is provided with CFU. User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported user C is network determined user busy	
Test purpose:	To verify that a call is released correctly if CFU was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE					
					←	486	←	486	←	486					
					→	ACK	→	ACK	→	ACK					
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SIS_XXSSCFU 12	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFU	
Configuration:	The user B is in network N2 and is provided with CFU. User B has a point-to-point Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported performed by the UE user C is network determined user busy User B has activated the Partial Rerouting service	
Test purpose:	To verify that a call is released correctly if CFU was not successful. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
			REL	←	→	INVITE	→	INVITE	→	INVITE			→	INVITE	
			RLC	→	←	486	←	486	←	486			←	486	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SIS_XXSSCFU 13	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFU	
Configuration:	The user B is in network N2 and is provided with CFU. User B has a point-to-point Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported performed by the UE user C is network determined user busy User B has activated the Partial Rerouting service	
Test purpose:	To verify that a call is released correctly if CFU was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
			REL	←	→	INVITE	→	INVITE	→	INVITE					
			RLC	→	←	486	←	486	←	486					
					→	ACK	→	ACK	→	ACK					
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SISI_XXSSCFU 14	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU. User D forwards the call to back to user B. User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C and D. User D forwards the call to back to user B. Ensure that the call is released.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

6.2.2.5.2 CFU - SII

SII_XXSSCFU 01	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU. User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

UE B	UE C		MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A			
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
	SETUP	←													
	ALERTING	→			→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFU 02	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = Yes). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

UE B	UE C		MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A			
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
	SETUP	←													
	ALERTING	→			→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFU 03	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = No). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

UE B	UE C		MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A			
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
	SETUP	←													
	ALERTING	→			→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFU 04	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

UE B		UE C	MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A			
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
	SETUP	←													
	ALERTING	→			→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFU 05	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU "calling user is notified of call diversion with diverted to number" = No, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

UE B		UE C		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A			
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
	SETUP	←													
	ALERTING	→			→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFU 06	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Partial Rerouting service. User B has a point-to-point Configuration.	
Selection criteria:	User B has activated the Partial Rerouting service Call forwarding unconditional supported performed by the UE	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

UE C	UE B		MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A			
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
	SETUP	←	REL	←											
	ALERTING	→	RLC	→	→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFU 07	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Partial Rerouting service "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes). User B has a point-to-point Configuration.	
Selection criteria:	User B has activated the Partial Rerouting service Call forwarding unconditional supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

UE C	UE B		MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A			
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
	SETUP	←	REL	←											
	ALERTING	→	RLC	→	→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFU 08	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Partial Rerouting service "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = No). User B has a point-to-point Configuration.	
Selection criteria:	User B has activated the Partial Rerouting service Call forwarding unconditional supported performed by the UE	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

UE C		UE B		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A			
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
	SETUP	←	REL	←											
	ALERTING	→	RLC	→	→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFU 09	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Partial Rerouting service "calling user is notified of call diversion with diverted to number" = No, "diverting number is released to the diverted-to user" = No). User B has a point-to-point Configuration.	
Selection criteria:	User B has activated the Partial Rerouting service Call forwarding unconditional supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

UE B		UE B		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A			
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→											
	SETUP	←	REL	←											
	ALERTING	→	RLC	→	→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFU 10	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU. User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	To verify that a call is released correctly if **CFU was not successful.** User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

UE B		UE C		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A			
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
	SETUP	←													
	RLC#17	→			→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SII_XXSSCFU 11	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and in network N2 and is provided with CFU. User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	To verify that a call is released correctly if CFU **was not successful**. User A calls user B, the call is forwarded to user C who is **network determined user busy.**	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

UE C		UE B		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A			
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SII_XXSSCFU 13	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Partial Rerouting service. User B has a point-to-point Configuration.	
Selection criteria:	Call forwarding unconditional supported User B has activated the Partial Rerouting service	
Test purpose:	To verify that a call is released correctly if **CFU** performed by the UE **was not successful.** User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

UE B		UE C		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A			
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
	SETUP	←	REL	←											
	RLC#17	→	RLC	→	→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SII_XXSSCFU 14	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Partial Rerouting service. User B has a point-to-point Configuration.	
Selection criteria:	Call forwarding unconditional supported User B has activated the Partial Rerouting service	
Test purpose:	To verify that a call is released correctly if CFU performed by the UE **was not successful.** User A calls user B, the call is forwarded to user C who is **network determined user busy.**	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

UE B	UE C	MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A	UE-C
										←	INVITE	
								←	INVITE			
						←	INVITE					
		SETUP	←	←	INVITE							
		FAC	→	→	181	→	181	→	181	→	181	
		REL	←									
		RLC	→	→	486	→	486	→	486	→	486	
				←	ACK	←	ACK	←	ACK	←	ACK	

SIIS_XXSSCFU 15	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU. User D in network N2 forwards the call to back to user B.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C and D. User D forwards the call to back to user B. Ensure that the call is released.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	BC = PIXIT	
Comments:		

SIIS_XXSSCFU 16	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:**
TSS reference:	SIP-ISDN-ISDN-SIP/Supplementary_services/CFU	
Selection criteria:	The user is A in network N1. The user B and the user C are in network N2. User B is provided with CFU. User D in network N1 forwards the call to back to user B. Network option: hop counter supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, C to D. User D forwards the call to back to user B. User D forwards the call to back to user B. Ensure that the call is released.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	BC = PIXIT	
Comments:		

6.2.2.6 CFB

6.2.2.6.1 CFB - SIS

SIS_XXSSCFB01	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFB- (network determined). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network CFB- (network determined)	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFB02	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFB-(network determined). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy (user determined)	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	RLC # 17											
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFB03	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is provided with CFU "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = Yes). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy (user determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	RLC #17											
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFB04	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is provided with CFU "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = No). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy (user determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	RLC #17											
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFB05	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFB "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = No, served user receives notification that the call has been forwarded" = No.) User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy (user determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	RLC #17											
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFB06	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFB The user B is provided with CFB "calling user is notified of call diversion with diverted to number" = No, "diverting number is released to the diverted-to user" = No, served user receives notification that the call has been forwarded" = No). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy (user determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation allowed - no TIR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	RLC #17											
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFB07	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is provided with CFU "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = Yes). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy (network determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no TIR) and user C is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFB08	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is provided with CFU "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = No). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy (network determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no TIR) and user C is informed of the forwarding number (user B has presentation allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFB09	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFB "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = No). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy (network determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - TIR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFB 10	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFB The user B is provided with CFB "calling user is notified of call diversion with diverted to number" = No, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = No). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy (network determined) CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation allowed - no TIR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFB 11	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Partial Rerouting service. User B has a point-to-point Configuration.	
Selection criteria:	User B has activated the Partial Rerouting service Call forwarding busy performed by the UE	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
			REL	←	→	INVITE	→	INVITE	→	INVITE			→	INVITE	
			RLC	→									←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFB 12	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Partial Rerouting service "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes. User B has a point-to-point Configuration.	
Selection criteria:	User B has activated the Partial Rerouting service Call forwarding busy	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no TIR) and user C is informed of the forwarding number (user B has presentation allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
			REL	←	→	INVITE	→	INVITE	→	INVITE			→	INVITE	
			RLC	→									←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFB 13	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Partial Rerouting service "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = No. User B has a point-to-point Configuration.	
Selection criteria:	User B has activated the Partial Rerouting service Call forwarding busy performed by the UE	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
			REL	←	→	INVITE	→	INVITE	→	INVITE			→	INVITE	
			RLC	→									←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFB 14	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Partial Rerouting service The user B is provided with Partial Rerouting service "calling user is notified of call diversion with diverted to number" = No, "diverting number is released to the diverted-to user" = No). User B has a point-to-point Configuration.	
Selection criteria:	User B has activated the Partial Rerouting service Call forwarding busy performed by the UE CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation allowed - no TIR) and user C is not informed of the forwarding number (user B has presentation not allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→											
			REL	←	→	INVITE	→	INVITE	→	INVITE			→	INVITE	
			RLC	→									←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFB 15	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFB	
Configuration:	The user B is in network N2 and is provided with CFB. User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network CFB supported user C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFB was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			RLC#17	→	→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
					←	486	←	486	←	486			←	486	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SIS_XXSSCFB 16	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFB	
Configuration:	The user B is in network N2 and is provided with CFB. User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network CFB supported user C is network determined user busy	
Test purpose:	To verify that a call is released correctly if CFB was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE					
					←	486	←	486	←	486					
					→	ACK	→	ACK	→	ACK					
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SIS_XXSSCFB 17	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFB	
Configuration:	The user B is in network N2 and is provided with Partial Rerouting service. User B has activated the Partial Rerouting service. User B has a point-to-point Configuration.	
Selection criteria:	CFB supported performed by the UE user C is network determined user busy User B has activated the Partial Rerouting service	
Test purpose:	To verify that a call is released correctly if CFU was not successful. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
			REL	←	→	INVITE	→	INVITE	→	INVITE			→	INVITE	
			RLC	→	←	486	←	486	←	486			←	486	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SIS_XXSSCFB 18	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFB	
Configuration:	The user B is in network N2 and is provided with Partial Rerouting service. User B has activated the Partial Rerouting service. User B has a point-to-point Configuration.	
Selection criteria:	CFB supported performed by the UE User C is network determined user busy User B has activated the Partial Rerouting service	
Test purpose:	To verify that a call is released correctly if CFU was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
			REL	←	→	INVITE	→	INVITE	→	INVITE					
			RLC	→	←	486	←	486	←	486					
					→	ACK	→	ACK	→	ACK					
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SISI_XXSSCFB 19	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFB. User D forwards the call to back to user B.	
Selection criteria:	Call forwarding by the network Call forwarding busy	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C and D. User D forwards the call to back to user B. Ensure that the call is released.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
ISDN Parameter values:	BC = PIXIT	
Comments:		

6.2.2.6.2 CFB - SII

SII_XXSSCFB 01	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B in network N2 is provided with CFB. User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy - NDUB	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
	SETUP	←													
	ALERTING	→			→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFB 02	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B in network N2 is provided with CFB. User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy - UDUB	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			RLC#17	→											
					→	181	→	181	→	181	→	181			
	SETUP	←													
	ALERTING	→			→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFB 03	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 provided with CFB "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = Yes). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy - NDUB CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2	ISDN 1		MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
	SETUP	←													
	ALERTING	→			→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFB 04	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 provided with CFB "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = No). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy - NDUB CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
	SETUP	←													
	ALERTING	→			→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFB 05	ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5	NGN reference to: TS 183 004 [45]
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. User B in network N2 is provided with CFU "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy - NDUB CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - TIR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
												←	INVITE			
										←	INVITE					
								←	INVITE							
						←	INVITE									
						→	181	→	181	→	181	→	181			
	SETUP	←														
	ALERTING	→				→	180	→	180	→	180	→	180			
	CONNECT	→				→	200 OK	→	200 OK	→	200 OK	→	200 OK			
						→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←				←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→				→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←														

SII_XXSSCFB 06	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. User B in network N2 is provided with CFU "calling user is notified of call diversion with diverted to number" = No, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy - NDUB CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2	ISDN 1		MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
	SETUP	←													
	ALERTING	→			→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFB 07	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B in network N2 is provided with CFB "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = Yes). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy - UDUB CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			RLC # 17	→											
					→	181	→	181	→	181	→	181			
	SETUP	←													
	ALERTING	→			→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFB 08	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B in network N2 is provided with CFB "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = No). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy - UDUB CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1	MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			RLC # 17	→											
					→	181	→	181	→	181	→	181			
	SETUP	←													
	ALERTING	→			→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFB 09	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. User B in network is provided with CFB "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy - UDUB CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			RLC # 17	→											
					→	181	→	181	→	181	→	181			
	SETUP	←													
	ALERTING	→			→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFB 10	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. User B in network N2 is provided with CFB "calling user is notified of call diversion with diverted to number" = No, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy - NDUB CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2	ISDN 1		MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
					←	INVITE									
					→	181	→	181	→	181	→	181			
	SETUP	←													
	ALERTING	→			→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFB 11	ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5	NGN reference to: TS 183 004 [45]
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B in network N2 is provided with CFB. User B has activated the Partial Rerouting service. User B has a point-to-point Configuration.	
Selection criteria:	User B has activated the Partial Rerouting service Call forwarding busy performed by the UE	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C
												←	INVITE		
										←	INVITE				
								←	INVITE						
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
	SETUP	←	REL	←											
	ALERTING	→	RLC	→	→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFB 12	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B in network N2 is provided with CFB Partial Rerouting service "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes). User B has a point-to-point Configuration.	
Selection criteria:	User B has activated the Partial Rerouting service Call forwarding busy performed by the UE	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
	SETUP	←	REL	←											
	ALERTING	→	RLC	→	→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFB 13	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B in network N2 is provided with CFB Partial Rerouting service "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = No). User B has a point-to-point Configuration.	
Selection criteria:	User B has activated the Partial Rerouting service Call forwarding busy	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
	SETUP	←	REL	←											
	ALERTING	→	RLC	→	→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFB 14	**ISDN reference to:** **EN 300 207-1 [i.5]** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B in network N2 is provided with CFB - Partial Rerouting service "calling user is notified of call diversion with diverted to number" = No, "diverting number is released to the diverted-to user" = No). User B has a point-to-point Configuration.	
Selection criteria:	User B has activated the Partial Rerouting service Call forwarding busy performed by the UE	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation allowed no - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2	ISDN 1		MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→											
	SETUP	←	REL	←											
	ALERTING	→	RLC	→	→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCFB 15	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFB. User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy supported	
Test purpose:	To verify that a call is released correctly if **CFB was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2	ISDN 1		MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			RLC # 17	→											
	SETUP	←			→	181	→	181	→	181	→	181			
	RLC #17	→													
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SII_XXSSCFB 16	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFB. User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy supported	
Test purpose:	To verify that a call is released correctly if CFB **was not successful**. User A calls user B, the call is forwarded to user C who is **network determined user busy.**	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			RLC # 17	→											
					→	181	→	181	→	181	→	181			
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SII_XXSSCFB 17	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFB. User B has a point-to-point Configuration.	
Selection criteria:	Call forwarding busy supported User B has activated the Partial Rerouting service	
Test purpose:	To verify that a call is released correctly if Partial Rerouting **was not successfully**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1	ISDN 2		MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
											→	100 Trying			
									←	INVITE					
									→	100 Trying					
							←	INVITE							
							→	100 Trying							
			SETUP	←	←	INVITE									
					→	100 Trying									
			FAC	→	→	181									
			REL	←			→	181							
			RLC	→					→	181					
											→	181			
	SETUP	←													
	RLC #17	→													
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SII_XXSSCFB 18	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFB- Partial Rerouting. User B has a point-to-point Configuration.	
Selection criteria:	Call forwarding busy supported User B has activated the Partial Rerouting service	
Test purpose:	To verify that a call is released correctly if Partial Rerouting - CFB performed by the UE **was not successful**. User A calls user B, the call is forwarded to user C who is **network determined user busy.**	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1	ISDN 2		MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
											→	100 Trying			
									←	INVITE					
									→	100 Trying					
							←	INVITE							
							→	100 Trying							
			SETUP	←	←	INVITE									
					→	100 Trying									
			FAC	→	→	181									
			REL	←			→	181							
			RLC	→					→	181					
											→	181			
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SII_XXSSCFB 19	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Partial Rerouting - CFB. User B has a point-to-point Configuration.	
Selection criteria:	Call forwarding busy supported User B has activated the Partial Rerouting service	
Test purpose:	To verify that a call is released correctly if Partial Rerouting - CFB performed by the UE was not successful. User A calls user B, the call is forwarded to user C who is **user determined user busy.**	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1	ISDN 2		MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
											→	100 Trying			
									←	INVITE					
									→	100 Trying					
							←	INVITE							
							→	100 Trying							
			SETUP	←	←	INVITE									
					→	100 Trying									
			FAC	→	→	181									
			REL	←			→	181							
			RLC	→					→	181					
											→	181			
	SETUP	←													
	RLC #17	→													
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SIIS_XXSSCFB 20	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP-ISDN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU. User D forwards the call to back to user B. User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network Call forwarding busy	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C and D. User D forwards the call to back to user B. Ensure that the call is released.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SIIS_XXSSCFB 21	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: ITU-T Q.1912.5 [51]**
TSS reference:	SIP-ISDN-ISDN-SIP/Supplementary_services/CFB	
Selection criteria:	The user is A in network N1. The user B and the user C are in network N2. User B is provided with CFB. User E forwards the call to back to user B. Network option: hop counter supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, C to D. User D forwards the call to back to user B. User D forwards the call to back to user B. Ensure that the call is released.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

6.2.2.7 CFNR

6.2.2.7.1 CFNR - SIS

SIS_XXSSCFNR 01	**ISDN reference to: EN 300 403-1 [i.3], clauses 9.2.2, 9.2.4.4, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFNR/	
Selection criteria:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR , option B, immediate release, and no notification. User B has a point-to-multipoint Configuration.	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
			←	REL											
			→	RLC											
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180			←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFNR02	**ISDN reference to:** **EN 300 403-1 [i.3],** **clauses 9.2.2, 9.2.4.4, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFNR/	
Selection criteria:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR , option A, late release, no notification. User B has a point-to-multipoint Configuration.	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180			←	180	
			←	REL	→	180	→	180	→	180	→	180			
			→	RLC	←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFNR03	**ISDN reference to: EN 300 403-1 [i.3], clauses 9.2.2, 9.2.4.4, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFNR/	
Configuration:	The user A and the user C are in network N1. The user B is provided with CFNR "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes ". User B has a point-to-multipoint Configuration.	
Selection criteria:	CFNR supported, option B, immediate release CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C. User A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed).	
ISDN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
			←	REL											
			→	RLC											
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180			←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFNR 04	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = No). User B has a point-to-multipoint Configuration.	
Selection criteria:	CFNR supported, option B, immediate release CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
			←	REL											
			→	RLC											
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180			←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFNR 05	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR "calling user is notified of call diversion with diverted to number" = No, "diverting number is released to the diverted-to user" = No). User B has a point-to-multipoint Configuration.	
Selection criteria:	CFNR supported, option B, immediate release CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is not informed of the forwarding number (user B has presentation not allowed).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
			←	REL											
			→	RLC											
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180			←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFNR 06	**ISDN reference to: EN 300 403-1 [i.3], clauses 9.2.2, 9.2.4.4, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFNR/	
Configuration:	The user A and the user C are in network N1. The user B is provided with CFNR "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes). User B has a point-to-multipoint Configuration.	
Selection criteria:	CFNR supported, option A, late release CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C. User A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed).	
ISDN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
			←	REL	←	180	←	180	←	180			←	180	
			→	RLC	→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFNR 07	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = No). User B has a point-to-multipoint Configuration.	
Selection criteria:	CFNR supported, option A, late release CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
			←	REL	←	180	←	180	←	180			←	180	
			→	RLC	→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFNR 08	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR "calling user is notified of call diversion with diverted to number" = No, "diverting number is released to the diverted-to user" = No, "). User B has a point-to-multipoint Configuration.	
Selection criteria:	CFNR supported, option A, late release CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is not informed of the forwarding number (user B has presentation not allowed).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
			←	REL	←	180	←	180	←	180			←	180	
			→	RLC	→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFNR 09	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Partial rerouting during alerting. User B has a point-to-point Configuration.	
Selection criteria:	Partial rerouting during alerting - no notification	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERT	→	180	→	180	→	180	→	180			
			→	FAC											
			←	REL											
			→	RLC											
			←	SETUP	→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180			←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFNR 10	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Partial rerouting during alerting "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes). User B has a point-to-point Configuration.	
Selection criteria:	Partial rerouting during alerting - full notification	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. User A is notified of call diversion and informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is informed of the forwarding number (user B has presentation allowed).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERT	→	180	→	180	→	180	→	180			
			→	FAC											
			←	REL											
			→	RLC	→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180			←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFNR 11	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Partial rerouting during alerting "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = No). User B has a point-to-point Configuration.	
Selection criteria:	Partial rerouting during alerting	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERT	→	180	→	180	→	180	→	180			
			→	FAC											
			←	REL											
					→	181	→	181	→	181	→	181			
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180			←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFNR 12	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Partial rerouting during alerting "calling user is notified of call diversion with diverted to number" = No, "diverting number is released to the diverted-to user" = No). User B has a point-to-point Configuration.	
Selection criteria:	Partial rerouting during alerting - no notification	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is not informed of the forwarding number (user B has presentation not allowed).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERT											
			→	FAC											
			←	REL											
			→	RLC	→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	180	←	180	←	180			←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCFNR 13	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR. User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network CFNR option B, immediate release, no notification user C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFNR was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERT											
			←	DISC											
			→	REL	→	INVITE	→	INVITE	→	INVITE			→	INVITE	
													←	100 Trying	
					←	486	←	486	←	486			←	486	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SIS_XXSSCFNR 14	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR. User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network CFNR supported option B, immediate release, no notification user C is network determined user busy	
Test purpose:	To verify that a call is released correctly if CFNR was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERT											
			←	DISC											
			→	REL	→	INVITE	→	INVITE	→	INVITE					
					←	486	←	486	←	486					
					→	ACK	→	ACK	→	ACK					
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SIS_XXSSCFNR 15	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR . User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network CFNR option A, late release, no notification user C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFNR was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy. User B continues to alert to the forwarding user.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERT											
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
					←	486	←	486	←	486			←	486	
					→	ACK	→	ACK	→	ACK			→	ACK	
			←	CONNECT	←	200 OK	←	200 OK	←	200 OK	←	200 OK			
					→	ACK	→	ACK	→	ASCK	→	ACK			
			←	DISC	←	BYE	←	BYE	←	BYE	←	BYE			
			→	REL	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
			←	RLC											

SIS_XXSSCFNR 16	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR. User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network CFNR supported option A, late release, no notification user C is network determined user busy	
Test purpose:	To verify that a call is released correctly if CFNR was not successful. User A calls user B, the call is forwarded to user C who is **user** determined user busy. User B continues to alert to the forwarding user.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERT											
			→	FAC											
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
					←	486	←	486	←	486			←	486	
					→	ACK	→	ACK	→	ACK			→	ACK	
			←	CONNECT	←	200 OK	←	200 OK	←	200 OK	←	200 OK			
					→	ACK	→	ACK	→	ASCK	→	ACK			
			←	DISC	←	BYE	←	BYE	←	BYE	←	BYE			
			→	REL	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
			←	RLC											

SIS_XXSSCFNR 17	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR. User B has a point-to-point Configuration.	
Selection criteria:	Call forwarding by the network CFNR supported, partial rerouting (option B, immediate release) user C is network determined network busy	
Test purpose:	To verify that a call is released correctly if CFNR was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERT											
			→	FAC											
			←	REL											
			→	RLC	→	INVITE	→	INVITE	→	INVITE					
					←	486	←	486	←	486					
					→	ACK	→	ACK	→	ACK					
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SIS_XXSSCFNR 18	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR. User B has a point-to-point Configuration.	
Selection criteria:	Call forwarding by the network CFNR, partial rerouting (option A, late release) user C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFNR was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy. User B continues to alert to the forwarding user.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERT											
			→	FAC											
					→	INVITE	→	INVITE	→	INVITE			→	INVITE	
					←	486	←	486	←	486			←	486	
					→	ACK	→	ACK	→	ACK			→	ACK	
			→	CONNECT	→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
			←	DISC	←	BYE	←	BYE	←	BYE	←	BYE			
			→	REL	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
			←	RLC											

SISI_XXSSCFNR 19	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP-ISDN/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR. User D forwards the call to back to user B. User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network CFNR supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C and D. User D forwards the call to back to user B. Ensure that the call is released.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

6.2.2.7.2 CFNR - SII

SII_XXSSCFNR 01	**ISDN reference to: EN 300 403-1 [i.3], clauses 9.2.2, 9.2.4.4, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFNR/	
Selection criteria:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR , **option B, immediate release**, and no notification. User B has a point-to-multipoint Configuration.	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1			ISDN 2	MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
			←	REL											
			→	RLC											
	←	SETUP													
	→	ALERTING													
	→	CONNECT			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
	←	DISC			←	BYE	←	BYE	←	BYE	←	BYE			
	→	REL			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	←	RLC													

SII_XXSSCFNR 02	**ISDN reference to: EN 300 403-1 [i.3], clauses 9.2.2, 9.2.4.4, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFNR/	
Selection criteria:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR , **option A, late release**, no notification. User B has a point-to-multipoint Configuration.	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
	←	SETUP	←	REL											
	→	ALERTING	→	RLC											
	→	CONNECT			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
	←	DISC			←	BYE	←	BYE	←	BYE	←	BYE			
	→	REL			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	←	RLC													

SII_XXSSCFNR 03	**ISDN reference to: EN 300 403-1 [i.3] clause 9.2.2, clause 9.2.4.4, clause 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFNR/	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 is provided with CFNR "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes. User B has a point-to-multipoint Configuration.	
Selection criteria:	CFNR supported, option B, immediate release CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C. User A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed).	
ISDN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
			←	REL											
			→	RLC											
					→	181	→	181	→	181	→	181			
	←	SETUP													
	→	ALERTING													
	→	CONNECT			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
	←	DISC			←	BYE	←	BYE	←	BYE	←	BYE			
	→	REL			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	←	RLC													

SII_XXSSCFNR 04	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = No). User B has a point-to-multipoint Configuration.	
Selection criteria:	CFNR supported, option B, immediate release CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
			←	REL											
			→	RLC											
					→	181	→	181	→	181	→	181			
	←	SETUP													
	→	ALERTING													
	→	CONNECT			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
	←	DISC			←	BYE	←	BYE	←	BYE	←	BYE			
	→	REL			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	←	RLC													

SII_XXSSCFNR 05	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR "calling user is notified of call diversion with diverted to number" = No, "diverting number is released to the diverted-to user" = No). User B has a point-to-multipoint Configuration.	
Selection criteria:	CFNR supported, option B, immediate release CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is not informed of the forwarding number (user B has presentation not allowed).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
			←	REL											
			→	RLC											
	←	SETUP													
	→	ALERTING													
	→	CONNECT			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
	←	DISC			←	BYE	←	BYE	←	BYE	←	BYE			
	→	REL			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	←	RLC													

SII_XXSSCFNR 06	**ISDN reference to:** **EN 300 403-1 [i.3],** **clauses 9.2.2, 9.2.4.4, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFNR/	
Configuration:	The user A and the user C are in network N1. The user B is provided with CFNR "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes). User B has a point-to-multipoint Configuration.	
Selection criteria:	CFNR supported, **option A, late release** CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C. User A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed).	
ISDN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
					→	181	→	181	→	181	→	181			
	←	SETUP	←	DISC											
	→	ALERTING	→	REL											
	→	CONNECT			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
	←	DISC			←	BYE	←	BYE	←	BYE	←	BYE			
	→	REL			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	←	RLC													

SII_XXSSCFNR 07	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = No). User B has a point-to-multipoint Configuration.	
Selection criteria:	CFNR supported, option A, late release CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
					→	181	→	181	→	181	→	181			
	←	SETUP	←	DISC											
	→	ALERTING	→	REL											
	→	CONNECT			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
	←	DISC			←	BYE	←	BYE	←	BYE	←	BYE			
	→	REL			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	←	RLC													

SII_XXSSCFNR 08	ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5	NGN reference to: TS 183 004 [45]
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR "calling user is notified of call diversion with diverted to number" = No, "diverting number is released to the diverted-to user" = No. User B has a point-to-multipoint Configuration.	
Selection criteria:	CFNR supported, option A, late release CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is not informed of the forwarding number (user B has presentation not allowed).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
	←	SETUP													
	→	ALERTING	←	DISC											
	→	CONNECT	→	REL	→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
	←	DISC			←	BYE	←	BYE	←	BYE	←	BYE			
	→	REL			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	←	RLC													

SII_XXSSCFNR 09	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR . User B has a point-to-point Configuration.	
Selection criteria:	Partial rerouting during alerting - no notification	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
			→	FAC											
			←	REL											
			→	RLC											
	←	SETUP													
	→	ALERTING													
	→	CONNECT			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
	←	DISC			←	BYE	←	BYE	←	BYE	←	BYE			
	→	REL			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	←	RLC													

SII_XXSSCFNR 10	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes). User B has a point-to-point Configuration.	
Selection criteria:	Partial rerouting during alerting - full notification	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. User A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number (user B has presentation allowed).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
			→	FAC											
			←	REL											
			→	RLC	→	181	→	181	→	181	→	181			
	←	SETUP													
	→	ALERTING													
	→	CONNECT			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
	←	DISC			←	BYE	←	BYE	←	BYE	←	BYE			
	→	REL			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	←	RLC													

SII_XXSSCFNR 11	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = No). User B has a point-to-point Configuration.	
Selection criteria:	Partial rerouting during alerting	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
			→	FAC											
			←	DISC											
			→	REL	→	181	→	181	→	181	→	181			
	←	SETUP													
	→	ALERTING													
	→	CONNECT			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
	←	DISC			←	BYE	←	BYE	←	BYE	←	BYE			
	→	REL			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	←	RLC													

SII_XXSSCFNR 12	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR "calling user is notified of call diversion with diverted to number" = No, "diverting number is released to the diverted-to user" = No). User B has a point-to-point Configuration.	
Selection criteria:	Partial rerouting during alerting - no notification	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is not informed of the forwarding number (user B has presentation not allowed).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
			→	FAC											
			←	REL											
			→	RLC											
	←	SETUP													
	→	ALERTING													
	→	CONNECT			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ASCK	←	ACK			
	←	DISC			←	BYE	←	BYE	←	BYE	←	BYE			
	→	REL			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	←	RLC													

SII_XXSSCFNR 13	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR . User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network **CFNR option B,** immediate release, no notification user C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFNR was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
			←	REL											
			→	RLC											
	←	SETUP													
	→	REL # 17													
	→	RLC			→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ASCK	←	ACK			

SII_XXSSCFNR 14	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR. User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network CFNR supported option B, immediate release, no notification user C is network determined user busy	
Test purpose:	To verify that a call is released correctly if CFNR was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
			←	REL											
			→	RLC	→	181	→	181	→	181	→	181			
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ASCK	←	ACK			

SII_XXSSCFNR 15	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR . User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network CFNR option A, late release, no notification user C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFNR was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy. User B continues to alert to the forwarding user.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
					→	181	→	181	→	181	→	181			
	←	SETUP													
	→	REL # 17													
	→	RLC													
			→	CONNECT	→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ASCK	←	ACK	←	ACK			
			←	DISC	←	BYE	←	BYE	←	BYE	←	BYE			
			→	REL	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
			←	RLC											

SII_XXSSCFNR 16	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR. User B has a point-to-multipoint Configuration.	
Selection criteria:	Call forwarding by the network CFNR supported option A, late release, no notification user C is network determined user busy	
Test purpose:	To verify that a call is released correctly if CFNR was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy. User B continues to alert to the forwarding user.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
			→	CONNECT	→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ASCK	←	ACK	←	ACK			
			←	DISC	←	BYE	←	BYE	←	BYE	←	BYE			
			→	REL	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
			←	RLC											

SII_XXSSCFNR 17	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR supported, partial rerouting. User B has a point-to-point Configuration.	
Selection criteria:	Call forwarding by the network CFNR supported, partial rerouting (option B, immediate release) user C is network determined network busy	
Test purpose:	To verify that a call is released correctly if CFNR was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERT											
			→	FAC											
			←	REL											
			→	RLC	→	INVITE	→	INVITE	→	INVITE					
					←	486	←	486	←	486					
					→	ACK	→	ACK	→	ACK					
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SII_XXSSCFNR 18	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR supported, partial rerouting. User B has a point-to-point Configuration.	
Selection criteria:	Call forwarding by the network CFNR, partial rerouting (option A, late release) user C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFNR was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy. User B continues to alert to the forwarding user.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			←	SETUP	←	INVITE									
			→	ALERTING	→	180	→	180	→	180	→	180			
			→	FAC	→	181	→	181	→	181	→	181			
	←	SETUP													
	→	REL # 17													
	→	RLC													
			→	CONNECT	→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ASCK	←	ACK	←	ACK			
			←	DISC	←	BYE	←	BYE	←	BYE	←	BYE			
			→	REL	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
			←	RLC											

6.2.2.8 Call Deflection

6.2.2.8.1 CD - SIS

SIS_XXSSCD 01	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CD	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Call Deflection immediate response (option B, immediate release). User B has a point-to-multipoint Configuration.	
Selection criteria:	User B has activated the, Call Deflection immediate response (option B, immediate release)	
Test purpose:	Ensure that when user A calls user B, the call is deflected immediate to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
			REL	←	→	INVITE	→	INVITE	→	INVITE			→	INVITE	
			RLC	→									←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCD 02	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CD	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with Call Deflection immediate response (option B, immediate release) "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes,). User B has a point-to-multipoint Configuration.	
Selection criteria:	User B has activated the Call Deflection immediate response (option B, immediate release)	
Test purpose:	Ensure that when user A calls user B, the call is deflected immediate to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no TIR) and user C is informed of the forwarding number (user B has presentation allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
			REL	←	→	INVITE	→	INVITE	→	INVITE			→	INVITE	
			RLC	→									←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCD 03	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CD	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with, Call Deflection immediate response (option B, immediate release) "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = No). User B has a point-to-multipoint Configuration.	
Selection criteria:	Call Deflection immediate response (option B, immediate release)	
Test purpose:	Ensure that when user A calls user B, the call is deflected immediate to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - TIR) and user C is not informed of the forwarding number (user B has presentation not allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
			REL	←	→	INVITE	→	INVITE	→	INVITE			→	INVITE	
			RLC	→									←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCD 04	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CD	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with Call Deflection immediate response (option B, immediate release) User C is provided with CD immediate response. "calling user is notified of call diversion with diverted to number" = No, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no). User B has a point-to-multipoint Configuration.	
Selection criteria:	User B has activated the Call Deflection immediate response (option B, immediate release)	
Test purpose:	Ensure that when user A calls user B, the call is deflected immediate to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation allowed - no TIR) and user C is not informed of the forwarding number (user B has presentation not allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→											
			REL	←	→	INVITE	→	INVITE	→	INVITE			→	INVITE	
			RLC	→									←	100 Trying	
					←	180	←	180	←	180	←	180	←	180	
					→	180	→	180	→	180	→	180			
					←	200 OK	←	200 OK	←	200 OK			←	200 OK	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					←	ACK	←	ACK	←	ACK	←	ACK			
					←	BYE	←	BYE	←	BYE			←	BYE	
					→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			→	200 OK BYE	
					→	BYE	→	BYE	→	BYE	→	BYE			
					←	200 OK BYE	←	200 OK BYE	←	200 OK BYE	←	200 OK BYE			

SIS_XXSSCD 05	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CD	
Configuration:	The user B is in network N2 and is provided with Call Deflection immediate response (option B, immediate release). User B has a point-to-multipoint Configuration.	
Selection criteria:	User B has activated the Call Deflection immediate response (option B, immediate release)	
Test purpose:	To verify that a call is released correctly if Call Deflection immediate response (option B, immediate release) was not successful. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
			REL	←	→	INVITE	→	INVITE	→	INVITE			→	INVITE	
			RLC	→	←	486	←	486	←	486			←	486	
					→	ACK	→	ACK	→	ACK			→	ACK	
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SIS_XXSSCD 06	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-SIP/Supplementary_services/CD	
Configuration:	User B has a point-to-multipoint Configuration.	
Selection criteria:	User B has activated the, Call Deflection immediate response (option B, immediate release)	
Test purpose:	To verify that a call is released correctly if Call Deflection immediate response (option B, immediate release) was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 1		ISDN 2		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
			REL	←	→	INVITE	→	INVITE	→	INVITE					
			RLC	→	←	486	←	486	←	486					
					→	ACK	→	ACK	→	ACK					
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

6.2.2.8.2 CD - SII

SII_XXSSCD 01	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CD	
Configuration:	The user B is provided has activated the CALL DEFLECTION, option B, immediate release. User B has a point-to-multipoint Configuration.	
Selection criteria:	User B has activated the CALL DEFLECTION, option B, immediate release	
Test purpose:	Ensure that when user A calls user B, the call is deflected immediate response to user C. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2	ISDN 1		MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
	SETUP	←	REL	←											
	ALERTING	→	RLC	→	→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCD 02	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CD	
Configuration:	The user B has activated the CALL DEFLECTION, option B, immediate release "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = Yes. User B has a point-to-multipoint Configuration.	
Selection criteria:	User B has activated the CD immediate response, option B, immediate release	
Test purpose:	Ensure that when user A calls user B, the call is deflected immediate to user C, user A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is informed of the forwarding number (user B has presentation allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C
												←	INVITE		
										←	INVITE				
								←	INVITE						
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
	SETUP	←	REL	←											
	ALERTING	→	RLC	→	→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCD 03	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CD	
Configuration:	The user B has activated the CALL DEFLECTION immediate response, option B, immediate release "calling user is notified of call diversion with diverted to number" = Yes, "diverting number is released to the diverted-to user" = No. User B has a point-to-multipoint Configuration.	
Selection criteria:	User B has activated the CD immediate response, option B, immediate release	
Test purpose:	Ensure that when user A calls user B, the call is deflected immediate response to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed - COLR) and user C is not informed of the forwarding number (user B has presentation not allowed). Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2	ISDN 1		MGCF			I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
	SETUP	←	REL	←											
	ALERTING	→	RLC	→	→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCD 04	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CD	
Configuration:	The user B has activated the CALL DEFLECTION immediate response, option B, immediate release "calling user is notified of call diversion with diverted to number" = No, "diverting number is released to the diverted-to user" = No. User B has a point-to-multipoint Configuration.	
Selection criteria:	User B has activated the CALL DEFLECTION immediate response, option B, immediate release	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation allowed - no COLR) and user C is not informed of the forwarding number (user B has presentation not allowed).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→											
	SETUP	←	REL	←											
	ALERTING	→	RLC	→	→	180	→	180	→	180	→	180			
	CONNECT	→			→	200 OK	→	200 OK	→	200 OK	→	200 OK			
					→	ACK	→	ACK	→	ACK	→	ACK			
	DISC	←			←	BYE	←	BYE	←	BYE	←	BYE			
	REL	→			→	200 OK BYE	→	200 OK BYE	→	200 OK BYE	→	200 OK BYE			
	RLC	←													

SII_XXSSCD 05	**ISDN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CD	
Configuration:	The user B is in network N2 and is provided with CD during alerting. User B has activated the CALL DEFLECTION during alerting , option B, immediate release. User B has a point-to-multipoint Configuration.	
Selection criteria:	User B has activated the CALL DEFLECTION during alerting , option B, immediate release	
Test purpose:	To verify that a call is released correctly if **CD performed by the UE was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→											
	SETUP	←	REL	←	→	181	→	181	→	181	→	181			
	RLC #17	→	RLC	→											
					→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

SII_XXSSCD 06	**ISDN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/CD	
Configuration:	The user B is in network N2 and is provided with the CALL DEFLECTION during alerting, option B, immediate release. User B has a point-to-multipoint Configuration.	
Selection criteria:	User B has activated the CALL DEFLECTION during alerting , option B, immediate release	
Test purpose:	To verify that a call is released correctly if CD performed by the UE was not successful. User A calls user B, the call is forwarded to user C who is network determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

ISDN 2		ISDN 1		MGCF		I-CSCF		S-CSCF		P-CSCF		UE-A		UE-C	
											←	INVITE			
									←	INVITE					
							←	INVITE							
			SETUP	←	←	INVITE									
			FAC	→	→	181	→	181	→	181	→	181			
			REL	←											
			RLC	→	→	486	→	486	→	486	→	486			
					←	ACK	←	ACK	←	ACK	←	ACK			

6.2.2.9 Three Party service 3PTY

SII_xxSS3PTY01	**ISDN reference to:** **EN 300 188-1 [i.6],** **clause 9.2**	**NGN reference to:** **ES 283 027 [48]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/3PTY	
Selection criteria:	The user B is in network N1 and is provided with 3PTY. The user A is in network N2 and user C is in the network N1 User B has a point-to-multipoint Configuration.	
Test purpose:	Ensure that user B can establish a three-way conversation call with user A and user C and release the Active-Idle connection (B-C). The call clearing procedure is performed from user A after the Active-Held call has been retrieved.	
ISDN Parameter values:	BC =PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SII_xxSS3PTY02	ISDN reference to: EN 300 188-1 [i.6], clause 9.2, figure A.2	NGN reference to: ES 283 027 [48]
TSS reference:	ISDN-ISDN/Supplementary_services/3PTY/	
Selection criteria:	The user B is in network N1 and is provided with 3PTY. The user A is in network N2 and user C is in the network N1 User B has a point-to-multipoint Configuration.	
Test purpose:	Ensure that user B can establish a three-way conversation call with user A and user C and release the Active-Held connection (B-C). The call clearing procedure is performed from user B.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SII_xxSS3PTY03	ISDN reference to: EN 300 188-1 [i.6], clause 9.2	NGN reference to: ES 283 027 [48]
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/3PTY	
Selection criteria:	The user B is in network N1 and is provided with 3PTY. The user A is in network N2 and user C is in the network N1 User B has a point-to-multipoint Configuration.	
Test purpose:	Ensure that user B can establish a three-way conversation call with user A and user C and release the Active-Idle connection (A-B). The call clearing procedure is performed from user B.	
ISDN Parameter values:	BC =PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SII_xxSS3PTY04	**ISDN reference to: EN 300 188-1 [i.6], clause 9.2**	**NGN reference to: ES 283 027 [48]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/3PTY	
Selection criteria:	The user B is in network N1 and is provided with 3PTY. The user A is in network N2 and user C is in the network N1 User B has a point-to-multipoint Configuration.	
Test purpose:	Ensure that user B can establish a three-way conversation call with user A and user C and release the Active-Idle connection (A-B). The call clearing procedure is performed from user C after the Active-Held call has been retrieved.	
ISDN Parameter values:	BC =PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SIS_xxSS3PTY05	**ISDN reference to: EN 300 188-1 [i.6], clause 9.2**	**NGN reference to: ES 283 027 [48]**
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/3PTY	
Selection criteria:	The user B is in network N1 and is provided with 3PTY. The user A and user C are in the network N2. User B has a point-to-multipoint Configuration.	
Test purpose:	Ensure that user B can establish a three-way conversation call with user A and user C and B release the Active-Idle connection (B-C). The call clearing procedure is performed from user A after the Active-Held call has been retrieved.	
ISDN Parameter values:	BC =PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SIS_xxSS3PTY06	ISDN reference to: EN 300 188-1 [i.6], clause 9.2, figure A.2	NGN reference to: ITU-T Q.1912.5 [51]
TSS reference:	ISDN-ISDN/Supplementary_services/3PTY/211702	
Selection criteria:	The user B is in network N1 and is provided with 3PTY.The user A and user C are in the network N2. User B has a point-to-multipoint Configuration.	
Test purpose:	Ensure that user B can establish a three-way conversation call with user A and user C and B release the Active-Held connection (B-C). The call clearing procedure is performed from user B.	
ISDN Parameter values:	BC = PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SIS_xxSS3PTY07	ISDN reference to: EN 300 188-1 [i.6], clause 9.2	NGN reference to: ITU-T Q.1912.5 [51]
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/3PTY	
Selection criteria:	The user B is in network N1 and is provided with 3PTY. The user A and user C are in the network N2. User B has a point-to-multipoint Configuration.	
Test purpose:	Ensure that user B can establish a three-way conversation call with user A and user C and B release the Active-Idle connection (A-B). The call clearing procedure is performed from user B.	
ISDN Parameter values:	BC =PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SIS_xxSS3PTY08	ISDN reference to: EN 300 188-1 [i.6], clause 9.2	NGN reference to: ITU-T Q.1912.5 [51]
TSS reference:	SIP-ISDN-ISDN/Supplementary_services/3PTY	
Selection criteria:	The user B is in network N1 and is provided with 3PTY. The user A and user C are in the network N2. User B has a point-to-multipoint Configuration.	
Test purpose:	Ensure that user B can establish a three-way conversation call with user A and user C and B release the Active-Idle connection (A-B). The call clearing procedure is performed from user C after the Active-Held call has been retrieved.	
ISDN Parameter values:	BC =PIXIT	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

6.2.2.10 TP

SI_XXSSTP01	ISDN reference to: EN 300 141-1 [i.7], clause 7 EN 300 196-1 [i.9], clause 7.1				NGN reference to: Q.1912.5 [51], annex B.10
TSS reference:	SIP-ISDN/Supplementary_services/TP				
Selection criteria:	The called user is provided with TP. User B has a point-to-multipoint Configuration.				
Test purpose:	Ensure that the user B can provide the suspend and resume procedure				
ISDN Parameter values:	SETUP = 3,1 kHz audio;				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	INVITE	←		←	SUSPEND
	200 OK INVITE	→		→	SUS_ACK
	ACK	←			
	INVITE	←		←	RESUME
	200 OK INVITE	→		→	RES_ACK
	ACK	←			
	BYE	←		←	DISC
	200 OK BYE	→		→	REL

SI_XXSSTP02	**ISDN reference to:** **EN 300 141-1 [i.7], clause 7** **EN 300 196-1 [i.9], clause 7.1**			**NGN reference to:** **Q.1912.5 [51], annex B.10**	
TSS reference:	SIP-ISDN/Supplementary_services/TP				
Selection criteria:	User B has a point-to-multipoint Configuration.				
Test purpose:	Ensure that the user B can provide the suspend and resume procedure and that the call can be released from the calling user during the call suspension.				
ISDN Parameter values:	SETUP = 3,1 kHz audio;				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	INVITE (sendonly)	←		←	SUSPEND
	200 OK INVITE (reconly)	→		→	SUS_ACK
	ACK	←			
	BYE	→			
	200 OK BYE	←			

SI_XXSSTP03	**ISDN reference to:** **EN 300 141-1 [i.7], clause 7** **EN 300 196-1 [i.9], clause 7.1**			**NGN reference to:** **Q.1912.5 [51], annex B.10**	
TSS reference:	SIP-ISDN/Supplementary_services/TP				
Selection criteria:	User B has a point-to-multipoint Configuration.				
Test purpose:	Ensure that the SUT stop temporarily one or more unicast media streams if a SUS message (ISDN subscriber initiated) was received. Ensure that the connection is cleared after T2 was expired in the PSTN.				
ISDN Parameter values:	SETUP = 3,1 kHz audio;				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONN
	ACK	→			
	INVITE	←		←	SUSPEND
	200 OK INVITE	→		→	SUS_ACK
			T2 expiry		
	BYE	←			
	200 OK BYE	→			

6.2.2.11 CUG

SI_XXSSCUG01	**ISDN reference to: EN 300 138-1 [i.10], clauses 9.2.2, 9.2.4**		**NGN reference to:**	
TSS reference:	SIP-ISDN/Supplementary_services/CUG/			
Selection criteria:	Term: The called user belongs to CUG with the following CUG supplementary options: not IA; not ICB. User B has a point-to-multipoint Configuration.			
Test purpose:	Ensure that when the called user belongs to a CUG with **incoming access not allowed** and **not incoming calls barred** within the CUG, after the receipt of an INVITE message the network initiate call clearing to the calling user with 500 Server internal errors.			
ISDN Parameter values:				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)			
Comments:				
	SIP		SUT	ISDN
	INVITE	→		
	500 Server Internal Error	←		
	ACK	→		

SI_XXSSCUG02	**ISDN reference to: EN 300 138-1 [i.10], clauses 9.2.2, 9.2.4**		**NGN reference to:**	
TSS reference:	SIP-ISDN/Supplementary_services/CUG/			
Selection criteria:	Term: The called user belongs to CUG with the following CUG supplementary options: not IA; ICB. User B has a point-to-multipoint Configuration.			
Test purpose:	Ensure that when the called user belongs to a CUG with **incoming access not allowed** and **incoming calls barred** within the CUG, after the receipt of an INVITE message the network initiate call clearing to the calling user with 500 Server internal errors.			
ISDN Parameter values:				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)			
Comments:				
	SIP		SUT	ISDN
	INVITE	→		
	500 Server Internal Error	←		
	ACK	→		

SI_XXSSCUG03	**ISDN reference to: EN 300 138-1 [i.10], clauses 9.2.2, 9.2.4**				**NGN reference to:**
TSS reference:	SIP-ISDN/Supplementary_services/CUG/				
Selection criteria:	Term: The called user belongs to CUG with the following CUG supplementary options: IA; not ICB . User B has a point-to-multipoint Configuration.				
Test purpose:	Ensure that when the called user belongs to a CUG with **incoming access allowed** and **not incoming calls barred** within the CUG, after the receipt of an INVITE message the call is successful on a "non CUG call" base.				
ISDN Parameter values:					
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→			
	100 Trying	←		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CON
	ACK	→			
			Conversation		
	BYE	→		→	DISC
	200 OK BYE	←		←	REL

SI_XXSSCUG04	**ISDN reference to: EN 300 138-1 [i.10], clauses 9.2.2, 9.2.4**				**NGN reference to:**
TSS reference:	SIP-ISDN/Supplementary_services/CUG/				
Selection criteria:	Term: The called user belongs to CUG with the following CUG supplementary options: IA; ICB. User B has a point-to-multipoint Configuration.				
Test purpose:	Ensure that when the called user belongs to a CUG with **incoming access allowed** and **incoming calls barred** within the CUG, after the receipt of an INVITE message the call is successful on a "non CUG call" base.				
ISDN Parameter values:					
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		ISDN
	INVITE	→			
	100 Trying	←		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CON
	ACK	→			
			Conversation		
	BYE	→		→	DISC
	200 OK BYE	←		←	REL

6.2.2.12 Hold

SI_XXSSHOLD 01	**ISDN reference to: EN 300 141-1 [i.7]**	**NGN reference to: Q.1912.5 [51], annex B.10**
TSS reference:	SIP-ISDN/SS/HOLD/	
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams	
ISDN selection criteria:	Support the generic notification procedure for HOLD supplementary service	
Test purpose:	Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold The calling party should be able to retrieve the other party The called party should be able to put the other party on hold The called party should be able to retrieve the other party	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>	
ISDN Parameter values:	SETUP = 3,1 kHz audio	
Comments:		

	SIP		MGCF		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERTING
	200 OK INVITE	←		←	CONNECT
	INVITE(sendonly)	→		→	NOTIFY Remote HOLD
	200 OK INVITE(recvonly)	←			
	INVITE(sendrecv)	→		→	NOTIFY - Remote RETRIEVAL
	200 OK INVITE(sendrecv)	←			
	INVITE(sendonly)	←		←	HOLD
	200 OK INVITE(recvonly)	→			
	INVITE(sendrecv)	←		←	RETRIEVE
	200 OK INVITE(sendrecv)	→			

SI_XXSSHOLD 02	**ISDN reference to: EN 300 141-1 [i.7]**	**NGN reference to: Q.1912.5 [51], annex B.10**
TSS reference:	SIP-ISDN/SS/HOLD/	
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams	
ISDN selection criteria:	Support the generic notification procedure for HOLD supplementary service Support the invocation of the service in the alerting state	
Test purpose:	Ensure that a party can put the other party on hold in the alerting state. Ensure that the party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold The calling party should be able to retrieve the other party	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>	
ISDN Parameter values:	SETUP = 3,1 kHz audio;	
Comments:		

	SIP		MGCF		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	UPDATE(sendonly)	→		→	NOTIFY - Remote HOLD
	200 OK UPDATE(recvonly)	←			
	UPDATE(sendrecv)	→		→	NOTIFY - Remote RETRIEVAL
	200 OK UPDATE(sendrecv)	←			

SI_XXSSHOLD 03	**ISDN reference to: EN 300 141-1 [i.7]**			**NGN reference to: Q.1912.5 [51], annex B.10**	
TSS reference:	SIP-ISDN/SS/HOLD/				
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams				
ISDN selection criteria:	Support the generic notification procedure for HOLD supplementary service				
Test purpose:	Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold The calling party should be able to retrieve the other party				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>				
ISDN Parameter values:	SETUP = 3,1 kHz audio				
Comments:					
	SIP		MGCF		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERTING
	200 OK INVITE	←		←	CONNECT
	UPDATE(sendonly)	→		→	Notify - Remote HOLD
	200 OK INVITE(recvonly)	←			
	UPDATE(sendrecv)	→		→	Notify - Remote RETRIEVAL
	200 OK UPDATE(recvonly)	←			

SI_XXSSHOLD 04	**ISDN reference to: EN 300 141-1 [i.7]**	**NGN reference to: Q.1912.5 [51], annex B.10**
TSS reference:	SIP-ISDN/SS/HOLD/	
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams The MGCF sends the update of the media stream in an UPDATE message	
ISDN selection criteria:	Support the generic notification procedure for HOLD supplementary service	
Test purpose:	Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold. The called party should be able to put the other party on hold The called party should be able to retrieve the other party	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>	
ISDN Parameter values:	SETUP = 3,1 kHz audio;	
Comments:		

	SIP		MGCF		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERT
	200 OK INVITE	←		←	CONNECT
	UPDATE(sendonly)	←		←	HOLD
	200 OK INVITE(recvonly)	→			
	UPDATE(sendrecv)	←		←	RETRIEVE
	200 OK UPDATE(recvonly)	→			

SI_XXSSHOLD 05	**ISDN reference to: EN 300 141-1 [i.7]**			**NGN reference to: Q.1912.5 [51], annex B.10**
TSS reference:	SIP-ISDN/SS/HOLD/			
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams			
ISDN selection criteria:	Support the generic notification procedure for HOLD supplementary service			
Test purpose:	Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party in held state can put the remote party put on hold. Ensure that a party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold The called party should be able to put the other party on hold The calling party should be able to retrieve the other party The called party should be able to retrieve the other party			
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition SDP: a=sendonly or a=inactive (put on hold) a=sendrecv or a=recvonly or omitted (retrieve the call) o= . . <version incremented>			
ISDN Parameter values:	SETUP = 3,1 kHz audio;			
Comments:				

	SIP		MGCF		ISDN
	INVITE	→		→	
	180 Ringing	←		←	
	200 OK INVITE	←		←	
	INVITE(sendonly)	→		→	NOTIFY- Remote HOLD
	200 OK INVITE(recvonly)	←			
	INVITE(inactive)	←		←	HOLD
	200 OK INVITE(inactive)	→			
	INVITE(recvonly)	→		→	Notify - Remote Retrieval
	200 OK INVITE(sendonly)	←			
	INVITE(sendrecv)	←		←	RETRIEVE
	200 OK INVITE(sendrecv)	→			

SI_XXSSHOLD 06	**ISDN reference to: EN 300 141-1 [i.7]**			**NGN reference to: Q.1912.5 [51], annex B.10**
TSS reference:	SIP-ISDN/SS/HOLD/			
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams			
ISDN selection criteria:	Support the generic notification procedure for HOLD supplementary service			
Test purpose:	Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party in held state can put the remote party put on hold. Ensure that a party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold The called party should be able to put the other party on hold The called party should be able to retrieve the other party The calling party should be able to retrieve the other party			
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition SDP: a=sendonly or a=inactive (put on hold) a=sendrecv or a=recvonly or omitted (retrieve the call) o= . . <version incremented>			
ISDN Parameter values:	SETUP = 3,1 kHz audio			
Comments:				

	SIP		MGCF		ISDN
	INVITE	→		→	
	180 Ringing	←		←	
	200 OK INVITE	←		←	
	INVITE(sendonly)	→		→	NOTIFY - Remote HOLD
	200 OK INVITE(recvonly)	←			
	INVITE(inactive)	←		←	HOLD
	200 OK INVITE(inactive)	→			
	INVITE(recvonly)	←		←	RETRIEVE
	200 OK INVITE(sendonly)	→			
	INVITE(sendrecv)	→		→	NOTIFY - Remote Retrieval
	200 OK INVITE(sendrecv)	←			

6.2.2.13 CONF

SI_XXSSCONF01	**ISDN reference to:** **EN 300 185-1 [i.8]**	**NGN reference to:** **TS 183 005 [46]** **TS 124 147 [i.11]**
TSS reference:	SIP-ISDN/SS/CONF/	
SIP selection criteria:	Support that a party can put the other party on hold after then calling user has provided all the information necessary for processing the call	
ISDN selection criteria:	Support of service conference call, add-on (conf)	
Test purpose:	Ensure that the SUT in the confirmed dialogue stop the temporarily sending one or more unicast media streams if a **HOLD** was sent and **resume** the session after the **BeginCONF** was received due to the CONF supplementary service. **If the media stream is either in state "sendonly" or "inactive" then: INVITE with the attribute line a_LINE_VA, or omitted attribute line, else:** no mapping	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition SDP: a= **a_LINE_VA (TABLE 9) or a line is omitted**	
ISDN Parameter values:	SETUP = 3,1 kHz audio;	
Comments:		

SIP		MGCF		ISDN
INVITE	→		→	SETUP
180 Ringing	←		←	ALERTING
200 OK INVITE	←		←	CONNECT
			←	BeginCONF
BYE	←		←	DISC
200 OK BYE	→		→	RELEASE
			←	REL_COMP

SI_XXSSCONF03	**ISDN reference to: EN 300 185-1 [i.8]**				**NGN reference to: TS 183 005 [46] TS 124 147 [i.11]**
TSS reference:	SIP-ISDN/SS/CONF/				
SIP selection criteria:	Support that a party can put the other party on hold after then calling user has provided all the information necessary for processing the call				
ISDN selection criteria:	Support of service conference call, add-on (conf)				
Test purpose:	Ensure that the SUT in the confirmed dialogue stops the temporarily sending one or more unicast media streams if a FACILITY message **isolatedCONF** and resume the media stream if **reattachedCONF** was received due to the CONF supplementary service.				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition SDP: a= **a_LINE_VA (TABLE 10) or a line is omitted**				
ISDN Parameter values:	SETUP = 3,1 kHz audio;				
Comments:					
	SIP		MGCF		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERTING
	200 OK INVITE	←		←	CONNECT
				←	BeginCONF
	INVITE(sendonly)	←		←	IsolatedCONF
	200 OK INVITE(recvonly)	→			
	ACK	←			
	INVITE(sendrecv)	←		←	ReattachCONF
	200 OK INVITE(sendrecv)	→			
	ACK	←			
	BYE	←		←	DISC
	200 OK BYE	→		→	RELEASE
				←	REL_COMP

SI_XXSSCONF05	**ISDN reference to: EN 300 185-1 [i.8]**	**NGN reference to: TS 183 005 [46] TS 124 147 [i.11]**
TSS reference:	SIP-ISDN/SS/CONF/	
SIP selection criteria:	Conference event package supported PICS [43] 1/1	
ISDN selection criteria:		
Test purpose:	*Conference notification information is mapped into "conference established"* Upon the receipt of a conference information document with the <conference-state-type> element *active* is set to 'true', the ISDN Network shall send a NOTIFY message with a notification '***conference established***'.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition NOTIFY 1: *<conference-state>* <active>**true**</active> if present	
ISDN Parameter values:	SETUP = 3,1 kHz audio	
Comments:		

	SIP		MGCF		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERTING
	200 OK INVITE	←		←	CONNECT
	NOTIFY 1	→		→	NOTIFY(conference established)
	200 OK NOTIFY	←			
	BYE	←		←	DISCONNECT
	200 OK BYE	→		→	RELEASE
				←	RELEASE COMPLETE

<table>
<tr><th>SI_XXSSCONF06</th><th colspan="3">ISDN reference to:
EN 300 185-1 [i.8]</th><th colspan="2">NGN reference to:
TS 183 005 [46]
TS 124 147 [i.11]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISDN/SS/CONF/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">Conference event package supported PICS [43] 1/1</td></tr>
<tr><td>ISDN selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5">Conference notification information is mapped into "other party added"
Upon the receipt of a conference information document with the <endpoint-type> and the element status of endpoint-status-type is set to 'connected' and it was not set to 'on-hold' before and the Contact URI in the element entity is not the address of the served PSTN/ISDN participant, the ISDN Network shall send a NOTIFY message 'other party added'.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

NOTIFY 1: <conference-state>
<active>true</active> if present

NOTIFY 2: <endpoint entity=endpoint SIPx URI
<status>connected</status></td></tr>
<tr><td>ISDN Parameter values:</td><td colspan="5">SETUP = 3,1 kHz audio</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="14"></td><td>SIP</td><td></td><td>MGCF</td><td></td><td>ISDN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>SETUP</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ALERTING</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>CONNECT</td></tr>
<tr><td colspan="5"></td></tr>
<tr><td>NOTIFY 1</td><td>→</td><td></td><td>→</td><td>NOTIFY(conference established)</td></tr>
<tr><td>200 OK NOTIFY</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>NOTIFY 2</td><td>→</td><td></td><td>→</td><td>NOTIFY(other party added)</td></tr>
<tr><td>200 OK NOTIFY</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>DISCONNECT</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>RELEASE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>RELEASE COMPLETE</td></tr>
<tr><td></td><td colspan="5">The Connection to SIP2 is not shown in this message flow.</td></tr>
</table>

SI_XXSSCONF11	**ISDN reference to:** **EN 300 185-1 [i.8]**			**NGN reference to:** **TS 183 005 [46], clause 4.7.1.1.1** **TS 124 147 [i.11]**	
TSS reference:	SIP-ISDN/SS/CONF/				
SIP selection criteria:	Conference event package supported PICS [43] 1/1				
ISDN selection criteria:					
Test purpose:	Conference notification information is mapped into "other party disconnected" Upon the receipt of a conference information document with the <endpoint-type> and the element status of endpoint-status-type is set to 'disconnected' and the element joining-method of joining-type is not set to 'focus-owner, the ISDN network shall send a NOTIFY message '**other party disconnected**'.				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition NOTIFY 3: <endpoint entity=endpoint SIPx URI <status>**disconnected**</status>				
ISDN Parameter values:	SETUP = 3,1 kHz audio				
Comments:					
	SIP		MGCF		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERTING
	200 OK INVITE	←		←	CONNECT
	NOTIFY 1	→		→	NOTIFY(conference established)
	200 OK NOTIFY	←			
	NOTIFY 2	→		→	NOTIFY (other party added)
	200 OK NOTIFY	←			
	NOTIFY 3	→		→	NOTIFY(other party disconnected)
	200 OK NOTIFY	←			
	BYE	←		←	DISCONNECT
	200 OK BYE	→		→	RELEASE
				←	RELEASE COMPLETE
	The session with SIP 2 is not shown in this message flow.				

SI_XXSSCONF12	**ISDN reference to:** **EN 300 185-1 [i.8]**			**NGN reference to:** **TS 183 005 [46]** **TS 124 147 [i.11]**	
TSS reference:	SIP-ISDN/SS/CONF/				
SIP selection criteria:	Conference event package not supported NOT PICS [43] 1/1				
ISDN selection criteria:					
Test purpose:	*Conference notification information is not mapped to ISDN* Upon the receipt of a conference information document the conference notification information is not mapped to the PSTN side. No NOTIFY is sent to the ISDN user.				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition NOTIFY 1: *<conference-state>* <active>**true**</active> if present NOTIFY 2: <endpoint entity=endpoint SIPx URI <status>**connected**</status>				
ISDN Parameter values:	SETUP = 3,1 kHz audio;				
Comments:					
	SIP		MGCF		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERTING
	200 OK INVITE	←		←	CONNECT
	NOTIFY 1	→			
	200 OK NOTIFY	←			
	NOTIFY 2				
	200 OK NOTIFY				
	BYE	←		←	DISCONNECT
	200 OK BYE	→		→	RELEASE
				←	RELEASE COMPLETE

SI_XXSSCONF13	**SIP reference: RFC 3261 [28]**				**NGN reference: Q.734.1 [14], clause 2.7**
TSS reference:	SIP-ISUP/SS/CONF/				
SIP selection criteria:	The stop and retrieve of media streams is not supported				
ISUP selection criteria:					
Test purpose:	Ensure that the SUT on receipt of FACILITY messages due to the CONF supplementary service, **no mapping, no disrupting of the SIP procedure.**				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition No mapping				
ISDN Parameter values:	SETUP = 3,1 kHz audio;				
Comments:					
	SIP		MGCF		ISUP
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERTING
	200 OK INVITE	←		←	CONNECT
				←	HOLD
				←	BeginCONF
				←	IsolatedCONF
				←	ReattachCONF
	BYE	←		←	DISCONNECT
	200 OK BYE	→		→	RELEASE
				←	RELEASE COMPLETE

SI_XXSSCONF14	**ISDN reference to: EN 300 185-1 [i.8]**				**NGN reference to: TS 183 005 [46], clause 4.7.1.1.1 TS 124 147 [i.11]**
TSS reference:	SIP-ISDN/SS/CONF/				
SIP selection criteria:					
ISDN selection criteria:					
Test purpose:	*The referring of MGCF is not possible call is established* Ensure that a REFER request received by the MGCF is not successful. The request is rejected with . 403 Forbidden. The CS -site is not affected.				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition REFER: Request URI contained the **conference URI** Refer-To contains the URI of **ISDNx**, method=invite Referred-By contains SIP or tel URI of **SIPx**				
ISDN Parameter values:	SETUP = 3,1 kHz audio				
Comments:					
	SIP		MGCF		ISDN
	INVITE	→		→	SETUP
	180 Ringing	←		←	ALERTING
	200 OK INVITE	←		←	CONNECT
	REFER	→			
	403 Forbidden	←			

SI_XXSSCONF15	**ISDN reference to: EN 300 185-1 [i.8]**				**NGN reference to: TS 183 005 [46], clause 4.7.1.1.1 TS 124 147 [i.11]**
TSS reference:	SIP-ISDN/SS/CONF/				
SIP selection criteria:					
ISDN selection criteria:					
Test purpose:	*The referring of MGCF is not possible call is not established* Ensure that a REFER request received by the MGCF is not successful. The request is rejected with . 403 Forbidden. The CS -site is not affected.				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition REFER: Request URI contained the **conference URI** Refer-To contains the URI of **ISDNx**, method=invite Referred-By contains SIP or tel URI of **SIPx**				
ISDN Parameter values:	SETUP = 3,1 kHz audio;				
Comments:					
	SIP		MGCF		ISDN
	REFER	→			
	403 Forbidden	←			

6.2.2.14 Call waiting (CW)

SI_XXSSCW01	**SIP reference: RFC 3261 [28]**				**ISUP reference: Q.1912.5 [51], annex B.9**
TSS reference:	SIP-ISDN/SS/CW/				
SIP selection criteria:					
ISDN selection criteria:					
Test purpose:	Ensure that when the ISDN SUT indicates that a **Call is a waiting call** the SIP signalling procedure is not disrupted.				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition No mapping				
ISDN Parameter values:	SETUP = 3,1 kHz audio;				
Comments:					
	SIP		SUT		ISDN
	INVITE	→			SETUP
	180 Ringing	←		←	ALERTING
	200 OK INVITE	←		←	CONN
	ACK	→			
			Conversation		
	BYE	→		→	DISC
	200 OK BYE	←		←	REL

6.2.2.15 Anonymous Call Rejection (ACR)

SI_XXSSACR01	**SIP reference: RFC 3261 [28]**	**ISUP reference: Q.1912.5 [51], annex B.8**
TSS reference:	SIP-ISDN/SS/ACR/	
SIP selection criteria:		
ISDN selection criteria:	An implementation according EN 383 001 [49]	
Test purpose:	Ensure that the SUT, if a destination user has subscribed the ACR supplementary service the call attempt is rejected with a 603 Decline: Reason header field Reason: Q.850 [22];cause=24 due to ACR supplementary service"	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition INVITE: Privacy-header = "id" 603 Decline: Reason header field Reason: Q.850 [22];cause=24	
ISDN Parameter values:	SETUP = 3,1 kHz audio	
Comments:		

SI_XXSSACR02	**SIP reference: RFC 3261 [28]**	**ISUP reference: Q.1912.5 [51], annex B.8**
TSS reference:	SIP-ISUP/SS/ACR/	
SIP selection criteria:		
ISDN selection criteria:	An implementation according ETSI EN 383 001 [49] Send a Calling Party Number with an Number Presentation restriction Indicator set to "presentation restricted by the network" if no P-Asserted -Identity header field has not been received or not in the format "+"CC+NDC+SN	
Test purpose:	Ensure that the SUT if a destination user has subscribed the ACR supplementary service the call attempt is successful	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition INVITE: No P-Asserted-Identity header field and no Privacy header field present	
ISDN Parameter values:	Calling party number Address presentation restriction is set to "Presentation restricted by the network"	
Comments:		

6.3 Test purposes for PSTN - SIP

6.3.1 Basic call

6.3.1.1 Basic call Successful

Successful
PSTN

PS_AU_01	**PSTN reference to: EN 300 899-1 [23], clause 2.1.1**	**NGN reference to: Q.1912.5 [51]**
TSS reference:	PSTN-SIP/Basic_call/Successful/3,1 kHz audio	
Selection criteria:		
Test purpose:	Ensure that the PSTN can in the overlap proceeding state handle the call establishment correctly when the SIP user answers with a Session Progress message following by a 180 Ringing message. Ensure that in the active call state the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters). In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in table 8 applies.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSTN		SUT		SIP
Off hook				
Dial number	→		→	INVITE
			←	183 Session Progress
Ringing tone			←	180 Ringing
			←	200 OK INVITE
			→	ACK
Communication				
On hook	→		→	BYE
			←	200 OK BYE

PS_AU_02	**PSTN reference to:** **EN 300 899-1 [23], clause 2.1.1**		**NGN reference to:** **Q.1912.5 [51]**		
TSS reference:	PSTN-SIP/Basic_call/Successful/3,1 kHz audio				
Selection criteria:					
Test purpose:	Ensure that the PSTN in the Outgoing call proceeding state can handle the call establishment correctly when the SIP user answers with a 200 OK message. Ensure that in the active call state the voice/data transfer is performed correctly (e.g. testing QoS parameters). In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in table 8 applies.				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	PSTN		SUT		SIP
	Off hook				
	Dial number	→		→	INVITE
				←	200 OK INVITE
				→	ACK
	Communication				
	On hook	→		→	BYE
				←	200 OK BYE

PS_AU_03	**PSTN reference to:** **EN 300 899-1 [23], clause 2.1.1**		**NGN reference to:** **Q.1912.5 [51]**		
TSS reference:	PSTN-SIP/Basic_call/Successful/3,1 kHz audio /				
Selection criteria:					
Test purpose:	Ensure that the call establishment and the call clearing procedure are performed correctly when the **calling user** clears after answering. The called user shall receive a BYE message. The Reason header should contain Cause Value #16. Ensure that in the Call Delivered call state the transfer of tone or announcement on the B- channel is performed correctly. In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in table 8 applies.				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	PSTN		SUT		SIP
	Off hook				
	Dial number	→		→	INVITE
	Ringing tone			←	180 Ringing
				←	200 OK INVITE
				→	ACK
	Communication				
	On hook	→		→	BYE
				←	200 OK BYE

PS_AU_04	PSTN reference to: EN 300 899-1 [23], clause 2.1.1	NGN reference to: Q.1912.5 [51]
TSS reference:	PSTN -SIP/Basic_call/Successful/3,1 kHz audio/	
Selection criteria:		
Test purpose:	Ensure that the call clearing procedure is performed correctly when the **called user** clears after answering with a BYE message indicating the Cause value # 16 "normal call clearing" in the Reason header field . Ensure that in the Call Delivered call state U4 and disconnect indication state (N12) the transfer of tone or announcement is performed correctly. In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in table 8 applies.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSTN		SUT		SIP
Off hook				
Dial number	→		→	INVITE
Ringing tone			←	180 Ringing
			←	200 OK INVITE
			→	ACK
Communication				
Special tone or announcement	←		←	BYE
On hook			→	200 OK BYE

Table 7

		m= line		b= line	a= line
	<media>	**<transport>**	**<fmt-list>**	**<modifier>:<band width-value>**	**rtpmap:<dynamic-PT> <encoding name>/ <clock rate>[/encoding parameters>**
VA_01	audio	RTP/AVP	0 and/or 8	AS:64	rtpmap:0 PCMU/8000 and/or rtpmap:8 PCMA/8000
VA_02	audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)
VA_03	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000

Table 8: Values of codecs for test purposes PS_AU_01 to PS_AU_04

VARIABLE	PT	Encoding	media type	clock rate	channels
VA_01	0	PCMU	A	8 000	1
VA_02	3	GSM	A	8 000	1
VA_03	4	G723	A	8 000	1
VA_04	5	DVI4	A	8 000	1
VA_05	7	LPC	A	8 000	1
VA_06	8	PCMA	A	8 000	1
VA_07	9	G722	A	8 000	1
VA_08	12	QCELP	A	8 000	1
VA_09	13	CN	A	8 000	1
VA_10	18	G729	A	8 000	1
VA_11	Dyn	G726-40	A	8 000	1
VA_12	Dyn	G726-32	A	8 000	1
VA_13	Dyn	G726-24	A	8 000	1
VA_14	Dyn	G726-16	A	8 000	1
VA_15	Dyn	G729D	A	8 000	1
VA_16	Dyn	G729E	A	8 000	1
VA_17	Dyn	GSM-EFR	A	8 000	1

6.3.1.2 Test purposes for PSTN - SIP Basic call Unsuccessful

Unsuccessful
PSTN

PS_AU_U01	**PSTN reference to: EN 300 899-1 [23], clause 2.1.1**			**NGN reference to: Q.1912.5 [51]**
TSS reference:	PSTN-SIP/Basic_call/Unsuccessful			
Selection criteria:				
Test purpose:	Ensure that when the called SIP user is busy, the calling user receives in-band information that the called user is busy.			
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)			
Comments:				
	PSTN		SUT		SIP
	Off hook				
	Dial number	→		→	INVITE
	Special tone or announcement	←		←	486 User Busy
				→	ACK
	On hook				

PS_AU_U02	**PSTN reference to: EN 300 899-1 [23], clause 2.1.1**				**NGN reference to: Q.1912.5 [51]**
TSS reference:	PSTN-SIP/Basic_call/Unsuccessful				
Selection criteria:					
Test purpose:	Ensure that when the calling user clears before answer from the called SIP user the call is cleared				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	PSTN		SUT		SIP
	Off hook				
	Dial number	→		→	INVITE
	Ringing tone			←	180 Ringing
	On hook			→	CANCEL
				←	200 OK Cancel
				←	487 Request Terminated
				→	ACK

PS_AU_U03	**PSTN reference to: EN 300 899-1 [23], clause 2.1.1**	**NGN reference to: Q.1912.5 [51]**
TSS reference:	PSTN-SIP/Basic_call/Unsuccessful/	
Selection criteria:		
Test purpose:	Ensure that when the called SIP user is alerted by not answering before timer Q.18 expires, the network initiate call clearing.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PS_XX_U04	**PSTN reference to: EN 300 899-1 [23], clause 2.1.1**	**NGN reference to: Q.1912.5 [51], RFC 3261 [28], RFC 4566 [25]**
TSS reference:	PSTN-SIP/Basic_call/Unsuccessful	
Selection criteria:		
Test purpose:	Ensure that when there is no answer from the called user (there is no response from INVITE messages), the network initiate call clearing to the calling user.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PS_XX_U05	**PSTN reference to:** **EN 300 899-1 [23], clause 2.1.1**	**NGN reference to:** **RFC 3261 [28] and RFC 4566 [25]**
TSS reference:	PSTN-SIP/Basic_call/Unsuccessful	
Selection criteria:		
Test purpose:	Ensure that the call will be released when the called number is incomplete. The SIP network initiates call clearing 484 Address Incomplete message.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

6.3.2 Test purposes for PSTN - SIP Supplementary services

6.3.2.1 CLIP

PS_XXSSCLIP01	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**			**NGN reference to:** **Q.1912.5 [51], clause 7.1.3** **EN 383 001 [49], clause 7.1.3** **ES 283 027 [48], clause 7.2.3.2.2.3**	
TSS reference:	ISDN-SIP/Supplementary_services/CLIP				
Selection criteria:	The called user is provided with CLIP				
Test purpose:	Ensure that when the Calling party number is provided by the network , the Calling party number information elements is correctly delivered to the called (served) with following mapping rules: The SIP P-Asserted-Identity header is Derived from Calling party information element Address Signal The Calling Party Number is mapped into the SIP From header				
PSTN Parameter: values:					
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	PSTN		SUT		UA S
	Off hook, Dial number			→	INVITE
	Ringing tone			←	180 Ringing
	Connection			←	200 OK INVITE
				→	ACK
			Conversation		
	On hook			→	BYE
				←	200 OK BYE

	Display_Options_PA	**Display name**
VA_1	"display-name" is supported in the P-Asserted header	Presented
VA_2	"display-name" is not supported in the P-Asserted header	Not presented
VA_3	"display-name" is supported in the From header	Presented
VA_4	"display-name" is not supported in the From header	Not presented

6.3.2.2 CLIR

PS_XXSSCLIR01	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**				**NGN reference to:** **Q.1912.5 [51], clause 7.1.3** **EN 383 001 [49], clause 7.1.3** **ES 283 027 [48], clause 7.2.3.2.2.3** **RFC 3323 [33]** **RFC 3325 [34]**
TSS reference:	ISDN-SIP/Supplementary_services/CLIR				
Selection criteria:	The calling user is provided with CLIR permanent mode subscription				
Test purpose:	Ensure that when the Calling party number is provided by the network, Sends a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received				
PSTN Parameter: values:					
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	PSTN		SUT		UA S
	Off hook, Dial number			→	INVITE
	Ringing tone			←	180 Ringing
	Connection			←	200 OK INVITE
				→	ACK
			Conversation		
	On hook			→	BYE
				←	200 OK BYE

PS_XXSSCLIR02	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**				**NGN reference to:** **Q.1912.5 [51], clause 7.1.3** **EN 383 001 [49], clause 7.1.3** **ES 283 027 [48], clause 7.2.3.2.2.3** **RFC 3323 [33]** **RFC 3325 [34]**
TSS reference:	ISDN-SIP/Supplementary_services/CLIR				
Selection criteria:	The calling user is provided with CLIR temporary mode subscription				
Test purpose:	Ensure that when the Calling party number is provided by the network, Sends a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received				
PSTN Parameter: values:					
SIP Parameter values:	a = line (PIXIT) b = line (PIXIT) m = line (PIXIT) Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition				
Comments:					
	PSTN		SUT		UA S
	Off hook, Dial number			→	INVITE
	Ringing tone			←	180 Ringing
	Connection			←	200 OK INVITE
				→	ACK
			Conversation		
	On hook			→	BYE
				←	200 OK BYE

6.3.2.3 CFU

PSP_XXSSCFU 01	**PSTN reference to:**	**NGN reference to:** **Q.1912.5 [51] annex B.6.**
TSS reference:	PSTN-SIP-PSTN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C.	
PSTN Parameter: values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSP_XXSSCFU 02	**PSTN reference to:**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	PSTN-SIP-PSTN/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU	
Selection criteria:	User B has activated the CALL DEFLECTION service Call forwarding unconditional supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C.	
PSTN Parameter: values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSP_XXSSCFU 03	**PSTN reference to:**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	PSTN-SIP-PSTN/Supplementary_services/CFU	
Configuration:	The user B is in network N2 and is provided with CFU	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported user C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFU was not successful**. User A calls user B, the call is forwarded to user C who is busy.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSP_XXSSCFU 04	PSTN reference to:	NGN reference to: TS 183 004 [45]
TSS reference:	PSTN-SIP-PSTN/Supplementary_services/CFU	
Configuration:	The user B is in network N2 and is provided with CFU	
Selection criteria:	Call forwarding unconditional supported user C is network determined user busy User B has activated the CALL DEFLECTION service	
Test purpose:	To verify that a call is released correctly if CFU was not successful. User A calls user B, the call is forwarded to user C who is busy.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSPP_XXSSCFU 05	PSTN reference to:	NGN reference to: TS 183 004 [45]
TSS reference:	PSTN-SIP-PSTN-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFU. User D forwards the call to back to user B.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C and D. User D forwards the call to back to user B. Ensure that the call is released.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSPP_XXSSCFU 06	**PSTN reference to:**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	PSTN-SIP-PSTN-ISDN/Supplementary_services/CFU	
Configuration:	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFU. User D forwards the call to back to user B.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported Network option: hop counter supported N<5	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C and D. User D forwards the call to back to user B. Ensure that the call is released.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSS_XXSSCFU 07	**PSTN reference to:**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFU	
Configuration:	The user B is provided with CFU	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSS_XXSSCFU 08	PSTN reference to:	NGN reference to: TS 183 004 [45]
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFU	
Configuration:	The user B is provided with CFU ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = Yes).	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C which is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSS_XXSSCFU 09	PSTN reference to:	NGN reference to: TS 183 004 [45]
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFU	
Configuration:	The user B is provided with CFU ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = No).	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C which is informed of the forwarding number (user B has presentation allowed). User B is not notified of call diversion.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSS_XXSSCFU 10	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFU	
Configuration:	User B is provided with CFU ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no).	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C which is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSS_XXSSCFU 11	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFU	
Configuration:	The user B is in network N2 and is provided with CFU	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	To verify that a call is released correctly if **CFU was not successful**. User A calls user B, the call is forwarded to user C who is busy.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSS_XXSSCFU 12	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFU	
Configuration:	The user B is in network N2 and is provided with CFU	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported User B has activated the CALL DEFLECTION service	
Test purpose:	To verify that a call is released correctly if CFU **was not successful**. User A calls user B, the call is forwarded to user C who is **busy**.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSSP_XXSSCFU 13	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP-PSTN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU. User D forwards the call to back to user B.	
Selection criteria:	Call forwarding busy by the network Call forwarding busy supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C and D. User D forwards the call to back to user B. Ensure that the call is released.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSSP_XXSSCFB 14	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP-PSTN/Supplementary_services/CFU	
Selection criteria:	The user is A in network N1. The user B and the user C are in network N2. User B is provided with CFU. User E forwards the call to back to user B. Network option: hop counter supported N<5	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, C to D. User D forwards the call to back to user B. User D forwards the call to back to user B. Ensure that the call is released.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

6.3.2.4 CFB

PSP_XXSSCFB 01	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-PSTN/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFB	
Selection criteria:	Call forwarding by the network Call forwarding busy supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C.	
PSTN Parameter: values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSP_XXSSCFB 02	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-PSTN/Supplementary_services/CFB	
Configuration:	The user B is in network N2 and is provided with CFB	
Selection criteria:	Call forwarding by the network Call forwarding busy supported user C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFB was not successful**. User A calls user B, the call is forwarded to user C who is busy.	
PSTN Parameter values:		
SIP Parameter values:	a = line (PIXIT) b = line (PIXIT) m = line (PIXIT) Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition	
Comments:		

PSS_XXSSCFB 03	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFB	
Configuration:	The user B is provided with CFB	
Selection criteria:	Call forwarding by the network Call forwarding busy supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSS_XXSSCFB 04	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFB	
Configuration:	The user B is provided with CFB ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = Yes).	
Selection criteria:	Call forwarding by the network Call forwarding busy supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C which is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSS_XXSSCFB 05	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFB	
Configuration:	The user B is provided with CFB ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes, "served user receives notification that the call has been forwarded" = No).	
Selection criteria:	Call forwarding by the network Call forwarding busy supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C which is informed of the forwarding number (user B has presentation allowed). User B is not notified of call diversion.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSS_XXSSCFB 06	**PSTN reference to:**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFB	
Configuration:	User B is provided with CFB ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = No, "served user receives notification that the call has been forwarded" = no).	
Selection criteria:	Call forwarding by the network Call forwarding busy supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C which is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSS_XXSSCFB 07	**PSTN reference to:**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFB	
Configuration:	The user B is in network N2 and is provided with CFB	
Selection criteria:	Call forwarding by the network Call forwarding busy supported	
Test purpose:	To verify that a call is released correctly if **CFB was not successful**. User A calls user B, the call is forwarded to user C who is user determined user busy.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSSP_XXSSCFB 08	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU. User D forwards the call to back to user B.	
Selection criteria:	Call forwarding busy by the network Call forwarding busy supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C and D. User D forwards the call to back to user B. Ensure that the call is released.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSSP_XXSSCFB 09	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP-PSTN/Supplementary_services/CFU	
Selection criteria:	The user is A in network N1. The user B and the user C are in network N2. User B is provided with CFU. User E forwards the call to back to user B. Network option: hop counter supported N<5	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, C to D. User D forwards the call to back to user B. User D forwards the call to back to user B. Ensure that the call is released.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

6.3.2.5 CFNR

PSP_XXSSCFNR01	**PSTN reference to: EN 300 403-1 [i.3], clauses 9.2.2, 9.2.4.4, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-PSTN/Supplementary_services/CFNR/	
Selection criteria:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR . Served user communication retention on invocation of diversion (forwarding or deflection) = No.	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSP_XXSSCFNR02	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-PSTN/Supplementary_services/CFNR/	
Configuration:	The user A and the user C are in network N1. The user B is provided with CFNR , Served user communication retention on invocation of diversion (forwarding or deflection) = Yes	
Selection criteria:	CFNR supported CF Notifications supported	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C The diverting user accepts the communication after sending the INVITE request, the communication path towards the diverted to user shall be released according to the rules and procedures in RFC 3261 [28].	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSP_XXSSCFNR 03	**PSTN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-PSTN/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR. Served user communication retention on invocation of diversion (forwarding or deflection) = No [Clear call to the served user on invocation of call diversion], Served user communication retention when forwarding is rejected at forwarded-to user = No action at the forwarding user .	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported user C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFNR was not successful**. User A calls user B, the call is forwarded to user C who is busy.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSP_XXSSCFNR 04	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-PSTN/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR Served user communication retention on invocation of diversion (forwarding or deflection) = Yes.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported user C is user determined user busy	
Test purpose:	User A calls user B, the call is forwarded to user C who is **network** determined user busy. The forwarding user User B continues to alert.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSP_XXSSCFNR 05	PSTN reference to:	NGN reference to: TS 183 004 [45]
TSS reference:	PSTN-SIP-PSTN/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR Served user communication retention on invocation of diversion (forwarding or deflection) = Yes.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported user C is user determined user busy	
Test purpose:	User A calls user B, the call is forwarded to user C who is **user** determined user busy. The forwarding user User B continues to alert.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSPP_XXSSCFNR 06	PSTN reference to:	NGN reference to: TS 183 004 [45]
TSS reference:	PSTN-SIP-PSTN-PSTN/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR. User D forwards the call to back to user B.	
Selection criteria:	Call forwarding by the network CFNR supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C and D. User D forwards the call to back to user B. Ensure that the call is released.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSS_XXSSCFNR 07	**PSTN reference to:** **EN 300 207-1 [i.5]** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	The user B is provided with CFNR. Served user communication retention on invocation of diversion (forwarding or deflection) = No	
Selection criteria:	Call forwarding by the network CFNR supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSS_XXSSCFNR 08	**PSTN reference to:**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	The user B is provided with CFNR. Served user communication retention on invocation of diversion (forwarding or deflection) = Yes	
Selection criteria:	Call forwarding by the network CFNR supported	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C The diverting user accepts the communication after sending the INVITE request, the communication path towards the diverted to user shall be released according to the rules and procedures in RFC 3261 [28].	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSS_XXSSCFNR 09	**PSTN reference to:**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	The user B is provided with CFNR	
Selection criteria:	User B has activated the CALL DEFLECTION service CFNR supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSS_XXSSCFNR 10	**PSTN reference to:**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR Served user communication retention on invocation of diversion (forwarding or deflection) = No [Clear call to the served user on invocation of call diversion], Served user communication retention when forwarding is rejected at forwarded-to user = No action at the forwarding user).	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	To verify that a call is released correctly if **CFNR was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSS_XXSSCFNR 11	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR Served user communication retention on invocation of diversion (forwarding or deflection) = No [Clear call to the served user on invocation of call diversion], Served user communication retention when forwarding is rejected at forwarded-to user = No action at the forwarding user).	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	To verify that a call is released correctly if CFNR **was not successful**. User A calls user B, the call is forwarded to user C who is **network determined user busy**.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSS_XXSSCFNR 12	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR. Served user communication retention on invocation of diversion (forwarding or deflection) = Yes.	
Selection criteria:	Call forwarding by the network CFNR supported	
Test purpose:	User A calls user B, the call is forwarded to user C who is **user** determined user busy. The forwarding user User B continues to alert.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSS_XXSSCFNR 13	PSTN reference to:	NGN reference to: TS 183 004 [45]
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. Served user communication retention on invocation of diversion (forwarding or deflection) = Yes.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	User A calls user B, the call is forwarded to user C who is **network** determined user busy. The forwarding user User B continues to alert.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

PSPP_XXSSCFNR 14	PSTN reference to:	NGN reference to: TS 183 004 [45]
TSS reference:	PSTN-SIP-PSTN-PSTN/Supplementary_services/CFNR	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR. User D forwards the call to back to user B.	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C and D. User D forwards the call to back to user B. Ensure that the call is released.	
PSTN Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

6.3.2.6 CFNL

PSP_XXSSCFNL 01	PSTN reference to:	NGN reference to: TS 183 004 [45]
TSS reference:	PSTN-SIP-PSTN/Supplementary_services/CFNL	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNL	
Selection criteria:	Call forwarding by the network CFNL supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C.	
PSTN Parameter values:		
SIP Parameter values:		
Comments:		

PSP_XXSSCFNL 02	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-PSTN/Supplementary_services/CFNL	
Configuration:	The user B is in network N2 and is provided with CFNL	
Selection criteria:	Call forwarding by the network CFNL supported user C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFNL was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
PSTN Parameter values:		
SIP Parameter values:		
Comments:		

PSP_XXSSCFNL 03	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-PSTN/Supplementary_services/CFNL	
Configuration:	The user B is in network N2 and is provided with CFNL	
Selection criteria:	Call forwarding by the network CFNL supported user C is network determined user busy	
Test purpose:	To verify that a call is released correctly if CFNL was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy. User A is notified of call diversion and user C is informed of the forwarding number (user B has presentation allowed).	
PSTN Parameter values:		
SIP Parameter values:		
Comments		

PSS_XXSSCFNL 04	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFNL	
Configuration:	The user B is provided with CFNL	
Selection criteria:	Call forwarding by the network CFNL supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C.	
PSTN Parameter values:		
SIP Parameter values:	a = line (PIXIT) b = line (PIXIT) m = line (PIXIT) Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition	
Comments:		

PSS_XXSSCFNL 05	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFNL	
Configuration:	The user B is in network N2 and is provided with CFNL	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	To verify that a call is released correctly if **CFNL was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
PSTN Parameter values:		
SIP Parameter values:	a = line (PIXIT) b = line (PIXIT) m = line (PIXIT) Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition	
Comments:		

PSS_XXSSCFNL 06	**PSTN reference to: EN 300 207-1 [i.5], clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	PSTN-SIP-SIP/Supplementary_services/CFNL	
Configuration:	The user B is in network N2 and is provided with CFNL	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	To verify that a call is released correctly if CFNL **was not successful**. User A calls user B, the call is forwarded to user C who is **network determined user busy**.	
PSTN Parameter values:		
SIP Parameter values:	a = line (PIXIT) b = line (PIXIT) m = line (PIXIT) Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition	
Comments:		

6.4 Test purposes for SIP-PSTN

6.4.1 Test purposes for SIP-PSTN, Basic call

6.4.1.1 Test purposes for SIP-PSTN, Basic call, Successful

SP_AU_01	**PSTN reference to:** **EN 300 899-1 [23],** **clause 3.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3**
TSS reference:	SIP-PSTN /Basic_call/Successful/3,1 kHz audio	
Selection criteria:		
Test purpose:	Ensure that call establishment and the mapping of the parameters defined in table 9 between INVITE message and the PSTN is performed correctly. Ensure that in the confirmed state the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	a = line (PIXIT) b = line (PIXIT) m = line (PIXIT) Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition	
Comments:	In the Request-URI a sip: URI with the user=phone parameter, and the user info part of that URI is an E.164 number encoded as specified by the telephone-subscriber rule of RFC 3966 [26].	

SP_AU_02	**PSTN reference to:** **EN 300 899-1 [23],** **clause 3.1.1**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3**
TSS reference:	SIP-PSTN/Basic_call/Successful/3,1 kHz audio	
Selection criteria:	FAX G3	
Test purpose:	Ensure that call establishment and the mapping of the defined parameters between INVITE message and PSTN is performed correctly Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line Based on T.38. b = line AS: 64 m = line: udptl; T38	
Comments:	In the Request-URI a sip: URI with the user=phone parameter, and the user info part of that URI is an E.164 number encoded as specified by the telephone-subscriber rule of RFC 3966 [26].	

SP_AU_03	PSTN reference to: EN 300 899-1 [23], clause 3.1.1	NGN reference to: Q.1912.5 [51], clause 6.11
TSS reference:	SIP-PSTN/Basic_call/Successful/3,1 kHz audio	
Selection criteria:		
Test purpose:	Ensure that the call establishment and the call clearing procedure are performed correctly when the **calling user** clears after answering with a BYE message. Ensure that in the confirmed state the transfer of tone or announcement on the media channel is performed correctly. In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in table 10 applies.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:	In the Request-URI a sip: URI with the user=phone parameter, and the user info part of that URI is an E.164 number encoded as specified by the telephone-subscriber rule of RFC 3966 [26].	

SP_AU_04	PSTN reference to: EN 300 899-1 [23], clause 3.1.1	NGN reference to: Q.1912.5 [51], clause 6.11
TSS reference:	SIP-PSTN/Basic_call/Successful/3,1 kHz audio	
Selection criteria:		
Test purpose:	Ensure that the call clearing procedure is performed correctly when the **called user** clears after answering The calling user shall receive a BYE message.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:	In the Request-URI a sip: URI with the user=phone parameter, and the user info part of that URI is an E.164 number encoded as specified by the telephone-subscriber rule of RFC 3966 [26].	

Table 9: Values for test purposes SP_AU_01 to SP_AU_04

m= line				b= line	a= line
VA	**<media>**	**<transport>**	**<fmt-list>**	**<modifier>: <bandwidth-value> (see note)**	**rtpmap:<dynamic-PT> <encoding name>/ <clock rate>[/encoding parameters>**
VA_01	audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A
VA_02	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMU/8000
VA_03	audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A
VA_04	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMA/8000
VA_05	image	Udptl	t38	N/A or up to 64 kbit/s	Based on T.38
VA_06	image	Tcptl	t38	N/A or up to 64 kbit/s	Based on T.38
NOTE: <bandwidth value> for <modifier> of AS is evaluated to be B kbit/s.					

Table 10: Values for test purposes SP_AU_01 and SP_AU_04

VARIABLE	PT	Encoding	media type	clock rate	channels
VA_01	0	PCMU	A	8,000	1
VA_02	3	GSM	A	8,000	1
VA_03	4	G723	A	8,000	1
VA_04	5	DVI4	A	8,000	1
VA_05	7	LPC	A	8,000	1
VA_06	8	PCMA	A	8,000	1
VA_07	9	G722	A	8,000	1
VA_08	12	QCELP	A	8,000	1
VA_09	13	CN	A	8,000	1
VA_10	18	G729	A	8,000	1
VA_11	Dyn	G726-40	A	8,000	1
VA_12	Dyn	G726-32	A	8,000	1
VA_13	Dyn	G726-24	A	8,000	1
VA_14	Dyn	G726-16	A	8,000	1
VA_15	Dyn	G729D	A	8,000	1
VA_16	Dyn	G729E	A	8,000	1
VA_17	Dyn	GSM-EFR	A	8,000	1

6.4.1.2 Test purposes for SIP-PSTN, Basic call, Unsuccessful

Unsuccessful

SP_XX_U01		**NGN reference to: Q.1912.5 [51]**
TSS reference:	SIP-PSTN/Basic_call/Unsuccessful	
Selection criteria:		
Test purpose:	Ensure that, when calling to **unallocated number**, the network initiate call clearing to the calling user with a 404Not Found message. If the Reason Header field is implemented the cause value #1 should be set in the Reason Header field.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:	In the Request-URI a sip: URI with the user=phone parameter, and the user info part of that URI is an E.164 number encoded as specified by the telephone-subscriber rule of RFC 3966 [26].	

SP_XX_U02		**NGN reference to:** **Q.1912.5 [51]**
TSS reference:	SIP-PSTN/Basic_call/Unsuccessful	
Selection criteria:		
Test purpose:	Ensure that the call will be released when there is **no route to destination**. The network initiates call clearing to the calling user with a 503 Service unavailable message. If the Reason Header field is implemented the cause value # 3 should be set in the Reason Header field.	
SIP Parameter values:	a = line (PIXIT) b = line (PIXIT) m = line (PIXIT) Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition	
Comments:	In the Request-URI a sip: URI with the user=phone parameter, and the user info part of that URI is an E.164 number encoded as specified by the telephone-subscriber rule of RFC 3966 [26].	

SP_XX_U03		**NGN reference to:** **Q.1912.5 [51]**
TSS reference:	SIP-PSTN/Basic_call/Unsuccessful	
Selection criteria:		
Test purpose:	Ensure that, when the **called user is busy** the network initiates call clearing to the calling user with a 486 Busy Here message.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:	In the Request-URI a sip: URI with the user=phone parameter, and the user info part of that URI is an E.164 number encoded as specified by the telephone-subscriber rule of RFC 3966 [26].	

SP_XX_U04		**NGN reference to:** **Q.1912.5 [51]**
TSS reference:	SIP-PSTN/Basic_call/Unsuccessful	
Selection criteria:		
Test purpose:	Ensure that when there is no answer from the called user (but user alerted), the network initiate call clearing to the calling user with a CANCEL or 408 Request Timeout message.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:	In the Request-URI a sip: URI with the user=phone parameter, and the user info part of that URI is an E.164 number encoded as specified by the telephone-subscriber rule of RFC 3966 [26].	

SP_XX_U05		**NGN reference to: Q.1912.5 [51]**
TSS reference:	SIP-PSTN/Basic_call/Unsuccessful	
Selection criteria:		
Test purpose:	Ensure that the call will be released when the called number is incomplete. The network initiates call clearing to the calling user with final response code in SIP_MESSAGE_VA. The cause value should be mapped to the Reason Header field.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:	In the Request-URI a sip: URI with the user=phone parameter, and the user info part of that URI is an E.164 number encoded as specified by the telephone-subscriber rule of RFC 3966 [26].	

Test case variable for test case SP_XX_U05		
	SIP_MESSAGE_VA	Unsuccessful destination (Address incomplete)
VA_1	404 Not Found	"Unassigned (unallocated) number"
VA_2	503 Service unavailable	"No route to destination"
VA_3	410 Gone	"Number changed"
VA_4	484 Address Incomplete	"Invalid number format (incomplete number")

SP_XX_U06		**NGN reference to: Q.1912.5 [51]**
TSS reference:	SIP-PSTN/Basic_call/Unsuccessful	
Selection criteria:		
Test purpose:	Ensure that the call will be released when the calling user clears the call with a CANCEL or BYE before answer from called user.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:	In the Request-URI a sip: URI with the user=phone parameter, and the user info part of that URI is an E.164 number encoded as specified by the telephone-subscriber rule of RFC 3966 [26].	

Table 11: Values for test purposes SP_XX_U01 to SP_XX_U06

m= line				**b= line**	**a= line**
VA	**<media>**	**<transport>**	**<fmt-list>**	**<modifier>: <bandwidth-value> (see note)**	**rtpmap:<dynamic-PT> <encoding name>/ <clock rate>[/encoding parameters>**
VA_01	audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A
VA_02	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMU/8000
VA_03	audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A
VA_04	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMA/8000
VA_05	image	Udptl	t38	N/A or up to 64 kbit/s	Based on T.38
VA_06	image	Tcptl	t38	N/A or up to 64 kbit/s	Based on T.38
NOTE: <bandwidth value> for <modifier> of AS is evaluated to be B kbit/s.					

Table 12: Values for test purposes SP_XX_U01 to SP_XX_U06

VARIABLE	PT	Encoding	media type	clock rate	channels
VA_01	0	PCMU	A	8 000	1
VA_02	3	GSM	A	8 000	1
VA_03	4	G723	A	8 000	1
VA_04	5	DVI4	A	8 000	1
VA_05	7	LPC	A	8 000	1
VA_06	8	PCMA	A	8 000	1
VA_07	9	G722	A	8 000	1
VA_08	12	QCELP	A	8 000	1
VA_09	13	CN	A	8 000	1
VA_10	18	G729	A	8 000	1
VA_11	Dyn	G726-40	A	8 000	1
VA_12	Dyn	G726-32	A	8 000	1
VA_13	Dyn	G726-24	A	8 000	1
VA_14	Dyn	G726-16	A	8 000	1
VA_15	Dyn	G729D	A	8 000	1
VA_16	Dyn	G729E	A	8 000	1
VA_17	Dyn	GSM-EFR	A	8 000	1

6.4.2 Test purposes for SIP - PSTN Supplementary services

6.4.2.1 OIP/ CLIP

<table>
<tr><th>SP_XXSSOIP01</th><th colspan="3">PSTN reference to:
EN 300 001 [i.12],
ETS 300 648 [i.13],
EN 300 659 [i.15]</th><th colspan="2">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6</th></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
the SIP P-Preferred-Identity containing a SIP URI (PIXIT) with an identity in the format of a tel URI has not been received
the priv-value component is set to "none"
the SIP From header field containing a URI with an identity in the format of a tel URI has not been received
the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served)</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>PSTN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td></td><td>Ringing</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td></td><td>Off Hook</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td></td><td>On hook</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td></td><td></td></tr>
</table>

SP_XXSSOIP02	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" No priv value is sent Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has not** been received • no priv value is received • the SIP From header field containing a URI with an identity in the format of a tel URI has been received • the Calling Party Number mapped from the From Header is correctly delivered to the called (served)	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP03	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" No priv value is sent No special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has not** been received • no priv value is received • the SIP From header field containing a URI with an identity in the format of a tel URI has been received • the Calling Party Number mapped from the P-Asserted ID is correctly delivered to the called (served) user.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

<table>
<tr><td>SP_XXSSOIP04</td><td>PSTN reference to:
EN 300 001 [i.12],
ETS 300 648 [i.13],
EN 300 659 [i.15]</td><td colspan="4">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6</td></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "not restricted"
No priv value is sent</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
the SIP P-Preferred-Identity containing a SIP URI (PIXIT) with an identity in the format of a tel URI has not been received
no priv value is received
the SIP From header field containing a URI with an identity in the format of a tel URI has not been received
the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>PSTN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td></td><td>Ringing</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td></td><td>Off Hook</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td></td><td>On hook</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td></td><td></td></tr>
</table>

SP_XXSSOIP05	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • the priv-value component is set to "none" • the Calling Party Number mapped from the From Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	

Comments:					
	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

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EN 300 001 [i.12],
ETS 300 648 [i.13],
EN 300 659 [i.15]</th><th colspan="5">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6</th></tr>
<tr><td>TSS reference:</td><td colspan="6">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="6">The user subscribes OIR "temporary mode" default "not restricted"
No Special arrangement applies</td></tr>
<tr><td>Test purpose:</td><td colspan="6">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has been received
the priv-value component is set to "none"
the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="6">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="6"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>PSTN</td><td></td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td></td><td>Ringing</td><td></td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td></td><td>Off Hook</td><td></td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td></td><td>On hook</td><td></td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td></td><td></td><td></td></tr>
</table>

SP_XXSSOIP07	**PSTN reference to: EN 300 001 [i.12], ETS 300 648 [i.13], EN 300 659 [i.15]**	**NGN reference to: Q.1912.5 [51], clause 6.1.3.6 EN 383 001 [49], clause 6.1.3.6 ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • no priv value is received • the Calling Party Number mapped from the From Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP08	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" No Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • no priv value is received • the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP09	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI **has not** been received • the priv-value component is set to "none" • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • the Calling Party Number mapped from the From Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP10	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" No Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI **has not** been received • the priv-value component is set to "none" • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP11	**PSTN reference to: EN 300 001 [i.12], ETS 300 648 [i.13], EN 300 659 [i.15]**	**NGN reference to: Q.1912.5 [51], clause 6.1.3.6 EN 383 001 [49], clause 6.1.3.6 ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI **has not** been received • No priv value is received • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • the Calling Party Number mapped from the From Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP12	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" No Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI **has not** been received • no priv value is received • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • the Calling Party Number mapped from the content of the P-Asserted Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SIP		SUT		PSTN
INVITE	→			Ringing
180 Ringing	←			Off Hook
200 OK INVITE	←			
ACK	→			
BYE	→			On hook
200 OK BYE	←			

SP_XXSSOIP 13	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted"	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received • the priv-value component is set to "none" • the SIP From header field containing a SIP URI (PIXIT) not in the format of a tel URI **has not been** received • the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP14	**PSTN reference to: EN 300 001 [i.12], ETS 300 648 [i.13], EN 300 659 [i.15]**	**NGN reference to: Q.1912.5 [51], clause 6.1.3.6 EN 383 001 [49], clause 6.1.3.6 ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted"	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received: • no priv value is received • the SIP From header field containing a SIP URI (PIXIT) in the format of a tel URI **has not been** received • the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP15	PSTN reference to: EN 300 001 [i.12], ETS 300 648 [i.13], EN 300 659 [i.15]	NGN reference to: Q.1912.5 [51], clause 6.1.3.6 EN 383 001 [49], clause 6.1.3.6 ES 283 027 [48], clause 7.2.3.1.2.6
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received • the priv-value component is set to "none" • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has been received • the Calling Party Number mapped from the From Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP16	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" No Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received • the priv-value component is set to "none" • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has been received • the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP17	**PSTN reference to: EN 300 001 [i.12], ETS 300 648 [i.13], EN 300 659 [i.15]**	**NGN reference to: Q.1912.5 [51], clause 6.1.3.6 EN 383 001 [49], clause 6.1.3.6 ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a URI (PIXIT) in the format of a tel URI has been received • no priv value is received • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received the Calling Party Number mapped from the From Header is correctly delivered to the called (served) user.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

<table>
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EN 300 001 [i.12],
ETS 300 648 [i.13],
EN 300 659 [i.15]</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6</th></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "not restricted"
No Special arrangement applies</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a URI (PIXIT) in the format of a tel URI has been received
• no priv value is received
• the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has been received and
• the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>PSTN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td></td><td>Ringing</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td></td><td>Off Hook</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td></td><td>On hook</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td></td><td></td></tr>
</table>

SP_XXSSOIP18	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • the priv-value component is set to "none" • the Calling Party Number mapped from the From Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP19	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" No Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • the priv-value component is set to "none" • the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP20	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • no priv value is received • the Calling Party mapped from the set to the content of the From Header Number is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP21	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted" No Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has been received • no priv value is received • the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP22	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted"	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI **has not** been received • the priv-value component is set to "none" • the SIP From header is set to **anonymous** • the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP23	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted"	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI **has not been** received • no priv value is received • the SIP From header is set to **anonymous** • the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

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EN 300 001 [i.12],
ETS 300 648 [i.13],
EN 300 659 [i.15]</td><td colspan="2">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6</td></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "restricted"
Special arrangement applies</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
the SIP P-Preferred-Identity containing a SIP URI (PIXIT) with an identity in the format of a tel URI has not been received
the priv-value component is set to "none"
the SIP From header field containing a SIP URI (PIXIT) not in the format of a tel URI has been received
the Calling Party Number set to the content of the From Header is correctly delivered to the called (served) user</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>PSTN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td></td><td>Ringing</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td></td><td>Off Hook</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td></td><td>On hook</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td></td><td></td></tr>
</table>

SP_XXSSOIP25	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "restricted" No Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has not** been received • the priv-value component is set to "none" • the SIP From header field containing a SIP URI (PIXIT) not in the format of a tel URI **has been** received • the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP26	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**		**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**		
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks				
Selection criteria:	The user subscribes OIR "temporary mode" default "restricted"				
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has not** been received • the priv-value component is set to "none" • the SIP From header field containing a SIP URI (PIXIT) not in the format of a tel URI **has not been** received • the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP27	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "restricted" Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • the priv-value component is set to "none" • the Calling Party Number mapped from the From Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	

Comments:					
	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP28	**PSTN reference to: EN 300 001 [i.12], ETS 300 648 [i.13], EN 300 659 [i.15]**				**NGN reference to: Q.1912.5 [51], clause 6.1.3.6 EN 383 001 [49], clause 6.1.3.6 ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks				
Selection criteria:	The user subscribes OIR "temporary mode" default "restricted" No Special arrangement applies				
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • the priv-value component is set to "none" • the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP29	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "restricted" Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI **has** not been received • the priv-value component is set to "none" • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received and • the Calling Party Number mapped to the From Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP30	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "restricted" No Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI **has** not been received • the priv-value component is set to "none" • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received and • the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

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EN 300 001 [i.12],
ETS 300 648 [i.13],
EN 300 659 [i.15]</td><td colspan="4">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6</td></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "restricted"</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received
the priv-value component is set to "none"
the SIP From header field containing a SIP URI (PIXIT) not in the format of a tel URI has not been received
the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>PSTN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td></td><td>Ringing</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td></td><td>Off Hook</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td></td><td>On hook</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td></td><td></td></tr>
</table>

<table>
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EN 300 001 [i.12],
ETS 300 648 [i.13],
EN 300 659 [i.15]</td><td colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6</td></tr>
<tr><td>TSS reference:</td><td colspan="6">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="6">The user subscribes OIR "temporary mode" default "restricted"
Special arrangement applies</td></tr>
<tr><td>Test purpose:</td><td colspan="6">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received:
• the priv-value component is set to "none"
• the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has been received:
• the Calling Party Number mapped from the From Header is correctly delivered to the called (served) user</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="6">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="6"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td colspan="2">PSTN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td></td><td colspan="2">Ringing</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td></td><td colspan="2">Off Hook</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td></td><td colspan="2"></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td></td><td colspan="2">On hook</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td></td><td colspan="2"></td></tr>
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SP_XXSSOIP33	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "restricted" No Special arrangement applies	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received • the priv-value component is set to "none" • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI has been received • the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP34	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**			**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks			
Selection criteria:	The user subscribes OIR "temporary mode" default "restricted"			
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI **has not been** received • the priv-value component is set to "none" • the SIP From header is set to **anonymous** • the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user			
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)			
Comments:				

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP35	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted"	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received • the priv-value component is set to "none" • the SIP From header is set to **anonymous** • the Calling Party Number set to the content of the P-Asserted-Identity Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	

Comments:					
	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIP36	**PSTN reference to: EN 300 001 [i.12], ETS 300 648 [i.13], EN 300 659 [i.15]**	**NGN reference to: Q.1912.5 [51], clause 6.1.3.6 EN 383 001 [49], clause 6.1.3.6 ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted"	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received • No priv value is received • the SIP From header is set to **anonymous** • the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

<table>
<tr><th>SP_XXSSOIP37</th><th colspan="3">PSTN reference to:
EN 300 001 [i.12],
ETS 300 648 [i.13],
EN 300 659 [i.15]</th><th colspan="2">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6</th></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "restricted"</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received
the priv-value component is set to "none"
the SIP From header is set to anonymous
the Calling Party Number mapped from the P-Asserted-Identity Header is correctly delivered to the called (served) user</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>PSTN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td></td><td>Ringing</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td></td><td>Off Hook</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td></td><td>On hook</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td></td><td></td></tr>
</table>

6.4.2.2 OIR/CLIR

<table>
<tr><th>SP_XXSSOIR01</th><th colspan="3">PSTN reference to:
EN 300 001 [i.12],
ETS 300 648 [i.13],
EN 300 659 [i.15]</th><th colspan="2">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6</th></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "restricted"</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) has been received
• the priv-value component is set to "id"
• the SIP From header field containing a SIP URI (PIXIT) has been received
• the Calling Party Number is not delivered to the called (served) user</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>PSTN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td></td><td>Ringing</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td></td><td>Off Hook</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td></td><td>On hook</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><th>SP_XXSSOIR02</th><th colspan="3">PSTN reference to:
EN 300 001 [i.12],
ETS 300 648 [i.13],
EN 300 659 [i.15]</th><th colspan="2">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6</th></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "restricted"</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) has been received
• the priv-value not present
• the SIP From header field containing a SIP URI (PIXIT) has been received
• the Calling Party Number is not delivered to the called (served) user</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>PSTN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td></td><td>Ringing</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td></td><td>Off Hook</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td></td><td>On hook</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><th>SP_XXSSOIR03</th><th>PSTN reference to:
EN 300 001 [i.12],
ETS 300 648 [i.13],
EN 300 659 [i.15]</th><th colspan="3">NGN reference to:
Q.1912.5 [51], clause 6.1.3.6
EN 383 001 [49], clause 6.1.3.6
ES 283 027 [48], clause 7.2.3.1.2.6</th></tr>
<tr><td>TSS reference:</td><td colspan="5">Private Extensions to SIP for Asserted Identity within Trusted Networks</td></tr>
<tr><td>Selection criteria:</td><td colspan="5">The user subscribes OIR "temporary mode" default "restricted"</td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Preferred-Identity containing a SIP URI (PIXIT) has been received
• priv-value component is set to "id"
• the SIP From header is set to anonymous
• the Calling Party Number is not delivered to the called (served) user</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">Dial string parameters options=PIXIT

PIXIT for supported header:
Case a) no 100 rel
Case b) Supported: 100 rel
Case c) Supported: 100 rel and precondition

a = line (PIXIT)
b = line (PIXIT)
m = line (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="5"></td></tr>
<tr><td rowspan="8"></td><td>SIP</td><td></td><td>SUT</td><td></td><td>PSTN</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td></td><td>Ringing</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td></td><td>Off Hook</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td></td><td>On hook</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td></td><td></td></tr>
</table>

SP_XXSSOIR04	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "temporary mode" default "restricted"	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) has been received • priv-value component is not present • the SIP From header is set to **anonymous** • the Calling Party Number is not delivered to the called (served) user	
SIP Parameter values:	a = line (PIXIT) b = line (PIXIT) m = line (PIXIT) Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition SIP URI PIXIT: Case a) sip: local-number-digits; phone-context=nat @hostportion; user=phone Case b) sip: global -number-digits @hostportion; user=phone	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIR05	**PSTN reference to: EN 300 001 [i.12], ETS 300 648 [i.13], EN 300 659 [i.15]**	**NGN reference to: Q.1912.5 [51], clause 6.1.3.6 EN 383 001 [49], clause 6.1.3.6 ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR temporary mode" default "not restricted	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) **has been** received • the priv-value component is set to "id" • the SIP From header field containing a SIP URI (PIXIT) **has been** received • the Calling Party Number is not delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SIP		SUT		PSTN
INVITE	→			Ringing
180 Ringing	←			Off Hook
200 OK INVITE	←			
ACK	→			
BYE	→			On hook
200 OK BYE	←			

SP_XXSSOIR06	**PSTN reference to: EN 300 001 [i.12], ETS 300 648 [i.13], EN 300 659 [i.15]**	**NGN reference to: Q.1912.5 [51], clause 6.1.3.6 EN 383 001 [49], clause 6.1.3.6 ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR temporary mode" default "not restricted	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) has been received • priv-value component is set to "id" • the SIP From header is set to **anonymous** • the Calling Party Number is not delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SIP		SUT		PSTN
INVITE	→			Ringing
180 Ringing	←			Off Hook
200 OK INVITE	←			
ACK	→			
BYE	→			On hook
200 OK BYE	←			

SP_XXSSOIR07	**PSTN reference to: EN 300 001 [i.12], ETS 300 648 [i.13], EN 300 659 [i.15]**	**NGN reference to: Q.1912.5 [51], clause 6.1.3.6 EN 383 001 [49], clause 6.1.3.6 ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR permanent mode	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) **has been** received • the priv-value component is set to "id" • the SIP From header field containing a SIP URI (PIXIT) **has been** received • the Calling Party Number is not delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIR08	**PSTN reference to: EN 300 001 [i.12], ETS 300 648 [i.13], EN 300 659 [i.15]**	**NGN reference to: Q.1912.5 [51], clause 6.1.3.6 EN 383 001 [49], clause 6.1.3.6 ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR permanent mode	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) **has been** received • the priv-value component is not present • the SIP From header field containing a SIP URI (PIXIT) **has been** received • the Calling Party Number is not delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SI_XXSSOIR09	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**				**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks				
Selection criteria:	The user subscribes OIR permanent mode				
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) has been received • priv-value component is set to "id" • the SIP From header is set to **anonymous** • the Calling Party Number is not delivered to the called (served) user				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIR10	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**				**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks				
Selection criteria:	The user subscribes OIR permanent mode				
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) has been received • priv-value component is not present • the SIP From header is set to **anonymous** • the Calling Party Number is not delivered to the called (served) user				
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)				
Comments:					
	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIR11	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	SIP URI or SIPS URI is used in the P-Preferred-Identity The user subscribes OIR "permanent mode"	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has not** been received • the priv-value component is set to "none" • the SIP From header field containing a SIP URI (PIXIT) not in the format of a tel URI **has been** received • the Calling Party Number is not delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIR12	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "permanent mode"	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has not** been received the priv-value component is set to "none" • the SIP From header field containing a SIP URI (PIXIT) not in the format of a tel URI **has not been** received • the Calling Party Number is not delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIR13	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "permanent mode"	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • the priv-value component is set to "none" • the Calling Party Number is not delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIR14	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "permanent mode"	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received • the priv-value component is set to "none" • the SIP From header field containing a SIP URI (PIXIT) with an identity in the format of a tel URI **has been** received • the Calling Party Number is not delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

SP_XXSSOIR15	**PSTN reference to:** **EN 300 001 [i.12],** **ETS 300 648 [i.13],** **EN 300 659 [i.15]**	**NGN reference to:** **Q.1912.5 [51], clause 6.1.3.6** **EN 383 001 [49], clause 6.1.3.6** **ES 283 027 [48], clause 7.2.3.1.2.6**
TSS reference:	Private Extensions to SIP for Asserted Identity within Trusted Networks	
Selection criteria:	The user subscribes OIR "permanent mode"	
Test purpose:	Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Preferred-Identity containing a SIP URI (PIXIT) in the format of a tel URI has been received • the priv-value component is set to "none" • the SIP From header is set to **anonymous** • the Calling Party Number is not delivered to the called (served) user	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

	SIP		SUT		PSTN
	INVITE	→			Ringing
	180 Ringing	←			Off Hook
	200 OK INVITE	←			
	ACK	→			
	BYE	→			On hook
	200 OK BYE	←			

6.4.2.3 CFU

SPS_XXSSCFU 01	PSTN reference to:	NGN reference to: TS 183 004 [45]
TSS reference:	SIP-PSTN-SIP/Supplementary_services/CFU	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SPS_XXSSCFU 02	PSTN reference to:	NGN reference to: TS 183 004 [45]
TSS reference:	SIP-PSTN-SIP/Supplementary_services/CFU	
Configuration:	The user B is in network N2 and is provided with CFU	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported User C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFU was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SPS_XXSSCFU 03	PSTN reference to:	NGN reference to: TS 183 004 [45]
TSS reference:	SIP-PSTN-SIP/Supplementary_services/CFU	
Configuration:	The user B is in network N2 and is provided with CFU	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported User C is network determined user busy	
Test purpose:	To verify that a call is released correctly if CFU was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SPP_XXSSCFU 04	**PSTN reference to:**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-PSTN-PSTN/Supplementary_services/CFU	
Configuration:	The user B is provided with CFU	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SPP_XXSSCFU 05	**PSTN reference to:**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-PSTN-PSTN/Supplementary_services/CFU	
Configuration:	The user B is in network N2 and is provided with CFU	
Selection criteria:	Call forwarding by the network Call forwarding unconditional supported	
Test purpose:	To verify that a call is released correctly if **CFU was not successful**. User A calls user B, the call is forwarded to user C who is busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

6.4.2.4 CFB

SPS_XXSSCFB01	**PSTN reference to:**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-PSTN-SIP/Supplementary_services/CFB	
Configuration:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFB- (network determined).	
Selection criteria:	Call forwarding by the network	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C.	
SIP Parameter values:		
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SPS_XXSSCFB 02	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-PSTN-SIP/Supplementary_services/CFB	
Configuration:	The user B is in network N2 and is provided with CFB	
Selection criteria:	Call forwarding by the network CFB supported User C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFU was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SPS_XXSSCFB 03	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-PSTN-SIP/Supplementary_services/CFB	
Configuration:	The user B is in network N2 and is provided with CFB	
Selection criteria:	Call forwarding by the network CFB supported User C is network determined user busy	
Test purpose:	To verify that a call is released correctly if CFB was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SPP_XXSSCFB 04	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-PSTN-PSTN/Supplementary_services/CFB	
Configuration:	The user B is provided with CFB	
Selection criteria:	Call forwarding by the network Call forwarding busy	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SPP_XXSSCFB 05	PSTN reference to:	NGN reference to: TS 183 004 [45]
TSS reference:	SIP-PSTN-PSTN/Supplementary_services/CFB	
Configuration:	The user B is in network N2 and is provided with CFB	
Selection criteria:	Call forwarding by the network CFB supported User C is network determined user busy	
Test purpose:	To verify that a call is released correctly if CFB was not successful. User A calls user B, the call is forwarded to user C who is busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

6.4.2.5 CFNR

SPS_XXSSCFNR01	PSTN reference to:	NGN reference to: TS 183 004 [45]
TSS reference:	SIP-PSTN-SIP/Supplementary_services/CFNR/	
Selection criteria:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR , option B, immediate release, no notification	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SPS_XXSSCFNR02	PSTN reference to:	NGN reference to: TS 183 004 [45]
TSS reference:	SIP-PSTN-SIP/Supplementary_services/CFNR/	
Selection criteria:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR , option A, late release, no notification	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SPS_XXSSCFNR 03	**PSTN reference to:**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-PSTN-SIP/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR	
Selection criteria:	Call forwarding by the network CFNR option B, immediate release, no notification User C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFNR was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SPS_XXSSCFNR 04	**PSTN reference to:**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-PSTN-SIP/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR	
Selection criteria:	Call forwarding by the network CFNR supported option B, immediate release, no notification User C is network determined user busy	
Test purpose:	To verify that a call is released correctly if CFNR was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SPS_XXSSCFNR 05	**PSTN reference to:**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-PSTN-SIP/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR	
Selection criteria:	Call forwarding by the network CFNR option A late release, no notification User C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFNR was not successful**. User A calls user B, the call is forwarded to user C who is **user** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SPS_XXSSCFNR 06	**PSTN reference to:**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-PSTN-SIP/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR	
Selection criteria:	Call forwarding by the network CFNR supported option A, late release, no notification User C is network determined user busy	
Test purpose:	To verify that a call is released correctly if CFNR was not successful. User A calls user B, the call is forwarded to user C who is **network** determined user busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SPP_XXSSCFNR 07	**PSTN reference to: EN 300 403-1 [i.3], clauses 9.2.2, 9.2.4.4, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-PSTN-PSTN/Supplementary_services/CFNR/	
Selection criteria:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR , option B, immediate release, no notification	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SPP_XXSSCFNR 08	**PSTN reference to: EN 300 403-1 [i.3], clauses 9.2.2, 9.2.4.4, 9.2.5**	**NGN reference to: TS 183 004 [45]**
TSS reference:	SIP-PSTN-PSTN/Supplementary_services/CFNR/	
Selection criteria:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR , option A, late release, no notification	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SPP_XXSSCFNR 09	**PSTN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-PSTN-PSTN/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR	
Selection criteria:	Call forwarding by the network CFNR option B, immediate release, no notification User C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFNR was not successful**. User A calls user B, the call is forwarded to user C who is busy.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

SPP_XXSSCFNR 10	**PSTN reference to:** **EN 300 207-1 [i.5],** **clauses 6.1, 9.2.2, 9.2.5**	**NGN reference to:** **TS 183 004 [45]**
TSS reference:	SIP-PSTN-PSTN/Supplementary_services/CFNR	
Configuration:	The user B is in network N2 and is provided with CFNR	
Selection criteria:	Call forwarding by the network CFNR option A, late release, no notification User C is user determined user busy	
Test purpose:	To verify that a call is released correctly if **CFNR was not successful**. User A calls user B, the call is forwarded to user C who is busy. User B continues to alert to the forwarding user.	
SIP Parameter values:	Dial string parameters options=PIXIT PIXIT for supported header: Case a) no 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel and precondition a = line (PIXIT) b = line (PIXIT) m = line (PIXIT)	
Comments:		

6.5 ISDN-ISDN

The tests for ISDN-ISDN are identical with the tests in EG 201 299-1 [57] V2.1.1.

Annex A (informative): Bibliography

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

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History

Document history		
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