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

Testing specifications – Testing specifications for next
generation networks

**VoLTE/ViLTE interconnection testing for
interworking and roaming scenarios**

Recommendation ITU-T Q.3953

ITU-T



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Recommendation ITU-T Q.3953

VoLTE/ViLTE interconnection testing for interworking and roaming scenarios

Summary

Recommendation ITU-T Q.3953 aims to verify the various interconnections and ensure that interoperability, interworking and roaming will respect national and international requirements and service level agreements (SLAs) among operators.

This Recommendation includes an electronic attachment with the test list announced in Annex A.

History

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Electronic attachment: The test list announced in Annex A

Recommendation ITU-T Q.3953

VoLTE/ViLTE interconnection testing for interworking and roaming scenarios

1 Scope

This Recommendation¹ aims to verify the various interconnections and ensure that interoperability, interworking and roaming will respect national and international requirements and service level agreements (SLAs) among operators.

2 References

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

- [ITU-T P.501] Recommendation ITU-T P.501 (2017), *Test signals for use in telephony*.
- [ITU-T P.863] Recommendation ITU-T P.863 (2014), *Perceptual objective listening quality assessment*.
- [ITU-T P.863.1] Recommendation ITU-T P.863.1 (2014), *Application guide for Recommendation ITU-T P.863*.
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- [ITU-T T.38] Recommendation ITU-T T.38 (2015), *Procedures for real-time Group 3 facsimile communication over IP networks*.

¹ This Recommendation contains an electronic attachment containing the test list announced in Annex A.

- [ITU-T V.152] Recommendation ITU-T V.152 (2010), *Procedures for supporting voice-band data over IP networks.*
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3 Definitions

None.

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

ACM	Address Complete Message
ACR	Anonymous Communication Rejection
AGCF	Access Gateway Control Function
ANM	Answer Message
APN	Access Point Name
BICC	Bearer Independent Call Control
CB	Communication Barring
CCBS	Completion of Communications to Busy Subscriber
CCNR	Completion of Communications by No Reply
CD	Communication Deflection
CDIV	Communication DIVersion
CDP	Charging Determinating Point
CDR	Communication Data Record
CFB	Communication Forwarding Busy
CFNL	Communication Forwarding Not Logged in
CFNR	Communication Forwarding No Reply
CFU	Communication Forwarding Unconditional
CONF	Conference
CPG	Call Progress Message
CS	Circuit Switched
CSFB	Circuit Switched FallBack
CSS	Composite Source Signal
CUG	Closed User Group
CW	Communication Waiting
DNS	Domain Name System
DTMF	Dual Tone Multi-Frequency
E2E	End-to-End
ECT	Explicit Communication Transfer
ENUM	Telephone Number Mapping
EPC	Evolved Packet Core
FDD	Frequency Division Duplexing
FFS	For Further Study
GRX/IPX	GPRS Roaming Exchange/IP exchange
GW	GateWay
HOLD	Communication Hold

HPMN	Home Public Mobile Network
IAM	Initial Address Message
IBCF	Interconnection Border Control Function
II-NNI	Inter-IMS Network to Network Interface
IMS	IP Multimedia Subsystem
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
IUT	Implementation Under Test
KPI	Key Performance Indicator
LBO	Local BreakOut
LBO-HR	Local BreakOut-Home Routing architecture
LBO-VR	Local BreakOut-VPMNr routing architectureLILawful Interception
LTE	Long-Term Evolution
MCID	Malicious Communication Identification
MME	Mobility Management Entity
MMTel	Multimedia Telephony Service
MOS	Mean Opinion Score
MWI	Message Waiting Indication
NDUB	Network Determined User Busy
NNI	Network to Network Interface
OIP	Originating Identification Presentation
OIR	Originating Identification Restriction
P-CSCF	Proxy-Call Session Control Function
PASP	Public Answering Safety Point
PES	PSTN Emulation Subsystem
PDD	Post Dialling Delay
PGW	Packet Data Network Gateway
PICS	Protocol Implementation Conformance Statement
PLMN	Public Land Mobile Network
PS	Packet Switched
PSTN	Public Switched Telephone Network
QoS	Quality of Service
R-NNI	Roaming Network to Network Interface
RAN	Radio Access Network
REL	Release
RLC	Release Complete
RTP	Real-time Transport Protocol

RTCP	Real Time Control Protocol
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SLA	Service Level Agreement
SRVCC	Single Radio Voice Call Continuity
SWB	Super WideBand
TD-SCDMA	Time Division Synchronous Code Division Multiple Access
TFO	Tandem Free Operation
TIP	Terminating Identification Presentation
TIR	Terminating Identification Restriction
TP	Test Purpose
TrFO	Transcoder Free Operation
TSS	Test Suite Structure
UAS	User Agent Server
UDUB	User Determined User Busy
UMTS	Universal Mobile Telecommunications System
URI	Uniform Resource Identifier
VGW	Voice GateWay
ViLTE	Video over LTE
VoLTE	Voice over LTE
VoPS	IMS Voice over PS session
VPLMN	Visited PLMN
WCDMA	Wideband Code Division Multiple Access

5 Conventions

None.

6 General principles of interconnection of VoLTE-based networks

Voice over long-term evolution (LTE) (VoLTE) and video over LTE (ViLTE) services deliver voice and video communication over packet-based networks which include LTE technology on the access layer. VoLTE/ViLTE services can be provided by either traditional fixed or mobile telecom operators that have implemented LTE technology as access technology on their core IP networks.

VoLTE/ViLTE services are so-called "managed" voice and video services which are based on standardized session initiation protocol (SIP)/IP multimedia subsystem (IMS) signalling and are provided by telecom operators, while over the top (OTT) applications are services which are provided in the public Internet by independent third parties, without standardized signalling protocols, traffic prioritization and guaranteed quality of service (QoS).

An IMS platform is used as a service control layer which is used for managing VoLTE/ViLTE sessions. The reference architecture of IMS is shown in Figure 6-1.

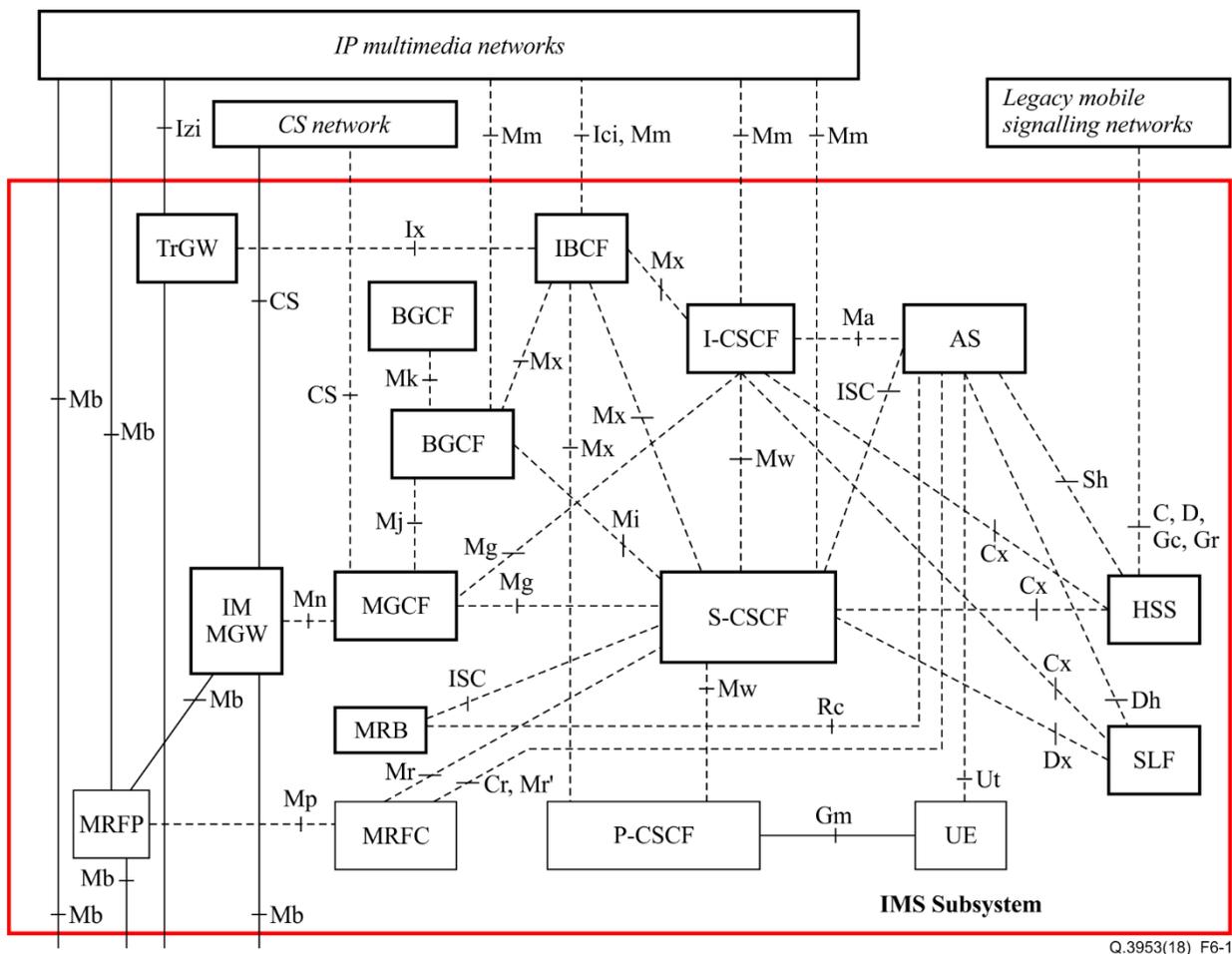


Figure 6-1 – Reference architecture of the IP multimedia core network subsystem

VoLTE/ViLTE interconnection implies interconnection of IMS platforms for providing VoLTE/ViLTE and legacy end-to-end (e2e) sessions.

The following are types of interconnections between IMS-based telecom operators:

- Interconnection for delivering sessions among users of different operators (hereafter interworking scenarios);
- Interconnection for providing roaming of users of home networks in visited networks (hereafter roaming scenarios).

There are also options for interconnections between VoLTE/ViLTE and IMS-based networks with existing legacy networks (e.g., public switched telephone network (PSTN), public land mobile network (PLMN)).

The telephone number mapping (ENUM)/domain name system (DNS) translation mechanism as specified in [IETF RFC 3761] can be used by all IMS nodes that require ITU-T E.164 address to SIP uniform resource identifier (URI) resolution. Subsequently, the GSMA published [b-GSMA IR.67].

6.1 E2E scenarios in terms of interworking, interconnection and roaming

The e2e tests shall be executed according to a test case selection expression and the type of end devices which are contained in the Excel test list included as an electronic attachment to this Recommendation. The Excel test list is a normative part of this Recommendation. The interconnection and roaming scenarios should be selected depending on the network infrastructure and company strategy.

The reference configuration depicted in Figure 6-2 shall be used to perform an interconnection test between two network operators. Depicted is the reference point to observe the message flow at the 'Ic' interface between the two networks.

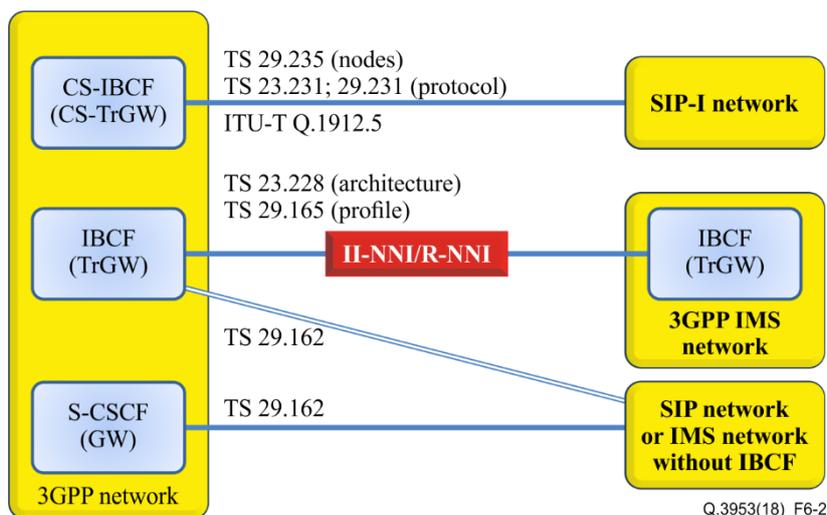


Figure 6-2 – Reference configuration for the interconnection test

6.2 Interconnection and roaming scenarios test selection description (Excel file)

The interconnection and roaming test scenarios selection procedure is divided into five steps:

6.2.1 Step 1 - Identification of the networks

During step 1, the table "Identification of the networks", depicted in Table 6-1 should (optionally) be completed.

Table 6-1 – Identification of the networks, with examples

	Network A	Network B
Network under Test identification	Telekom Austria	Deutsche Telekom
Responsibility		
Name:	Martin Brand	Gerhard Ott
Telephone number:		
Facsimile number:		
Additional information:		
Product Supplier	Nokia	Huawei
Date of the statement:		
Dates of Testing (from .. to ..)		

6.2.2 Step 2 - Selection expression

During step 2, the "Selection expression" form sheet should be completed. The Selection expression depicted in Table 6-2 was developed to select the scope of the compatibility test between network operator A and network operator B. By doing so, test purposes are selected automatically. The table shall be filled out (yes/no). This table can be used as a protocol implementation conformance statement (PICS) form as used in a conformance test.

Table 6-2 – Selection expression applicable in the test purposes

SELECTION EXPRESSION:	Support Network A	Support Network B
Network capabilities		

Table 6-2 – Selection expression applicable in the test purposes

SELECTION EXPRESSION:	Support Network A	Support Network B
SE 1: The originating network (Network A) sends the P-Charging-Vector header?		
SE 2: The originating network (Network A) sends a subset of parameters in the P-Charging-Vector header?		
SE 3: The P-Early-Media header is supported?		
SE 4: Overlap procedure using multiple INVITE method is supported?		
SE 5: Overlap sending using in-dialog method is supported?		
SE 6: Network A supports the PSTN XML schema?		
SE 7: The resource reservation procedure is supported?		
SE 8: Does the network perform the "fall back" procedure (PSTN or media gateway control unction (MGCF))?		
SE 9: The network is untrusted?		
SE 10: Originating network does not have a number portability database, the number portability lookup is done in the interconnected network?		
SE 11: The network supports the REFER method?		
SE 12: The network supports the three-party call control procedure (REFER interworking)?		
SE 13: The number portability is supported?		
SE 14: Carrier selection is performed?		
SE 15: The network is a long distance carrier?		
SE 16: SIP support of charging is supported?		
SE 17: The interworking ISDN user part (ISUP) – SIP I is performed in the network?		
Supplementary services		
SE 18: The network supports originating identification presentation (OIP)?		
SE 19: The network supports the "Special arrangement" procedure for the originating user?		
SE 20: The network supports originating identification restriction (OIR)?		
SE 21: The network supports terminating identification presentation (TIP)?		
SE 22: The network supports the "Special arrangement" procedure for the terminating user?		
SE 23: The network supports terminating identification restriction (TIR)?		
SE 24: The network supports the session communication hold (HOLD) procedure?		
SE 25: The network supports communication forwarding unconditional (CFU)?		
SE 26: The network supports communication forwarding busy (CFB)?		
SE 27: The network supports communication forwarding no reply (CFNR)?		
SE 28: The network supports communication forwarding not logged in (CFNL)?		
SE 29: The network supports communication deflection?		
SE 30: The network supports the CDIV notification procedure?		
SE 31: The network supports conference (CONF)?		
SE 32: The network supports the communication barring (CB) procedure – (Black list for incoming calls)?		
SE 33: The network supports the anonymous communication rejection (ACR)?		
SE 34: The network supports the closed user group (CUG)?		
SE 35: The network supports the communication waiting (CW) service?		
SE 36: The network supports the T _{AS-CW} timer?		
SE 37: The network supports explicit communication transfer (ECT)?		

Table 6-2 – Selection expression applicable in the test purposes

SELECTION EXPRESSION:	Support Network A	Support Network B
SE 38: The network supports malicious communication identification (MCID)?		
SE 39: The network supports message waiting indication (MWI)?		
SE 40: The network supports completion of communications to busy subscriber (CCBS)?		
SE 41: The network supports completion of communications by no reply (CCNR)?		
Terminal capabilities		
SE 42: Void		
SE 43: The end device supports fax transmission via ITU-T G.711 codec?		
SE 44: The end device supports fax transmission via ITU-T V.152 codec?		
SE 45: The end device supports fax transmission via m-line ITU-T T.38 codec?		
SE 46: A SIP end device is used supporting integrated services digital network (ISDN) user equipment and the PSTN XML schema is used?		
SE 47: End device is located in the PSTN or PLMN?		
SE 48: The terminating UE supports the from-change tag procedure and sends a second user identity in an UPDATE request after the dialogue is confirmed?		
SE 49: The end device performs ECT using the 'blind/assured transfer'?		
SE 50: The end device performs ECT using the 'consultative transfer'?		
SE 51: The end device supports the resource reservation procedure?		
PSTN/PLMN supplementary services		
SE 52: CLIP/CLIR is supported in the PSTN/PLMN part of the network?		
SE 53: COLP/COLR is supported in the PSTN/PLMN part of the network?		
SE 54: HOLD is supported in the PSTN/PLMN part of the network?		
SE 55: CDIV is supported in the PSTN/PLMN part of the network?		
SE 56: CONF/3PTY is supported in the PSTN/PLMN part of the network?		
SE 57: ACR is supported in the PSTN/PLMN part of the network?		
SE 58: CUG is supported in the PSTN/PLMN part of the network?		
SE 59: CW is supported in the PSTN/PLMN part of the network?		
SE 60: ECT is supported in the PSTN/PLMN part of the network?		
SE 61: MCID is supported in the PSTN/PLMN part of the network?		
SE 61A: Call completion is supported in the PSTN/PLMN part of the network?		
SE 62: Subaddressing (SUB) is supported in the PSTN/PLMN part of the network?		
SE 63: User-to-user signalling (UUS) is supported in the PSTN/PLMN part of the network?		
SE 64: Test purpose (TP) is supported in the PSTN/PLMN part of the network?		
Dual tone multi-frequency (DTMF) transmission		
SE 65: The network supports DTMF transmission in the real-time transport protocol (RTP) stream		
SE 66: The network supports DTMF transmission indicating in the SDP offer in the RTP stream		
SE 67: The network supports DTMF transmission by the SIP INFO/NOTIFY method for DTMF tone generation		

6.2.3 Step 3 - Access and end devices types

During step 3, the "Access and end devices types" form sheet should be completed. With the specified test purposes in this Recommendation, the compatibility between the interconnected networks and the used access and end devices selection expression shall be assured. Each test purpose can be performed by using a physical end device to assure end-to-end compatibility between the two interconnected networks.

Table 6-3 – Overview of access and end devices types

List of Type of End devices in both networks		
	Network A	Network B
	Telekom Austria	Deutsche Telekom
SIP-VoIP		x
POTS		x
ISDN		
GSM		
VoUMTS		
VoLTE		
PSTN	x	x

Highlight color	Explanation	Reference
	The user equipment is a SIP hardphone or a SIP soft client on a PC in the fixed network The user equipment is a 4G mobile device in an LTE network The user equipment is a 3G mobile device in an UMTS network	TS 124 229
	The user equipment is an integrated end device in the fixed network - access via a legacy analogue device	TS 183 043
	The user equipment is an integrated end device in the fixed network - access via a legacy ISDN device	TS 183 036
	The user equipment is a 2G mobile device in an GSM network. SS7 / SIP interworking applies The user equipment is located in a fixed SS7 network (analogue or ISDN)	ITU-T Q.761 - Q764 TS129 163 ITU-T Q.1912.5

6.2.4 Step 4 - Activation

In step 4, the test list (Table 6-4) should activate the filter "Selected" in row "G" (deactivate the "no" entry). In addition to hide the title of the test case deselect the "empty" entry.

Table 6-4 – Test list – Example

IMS interconnection tests at the Ic Interface; Test Suite Structure and Test Purposes (TSS&TP)								
Test case number	Test name	Dir	Originating end device	Terminating end device	Selected	Executed	Verdict	Observation
BCALL								
BCALL/successful								
SS_bcall_002_a_pstn_sip	Basic call normal call clearing from the calling user.	NA -> NB	PSTN	SIP-VoIP	yes	no		2
SS_bcall_002_a_pstn_pots	Basic call normal call clearing from the calling user.	NA -> NB	PSTN	POTS	yes	no		2
SS_bcall_002_a_pstn_pstn	Basic call normal call clearing from the calling user.	NA -> NB	PSTN	PSTN	yes	no		2
SS_bcall_002_b_sip_pstn	Basic call normal call clearing from the calling user.	NB -> NA	SIP-VoIP	PSTN	yes	yes		
SS_bcall_002_b_pots_pstn	Basic call normal call clearing from the calling user.	NB -> NA	POTS	PSTN	yes	yes		
SS_bcall_002_b_pstn_pstn	Basic call normal call clearing from the calling user.	NB -> NA	PSTN	PSTN	yes	yes		
SS_bcall_003_a_pstn_sip	Request line in the INVITE.	NA -> NB	PSTN	SIP-VoIP	yes	yes		
SS_bcall_003_a_pstn_pots	Request line in the INVITE.	NA -> NB	PSTN	POTS	yes	yes		

6.2.5 Step 5 - Selection of roaming scenarios (not contained in the Excel file)

According to the general principles described in step 3, Access and end devices types, four key e2e interworking scenarios can be identified (see Table 6-5):

- 1) VoLTE – IMS interconnection scenarios;
- 2) VoLTE – Legacy network scenarios;
- 3) VoLTE – VoLTE and ViLTE - ViLTE interconnection scenarios;
- 4) VoLTE – VoLTE and ViLTE - ViLTE roaming scenarios.

As for the roaming scenarios, the identical interworkings are used, the execution of the complete interconnection test list (step 4) is not useful. For the applicable roaming scenario(s) the first two test cases (SS_bcall_NNI_01 and SS_bcall_NNI_02) for different roaming scenarios and codecs should be repeated.

For simplified execution of the tests, the test cases can be grouped. An example of a VoLTE – 2G/3G interconnection scenario is depicted in Appendix I.

Table 6-5 – E2E scenarios in terms of interconnection and roaming

No.	Scenario	Description	Roaming options	Calling options
VoLTE – IMS interworking scenarios				
1	Scenario 1	The user UE1 (a) is in the IMS network A, UE2 (a) in home public mobile network (HPMN), HPMN (a)		The test shall be performed in both directions
2	Scenario 1A	The user UE 1 (a) is in the IMS network A, UE2 (a) in HPMN (a) with circuit switched fallback (CSFB). Note – This occurs only in the case where IMS voice over PS session (VoPS) is not supported in the HMPMN's LTE N/W		The test shall be performed in both directions
3	Scenario 1B	The user UE1 (a) is in the IMS network A, UE2 (a) in HPMN (a) is moving from 4G to 3G coverage with single radio voice call continuity (SRVCC)		The test shall be performed in both directions
4	Scenario 1C	The user UE1 (a) is in the IMS network A, UE2 (a) roamed in VPMN (b)	– Local breakout VPMN routing architecture (LBO-VR) – LBO home routing architecture (LBO-HR) – S8HR VoLTE roaming architecture	The test shall be performed in both directions
5	Scenario 1D	The user UE1 (a) is in the IMS network A, UE2 (a) roamed in VPMN (b) moving from 4G to 3G coverage with SRVCC	– LBO-VR – LBO-HR – S8HR VoLTE roaming architecture	The test shall be performed in both directions
VoLTE – Legacy network scenarios				
6	Scenario 2	User UE1 (a) is in the legacy network A , UE2 (a) is in HPMN (a),		The test shall be performed in both directions
7	Scenario 2A	User UE1 (a) is in the legacy network A, UE2 (a) is in HPMN (a), with CSFB Note – This occurs only in the case if VoPS is not supported in the HMPMN's LTE N/W		The test shall be performed in both directions
8	Scenario 2B	User UE1 (a) is in the legacy network A , UE2 (a) is in HPMN (a), roamed in VPMN (b) moving from 4G to 3G coverage with SRVCC		The test shall be performed in both directions
9	Scenario 3	User UE1(a) is in the legacy network A , UE2 (a) is in VPMN (b),	– LBO-VR – LBO-HR – S8HR VoLTE roaming architecture	
VoLTE – VoLTE and ViLTE – ViLTE interconnection scenarios				
10	Scenario 4	UE1 (a) is in HPMN (a), UE2 (b) is in HPMN (b)		
11	Scenario 4A	UE1 (a) is in HPMN (a), UE2 (b) is in HPMN (b) with CSFB. Note – This occurs only in the case where VoPS is not supported in HPMN		
12	Scenario 4B	UE1 (a) is in HPMN (a), UE2 (b) is in HPMN (b) moving from 4G to 3G coverage with SRVCC		
13	Scenario 4C	UE1 (a) is in HPMN (a) with CSFB. UE2 (b) is in HPMN (b).		
14	Scenario 4D	UE1 (a) is in HPMN (a) moving from 4G to 3G coverage with SRVCC, UE2 (b) is in HPMN (b).		
VoLTE – VoLTE and ViLTE – ViLTE roaming scenarios				
15	Scenario 5	UE1 (a) is in HPMN (a), UE3 (a) roamed in VPMN (b)	– LBO-VR – LBO-HR	User A is calling user B User B is calling user A

Table 6-5 – E2E scenarios in terms of interconnection and roaming

No.	Scenario	Description	Roaming options	Calling options
			– S8HR VoLTE roaming architecture	
16	Scenario 5A	UE1 (a) is in HPMN (a), UE3 (a) roamed in VPMN (b) with CSFB Note – This occurs only in the case where VoPS is not supported in HPMN		User A is calling user B User B is calling user A
17	Scenario 5B	UE1 (a) is in HPMN (a), UE3 (a) roamed in VPMN (b) moving from 4G to 3G coverage with SRVCC ;	– LBO-VR – LBO-HR – S8HR VoLTE roaming architecture	User A is calling user B User B is calling user A
18	Scenario 5C	UE1 (a) in HPMN (a) with CSFB , UE3 (a) roamed in VPMN (b)	Roaming options: – LBO-VR – LBO-HR – S8HR VoLTE roaming architecture	User A is calling user B User B is calling user A
19	Scenario 5D	UE1 (a) is in HPMN (a) moving from 4G to 3G coverage with SRVCC , UE3 (a) roamed in VPMN (b)	Roaming options: – LBO-VR – LBO-HR – S8HR VoLTE roaming architecture	User A is calling user B User B is calling user A
20	Scenario 6	UE1 (a) calls UE3 (a), both roamed in VPMN (b)	– LBO-VR – LBO-HR – S8HR VoLTE roaming architecture	User A is calling user B User B is calling user A
21	Scenario 6A	UE1 (a) calls UE3 (a), both roamed in VPMN (b), UE1 (a) with CSFB Note – This occurs only in the case where VoPS is not supported in VPMN of UE1	– LBO-VR – LBO-HR – S8HR VoLTE roaming architecture	User A is calling user B User B is calling user A
22	Scenario 6B	UE1 (a) calls UE3 (a), both roamed in VPMN (b), UE1 (a) is moving from 4G to 3G coverage with SRVCC	– LBO-VR – LBO-HR – S8HR VoLTE roaming architecture	User A is calling user B User B is calling user A
23	Scenario 6C	UE1 (a) calls UE3 (a), both boamed in VPMN (b), UE2 (a) is moving from 4G to 3G coverage with SRVCC	– LBO-VR – LBO-HR – S8HR VoLTE roaming architecture	User A is calling user B User B is calling user A
24	Scenario 7	UE1 (a) roamed in VPMN (b), UE2 (b) roamed in VPMN (a)	– LBO-VR – LBO-HR – S8HR VoLTE roaming architecture	User A is calling user B User B is calling user A
25	Scenario 7A	UE1 (a) roamed in VPMN (b), UE2 (b) roamed in VPMN (a), UE1 (a) with CSFB	– LBO-VR – LBO-HR – S8HR VoLTE roaming architecture	User A is calling user B User B is calling user A
26	Scenario 7B	UE1 (a) roamed in VPMN (b), UE2 (b) roamed in VPMN (a) and UE2 (b) is moving from 4G to 3G coverage with SRVCC	– LBO-VR – LBO-HR – S8HR VoLTE roaming architecture	
27	Scenario 7C	UE1 (a) roamed in VPMN (b), UE2 (b) roamed in VPMN (a), and UE1 (a) is moving from 4G to 3G coverage with SRVCC	– LBO-VR – LBO-HR – S8HR VoLTE roaming architecture	
28	Scenario 8	The VoLTE subscriber UE1 (a) in the HPMN (a) is calling the 2G/3G user UE2 (b) in HPMn (b)		User A is calling user B User B is calling user A

Table 6-5 – E2E scenarios in terms of interconnection and roaming

No.	Scenario	Description	Roaming options	Calling options
29	Scenario 9	The VoLTE subscribers UE1 (a) and UE 2(a) are subscribed in the HPMN (a). Subscribers UE1 (a) is roaming in 2G/3G VPMN (b).		User A is calling user B User B is calling user A
30	Scenario 10	The VoLTE subscribers UE1 (a) and UE 2(a) are subscribed in the HPMN (a). Subscribers UE1 (a) and UE2 (a) are roaming in 2G/3G VPMN (b).		User A is calling user B User B is calling user A

7 IMS roaming and interconnection options

According to [b-GSMA IR.65], there are different possible options for IMS interconnection which should meet the following requirements:

- 1) Routing of media for voice and video over IMS when call originator is roaming should be at least as optimal as that of current circuit switched (CS) domain;
- 2) The charging model for roaming used in CS domain shall be maintained in VoIMS;
- 3) Allow the HPMN to decide, based on service and commercial considerations and regulatory obligations, to enforce the routing of the originated traffic to itself (home routing).

UE has obtained IP connectivity in the visited network and might have access to home's IMS services via one of the following options:

Option 1 – Target IMS roaming solution, IMS is required in both the visited-PLMN (VPLMN) and HPLMN

UE has obtained IP connectivity from the visited network and is connected to the proxy-call session control function (P-CSCF) in the visited network which establishes connections using the home IMS platform; traffic is routed directly by the visited network;

Option 2 – Data local breakout, but IMS home routed, IMS is not needed in VPLMN

UE has obtained IP connectivity from the visited network and is connected to the P-CSCF in the visited network which itself is connected to the home IMS platform; traffic is routed via the home network;

Option 3 – UE has obtained IP connectivity from the home network and is directly connected to the home IMS platform; traffic is routed via the home network. Data and IMS are both home routed, IMS is not needed in VPLMN.

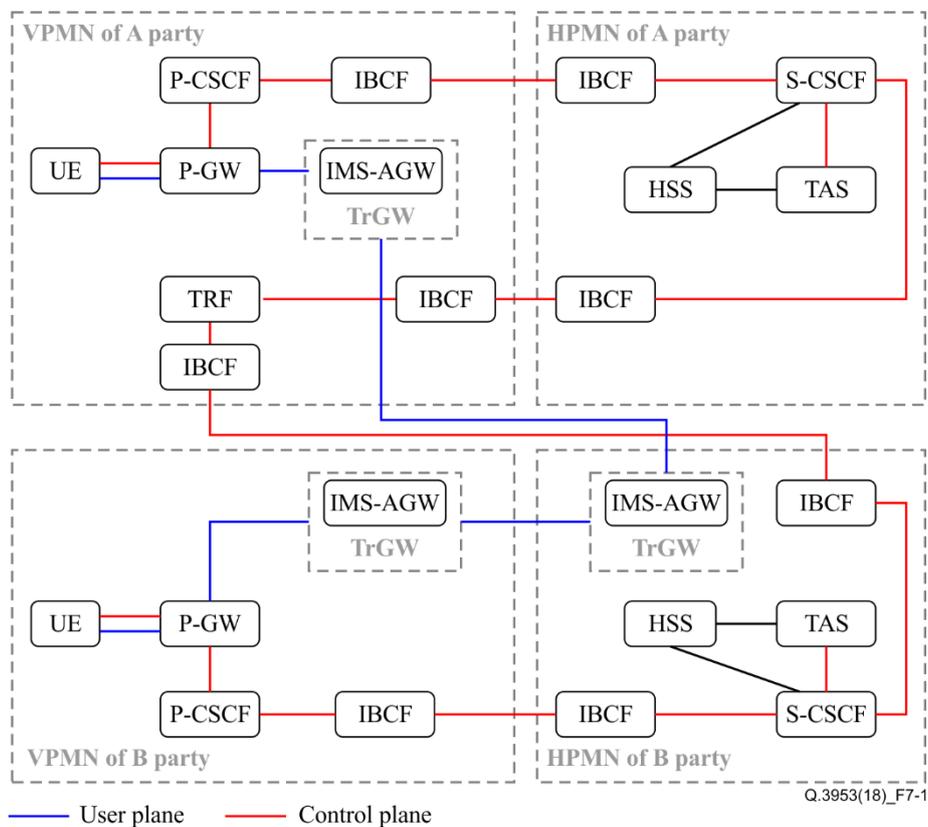


Figure 7-1 – Local breakout VPMN routing architecture

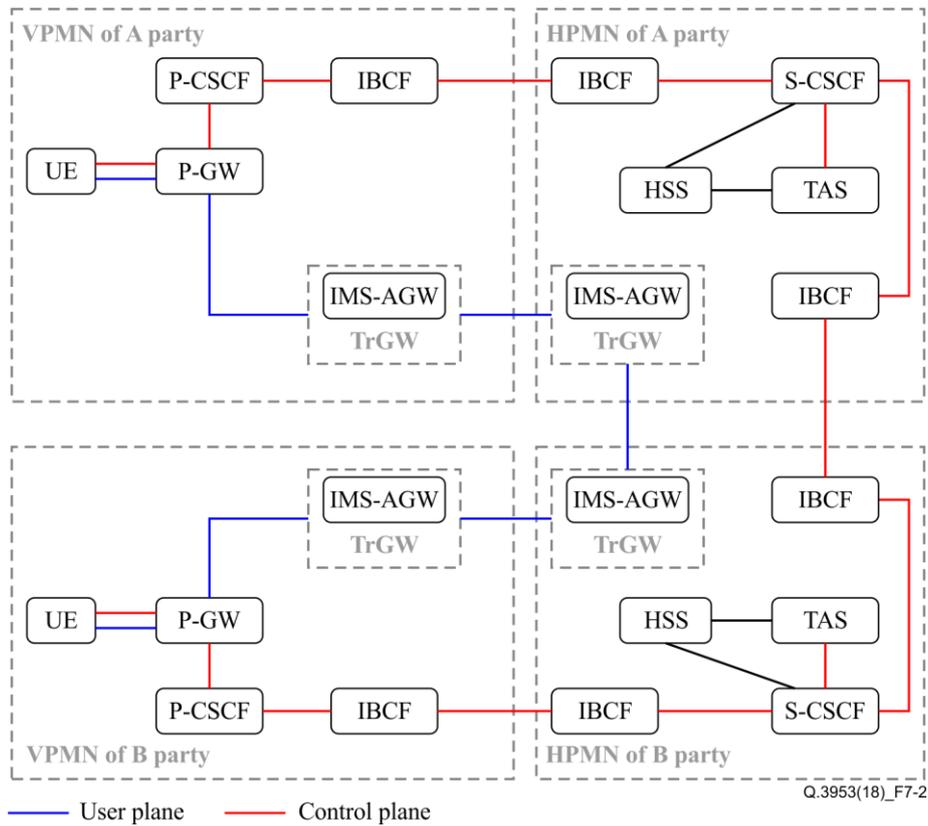


Figure 7-2 – LBO home routing architecture

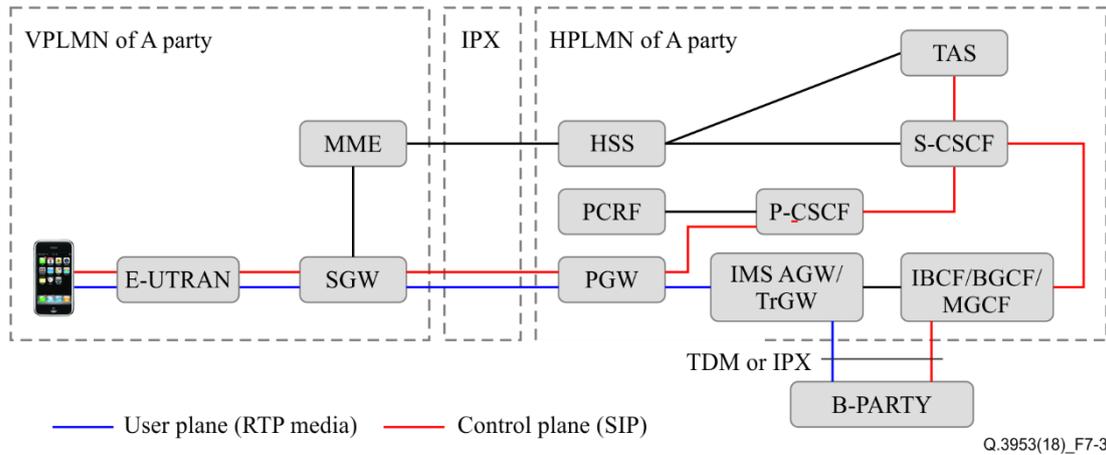


Figure 7-3 – S8HR VoLTE roaming architecture

For S8HR VoLTE roaming architecture, regulatory requirements (legal interception and emergency calling) have triggered additional specification work currently being performed in 3GPP. Deliverables are confirmed to be included in 3GPP Release 14.

Table 7-1 – Comparison of VoLTE roaming architecture

Item	Option 1– LBO-VR Target IMS roaming solution; IMS is required in both VPLMN and HPLMN	Option 2 – LBO-HR Data local breakout, but IMS home routed; IMS is not needed in VPLMN	Option 3 – S8HR Data and IMS are both home routed; IMS is not needed in VPLMN
HPLMN with VoLTE implementation	Required	Required	Required
VPLMN with VoLTE implementation	Required	Not required	Not required
IMS service over GRX	Not required	Required	Required
Charging depending on evolved packet core (EPC)	Optional (charge on IMS service layer)	Required	Required
Policy and charging control mode	HPLMN hPCRF can control the VPLMN vPCRF via S9 Interface (S9 interface is optional)	HPLMN hPCRF controls VPLMN vPCRF via S9 Interface or via roaming agreement and support of common QCI	HPLMN hPCRF controls HPLMN packet data network gateway (PGW), 2/3G and 4G (e.g., Web browsing) data roaming via S8
SRVCC support capability	Fully Supported	Supported	Partially supported
VoLTE local emergency call	Supported	Supported	Not supported
VoLTE local lawful interception (LI)	Supported	Supported	Not supported (LI will be possible at the S-GW; Under development in Rel. 14 of 3GPP).
LBO with optimal media routing (OMR)	Supported	Not supported	Not supported

8 VoLTE interconnect and roaming tests

8.1 Test suite structure

Table 8-1 – Test suite structure

BCALL	Successful network to network interface (NNI)	SS_bcall_NNI_xxx	
	DTMF	SS_DTMF_xxx	
	Fax transmission	SS_bcall_FAX_xxx (Note 2)	
	Successful SIP-I	SS_bcall_SIP-I_xxx	
	Codec_Negotiation	SS_codec_xxx	
	Resource_Reservation	SS_resource_xxx	
	Unsuccessful NNI	SS_unsucc_NNI_NNI_xxx	
	Unsuccessful SIP-I	SS_unsucc_NNI_SIP-I_xxx	
	Successful Video	SS_bcall_video_xxx	
SIP-SIP	Service	OIP	SS_oip_NNI_xxx
		OIP	SS_oip_SIP-I_xxx
		OIR	SS_oir_NNI_xxx
		OIR	SS_oir_SIP-I_xxx
		TIP	SS_tip_xxx (Note 1)

Table 8-1 – Test suite structure

		TIR	SS_tir_xxx (Note 1)
		HOLD	SS_hold_xxx (Note 1)
		CFU	SS_cfu_xxx (Note 1)
		CFB	SS_cfb_xxx (Note 1)
		CFNR	SS_cfnr_xxx (Note 1)
		CFNL	SS_cfnl_xxx (Note 1)
		CD	SS_cd_xxx (Note 1)
		CONF	SS_conf_xxx (Note 1)
		ACR-CB	SS_acr-cb_xxx (Note 1)
		CUG	SS_cug_xxx (Note 1)
		CW	SS_cw_xxx (Note 1)
		ECT	SS_ect_xxx (Note 1)
		MCID	SS_mcid_xxx (Note 1)
		MWI	SS_mwi_xxx (Note 1)
		CC	SS_cc_xxx (Note 1)
	SIP-I	UUS	SS_uus_xxx (Note 1)
		SUB	SS_sub_xxx (Note 1)
		TP	SS_tp_xxx (Note 1)
	NubP	SS_NP_xxx	(Note 1)
	ACCOUNTING	SS_acc_xxx	(Note 1)
	CS	SS_csel_xxx	(Note 1)
	EmC	SS_ecall_xxx	(Note 1)
	SIP_charging	SS_sipc_xxx	(Note 1)
NOTE 1 – The tests are specified in [ETSI TS 101 585].			
NOTE 2 – This clause contains three basic fax transmission tests without QoS requirements for the case when both networks are supporting the same transmission capabilities.			

NOTE 1 – The tests are specified in [ITU-T Q.3940].

NOTE 2 – This chapter contains three basic fax transmission tests without QoS requirements for the case when both networks are supporting the same transmission capabilities.

The "minimal requirements" for fax support between SIP-enabled devices for real-time fax over IP are described in [ITU-T Q.4016].

8.2 VoLTE consideration

8.2.1 EPC consideration

EPC network should support general VoLTE related functions, for example, basic voice and video calls.

- Visited network mobility management entity (MME) shall support VoLTE capability indication to UE, "IMS voice over PS" in order to select voice solution;
- Visited network MME may support the Sv interface and eSRVCC handover control function;
- SAE-GW shall support the establishment of dedicated bearer with QCI=8/9, QCI=1 and QCI=2;
- SAE-GW shall support IPv6 PDN type;
- PGW shall support the P-CSCF discovery function and allocating P-CSCF IP address to UE;
- PCRF shall guarantee end-to-end QoS by interworking with IMS and application function (AF) via Rx interface.

8.2.2 EPC configuration requirements

MME

- MME in visit network shall configure PLMN for inbound roamer, such that inbound roamer could attach to the network;
- MME in visit network shall configure "IMS voice over PS" for inbound roamer;
- LBO VR: MME in the visit network shall be able to resolve IMS access point name (APN) to PGW address in visit network;
- S8HR: MME in the visit network shall resolve IMS APN into PGW address in home network instead of visit network. If home network cannot resolve IMS APN to PGW, manually configured to the MME in visited network would be also acceptable for this trial.

8.3 Device and U/SIM consideration

8.3.1 Multi-mode and multi-band terminal

To meet requirements for domestic frequency access and international roaming, five radio modes including LTE frequency division duplexing (FDD), TD-LTE, time division synchronous code division multiple access (TD-SCDMA), wideband code division multiple access (WCDMA) and GSM should be supported. The multi-mode and multi-band requirements are as follow:

- GSM: Band3, Band8 and Band2 are mandatory. Furthermore, Band5 is recommended.
- TD-SCDMA: Band34 and Band39 are mandatory.
- TD-LTE: Band39, Band40 and Band41 (supporting at least 2575-2635 MHz) are mandatory. If the terminals support Band41 without Band38, it must support the frequency mutual identification via mFBI.
- WCDMA: Band1, Band2 and Band5 are mandatory.
- LTE FDD: Band3 and Band7 are mandatory. Besides, Band1, Band17, Band4 and Band20 are recommended.

Additionally, compatible bands based on bilateral/multi-lateral discussions will be supported by the Device. Other multi-mode and multi-band terminals could also be introduced in according to trial demand.

8.3.2 General requirements for VoLTE terminal

To achieve an excellent user experience, VoLTE terminals shall have performance requirements that are comparable with commercially available terminals in the following aspects: operation system, hardware, software, mean time between failure (MTBF), standby time, communication duration.

Regarding the outbound roaming requirement, VoLTE handsets should support CSFB from LTE to WCDMA/GSM and support four kinds of functions for VoLTE:

8.3.2.1 Radio access network features

- Support semi-persistent scheduling (SPS),TTI-Bundling, RoHC, Connected-DRX and its combinations; Support interoperation from LTE to GSM via eSRVCC, aSRVCC, mid-call SRVCC; Support SRVCC related measurement capability and capability report;
- Support UE-based fast return to LTE after SRVCC CS call ends;
- Support IPv4, IPv6 and IPv4v6 dual-stacks;
- Support multi-PDN connections; Delete IMS PDN when moving out of VoLTE coverage;
- Support EPS bearer combinations for VoLTE service; Support e2e QoS.

8.3.2.2 IMS function on control plane

- Support the standard SIP/IMS protocol in order to inter-work with the global IMS networks;

- Support derivation of IMS identifiers from USIM, if ISIM is not introduced;
- Support IMS exceptions handling;
- Support RTP/RTCP control protocol (RTCP);
- Support the IMS authorization and authentication, etc.;
- Support early media;
- Support precondition;
- Support upgrade and downgrade between voice and video call;
- Support supplementary service configuration via Ut/XCAP.

8.3.2.3 IMS function on media plane

- Audio codec: Entities in the IMS core network that terminate the user plane supporting speech communication and supporting tandem free operation (TFO) and/or transcoder free operation (TrFO) shall support AMR speech codec modes 12.2, 7.4, 5.9 and 4.75; Entities in the IMS core network that terminate the user plane supporting wideband speech communication and supporting TFO and/or TrFO shall support AMR-WB speech codec modes 12.65, 8.85 and 6.60;
- Video codec: ITU-T H.264 640*480@30fps; 720P@30fps is recommended;
- Quality enhanced features: Noise suppression, echo cancellation, jitter buffer, lip sync.

8.3.2.4 Services requirements

- Voice call
 - Support standard voice call and HD voice call;
 - Support voice domain transition between VoLTE and CSFB; Support voice continuity among different scenarios;
 - Support SilentRedial.
- Message
 - Support SMS over IP, SMS over CS;
 - Support MMS.
- Video call
 - Support video call when UE within VoLTE coverage.
- Supplementary services
 - Supported enhanced conference call;
 - Support IMS supplementary services.
- IMS emergency service

9 Test purposes

The application usage procedures in the abstract test suite (ATS) shall be compliant with [ETSI TS 129 165], [ETSI TS 124 229] and [IETF RFC 3261]. The validation of the registration procedure is out of scope of this Recommendation.

9.1 Testing of SIP protocol requirements

9.1.1 Test purposes for Basic call, successful

Test case number	SS_bcall_NNI_001	
Test case group	BCALL/successful	
Reference	[ETSI TS 124 229]	
SELECTION EXPRESSION		
Test purpose	<p>Basic call normal call clearing from the called user.</p> <p>Ensure that the UE can successfully activate the voice call via dedicated voice bearer. The test call is successful in the case where call setup time does not exceed the values of the performance design objectives listed in Table 9-1 and call is stable in unanswered and answered phases, the call remains in intelligible/high quality conversation phase for 80 seconds. The voice quality test procedures are described in clause 10.2.</p> <p>The test scenarios are listed in Table 6-5.</p> <p>The call is released from the called user.</p>	
Configuration		
SIP parameter		
Message flow		
SIP (Network A)	Interconnection interface INVITE → ← 100 Trying ← 180 Ringing ← 200 OK INVITE ACK → Communication ← BYE 200 OK BYE →	SIP (Network B)
Comments	<ul style="list-style-type: none"> • UE1 (a) establishes call to UE2 (b). • Call answered and held for 80 seconds. • Quality assessed. • Repeat this test in reverse direction. 	

Test case number	SS_bcall_NNI_002																											
Test case group	BCALL/successful																											
Reference	[ETSI TS 124 229]																											
SELECTION EXPRESSION																												
Test purpose	<p>Basic call normal call clearing from the calling user.</p> <p>Ensure that the UE can successfully activate the voice call via dedicated voice bearer. The test call is successful in the case if the call setup time does not exceed the values listed in Table 9-1 and call the is stable in unanswered and answered phases, the call remains in intelligible/high quality conversation phase for 80 seconds. The voice quality test procedures are described clause 10.2.</p> <p>The test scenarios are listed in Table 6-5.</p> <p>The call is released from the calling user.</p>																											
Configuration																												
SIP parameter																												
Message flow	<table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 33%;">SIP (Network A)</th> <th style="text-align: center; width: 33%;">Interconnection interface</th> <th style="text-align: right; width: 33%;">SIP (Network B)</th> </tr> </thead> <tbody> <tr> <td></td> <td style="text-align: center;">INVITE →</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 100 Trying</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 180 Ringing</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 200 OK INVITE</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">ACK →</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">Communication</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">BYE →</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 200 OK BYE</td> <td></td> </tr> </tbody> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE →			← 100 Trying			← 180 Ringing			← 200 OK INVITE			ACK →			Communication			BYE →			← 200 OK BYE	
SIP (Network A)	Interconnection interface	SIP (Network B)																										
	INVITE →																											
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	← 200 OK INVITE																											
	ACK →																											
	Communication																											
	BYE →																											
	← 200 OK BYE																											
Comments	<ul style="list-style-type: none"> • UE1 (a) establishes call to UE2 (b). • Call answered and held for 80 seconds. • Quality assessed. • Repeat this test in reverse direction. 																											

Table 9-1 – Call setup time (post dialling delay (PDD)) See [ETSI ES 202 765-2]

Meaning of timers	Parameter ITU-T Q.543 Detailed description	IMS, PES equivalent	Reference load A		Reference load B	
			Mean value	95% probability of not exceeding	Mean value	95% probability of not exceeding
VoLTE – VoLTE and IMS to VoLTE						
Call setup time: The definition of call setup time for VoLTE is defined in [ETSI TS 102 250-2].						
			≤ 1950 ms	≤ 2100 ms	≤ 2250 ms	≤ 2400 ms (Note 1) (Note 2) (Note 3)
VoLTE to IMS (Note 4)						
Call setup time (PDD)						
To determine the call setup time in a VoIP implementation, the time in seconds from the sending of the INVITE signal through the "A" side until the receipt of the 200 OK signal is measured on the "A" side, or the time in seconds from the sending of the INVITE signal through the "A" side until the receipt of the 180 Ringing signal on the "A" side is recorded.						
			≤ 420 ms	≤ 580 ms	≤ 750 ms	≤ 900 ms (Note 4)
IMS – IMS						
Call setup time (PDD)						
			≤ 350 ms	≤ 500 ms	≤ 650 ms	≤ 800 ms
NOTE 1 – Paging cycle 128 seconds.						
NOTE 2 – S1-Control plane delay: 2 ms – 15 ms (S1 is the interface between eNode Bs and MME and S-GW).						
NOTE 3 – The maximum value should not exceed 5,9 seconds.						
NOTE 4 – The values are based on the condition that the originating VoLTE – UE is in the state ECM Connected. In the case when the oLTE – UE is in ther state ECM Idle, the time duration is about 100 ms higher.						

Test case number	SS_bcall_NNI_007															
Test case group	BCALL/successful															
Reference	8/[IETF RFC 5009]															
SELECTION EXPRESSION	[Network A] SE 3 AND [Network B] SE3															
Test purpose	P-Early-Media header supported in early dialogue. Ensure that an early dialogue is established by sending a 183 Session Progress or 180 Ringing from Network B and the P-Early-Media header is present authorizes early media. The early media voice quality test procedures are described clause 8.2.															
Configuration																
SIP parameter	INVITE P-Early-Media: supported SDP 183 P-Early-Media: [any value authorizes early media] SDP OR 180 P-Early-Media: [any value authorizes early media] SDP															
Message flow	<table border="0" style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 30%; text-align: center;">SIP (Network A)</td> <td style="width: 40%; text-align: center;">Interconnection interface</td> <td style="width: 30%; text-align: center;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE →</td> <td></td> </tr> <tr> <td>CASE A</td> <td style="text-align: center;">← 183 Session Progress</td> <td></td> </tr> <tr> <td>CASE B</td> <td style="text-align: center;">← 180 Ringing</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">Apply post test routine</td> <td></td> </tr> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE →		CASE A	← 183 Session Progress		CASE B	← 180 Ringing			Apply post test routine	
SIP (Network A)	Interconnection interface	SIP (Network B)														
	INVITE →															
CASE A	← 183 Session Progress															
CASE B	← 180 Ringing															
	Apply post test routine															
Comments	<p>Establish a communication from Network A to Network B</p> <p>Check: Is a 183 or 180 send from Network B to establish an early dialogue?</p> <p>Check: Is an SDP present in the 183 as an SDP answer?</p> <p>Check: A bearer transmission is possible in backward directions (optional).</p> <p>NOTE 1 – The absence of the direction parameter of an 'a' line represents the default value 'sendrecv'</p> <p>NOTE 2 – The presence of the P-Early-Media header in the INVITE request indicates the support of "early media authorization" in the originating network.</p> <p>NOTE 3 – The presence of the P-Early-Media header in the 183 or 180 indicates the support of the P-Early-Media header and authorizes the media in the early dialogue.</p> <p>Repeat this test in reverse direction.</p> <p>Repeat this test by a call setup to an announcement application.</p>															

Test case number	SS_bcall_NNI_011												
Test case group	BCALL/successful												
Reference	5.10/[ETSI TS 124 229]												
SELECTION EXPRESSION													
Test purpose	Via header in the INVITE. Ensure that the Via header is present in the INVITE establishes a communication between a user of Network A and a user of Network B and the topmost header is set to the IBCF of Network A and contains a branch parameter.												
Configuration													
SIP parameter	INVITE Via: <Address of IBCF in network A>; branch=[any value]												
Message flow	<table style="width: 100%; border: none;"> <tr> <td style="text-align: center; width: 33%;">SIP (Network A)</td> <td style="text-align: center; width: 33%;">Interconnection interface</td> <td style="text-align: center; width: 33%;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">→</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">Apply post test routine</td> <td></td> </tr> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE			→			Apply post test routine	
SIP (Network A)	Interconnection interface	SIP (Network B)											
	INVITE												
	→												
	Apply post test routine												
Comments	Establish a communication from Network A to Network B Check: The topmost Via header contains the address of IBCF in Network A and a branch parameter. Repeat this test in reverse direction. Repeat this test with all chosen end devices.												

Test case number	SS_bcall_NNI_012																		
Test case group	BCALL/successful																		
Reference	5.10/[ETSI TS 124 229]																		
SELECTION EXPRESSION																			
Test purpose	Record-Route header in the 180 Ringing. Ensure if a Record-Route header was present in the initial INVITE that the Record-Route header is present in the 180 Ringing provisional response as the first response from Network B upon a connection establish setup from Network A.																		
Configuration																			
SIP parameter	INVITE Record-Route 180: Record-Route																		
Message flow	<table style="width: 100%; border: none;"> <tr> <td style="text-align: center; width: 33%;">SIP (Network A)</td> <td style="text-align: center; width: 33%;">Interconnection interface</td> <td style="text-align: center; width: 33%;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">←</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">180 Ringing</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">→</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">Apply post test routine</td> <td></td> </tr> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE			←			180 Ringing			→			Apply post test routine	
SIP (Network A)	Interconnection interface	SIP (Network B)																	
	INVITE																		
	←																		
	180 Ringing																		
	→																		
	Apply post test routine																		
Comments	Establish a communication from Network A to Network B Check: If the Record-Route header is present is in the 180 Ringing. The Record-Route header is optional. Repeat this test in reverse direction. Repeat this test with all chosen end devices.																		

Test case number	SS_bcall_NNI_013																
Test case group	BCALL/successful																
Reference	5.10/[ETSI TS 124 229]																
SELECTION EXPRESSION																	
Test purpose	Route header in the BYE of the originating user. Ensure that if a Record-Route header was present in the initial INVITE the Route header may be present in the BYE request sent from the originating user equipment in Network A the topmost Route header or entry is set to the IBCF of Network B.																
Configuration																	
SIP parameter	BYE: Route: <Address of IBCF in network B>;lr,																
Message flow	<table style="width: 100%; border: none;"> <tr> <td style="text-align: center; width: 33%;">SIP (Network A)</td> <td style="text-align: center; width: 33%;">Interconnection interface</td> <td style="text-align: center; width: 33%;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;">A confirmed session already exists</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">BYE →</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 200 OK BYE</td> <td></td> </tr> <tr> <td></td> <td colspan="2" style="text-align: center;">Apply post test routine</td> </tr> </table>		SIP (Network A)	Interconnection interface	SIP (Network B)		A confirmed session already exists			BYE →			← 200 OK BYE			Apply post test routine	
SIP (Network A)	Interconnection interface	SIP (Network B)															
	A confirmed session already exists																
	BYE →																
	← 200 OK BYE																
	Apply post test routine																
Comments	Establish a communication from Network A to Network B Check: If the Route header is present in the BYE, the topmost header or entry is set to the address of the IBCF of Network B. Repeat this test in reverse direction. Repeat this test with all chosen end devices.																

Test case number	SS_bcall_NNI_014																
Test case group	BCALL/successful																
Reference	5.10/[ETSI TS 124 229]																
SELECTION EXPRESSION																	
Test purpose	Route header in the BYE of the terminating user. Ensure that if a Record-Route header was present in the initial INVITE the Route header may be present in the BYE request sent from the terminating user equipment in Network B the topmost Route header or entry is set to the IBCF of Network A.																
Configuration																	
SIP parameter	BYE: Route: <Address of IBCF in network A>;lr,																
Message flow	<table style="width: 100%; border: none;"> <tr> <td style="text-align: center; width: 33%;">SIP (Network A)</td> <td style="text-align: center; width: 33%;">Interconnection interface</td> <td style="text-align: center; width: 33%;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;">A confirmed session already exists</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← BYE</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">200 OK BYE →</td> <td></td> </tr> <tr> <td></td> <td colspan="2" style="text-align: center;">Apply post test routine</td> </tr> </table>		SIP (Network A)	Interconnection interface	SIP (Network B)		A confirmed session already exists			← BYE			200 OK BYE →			Apply post test routine	
SIP (Network A)	Interconnection interface	SIP (Network B)															
	A confirmed session already exists																
	← BYE																
	200 OK BYE →																
	Apply post test routine																
Comments	Establish a communication from Network A to Network B Check: If the Route header is present in the BYE, the topmost header or entry is set to the address of the IBCF of Network A. Repeat this test in reverse direction. Repeat this test with all chosen end devices.																

Test case number	SS_bcall_NNI_015																		
Test case group	BCALL/successful																		
Reference	5.10/[ETSI TS 124 229]																		
SELECTION EXPRESSION																			
Test purpose	Route header in the ACK. Ensure that if a Record-Route header was present in the initial INVITE the Route header may be present in ACK from Network A upon a connection establishment from Network A is completed the topmost Route header or entry is set to the IBCF of Network B.																		
Configuration																			
SIP parameter	ACK: Route: <Address of IBCF in network B>;lr,																		
Message flow	<table style="width: 100%; border: none;"> <tr> <td style="text-align: center; width: 33%;">SIP (Network A)</td> <td style="text-align: center; width: 33%;">Interconnection interface</td> <td style="text-align: center; width: 33%;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE →</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 180 Ringing</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 200 OK INVITE</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">ACK →</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">Apply post test routine</td> <td></td> </tr> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE →			← 180 Ringing			← 200 OK INVITE			ACK →			Apply post test routine	
SIP (Network A)	Interconnection interface	SIP (Network B)																	
	INVITE →																		
	← 180 Ringing																		
	← 200 OK INVITE																		
	ACK →																		
	Apply post test routine																		
Comments	Establish a communication from Network A to Network B Check: If the Route header is present in the ACK, the topmost header or entry is set to the address of the IBCF of Network B. Repeat this test in reverse direction. Repeat this test with all chosen end devices.																		

Test case number	SS_bcall_NNI_016									
Test case group	BCALL/successful									
Reference	[IETF RFC 3261] and [IETF RFC 3264]									
SELECTION EXPRESSION										
Test purpose	Handling of SDP parameters in the INVITE. Ensure that call establishment and the correct handling of the SDP parameters of the INVITE message defined as: TYPE_SDP is performed correctly. Ensure that in the active call state the voice/data transfer on the media channels is performed correctly (e.g., testing QoS parameters). In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in Table 9.1.1-1 applies.									
Configuration										
SIP parameter	INVITE: Content-Type: application/sdp m=audio <Port number> RTP/AVP TYPE_SDP= PIXIT (Table 9-1) <i>or</i> m= Image <Port number> Udptl <i>or</i> Tcptl TYPE_SDP= PIXIT (Table 9-1) a=TYPE_SDP= PIXIT (Table 9-1) b=TYPE_SDP= PIXIT (Table 9-1)									
Message flow	<table style="width: 100%; border: none;"> <tr> <td style="text-align: center; width: 33%;">SIP (Network A)</td> <td style="text-align: center; width: 33%;">Interconnection Interface</td> <td style="text-align: center; width: 33%;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE →</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">Apply post test routine</td> <td></td> </tr> </table>	SIP (Network A)	Interconnection Interface	SIP (Network B)		INVITE →			Apply post test routine	
SIP (Network A)	Interconnection Interface	SIP (Network B)								
	INVITE →									
	Apply post test routine									
Comments	Establish a communication from Network A to Network B Check: Is the preferred codec set to TYPE_SDP? Check: If present: Is the a line set to TYPE_SDP? Check: If present: Is the b line set to TYPE_SDP? Check: Is the codec list consistent with the attribute(s) (bandwidth) regarding the media description? Repeat this test in reverse direction. Repeat this test with all chosen end devices.									

Test case number	SS_bcall_NNI_017									
Test case group	BCALL/successful									
Reference	[IETF RFC 3261] and [IETF RFC 3264]									
SELECTION EXPRESSION										
Test purpose	<p>The SDP answer is sent in the 200 OK.</p> <p>Ensure that the call establishment performed correctly.</p> <p>The initial INVITE contains an SDP with the offer 1 according Table 9-1.</p> <p>Ensure that answer related to the SDP offer is contained in the 200 OK INVITE message.</p> <p>Ensure that in the confirmed state the voice transfer on the media and B-channels is performed correctly..</p>									
Configuration										
SIP parameter										
Message flow	<table style="width: 100%; border: none;"> <tr> <td style="width: 30%; text-align: center;">SIP (Network A)</td> <td style="width: 40%; text-align: center;">Interconnection Interface</td> <td style="width: 30%; text-align: center;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;"> INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → </td> <td></td> </tr> <tr> <td colspan="3" style="text-align: center;">Apply post test routine</td> </tr> </table>	SIP (Network A)	Interconnection Interface	SIP (Network B)		INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK →		Apply post test routine		
SIP (Network A)	Interconnection Interface	SIP (Network B)								
	INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK →									
Apply post test routine										
Comments	<p>Establish a communication from Network A to Network B.</p> <p>Check: Is the SDP answer contained in the 200 OK INVITE?</p> <p>See (Note)</p> <p>Repeat this test in reverse direction.</p> <p>Repeat this test with all chosen end devices.</p>									
NOTE – An SDP answer could be present in a provisional response.										

Test case number	SS_bcall_NNI_018									
Test case group	BCALL/successful									
Reference	[IETF RFC 3261] and [IETF RFC 3264]									
SELECTION EXPRESSION										
Test purpose	<p>First response 200 OK INVITE.</p> <p>Ensure that call establishment and the correctly if the called user answers with a 200 OK message.</p>									
Configuration										
SIP parameter										
Message flow	<table style="width: 100%; border: none;"> <tr> <td style="width: 30%; text-align: center;">SIP (Network A)</td> <td style="width: 40%; text-align: center;">Interconnection interface</td> <td style="width: 30%; text-align: center;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;"> INVITE → ← 200 OK INVITE ACK → </td> <td></td> </tr> <tr> <td colspan="3" style="text-align: center;">Apply post test routine</td> </tr> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE → ← 200 OK INVITE ACK →		Apply post test routine		
SIP (Network A)	Interconnection interface	SIP (Network B)								
	INVITE → ← 200 OK INVITE ACK →									
Apply post test routine										
Comments	<p>Establish a communication from Network A to Network B</p> <p>Check: Is it possible to confirm a session without early dialogue?</p> <p>Repeat this test in reverse direction.</p> <p>Repeat this test with all chosen end devices.</p>									

Table 9-1

TYPE_SDP		m= line		b= line	a= line
VA	<media >	<transport>	<fmt-list>	<modifier>:<bandwidth-value> (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 or rtpmap 0 PCMU/8000
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 or rtpmap 8 PCMA/8000
VA_04	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMA/8000
VA_05	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> CLEARMODE
VA_06	audio	RTP/AVP	Dynamic PT		rtpmap:<dynamic-PT> AMR-WB/16000/1
VA_07	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> AMR/8000/1

NOTE – <bandwidth value> for <modifier> of AS is evaluated to be B kbit/s.

Test case number	SS_bcall_NNI_019																																							
Test case group	BCALL/successful																																							
Reference	4.9, N/[ETSI TS 124 229]																																							
SELECTION EXPRESSION	[Network A] SE 47 AND [Network A] SE 4 AND [Network B] SE 4																																							
Test purpose	Overlap sending, the Multiple INVITE method is used. Ensure that call establishment using overlap sending is performed correctly. Ensure that in the confirmed state the voice transfer on the media and B-channels is performed correctly.																																							
Configuration																																								
SIP parameter																																								
Message flow	<table border="0" style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%;">SIP (Network A)</th> <th style="text-align: center; width: 40%;">Interconnection interface</th> <th style="text-align: right; width: 30%;">SIP (Network B)</th> </tr> </thead> <tbody> <tr> <td></td> <td style="text-align: center;">INVITE (CSq 1)</td> <td style="text-align: right;">➔</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE (CSq 2)</td> <td style="text-align: right;">➔</td> </tr> <tr> <td style="text-align: right;">←</td> <td style="text-align: center;">484 Address Incomplete (CSq 1)</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">ACK</td> <td style="text-align: right;">➔</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE (CSq 3)</td> <td style="text-align: right;">➔</td> </tr> <tr> <td style="text-align: right;">←</td> <td style="text-align: center;">484 Address Incomplete (CSq 2)</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">ACK</td> <td style="text-align: right;">➔</td> </tr> <tr> <td></td> <td style="text-align: center;">.....</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE (CSq 4)</td> <td style="text-align: right;">➔</td> </tr> <tr> <td style="text-align: right;">←</td> <td style="text-align: center;">484 Address Incomplete (CSq 3)</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">ACK</td> <td style="text-align: right;">➔</td> </tr> <tr> <td style="text-align: right;">←</td> <td style="text-align: center;">180 Ringing (CSq 4)</td> <td></td> </tr> </tbody> </table> <p style="text-align: center;">Apply post test routine</p>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE (CSq 1)	➔		INVITE (CSq 2)	➔	←	484 Address Incomplete (CSq 1)			ACK	➔		INVITE (CSq 3)	➔	←	484 Address Incomplete (CSq 2)			ACK	➔				INVITE (CSq 4)	➔	←	484 Address Incomplete (CSq 3)			ACK	➔	←	180 Ringing (CSq 4)	
SIP (Network A)	Interconnection interface	SIP (Network B)																																						
	INVITE (CSq 1)	➔																																						
	INVITE (CSq 2)	➔																																						
←	484 Address Incomplete (CSq 1)																																							
	ACK	➔																																						
	INVITE (CSq 3)	➔																																						
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	ACK	➔																																						
																																							
	INVITE (CSq 4)	➔																																						
←	484 Address Incomplete (CSq 3)																																							
	ACK	➔																																						
←	180 Ringing (CSq 4)																																							
Comments	Establish a communication from ISDN to SIP using the overlap operation in ISDN Check: All INVITE requests contain the same Call ID and From header values. SIP answers with 180 Ringing. Repeat this test in reverse direction.																																							

Test case number	SS_bcall_NNI_020	
Test case group	BCALL/successful	
Reference	4.9, N/[ETSI TS 124 229]	
SELECTION EXPRESSION	[Network A] SE 47 AND [Network A] SE 5 AND [Network B] SE 5	
Test purpose	Overlap sending, the in-Dialogue method is used Ensure that call establishment using overlap sending is performed correctly. Ensure that in the confirmed state the voice transfer on the media and B-channels is performed correctly.	
Configuration		
SIP parameter	INVITE 2: Supported: 100rel 183:Require: 100rel INFO: Content-Type: application/x-session-info SubsequentDigit: <additional digits>	
Message flow		
SIP (Network A)	Interconnection interface	SIP (Network B)
	INVITE (CSq 1) 1	→
←	484 Address Incomplete (CSq 1)	
	ACK	→
	INVITE (CSq 2) 2	→
←	183 Session Progress (CSq 2)	
	PRACK	→
←	200 OK PRACK	
	INFO	→
←	200 OK INFO	
	
	INFO	→
←	200 OK INFO	
←	180 Ringing (CSq 2)	
	Apply post test routine	
Comments	Establish a communication from ISDN to SIP using the overlap operation in ISDN Check: All INVITE requests contains the same Call ID and From header values. Check: The 183 Session Progress that establishes an early dialogue contains a Require header set to 100rel. Check: All INFO requests contain the Content-Type header set to 'application/x-session-info'. Check: All INFO requests contains the 'SubsequentDigit:' multipurpose Internet mail extensions (MIME) body containing the additional digits. The UE B answers with 180 Ringing response after the INVITE was received. Repeat this test in reverse direction.	

Test case number	SS_bcall_NNI_025																		
Test case group	BCALL/successful																		
Reference	5.1.2.3/[ETSI TS 183 036]																		
SELECTION EXPRESSION	[Network B] (SE 46 OR SE 47) AND [Network B] SE 6																		
Test purpose	PSTN XML ProgressIndicator element in the 200. User B is located in Network B and an ISDN end device is used. Ensure that the 200 OK INVITE response contains a PSTN XML MIME body and at least one ProgressIndicator element is present.																		
Configuration	User B is an ISDN access either in the PSTN or the SIP – ISDN interworking according [ETSI TS 183 036] applies																		
SIP parameter	200: Content-Type: application/vnd.etsi.pstn+xml Content-Disposition: signal;handling=optional <?xml version="1.0" encoding="utf-8"?> PSTN ProgressIndicator ProgressOctet3 CodingStandard>00< Location>yyyy< ProgressOctet4 ProgressDescription>0000111<																		
Message flow	<table style="width: 100%; border: none;"> <tr> <td style="text-align: center; width: 30%;">SIP (Network A)</td> <td style="text-align: center; width: 40%;">Interconnection interface</td> <td style="text-align: center; width: 30%;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE →</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 180 Ringing</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 200 OK INVITE</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">ACK →</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">Apply post test routine</td> <td></td> </tr> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE →			← 180 Ringing			← 200 OK INVITE			ACK →			Apply post test routine	
SIP (Network A)	Interconnection interface	SIP (Network B)																	
	INVITE →																		
	← 180 Ringing																		
	← 200 OK INVITE																		
	ACK →																		
	Apply post test routine																		
Comments	Check: Is a PSTN XML MIME body contained in the 200 OK INVITE response? Check: Is a ProgressIndicator element present and the ProgressDescription element is set to '0000110'? Repeat this test in reverse direction.																		

Test case number	SS_bcall_NNI_026	
Test case group	BCALL/successful	
Reference	5.1.1.1.2/[ETSI TS 183 036]	
SELECTION EXPRESSION	[Network A] (SE 46 OR SE 47) AND [Network A] SE 6	
Test purpose	PSTN XML BearerCapability Fallback connection type element in the INVITE. User A is located in Network A and an ISDN end device is used. Ensure that the INVITE request contains a PSTN XML MIME body and one BearerCapability element is present the InformationTransferCabability element is set to '00000' and one InformationTransferCabability element is set to '10001'.	
Configuration	User A is an ISDN access either in the PSTN or the SIP – ISDN interworking according [ETSI TS 183 036] applies	
SIP parameter	INVITE: Content-Type: application/vnd.etsi.pstn+xml Content-Disposition: signal;handling=optional <?xml version="1.0" encoding="utf-8"?> PSTN BearerCapability BCoctet3 CodingStandard>00< InformationTransferCabability>00000< BearerCapability BCoctet3 CodingStandard>00< InformationTransferCabability>10001<	
Message flow	<div style="display: flex; justify-content: space-between; align-items: center;"> <div style="text-align: center;">SIP (Network A)</div> <div style="text-align: center;"> Interconnection interface INVITE → Apply post test routine </div> <div style="text-align: center;">SIP (Network B)</div> </div>	
Comments	Check: Is a PSTN XML MIME body contained in the INVITE request? Check: Is the first BearerCapability InformationTransferCabability element set as indicated to '00000'? Check: Is the second BearerCapability InformationTransferCabability element set as indicated to '10001'? Check: Is the InformationTransferCabability element value consistent with the codec list in the SDP? Check: Is the InformationTransferCabability element value consistent with the bandwidth information in the SDP? Repeat this test in reverse direction.	

Test case number	SS_bcall_NNI_027																		
Test case group	BCALL/successful																		
Reference	5.1.2.3/[ETSI TS 183 036]																		
SELECTION EXPRESSION	[Network B] (SE 46 OR SE 47) AND [Network B] SE 6																		
Test purpose	Fall back does not occur. User B is located in Network B and an ISDN end device is used. The Fallback connection type was requested in the initial INVITE request. Ensure that the 200 OK INVITE response contains a PSTN XML MIME body and a BearerCapability element is present the InformationTransferCabability element set to '10001'.																		
Configuration	User B is an ISDN access either in the PSTN or the SIP – ISDN interworking according [ETSI TS 183 036] applies																		
SIP parameter	200: Content-Type: application/vnd.etsi.pstn+xml Content-Disposition: signal;handling=optional <?xml version="1.0" encoding="utf-8"?> PSTN BearerCapability BCoectet3 CodingStandard>00< InformationTransferCabability>10001<																		
Message flow	<table style="width: 100%; border: none;"> <tr> <td style="text-align: center; width: 33%;">SIP (Network A)</td> <td style="text-align: center; width: 33%;">Interconnection interface</td> <td style="text-align: center; width: 33%;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE →</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 180 Ringing</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 200 OK INVITE</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">ACK →</td> <td></td> </tr> <tr> <td></td> <td colspan="2" style="text-align: center;">Apply post test routine</td> </tr> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE →			← 180 Ringing			← 200 OK INVITE			ACK →			Apply post test routine	
SIP (Network A)	Interconnection interface	SIP (Network B)																	
	INVITE →																		
	← 180 Ringing																		
	← 200 OK INVITE																		
	ACK →																		
	Apply post test routine																		
Comments	<p>Check: Is a PSTN XML MIME body contained in the 200 OK INVITE response?</p> <p>Check: Is a BearerCapability element present, the InformationTransferCabability element set to '10001'?</p> <p>Check: Is the InformationTransferCabability element value consistent with the codec list in the SDP?</p> <p>Check: Is the InformationTransferCabability element value consistent with the bandwidth information in the SDP?</p> <p>Repeat this test in reverse direction.</p>																		

Test case number	SS_bcall_NNI_029	
Test case group	BCALL/successful	
Reference	[ETSI TS 124 628]	
SELECTION EXPRESSION	[Network B] SE 17b	
Test purpose	Handling of multiple early dialogues. Ensure that in case of forking in Network B the early dialogues are handled in a proper way. When a 200 OK INVITE is received, the remaining early dialogues shall be cancelled.	
Configuration	User B has registered three end devices under the same identity.	
SIP parameter		
Message flow		
SIP (Network A)	Interconnection interface	SIP (Network B)
	INVITE	→
	← 180 Ringing 1	
	← 180 Ringing 2	
	← 180 Ringing 3	
	← 200 OK INVITE 3	
	ACK	→
	Communication	
CASE A	← BYE 3	
	200 OK BYE 3	→
CASE B		BYE 3
	← 200 OK BYE 3	→
Comments	Establish a communication from Network A to Network B Check: Ensure that several provisional responses with different 'To' tags are sent from Network B to Network A. Repeat this test in reverse direction.	

9.1.2 – Test purposes for Basic call, DTMF transport

Test case number	SS_DTMF_1
Test case group	BCALL/successful
Reference	5.1.2.3/[IETF RFC 4733]
SELECTION EXPRESSION	SE 35 SE 36 SE 37
Test purpose	Transmission of DTMF Ensure that the ability of transmission of DTMF can be performed by the originating and destination user. The transmission can be done by: <ul style="list-style-type: none"> • DTMF in the RTP stream • Either by indicating in the SDP offer in the RTP stream • Or by the SIP INFO/NOTIFY method for DTMF tone generation DTMF test: The DTMF test should consist DTMF tones (70 ms signal, 100 ms pause) and shall contain the tones 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, *, #. The transmission shall be tested in both directions.

Configuration																																					
SIP parameter	<p>INVITE: CASE A</p> <p>m=audio <Port> RTP/AVP <dynamic-PT></p> <p>CASE B</p> <p>m=audio <Port> RTP/AVP <dynamic-PT> a=rtpmap <dynamic-PT> telephone-event/8000 a=rtpmap <dynamic-PT> 0-15</p> <p>CASE C</p> <p>INFO Content-Type: application/dtmf 'x'</p> <p>or</p> <p>INFO Content-Type: application/dtmf-relay Signal=x Duration=y</p>																																				
Message flow	<table border="0" style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%; border-bottom: 1px solid black;">SIP (Network A)</th> <th style="text-align: center; width: 40%; border-bottom: 1px solid black;">Interconnection interface</th> <th style="text-align: right; width: 30%; border-bottom: 1px solid black;">SIP (Network B)</th> </tr> </thead> <tbody> <tr> <td></td> <td style="text-align: center;">INVITE →</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 180 Ringing</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 200 OK INVITE</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">ACK →</td> <td></td> </tr> <tr> <td>CASE A</td> <td style="text-align: center;">RTP DTMF inband</td> <td></td> </tr> <tr> <td>CASE B</td> <td style="text-align: center;">RTP DTMF events</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">INFO →</td> <td></td> </tr> <tr> <td>CASE C</td> <td style="text-align: center;">← 200 OK INFO</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">INFO →</td> <td></td> </tr> <tr> <td>CASE D</td> <td style="text-align: center;">← 200 OK INFO</td> <td></td> </tr> <tr> <td colspan="3" style="text-align: center;">Apply post test routine</td> </tr> </tbody> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE →			← 180 Ringing			← 200 OK INVITE			ACK →		CASE A	RTP DTMF inband		CASE B	RTP DTMF events			INFO →		CASE C	← 200 OK INFO			INFO →		CASE D	← 200 OK INFO		Apply post test routine		
SIP (Network A)	Interconnection interface	SIP (Network B)																																			
	INVITE →																																				
	← 180 Ringing																																				
	← 200 OK INVITE																																				
	ACK →																																				
CASE A	RTP DTMF inband																																				
CASE B	RTP DTMF events																																				
	INFO →																																				
CASE C	← 200 OK INFO																																				
	INFO →																																				
CASE D	← 200 OK INFO																																				
Apply post test routine																																					
Comments	<p>Establish a communication from Network A to Network B</p> <p>Check: Case B: Is the dynamic payload type 'telephone-event' present in the SDP offer?</p> <p>Check: Case B: Is the dynamic payload type 'telephone-event' covered in the RTP stream if the telephone event occurs?</p> <p>Check: Case C: Does the Content-Type header field in the INFO request conveying the DTMF signal set to 'application/dtmf'?</p> <p>Check: Case C: Does the MIME body of the INFO request covering the TMF signal contain the events regarding the used content type?</p> <p>Check: Case D: Does the Content-Type header field in the INFO request conveying the DTMF signal set to 'application/dtmf-relay'?</p> <p>Check: Case C: Does the MIME body of the INFO request covering the TMF signal contain the events and duration regarding the used content type?</p> <p>Repeat this test in reverse direction.</p>																																				

Test case number	SS_bcall_FAX_2									
Test case group	BCALL/successful									
Reference	[IETF RFC 3264] and [ITU-T V.152]									
SELECTION EXPRESSION	[Network A] SE 44 AND [Network A] SE 44									
Test purpose	<p>Fax transmission using the ITU-T V.152 codec.</p> <p>Ensure that a fax transmission is possible from Network A to Network B and the relevant codec is the ITU-T V.152 codec.</p> <p>Ensure in the active call state the property of fax transmission.</p> <p>The call establishment procedures based on SIP/SDP and ITU-T H.248 for a real-time fax over IP service are described in [ITU-T Q.4016].</p>									
Configuration										
SIP parameter	<p>INVITE: SDP m=audio <Port> RTP/AVP 8 <dynamic-PT> a=rtpmap <dynamic-PT> PCMA/8000 a=gpmd; vbd=yes</p> <p>180/200 OK INVITE: SDP m=audio <Port> RTP/AVP <dynamic-PT> a=rtpmap <dynamic-PT> PCMA/8000 a=gpmd; vbd=yes</p>									
Message flow	<table style="width: 100%; border: none;"> <tr> <td style="text-align: center; vertical-align: top;">SIP (Network A)</td> <td style="text-align: center; vertical-align: top;">Interconnection interface</td> <td style="text-align: center; vertical-align: top;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;"> INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → </td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">Apply post test routine</td> <td></td> </tr> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK →			Apply post test routine	
SIP (Network A)	Interconnection interface	SIP (Network B)								
	INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK →									
	Apply post test routine									
Comments	<p>Establish a communication from Network A to Network B</p> <p>Check: Contains the SDP offer in the initial INVITE a voiceband data codec?</p> <p>Check: Contains the SDP answer in the 180 or 200 OK INVITE a voiceband data codec?</p> <p>Check: Is fax transmission successful?</p> <p>Repeat this test in reverse direction.</p>									

9.1.4 SIP-I tests

Test case number	SS_bcall_SIP-I_01	
Test case group	BCALL/successful	
Reference	7.1/[ITU-T Q.1912.5]	
SELECTION EXPRESSION	[Network A] SE 17 AND [Network A] SE 47	
Test purpose	<p>SIP-I support, Basic call, initial address message (IAM) present in the INVITE request. Ensure that when a call initiated in the PSTN or the PLMN and the ISUP – SIP-I interworking is applicable in the originating network, an ISUP IAM is encapsulated in the initial INVITE request.</p> <p>Ensure that all the mandatory parameters in the IAM are present and the values are valid and the Transmission medium requirement parameter is consistent with the SDP.</p>	
Configuration		
SIP parameter	INVITE: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Nature of connection indicators Forward call indicators Calling party's category Transmission medium requirement Called party number <i>Calling party number (optional)</i> <i>Optional forward call indicators (optional)</i> <i>Hop counter (optional)</i> <i>User service information (optional)</i> <i>Access transport (optional)</i> --[any boundary name]--	
Message flow		
SIP (Network A)	Interconnection interface INVITE (IAM) → ← 100 Trying Apply post test routine	SIP (Network B)
Comments	Establish a communication from Network A to Network B Check: Is an ISUP IAM encapsulated in the INVITE request? Check: Are all the mandatory ISUP parameters present in the IAM and are the values valid? Check: Are the values of the optional parameters in the encapsulated IAM valid? Check: Is the 'm' line with corresponding attributes in the SDP consistent with the Transmission medium requirement parameter? Check: Is the Transmission medium requirement value consistent with the bandwidth information in the SDP? Repeat this test with all possible IAM USI and ATP combinations as indicated in Table 9-3 Repeat this test in reverse direction.	

Table 9-3 – IAM parametrization

ITC_value	IAM USI	ATP
ITC_VA_1	Speech	HLC: telephony
ITC_VA_2	3,1 kHz audio	No HLC
ITC_VA_3	3,1 kHz audio	HLC: facsimile group 2/3
ITC_VA_4	3,1 kHz audio	LLC: 3,1 kHz audio, voiceband data via modem, synchronous mode, user rate 2,4 kbit/s
ITC_VA_5	unrestricted digital information	HLC: facsimile group 4
ITC_VA_6	unrestricted digital information	HLC: facsimile group 4, LLC: telematic_term
ITC_VA_7	Speech	No HLC
ITC_VA_8	unrestricted digital information	No HLC

Test case number	SS_bcall_ SIP-I_02	
Test case group	BCALL/successful	
Reference	Clause 7.2.1/[ITU-T Q.1912.5]	
SELECTION EXPRESSION	[Network A] SE 4 AND SE 17 AND [Network A] SE 47	
Test purpose	SIP-I support, Basic call, overlap signalling. Ensure that when overlap signalling applies in the ISUP -SIP-I interworking in the originating network, several INVITE requests with the same Cal-ID and From tag are sent from Network A to Network B. Ensure that the original IAM is encapsulated in any INVITE request.	
Configuration		
SIP parameter		
Message flow		
SIP (Network A)	Interconnection interface	SIP (Network B)
	INVITE (1) →	
←	484 Address Incomplete (1)	→
	ACK →	
	INVITE (2) →	
←	484 Address Incomplete (2)	→
	ACK →	
	INVITE (3) →	
←	484 Address Incomplete (3)	→
	ACK →	
	.	
	INVITE (4) →	
←	180 Ringing (4)	
	Apply post test routine	
Comments	Establish a communication from Network A to Network B using the overlap procedure in Network A Check: Are the INVITE requests sent with the same From tag and the Call-ID? Check: After the 180 applies, are all previous INVITE transactions terminated with a 484 final response? Check: Is the encapsulated IAM present in the initial INVITE request also encapsulated in any following INVITE request required for the call setup? Repeat this test in reverse direction.	

Test case number	SS_bcall_SIP-I_03	
Test case group	BCALL/successful	
Reference	6.5/[ITU-T Q.1912.5]	
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47	
Test purpose	SIP-I support, Basic call, address complete message (ACM) present in the 180 response. Ensure that on receipt of a 180 Ringing provisional response and an SIP-I – ISUP interworking is applicable in the terminating network the Backward call indicators parameter in the encapsulated ACM is present and the values are valid. Ensure that the values of the optional parameters in the encapsulated ACM are valid.	
Configuration		
SIP parameter	180: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM Backward call indicators Called party's status indicator= subscriber free --[any boundary name]--	
Message flow		
SIP (Network A)	Interconnection interface INVITE → ← 100 Trying ← 180 Ringing(ACM) Apply post test routine	SIP (Network B)
Comments	Establish a communication from Network A to Network B Check: Is an ISUP ACM message encapsulated in the 180 Ringing provisional response? Check: Is the mandatory Backward call indicators parameter present in the encapsulated ISUP ACM and are the values valid? Check: Are the values of optional parameters in the encapsulated ISUP ACM valid? Check: If an SDP answer is present in the 180, are the codec and the bandwidth information in the 'a' attributes consistent with Transmission medium requirement in the encapsulated IAM of the INVITE request? Check: Can the ringing tone be heard from the terminating side? Repeat this test in reverse direction.	

Test case number	SS_bcall_SIP-I_04	
Test case group	BCALL/successful	
Reference	6.5/[ITU-T Q.1912.5]	
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47	
Test purpose	<p>SIP-I support. Basic call, early ACM present in the 183 response.</p> <p>Ensure that on receipt of a 183 Session Progress provisional response and an SIP-I - ISUP interworking is applicable in the terminating network the Backward call indicators parameter in the encapsulated ACM is present and the value of the Called party's status indicator is set to 'no indication'.</p> <p>Ensure that the values of the optional parameters in the encapsulated ACM are valid.</p>	
Configuration	Select a proper destination that sends an early ACM in the PSTN/PLMN (e.g., announcement)	
SIP parameter	<p>183:</p> <p>Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name]</p> <p>Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required</p> <p>ACM</p> <p>Backward call indicators</p> <p>Called party's status indicator= no indication Optional backward call indicator Inband info or appropriate pattern is now available Access Transport (optional) Progress Indicator Progress description = Destination address is non ISDN --[any boundary name]--</p>	
Message flow		
SIP (Network A)	<p>Interconnection interface</p> <p>INVITE →</p> <p>← 100 Trying</p> <p>← 183 Session Progress (ACM)</p> <p>Apply post test routine</p>	SIP (Network B)
Comments	<p>Establish a communication from Network A to Network B</p> <p>Check: Is an ISUP ACM message encapsulated in the 183 Session Progress provisional response?</p> <p>Check: Is the mandatory Backward call indicators parameter present in the encapsulated ISUP ACM and are the values valid?</p> <p>Check: Is the Called party's status indicator in the encapsulated ISUP ACM set to 'no indication'?</p> <p>Check: Can an early media (e.g., announcement) be heard from the terminating side?</p> <p>Check: Are the values of optional parameters in the encapsulated ISUP ACM valid?</p> <p>Repeat this test in reverse direction.</p>	

Test case number	SS_bcall_SIP-I_04 A	
Test case group	BCALL/successful	
Reference	6.5/[ITU-T Q.1912.5]	
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47	
Test purpose	<p>SIP-I support. Basic call, early ACM present in the 183 response.</p> <p>Ensure that on receipt of a 183 Session Progress provisional response and an SIP-I – ISUP interworking is applicable in the terminating network the Backward call indicators parameter in the encapsulated ACM is present and the value of the Called party's status indicator is set to 'no indication'.</p> <p>Ensure that the values of the optional parameters in the encapsulated ACM are valid.</p>	
Configuration	Select a proper destination that sends an early ACM in the PSTN/PLMN (e.g., announcement)	
SIP parameter	183: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM Backward call indicators Called party's status indicator= no indication Access transport = PI # 3 originating address is non-ISDN --[any boundary name]--	
Message flow		
SIP (Network A)	Interconnection interface INVITE → ← 100 Trying ← 183 Session Progress (ACM) Apply post test routine	SIP (Network B)
Comments	Establish a communication from Network A to Network B Check: Is an ISUP ACM message encapsulated in the 183 Session Progress provisional response? Check: Is the mandatory Backward call indicators parameter present in the encapsulated ISUP ACM and are the values valid? Check: Is the Called party's status indicator in the encapsulated ISUP ACM set to 'no indication'? Check: Are the values of optional parameters in the encapsulated ISUP ACM valid? Repeat this test in reverse direction.	

Test case number	SS_bcall_SIP-I_05	
Test case group	BCALL/successful	
Reference	6.6/[ITU-T Q.1912.5]	
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47	
Test purpose	SIP-I support. Basic call, call progress message (CPG) present in a 180 response. Ensure that on receipt of a 180 Ringing provisional response and an SIP-I – ISUP interworking is applicable in the terminating network the Event indicator in the encapsulated CPG is present and set to 'ALERTING'. Ensure that the values of the optional parameters in the encapsulated CPG are valid.	
Configuration	Select a proper destination that sends at first an early ACM and after then a CPG 'ALERTING' in the PSTN/PLMN (e.g., PBX).	
SIP parameter	180: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Event indicator = ALERTING --[any boundary name]--	
Message flow		
SIP (Network A)	Interconnection interface INVITE ← 100 Trying ← 183 Session Progress (ACM) ← 180 Ringing (CPG) Apply post test routine	SIP (Network B) →
Comments	Establish a communication from Network A to Network B Check: Is an ISUP CPG message encapsulated in the 180 Ringing provisional response? Check: Is the mandatory Event indicator present in the encapsulated ISUP CPG set to 'ALERTING'? Check: Are the values of optional parameters in the encapsulated ISUP CPG valid? Repeat this test in reverse direction.	

Test case number	SS_bcall_SIP-I_05 A	
Test case group	BCALL/successful	
Reference	Clause 6.6/[ITU-T Q.1912.5]	
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47	
Test purpose	SIP-I support. Basic call, CPG present in a 180 response. Ensure that on receipt of a 180 Ringing provisional response and an SIP-I – ISUP interworking is applicable in the terminating network the Event indicator in the encapsulated CPG is present and set to 'ALERTING'. Ensure that the values of the optional parameters in the encapsulated CPG are valid.	
Configuration	Select a proper destination that sends at first an early ACM and after then a CPG 'ALERTING' in the PSTN/PLMN (e.g., PBX).	
SIP parameter	180: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Event indicator = ALERTING Access transport = PI # 3 originating address is non-ISDN --[any boundary name]--	
Message flow		
SIP (Network A)	Interconnection interface	SIP (Network B)
	INVITE	➔
	←	100 Trying
	←	183 Session Progress (ACM)
	←	180 Ringing (CPG)
	Apply post test routine	
Comments	Establish a communication from Network A to Network B Check: Is an ISUP CPG message encapsulated in the 180 Ringing provisional response? Check: Is the mandatory Event indicator present in the encapsulated ISUP CPG set to 'ALERTING'? Check: Are the values of optional parameters in the encapsulated ISUP CPG valid? Repeat this test in reverse direction.	

Test case number	SS_bcall_SIP-I_06	
Test case group	BCALL/successful	
Reference	Clause 6.7/[ITU-T Q.1912.5]	
SELECTION EXPRESSION	[Network B] SE 17 [Network B] AND SE 47	
Test purpose	SIP-I support. Basic call, answer message (ANM) present in a 200 OK INVITE response. Ensure that on receipt of a 200 OK INVITE final response and an SIP-I – ISUP interworking is applicable in the terminating network the ISUP ANM is encapsulated in the 200 OK. Ensure that the values of the optional parameters in the encapsulated ANM are valid.	
Configuration		
SIP parameter	200 OK: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ANM --[any boundary name]--	
Message flow		
SIP (Network A)	Interconnection interface	SIP (Network B)
	INVITE	➔
	100 Trying	
	180 Ringing (ACM)	
	200 OK INVITE (ANM)	
	ACK	➔
	Apply post test routine	
Comments	Establish a confirmed communication from Network A to Network B Check: Is an ISUP ANM encapsulated in the 200 OK INVITE? Check: Are the values of optional parameters in the encapsulated ISUP ANM valid? Check: Ensure the property of speech. Check: Are the codec and the bandwidth information in the 'a' attributes consistent with Transmission medium requirement in the encapsulated IAM of the INVITE request? Repeat this test in reverse direction.	

Test case number	SS_bcall_SIP-I_07	
Test case group	BCALL/successful	
Reference	5.4.3.4, 6.11.2/[ITU-T Q.1912.5]	
SELECTION EXPRESSION	[Network A] SE 17 [Network A] AND SE 47	
Test purpose	<p>SIP-I support. Basic call, release (REL) present in a BYE request sent from the originating network.</p> <p>Ensure that a ISUP REL message is encapsulated in a BYE request sent in the release procedure initiated from the originating user when ISUP – SIP-I interworking is applicable in the originating network.</p> <p>Ensure the validity of the cause indicator in the encapsulated REL.</p> <p>Ensure that the ISUP release complete (RLC) is encapsulated in the 200 OK BYE.</p>	
Configuration		
SIP parameter	<p>BYE.</p> <p>Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required</p> <p>REL</p> <p>Cause value: --[any boundary name]--</p> <p>200 OK BYE</p> <p>Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required</p> <p>RLC</p> <p>--[any boundary name]--</p>	
Message flow		
SIP (Network A)	Interconnection interface	SIP (Network B)
	INVITE →	
	← 100 Trying	
	← 180 Ringing	
	← 200 OK INVITE	
	ACK →	
	Communication	
	BYE (REL) →	
	← 200 OK BYE (RLC)	
Comments	<p>Establish a confirmed communication from Network A to Network B</p> <p>The originating user terminates the communication</p> <p>Check: Is the ISUP REL encapsulated in the BYE request?</p> <p>Check: Are the cause indicators in the encapsulated ISUP REL valid?</p> <p>Check: If a Reason header is present in the BYE request, is the 'cause' value of Reason header equal to the 'Cause value' in the encapsulated REL?</p> <p>Check: Is the ISUP RLC encapsulated in the 200 OK BYE?</p> <p>Repeat this test in reverse direction.</p>	

Test case number	SS_bcall_SIP-I_08	
Test case group	BCALL/successful	
Reference	5.4.3.4, 6.11.2/[ITU-T Q.1912.5]	
SELECTION EXPRESSION	[Network B] SE 17 [Network B] AND SE 47	
Test purpose	<p>SIP-I support. Basic call, REL present in a BYE request sent from the terminating network.</p> <p>Ensure that a ISUP REL message is encapsulated in a BYE request sent in the release procedure initiated from the terminating user when SIP-I – ISUP interworking is applicable in the terminating network.</p> <p>Ensure the validity of the cause indicator in the encapsulated REL.</p> <p>Ensure that the ISUP RLC is encapsulated in the 200 OK BYE.</p>	
Configuration		
SIP parameter	<p>BYE.</p> <p>Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required</p> <p>REL</p> <p>Cause value: --[any boundary name]--</p> <p>200 OK BYE</p> <p>Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required</p> <p>RLC</p> <p>--[any boundary name]--</p>	
Message flow		
SIP (Network A)	Interconnection interface INVITE → ← 100 Trying ← 180 Ringing ← 200 OK INVITE ACK → Communication ← BYE (REL) → 200 OK BYE (RLC) →	SIP (Network B)
Comments	<p>Establish a confirmed communication from Network A to Network B</p> <p>The terminating user terminates the communication</p> <p>Check: Is the ISUP REL encapsulated in the BYE request?</p> <p>Check: Are the cause indicators in the encapsulated ISUP REL valid?</p> <p>Check: If a Reason header is present in the BYE request, is the 'cause' value of Reason header equal to the 'Cause value' in the encapsulated REL?</p> <p>Check: Is the ISUP RLC encapsulated in the 200 OK BYE?</p> <p>Repeat this test in reverse direction.</p>	

9.1.5 Codec negotiation

Test case number	SS_codec_001															
Test case group	BCALL/Codec_Negotiation															
Reference	[IETF RFC 4566], [IETF RFC 3261] and [IETF RFC 3264]															
SELECTION EXPRESSION																
Test purpose	<p>Session update requested by the calling user.</p> <p>During the session, the calling user decides to change the characteristics of the media session. This is accomplished by sending a re-INVITE or UPDATE containing a new media description. This re-INVITE or UPDATE references the existing dialogue so that the other party knows that it has 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.</p> <p>In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in Table 9.1.2-1 applies.</p>															
Configuration																
SIP parameter	SDP1: codec x chosen from Table 9-4 SDP3: codec y chosen from Table 9-4															
<p>Message flow</p> <table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="width: 30%; text-align: left;">SIP (Network A)</th> <th style="width: 40%; text-align: center;">Interconnection interface</th> <th style="width: 30%; text-align: right;">SIP (Network B)</th> </tr> </thead> <tbody> <tr> <td colspan="3" style="text-align: center;">A confirmed session already exists (SDP1)</td> </tr> <tr> <td style="vertical-align: top;">CASE A</td> <td style="text-align: center;"> INVITE (SDP3) → ← 200 OK INVITE (SDP4) ACK → </td> <td></td> </tr> <tr> <td style="vertical-align: top;">CASE B</td> <td style="text-align: center;"> UPDATE (SDP3) → ← 200 OK UPDATE (SDP4) </td> <td></td> </tr> <tr> <td colspan="3" style="text-align: center;">Apply post test routine</td> </tr> </tbody> </table>		SIP (Network A)	Interconnection interface	SIP (Network B)	A confirmed session already exists (SDP1)			CASE A	INVITE (SDP3) → ← 200 OK INVITE (SDP4) ACK →		CASE B	UPDATE (SDP3) → ← 200 OK UPDATE (SDP4)		Apply post test routine		
SIP (Network A)	Interconnection interface	SIP (Network B)														
A confirmed session already exists (SDP1)																
CASE A	INVITE (SDP3) → ← 200 OK INVITE (SDP4) ACK →															
CASE B	UPDATE (SDP3) → ← 200 OK UPDATE (SDP4)															
Apply post test routine																
Comments	Establish a communication from Network A to Network B using SDP1 chosen from Table 9-4 Check: The calling user changes the media description using INVITE request containing SDP3 codec chosen from Table 9-4 different to SDP1. Check: Is the codec list consistent with the attribute(s) (bandwidth) regarding the media description? Repeat this test in reverse direction.															

Test case number	SS_codec_002															
Test case group	BCALL/Codec_Negotiation															
Reference	[IETF RFC 4566], [IETF RFC 3261] and [IETF RFC 3264]															
SELECTION EXPRESSION																
Test purpose	<p>Session update requested by the called user.</p> <p>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 dialogue so that the other party knows that it has to modify an existing session instead of establishing a new session. The other party sends a 200 (OK) to accept the change.</p> <p>The requestor responds to the 200 (OK) with an ACK.</p> <p>In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in Table 9-4 applies.</p>															
Configuration																
SIP parameter	SDP1: codec x chosen from Table 9-4 SDP2: codec y chosen from Table 9-4															
Message flow	<table border="0" style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%;">SIP (Network A)</th> <th style="text-align: center; width: 40%;">Interconnection interface</th> <th style="text-align: right; width: 30%;">SIP (Network B)</th> </tr> </thead> <tbody> <tr> <td colspan="3" style="text-align: center;">A confirmed session already exists (SDP1)</td> </tr> <tr> <td style="vertical-align: top;">CASE A</td> <td style="text-align: center;"> INVITE (SDP3) → ← 200 OK INVITE (SDP4) ACK → </td> <td></td> </tr> <tr> <td style="vertical-align: top;">CASE B</td> <td style="text-align: center;"> UPDATE (SDP3) → ← 200 OK UPDATE (SDP4) </td> <td></td> </tr> <tr> <td colspan="3" style="text-align: center;">Apply post test routine</td> </tr> </tbody> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)	A confirmed session already exists (SDP1)			CASE A	INVITE (SDP3) → ← 200 OK INVITE (SDP4) ACK →		CASE B	UPDATE (SDP3) → ← 200 OK UPDATE (SDP4)		Apply post test routine		
SIP (Network A)	Interconnection interface	SIP (Network B)														
A confirmed session already exists (SDP1)																
CASE A	INVITE (SDP3) → ← 200 OK INVITE (SDP4) ACK →															
CASE B	UPDATE (SDP3) → ← 200 OK UPDATE (SDP4)															
Apply post test routine																
Comments	Establish a connection from SIP UE 1 to SIP UE 2 using SDP1 chosen from Table 9.-4 Check: The called user changes the media description using INVITE request containing SDP2 codec chosen from Table 9-4 different to SDP1. Check: Is the codec list consistent with the attribute(s) (bandwidth) regarding the media description? Repeat this test in reverse direction.															

Table 9-4

VARIABLE	PT	Encoding	Media type	Clock rate	Channels	Supported in Network A	Supported in Network B
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	6	DVI4	A	16 000	1		
VA_06	7	LPC	A	8 000	1		
VA_07	8	PCMA	A	8 000	1		
VA_08	9	G722	A	8 000	1		
VA_09	10	L16	A	44 100	2		
VA_10	11	L16	A	44 100	1		
VA_13	12	QCELP	A	8 000	1		
VA_12	13	CN	A	8 000	1		
VA_13	14	MPA	A	90 000			
VA_14	15	G728	A	18 000	1		
VA_15	16	DVI4	A	11 025	1		
VA_16	17	DVI4	A	22 050	1		
VA_17	18	G729	A	8 000	1		
VA_18	Dyn	G726-40	A	8 000	1		
VA_19	Dyn	G726-32	A	8 000	1		
VA_20	Dyn	G726-24	A	8 000	1		
VA_21	Dyn	G726-16	A	8 000	1		
VA_22	Dyn	G729D	A	8 000	1		
VA_23	Dyn	G729E	A	8 000	1		
VA_24	Dyn	GSM-EFR	A	8 000	1		
VA_25	25	CeIB	V	90 000			
VA_26	26	JPEG	V	90 000			
VA_27	28	Nv	V	90 000			
VA_28	31	H261	V	90 000			
VA_29	32	MPV	V	90 000			
VA_30	33	MP2T	V	90 000			
VA_31	34	H263	V	90 000			
VA_32	Dyn	H263-1998	V	90 000			
VA_33	Dyn	AMR	A	8 000	1		
VA_34	Dyn	AMR-WB	A	16 000	1		
VA_35	Dyn	telephone-event	A	8 000	1		

Test case number	SS_resource_001																											
Test case group	BCALL/Resource_Reservation																											
Reference	[IETF RFC 4566], [IETF RFC 3261], [IETF RFC 3264] and [IETF RFC 3312]																											
SELECTION EXPRESSION	([Network A] SE 7 AND [Network B] SE 7) AND ([User A] SE 42 AND [User B] SE 42)																											
Test purpose	<p>Resource reservation successful, segmented status. Ensure that the network is able to reserve resources for quality of service when requested from the initiating user.</p> <ul style="list-style-type: none"> In the INVITE the UE requests to establish QoS preconditions for all the media streams. In the 183 Session Progress the user agent server (UAS) supports the QoS preconditions and requests that user agent client (UAC) sends a confirmation when the QoS preconditions are met. The UPDATE includes in the SDP the information about the successful QoS bidirectional mode, due to the successful bidirectional packet data protocol (PDP) context established. 200 OK UPDATE the SDP contains an indication that the UE successfully reserved the QoS in the send and receive directions. 																											
Configuration																												
SIP parameter	<p>INVITE: Supported: 100rel precondition SDP1: m=audio <Port number> RTP/AVP <codec> a=curr:qos local none a=curr:qos remote none a=des:qos mandatory/optional local sendrecv a=des:qos none remote sendrecv</p> <p>183 Session Progress: Supported/Require: 100rel precondition SDP2: m=audio <Port number> RTP/AVP <codec> a=curr:qos local none a=curr:qos remote none a=des:qos mandatory/optional local sendrecv a=des:qos mandatory/optional remote sendrecv</p> <p>UPDATE SDP3: m=audio <Port number> RTP/AVP <codec> a=curr:qos local sendrecv a=curr:qos remote none a=des:qos mandatory/optional local sendrecv a=des:qos mandatory/optional remote sendrecv</p> <p>200 OK UPDATE SDP4: a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory/optional local sendrecv a=des:qos mandatory/optional remote sendrecv</p>																											
Message flow	<table border="0" style="width: 100%; text-align: center;"> <tr> <td style="width: 30%;">SIP (Network A)</td> <td style="width: 40%;">Interconnection interface</td> <td style="width: 30%;">SIP (Network B)</td> </tr> <tr> <td></td> <td>INVITE (SDP1) →</td> <td></td> </tr> <tr> <td>←</td> <td>183 Session Progress (SDP2)</td> <td>→</td> </tr> <tr> <td></td> <td>PRACK →</td> <td></td> </tr> <tr> <td>←</td> <td>200 OK PRACK</td> <td></td> </tr> <tr> <td></td> <td>Resource reservation</td> <td></td> </tr> <tr> <td></td> <td>UPDATE (SDP3) →</td> <td></td> </tr> <tr> <td>←</td> <td>200 OK UPDATE (SDP4)</td> <td></td> </tr> <tr> <td></td> <td>Apply post test routine</td> <td></td> </tr> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE (SDP1) →		←	183 Session Progress (SDP2)	→		PRACK →		←	200 OK PRACK			Resource reservation			UPDATE (SDP3) →		←	200 OK UPDATE (SDP4)			Apply post test routine	
SIP (Network A)	Interconnection interface	SIP (Network B)																										
	INVITE (SDP1) →																											
←	183 Session Progress (SDP2)	→																										
	PRACK →																											
←	200 OK PRACK																											
	Resource reservation																											
	UPDATE (SDP3) →																											
←	200 OK UPDATE (SDP4)																											
	Apply post test routine																											
Comments	<p>Establish a communication from Network A to Network B</p> <p>Check: Is the quality of service for the current state local and remote set to 'none' indicated in the SDP in the INVITE?</p> <p>Check: Is the quality of service for the desired state local and remote set to 'mandatory' and 'sendrecv' in the 183?</p> <p>Check: Is the quality of service for the current state local set to 'sendrecv' indicated in the SDP in the UPDATE?</p> <p>Check: Is the quality of service for the current state local and remote set to 'sendrecv' indicated in the SDP in the 200 OK UPDATE?</p> <p>Check: Is the codec in the codec list consistent with the attribute(s) (bandwidth) regarding the media description? At least a ITU-T G.711 codec is required.</p> <p>Repeat this test in reverse direction.</p> <p>Note – This test case is applicable with a VoLTE originator and termination</p>																											

Test case number	SS_resource_002																								
Test case group	BCALL/Resource_Reservation																								
Reference	[IETF RFC 4566], [IETF RFC 3261], [IETF RFC 3264] and [IETF RFC 3312]																								
SELECTION EXPRESSION	(Network A) SE 7 AND ([User A] SE 42 AND NOT [User B] SE 42)																								
Test purpose	<p>Resource reservation not supported.</p> <p>Ensure that the network is able to reserve resources for quality of service when requested from the initiating user. The terminating user does not support the precondition procedure.</p> <ul style="list-style-type: none"> In the INVITE the UE requests to establish QoS preconditions for all the media streams. In the 183 Session Progress: no support by the terminating UA is indicated. Or In the 180 Ringing: no support by the terminating UA is indicated. Or In the 200 OK INVITE: no support by the terminating UA is indicated. 																								
Configuration																									
SIP parameter	<p>INVITE: Supported: 100rel precondition SDP1: m=audio <Port number> RTP/AVP <codec> a=curr:qos local none a=curr:qos remote none a=des:qos mandatory/optional local sendrecv a=des:qos none remote sendrecv</p> <p>183 Session Progress: SDP2: m=audio <Port number> RTP/AVP <codec> Or 180 Ringing: SDP2: m=audio <Port number> RTP/AVP <codec> Or 200 OK: SDP2: m=audio <Port number> RTP/AVP <codec></p>																								
Message flow	<table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%;">SIP (Network A)</th> <th style="text-align: center; width: 40%;">Interconnection interface</th> <th style="text-align: right; width: 30%;">SIP (Network B)</th> </tr> </thead> <tbody> <tr> <td></td> <td style="text-align: center;">INVITE (SDP1)</td> <td style="text-align: right;">→</td> </tr> <tr> <td>CASE A</td> <td style="text-align: center;">←</td> <td style="text-align: right;">183 Session Progress (SDP2)</td> </tr> <tr> <td>CASE B</td> <td style="text-align: center;">←</td> <td style="text-align: right;">180 Ringing (SDP2)</td> </tr> <tr> <td>CASE C</td> <td style="text-align: center;">←</td> <td style="text-align: right;">180 Ringing</td> </tr> <tr> <td></td> <td style="text-align: center;">←</td> <td style="text-align: right;">200 OK INVITE (SDP2)</td> </tr> <tr> <td></td> <td style="text-align: center;">ACK</td> <td style="text-align: right;">→</td> </tr> <tr> <td></td> <td colspan="2" style="text-align: center;">Apply post test routine</td> </tr> </tbody> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE (SDP1)	→	CASE A	←	183 Session Progress (SDP2)	CASE B	←	180 Ringing (SDP2)	CASE C	←	180 Ringing		←	200 OK INVITE (SDP2)		ACK	→		Apply post test routine	
SIP (Network A)	Interconnection interface	SIP (Network B)																							
	INVITE (SDP1)	→																							
CASE A	←	183 Session Progress (SDP2)																							
CASE B	←	180 Ringing (SDP2)																							
CASE C	←	180 Ringing																							
	←	200 OK INVITE (SDP2)																							
	ACK	→																							
	Apply post test routine																								
Comments	<p>Establish a communication from Network A to Network B</p> <p>Check: Is the quality of service for the current state local and remote set to 'none' indicated in the SDP in the INVITE?</p> <p>Check: Is the support of Precondition not indicated in the 183 Session Progress (optional)</p> <p>Check: Is the support of Precondition not indicated in the 180 Ringing (optional)</p> <p>Check: Is the support of Precondition not indicated in the 200 OK INVITE</p> <p>Note – This test case is applicable with a VoLTE originator</p>																								

9.1.7 Test purposes for SIP-SIP, Basic call, unsuccessful

9.1.7.1 NNI

Test case number	SS_unsucc_NNI_001												
Test case group	BCALL/unsuccessful												
Reference	[IETF RFC 3261]												
SELECTION EXPRESSION													
Test purpose	Called number is not allocated in the assumed network. Ensure that, when calling to unallocated number, the network initiate call clearing to the calling user with a 404 Not Found message.												
Configuration													
SIP parameter													
Message flow	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 30%; text-align: right;">SIP (Network A)</td> <td style="width: 40%; text-align: center;">Interconnection interface</td> <td style="width: 30%; text-align: left;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE</td> <td style="text-align: right;">→</td> </tr> <tr> <td></td> <td style="text-align: center;">← 404 Not Found</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">ACK</td> <td style="text-align: right;">→</td> </tr> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE	→		← 404 Not Found			ACK	→
SIP (Network A)	Interconnection interface	SIP (Network B)											
	INVITE	→											
	← 404 Not Found												
	ACK	→											
Comments	<p>Establish a communication from Network A to Network B, called user number is not allocated in Network B</p> <p>Check: Is a 404 Not Found sent from Network B to Network A?</p> <p>Repeat this test in reverse direction.</p> <p>Repeat this test with all chosen end devices.</p>												

Test case number	SS_unsucc_NNI_002												
Test case group	BCALL/unsuccessful												
Reference	[IETF RFC 3261]												
SELECTION EXPRESSION													
Test purpose	Network B is unable to process the request. Ensure that the call will be released if the Service is unavailable. The network initiates call clearing to the calling user with a 503 Service unavailable message.												
Configuration													
SIP parameter													
Message flow	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 30%; text-align: right;">SIP (Network A)</td> <td style="width: 40%; text-align: center;">Interconnection interface</td> <td style="width: 30%; text-align: left;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE</td> <td style="text-align: right;">→</td> </tr> <tr> <td></td> <td style="text-align: center;">← 503 Service unavailable</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">ACK</td> <td style="text-align: right;">→</td> </tr> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE	→		← 503 Service unavailable			ACK	→
SIP (Network A)	Interconnection interface	SIP (Network B)											
	INVITE	→											
	← 503 Service unavailable												
	ACK	→											
Comments	<p>Establish a communication from Network A to Network B, Network B is unable to process the request.</p> <p>Check: Is a 503 Service unavailable sent from Network B to Network A?</p> <p>Repeat this test in reverse direction.</p> <p>Repeat this test with all chosen end devices.</p>												

Test case number	SS_unsucc_NNI__003												
Test case group	BCALL/unsuccessful												
Reference	[IETF RFC 3261]												
SELECTION EXPRESSION													
Test purpose	The called user is network determined busy. Ensure that, when the called user is busy, the network initiates call clearing to the calling user with a 486 Busy Here message.												
Configuration													
SIP parameter													
Message flow	<table style="width: 100%; border: none;"> <tr> <td style="text-align: center; width: 30%;">SIP (Network A)</td> <td style="text-align: center; width: 40%;">Interconnection interface</td> <td style="text-align: center; width: 30%;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE →</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 486 Busy Here</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">ACK →</td> <td></td> </tr> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE →			← 486 Busy Here			ACK →	
SIP (Network A)	Interconnection interface	SIP (Network B)											
	INVITE →												
	← 486 Busy Here												
	ACK →												
Comments	Establish a communication from Network A to Network B, user B is network determined user busy. Check: Is a 486 Busy Here sent from Network B to Network A? Repeat this test in reverse direction.												

Test case number	SS_unsucc_NNI__004												
Test case group	BCALL/unsuccessful												
Reference	[IETF RFC 3261]												
SELECTION EXPRESSION													
Test purpose	The called user is user determined busy. Ensure that, when the called user is busy, the user initiates call clearing to the calling user with a 486 Busy Here message.												
Configuration													
SIP parameter													
Message flow	<table style="width: 100%; border: none;"> <tr> <td style="text-align: center; width: 30%;">SIP (Network A)</td> <td style="text-align: center; width: 40%;">Interconnection interface</td> <td style="text-align: center; width: 30%;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE →</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 486 Busy Here</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">ACK →</td> <td></td> </tr> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE →			← 486 Busy Here			ACK →	
SIP (Network A)	Interconnection interface	SIP (Network B)											
	INVITE →												
	← 486 Busy Here												
	ACK →												
Comments	Establish a communication from Network A to Network B, user B is user determined user busy. Check: Is a 486 Busy Here sent from Network B to Network A? Repeat this test in reverse direction.												

Test case number	SS_unsucc_NNI__005												
Test case group	BCALL/unsuccessful												
Reference	[IETF RFC 3261]												
SELECTION EXPRESSION													
Test purpose	The called user is not available on the called number. Ensure that when the number is changed, the network initiate call clearing to the calling user with a 410 Gone message.												
Configuration													
SIP parameter													
Message flow	<table style="width: 100%; border: none;"> <tr> <td style="text-align: center; width: 30%;">SIP (Network A)</td> <td style="text-align: center; width: 40%;">Interconnection interface</td> <td style="text-align: center; width: 30%;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE →</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 410 Gone</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">ACK →</td> <td></td> </tr> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE →			← 410 Gone			ACK →	
SIP (Network A)	Interconnection interface	SIP (Network B)											
	INVITE →												
	← 410 Gone												
	ACK →												
Comments	Establish a communication from Network A to Network B, user B is not allocated in Network B. Check: Is a 410 Gone sent from Network B to Network A? Repeat this test in reverse direction.												

Test case number	SS_unsucc_NNI_006												
Test case group	BCALL/unsuccessful												
Reference	[IETF RFC 3261]												
SELECTION EXPRESSION													
Test purpose	The number of the called user is incomplete. Ensure that the call will be released when the called number is incomplete. The network initiates call clearing to the calling user with 484 Not Found message.												
Configuration													
SIP parameter													
Message flow	<table style="width: 100%; border: none;"> <tr> <td style="text-align: center; width: 30%;">SIP (Network A)</td> <td style="text-align: center; width: 40%;">Interconnection interface</td> <td style="text-align: center; width: 30%;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE</td> <td style="text-align: center;">→</td> </tr> <tr> <td></td> <td style="text-align: center;">← 484 Address Incomplete</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">ACK</td> <td style="text-align: center;">→</td> </tr> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE	→		← 484 Address Incomplete			ACK	→
SIP (Network A)	Interconnection interface	SIP (Network B)											
	INVITE	→											
	← 484 Address Incomplete												
	ACK	→											
Comments	Establish a communication from Network A to Network B, the called number is incomplete. Check: Is a 484 Address Incomplete sent from Network B to Network A? Repeat this test in reverse direction.												

Test case number	SS_unsucc_NNI_007																														
Test case group	BCALL/unsuccessful																														
Reference	[IETF RFC 4566], [IETF RFC 3261] and [IETF RFC 3264]																														
SELECTION EXPRESSION																															
Test purpose	Session update requested by the calling user is unsuccessful, existing session remains unchanged. 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 dialogue so that the other party knows that it has to modify an existing session instead of establishing a new session. Ensure that if the other party does not accept the change, it sends an error response such as 488 Not Acceptable Here, which also receives an ACK. The session remains unchanged.																														
Configuration																															
SIP parameter	INVITE: codec not supported in Network B																														
Message flow	<table style="width: 100%; border: none;"> <tr> <td style="text-align: center; width: 30%;">SIP (Network A)</td> <td style="text-align: center; width: 40%;">Interconnection interface</td> <td style="text-align: center; width: 30%;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE</td> <td style="text-align: center;">→</td> </tr> <tr> <td></td> <td style="text-align: center;">← 180 Ringing</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 200 OK INVITE</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">ACK</td> <td style="text-align: center;">→</td> </tr> <tr> <td></td> <td style="text-align: center;">Communication</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← INVITE</td> <td style="text-align: center;">→</td> </tr> <tr> <td></td> <td style="text-align: center;">← 488 Not Acceptable Here</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">ACK</td> <td style="text-align: center;">→</td> </tr> <tr> <td></td> <td style="text-align: center;">Apply post test routine</td> <td></td> </tr> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE	→		← 180 Ringing			← 200 OK INVITE			ACK	→		Communication			← INVITE	→		← 488 Not Acceptable Here			ACK	→		Apply post test routine	
SIP (Network A)	Interconnection interface	SIP (Network B)																													
	INVITE	→																													
	← 180 Ringing																														
	← 200 OK INVITE																														
	ACK	→																													
	Communication																														
	← INVITE	→																													
	← 488 Not Acceptable Here																														
	ACK	→																													
	Apply post test routine																														
Comments	Establish a communication from Network A to Network B. User A in Network A attempts to change the session by sending an SDP offer to the UE in Network B. Network B does not support the codec sent in the offer. Check: Is a 488 Not Acceptable Here sent from Network B to Network A? Repeat this test in reverse direction.																														

Test case number	SS_unsucc_NNI_008																														
Test case group	BCALL/unsuccesful																														
Reference	[IETF RFC 4566], [IETF RFC 3261] and [IETF RFC 3264]																														
SELECTION EXPRESSION																															
Test purpose	<p>Session update requested by the called user is unsuccessful, existing session remains unchanged.</p> <p>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 dialogue so that the other party knows that it has to modify an existing session instead of establishing a new session. Ensure that if the other party does not accept the change, it sends an error response such as 488 Not Acceptable Here, which also receives an ACK. The session remains unchanged. The 488 Not Acceptable Here may be sent by a simulation equipment.</p>																														
Configuration																															
SIP parameter	INVITE : codec not supported in Network A																														
Message flow	<table border="0" style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%;">SIP (Network A)</th> <th style="text-align: center; width: 40%;">Interconnection interface</th> <th style="text-align: right; width: 30%;">SIP (Network B)</th> </tr> </thead> <tbody> <tr> <td></td> <td style="text-align: center;">INVITE</td> <td style="text-align: right;">→</td> </tr> <tr> <td></td> <td style="text-align: center;">180 Ringing</td> <td></td> </tr> <tr> <td style="text-align: left;">←</td> <td style="text-align: center;">200 OK INVITE</td> <td></td> </tr> <tr> <td style="text-align: left;">←</td> <td style="text-align: center;">ACK</td> <td style="text-align: right;">→</td> </tr> <tr> <td></td> <td style="text-align: center;">Communication</td> <td></td> </tr> <tr> <td style="text-align: left;">←</td> <td style="text-align: center;">INVITE</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">488 Not Acceptable Here</td> <td style="text-align: right;">→</td> </tr> <tr> <td style="text-align: left;">←</td> <td style="text-align: center;">ACK</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">Apply post test routine</td> <td></td> </tr> </tbody> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE	→		180 Ringing		←	200 OK INVITE		←	ACK	→		Communication		←	INVITE			488 Not Acceptable Here	→	←	ACK			Apply post test routine	
SIP (Network A)	Interconnection interface	SIP (Network B)																													
	INVITE	→																													
	180 Ringing																														
←	200 OK INVITE																														
←	ACK	→																													
	Communication																														
←	INVITE																														
	488 Not Acceptable Here	→																													
←	ACK																														
	Apply post test routine																														
Comments	<p>Establish a communication from Network A to Network B.</p> <p>User B in Network B attempts to change the session by sending an SDP offer to the UE in Network A.</p> <p>Network A does not support the codec sent in the offer.</p> <p>Check: Is a 488 Not Acceptable Here sent from Network B to Network A?</p> <p>Repeat this test in reverse direction.</p>																														

Test case number	SS_unsucc_NNI_009																					
Test case group	BCALL/unsuccesful																					
Reference	[IETF RFC 3261]																					
SELECTION EXPRESSION																						
Test purpose	<p>Call clearing due to no answer from the called user initiated by the calling user.</p> <p>Ensure that when there is no answer from the called user, the calling user initiates call clearing to the called user with CANCEL or BYE.</p>																					
Configuration																						
SIP parameter																						
Message flow	<table border="0" style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%;">SIP (Network A)</th> <th style="text-align: center; width: 40%;">Interconnection interface</th> <th style="text-align: right; width: 30%;">SIP (Network B)</th> </tr> </thead> <tbody> <tr> <td></td> <td style="text-align: center;">INVITE</td> <td style="text-align: right;">→</td> </tr> <tr> <td style="text-align: left;">←</td> <td style="text-align: center;">180 Ringing</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">CANCEL/BYE</td> <td style="text-align: right;">→</td> </tr> <tr> <td style="text-align: left;">←</td> <td style="text-align: center;">200 OK CANCEL/BYE</td> <td></td> </tr> <tr> <td style="text-align: left;">←</td> <td style="text-align: center;">487 Request Terminated</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">ACK</td> <td style="text-align: right;">→</td> </tr> </tbody> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE	→	←	180 Ringing			CANCEL/BYE	→	←	200 OK CANCEL/BYE		←	487 Request Terminated			ACK	→
SIP (Network A)	Interconnection interface	SIP (Network B)																				
	INVITE	→																				
←	180 Ringing																					
	CANCEL/BYE	→																				
←	200 OK CANCEL/BYE																					
←	487 Request Terminated																					
	ACK	→																				
Comments	<p>Check: Is a CANCEL or BYE request is sent by the the originating user?</p> <p>Check: Is a 487 Request Terminating sent by the terminating user?</p> <p>Check: Are the media streams terminated after the 200 OK CANCEL/BYE was sent?</p> <p>Repeat this test in reverse direction.</p>																					

Test case number	SS_unsucc_NNI_010												
Test case group	BCALL/unsuccessful												
Reference	[IETF RFC 4566], [IETF RFC 3261] and [IETF RFC 3264]												
SELECTION EXPRESSION													
Test purpose	Codec not supported by the called user. The initial INVITE contains an SDP with codecs that are not supported by the called user. Ensure that, when the called user does not accept the Media session, the called user initiates call clearing to the calling user with 488 Not Acceptable Here or 606 Not Acceptable, which also receives an ACK.												
Configuration													
SIP parameter	INVITE : codec not supported at user (Network B)												
Message flow	<table border="0" style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%;">SIP (Network A)</th> <th style="text-align: center; width: 40%;">Interconnection interface</th> <th style="text-align: right; width: 30%;">SIP (Network B)</th> </tr> </thead> <tbody> <tr> <td></td> <td style="text-align: center;">INVITE →</td> <td></td> </tr> <tr> <td>CASE A</td> <td style="text-align: center;">← 488 Not Acceptable Here ACK →</td> <td></td> </tr> <tr> <td>CASE B</td> <td style="text-align: center;">← 606 Not Acceptable ACK →</td> <td></td> </tr> </tbody> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE →		CASE A	← 488 Not Acceptable Here ACK →		CASE B	← 606 Not Acceptable ACK →	
SIP (Network A)	Interconnection interface	SIP (Network B)											
	INVITE →												
CASE A	← 488 Not Acceptable Here ACK →												
CASE B	← 606 Not Acceptable ACK →												
Comments	Establish a call setup from Network A to Network B. User B in Network B does not support the codec offered in the SDP received from Network A. Check: Is a 488 Not Acceptable Here or 606 Not Acceptable sent from Network B to Network A. Repeat this test in reverse direction.												

Test case number	SS_unsucc_NNI_011																					
Test case group	BCALL/unsuccessful																					
Reference	[IETF RFC 3261]																					
SELECTION EXPRESSION																						
Test purpose	Call clearing due to no answer from the called user initiated by the originating network. Ensure that when there is no answer from the called user, the originating network initiates the call clearing after timeout of SIP timer C and sends a CANCEL or BYE to the called user.																					
Configuration																						
SIP parameter																						
Message flow	<table border="0" style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%;">SIP (Network A)</th> <th style="text-align: center; width: 40%;">Interconnection interface</th> <th style="text-align: right; width: 30%;">SIP (Network B)</th> </tr> </thead> <tbody> <tr> <td></td> <td style="text-align: center;">→ INVITE →</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 180 Ringing Start timer C</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">Timeout timer C</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← CANCEL/BYE →</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 200 OK CANCEL/BYE →</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 487 Request Terminated ACK →</td> <td></td> </tr> </tbody> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		→ INVITE →			← 180 Ringing Start timer C			Timeout timer C			← CANCEL/BYE →			← 200 OK CANCEL/BYE →			← 487 Request Terminated ACK →	
SIP (Network A)	Interconnection interface	SIP (Network B)																				
	→ INVITE →																					
	← 180 Ringing Start timer C																					
	Timeout timer C																					
	← CANCEL/BYE →																					
	← 200 OK CANCEL/BYE →																					
	← 487 Request Terminated ACK →																					
Comments	Check: Is a CANCEL or BYE request sent by the originating network? Check: Is a 487 Request Terminating send from the terminating user? Check: Are the media streams terminated after the 200 OK CANCEL/BYE was sent? Repeat this test in reverse direction.																					

Test case number	SS_unsucc_NNI_011A																		
Test case group	BCALL/unsuccessful																		
Reference	[IETF RFC 4028]																		
SELECTION EXPRESSION	[Network A] SE 17a AND [Network B] SE 17a																		
Test purpose	<p>Negotiation of session timer. Ensure that the interconnected networks are able to negotiate the session time to refresh the session. If the session refresh duration is too short for one of the involved entities, a 422 Session Interval Too Small unsuccessful final response is sent in backward direction to update the session duration time. A new INVITE is sent and a Min-SE header present proposes a longer session duration.</p>																		
Configuration	The session time in Network B is smaller than the session time used in Network A																		
Comment	This test case is only applicable if the session refresh time is different in Network A and Network B. This situation is also load dependent.																		
SIP parameter	INVITE 1: Supported: timer Session-Expires: x 422: Min-SE. x + y INVITE 2 Session-Expires: x + y																		
Message flow	<table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%;">SIP (Network A)</th> <th style="text-align: center; width: 40%;">Interconnection interface</th> <th style="text-align: right; width: 30%;">SIP (Network B)</th> </tr> </thead> <tbody> <tr> <td></td> <td style="text-align: center;">INVITE 1</td> <td style="text-align: right;">→</td> </tr> <tr> <td style="text-align: left;">←</td> <td style="text-align: center;">422 Session Interval Too Small ACK</td> <td style="text-align: right;">→</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE 2</td> <td style="text-align: right;">→</td> </tr> <tr> <td style="text-align: left;">←</td> <td style="text-align: center;">180 Ringing</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">Apply post test routine</td> <td></td> </tr> </tbody> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE 1	→	←	422 Session Interval Too Small ACK	→		INVITE 2	→	←	180 Ringing			Apply post test routine	
SIP (Network A)	Interconnection interface	SIP (Network B)																	
	INVITE 1	→																	
←	422 Session Interval Too Small ACK	→																	
	INVITE 2	→																	
←	180 Ringing																		
	Apply post test routine																		
Comments	Establish a communication setup from Network A to Network B Check: Is the supported header in the initial INVITE set to 'timer' Check: Is a 422 Session Interval Too Small sent by the terminating Network? Check: Is the Session-Expires header in the second initial INVITE request sent from Network A set to the value indicated in the 422 final response? Repeat this test in reverse direction.																		

9.1.7.2 SIP-I

Test case number	SS_unsucc_SIP-I_01													
Test case group	BCALL/unsuccessful													
Reference	6.11.2/[ITU-T Q.1912.5]													
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47													
Test purpose	<p>SIP-I support. Called number is not allocated in the PSTN/PLMN network. Ensure that, when calling to an unallocated number in the PSTN/PLMN part of Network B and ISUP – SIP-I interworking applies in Network B, the network initiate call clearing to the calling user with a 404 Not Found message. A ISUP REL message is encapsulated and the Cause value indicator is set to '1'.</p>													
Configuration	The called user number is not assigned to the PSTN/PLMN part in Network B													
SIP parameter	404: Reason: Q.850;cause=1 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 1 --[any boundary name]--													
Message flow	<table border="0" style="width: 100%; text-align: center;"> <tr> <td style="width: 30%;">SIP (Network A)</td> <td style="width: 40%;">Interconnection interface</td> <td style="width: 30%;">SIP (Network B)</td> </tr> <tr> <td></td> <td>INVITE</td> <td>➔</td> </tr> <tr> <td></td> <td>← 404 Not Found(REL)</td> <td></td> </tr> <tr> <td></td> <td>ACK</td> <td>➔</td> </tr> </table>		SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE	➔		← 404 Not Found(REL)			ACK	➔
SIP (Network A)	Interconnection interface	SIP (Network B)												
	INVITE	➔												
	← 404 Not Found(REL)													
	ACK	➔												
Comments	Establish a communication from Network A to Network B, called user number is not allocated in the PSTN/PLMN part of Network B Check: Is a 404 Not Found sent from Network B to Network A? Check: is a ISUP REL encapsulated and the Cause value indicator is set to '1'? Check: If a Reason header is present, is the cause value equal to the value in the Cause value of the encapsulated ISUP REL? Repeat this test in reverse direction.													

Test case number	SS_unsucc_SIP-I_02												
Test case group	BCALL/unsuccessful												
Reference	6.11.2/[ITU-T Q.1912.5]												
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47												
Test purpose	SIP-I support. The called user is busy. Ensure that, when the called user in the PSTN/PLMN part of Network B and ISUP – SIP-I interworking applies in Network B is busy, the network initiates call clearing to the calling user with a 486 Busy Here message. A ISUP REL message is encapsulated and the Cause value indicator is set to '17'.												
Configuration	The called user is busy in the PSTN/PLMN part in Network B												
SIP parameter	486: Reason: Q.850;cause=17 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 17 --[any boundary name]--												
Message flow	<table style="width: 100%; border: none;"> <tr> <td style="text-align: center; width: 30%;">SIP (Network A)</td> <td style="text-align: center; width: 40%;">Interconnection interface</td> <td style="text-align: center; width: 30%;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE →</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">← 486 Busy Here (REL)</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">ACK →</td> <td></td> </tr> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE →			← 486 Busy Here (REL)			ACK →	
SIP (Network A)	Interconnection interface	SIP (Network B)											
	INVITE →												
	← 486 Busy Here (REL)												
	ACK →												
Comments	Establish a communication from Network A to Network B, user B in the PSTN/PLMN part of Network B is busy. Check: Is a 486 Busy Here sent from Network B to Network A? Check: Is a ISUP REL encapsulated and the Cause value indicator is set to '17'? Check: If a Reason header is present, is the cause value equal to the value in the Cause value of the encapsulated ISUP REL? Repeat this test in reverse direction.												

Test case number	SS_unsucc_SIP-I_3												
Test case group	BCALL/unsuccessful												
Reference	6.11.2/[ITU-T Q.1912.5]												
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47												
Test purpose	SIP-I support. The called user rejects the call. Ensure that, when the called user in the PSTN/PLMN part of Network B and ISUP – SIP-I interworking applies in Network B rejects the communication setup, the network initiates call clearing to the calling user with a 480 Temporarily Unavailable final response. A ISUP REL message is encapsulated and the Cause value indicator is set to '21'.												
Configuration													
SIP parameter	480: Reason: Q.850;cause=21 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 21 --[any boundary name]--												
Message flow	<table style="width: 100%; border: none;"> <tr> <td style="text-align: center; vertical-align: top;">SIP (Network A)</td> <td style="text-align: center; vertical-align: top;">Interconnection interface</td> <td style="text-align: center; vertical-align: top;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE</td> <td style="text-align: center;">→</td> </tr> <tr> <td style="text-align: center;">←</td> <td style="text-align: center;">480 Temporarily Unavailable (REL)</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">ACK</td> <td style="text-align: center;">→</td> </tr> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE	→	←	480 Temporarily Unavailable (REL)			ACK	→
SIP (Network A)	Interconnection interface	SIP (Network B)											
	INVITE	→											
←	480 Temporarily Unavailable (REL)												
	ACK	→											
Comments	Establish a communication from Network A to Network B, user B in the PSTN/PLMN part of Network B rejects the communication setup. Check: Is a 480 Temporarily Unavailable sent from Network B to Network A? Check: is a ISUP REL encapsulated and the Cause value indicator is set to '21'? Check: If a Reason header is present, is the cause value equal to the value in the Cause value of the encapsulated ISUP REL? Repeat this test in reverse direction.												

Test case number	SS_unsucc_SIP-I_04																																																						
Test case group	BCALL/unsuccessful																																																						
Reference	7.7.1/[ITU-T Q.1912.5]																																																						
SELECTION EXPRESSION	[Network A] SE 17 AND [Network B] SE 47																																																						
Test purpose	<p>SIP-I support. Call clearing due to no answer from the called user initiated by the calling user.</p> <p>Ensure when the early dialogue is not confirmed by the called user, the calling user located in the PSTN/PLMN part of Network A and ISUP – SIP-I interworking applies in Network A initiates call clearing to the called user with CANCEL or BYE.</p> <p>An ISUP REL message is encapsulated in the BYE request and the Cause value indicator is set to '16'.</p>																																																						
Configuration																																																							
SIP parameter	<pre> 480: Reason: Q.850;cause=16 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 16 --[any boundary name]-- </pre>																																																						
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Comments	<p>Establish a communication from Network A to Network B, user B does not confirm the communication.</p> <p>The originating user in the PSTN/PLMN part of Network A terminates the early dialogue.</p> <p>Check: Is a CANCEL or BYE request is sent from the originating network?</p> <p>Check: Is a ISUP REL encapsulated in a BYE request?</p> <p>Check: Is the Cause value of the encapsulated REL set to '16'?</p> <p>Check: If a Reason header is present, is the cause value equal to the value in the Cause value of the encapsulated ISUP REL?</p> <p>Check: Is a 487 Request Terminating send from the terminating user?</p> <p>Check: Are the media streams terminated after the 200 OK CANCEL/BYE was sent?</p> <p>Note – A ISUP REL is not encapsulated in a CANCEL request.</p> <p>Repeat this test in reverse direction.</p>																																																						

Test case number	SS_unsucc_SIP-I_04																																							
Test case group	BCALL/unsuccessful																																							
Reference	7.7.1/[ITU-T Q.1912.5]																																							
SELECTION EXPRESSION	[Network A] SE 17 AND [Network B] SE 47																																							
Test purpose	<p>SIP-I support. Call clearing due to no answer from the called user initiated by the originating network.</p> <p>Ensure when the early dialogue is not confirmed by the called user, the originating network initiate the call clearing after timeout of ISUP timer T9 if the calling user is located in the PSTN/PLMN part of Network A and ISUP – SIP-I interworking applies in Network A and the originating network sends a CANCEL or BYE to the called user.</p> <p>An ISUP REL message is encapsulated in the BYE request and the Cause value indicator is set to '19'.</p>																																							
Configuration																																								
SIP parameter	<p>480:</p> <p>Reason: Q.850;cause=19 (optional)</p> <p>Content-Type: multipart/mixed;boundary=[any boundary name]</p> <p>--[any boundary name]</p> <p>Content-Type: application/isup;version=itu-t92</p> <p>Content-Disposition: signal;handling=required</p> <p>REL</p> <p>Cause value: 19</p> <p>--[any boundary name]--</p>																																							
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Comments	<p>Establish a communication from Network A to Network B, user B does not answer the communication setup.</p> <p>The ISUP timer T9 in the PSTN/PLMN expires</p> <p>Check: Is a CANCEL or BYE request is sent by the originating network?</p> <p>Check: Is a ISUP REL encapsulated in a BYE request?</p> <p>Check: Is the Cause value of the encapsulated REL set to '19'?</p> <p>Check: If a Reason header is present, is the cause value equal to the value in the Cause value of the encapsulated ISUP REL?</p> <p>Check: Is a 487 Request Terminating send from the terminating user?</p> <p>Check: Are the media streams terminated after the 200 OK CANCEL/BYE was sent?</p> <p>Note – A ISUP REL is not encapsulated in a CANCEL request.</p> <p>Repeat this test in reverse direction.</p>																																							

Test case number	SS_oip_SIP-I_001												
Test case group	SIP-SIP/Service/OIP												
Reference	7.1.3/[ITU-T Q.1912.5]												
SELECTION EXPRESSION	[Network A] SE 17 [Network A] AND SE 47 AND SE 52												
Test purpose	<p>SIP-I support. ISUP calling party number presentation allowed in the encapsulated IAM. Ensure when bearer independent call control (BICC)/ISUP – SIP-I interworking applies in the originating network the BICC/ISUP IAM is encapsulated in the INVITE request. The P-Asserted-Identity header field is derived from the Calling party number in the encapsulated IAM. The 'Presentation restriction' indicator in the encapsulated IAM is set to 'allowed' no Privacy value 'id' is present in the INVITE request.</p>												
Configuration													
SIP parameter	INVITE P-Asserted-Identity=[derived from the ISUP calling party number] Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Calling party number Screening indicator Network provided or user provided, verified and passed Presentation restriction allowed Address signal --[any boundary name]--												
Message flow	<table style="width: 100%; border: none;"> <tr> <td style="text-align: center; width: 33%;">SIP (Network A)</td> <td style="text-align: center; width: 33%;">Interconnection interface</td> <td style="text-align: center; width: 33%;">SIP (Network B)</td> </tr> <tr> <td></td> <td style="text-align: center;">INVITE (IAM)</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">→</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">Apply post test routine</td> <td></td> </tr> </table>	SIP (Network A)	Interconnection interface	SIP (Network B)		INVITE (IAM)			→			Apply post test routine	
SIP (Network A)	Interconnection interface	SIP (Network B)											
	INVITE (IAM)												
	→												
	Apply post test routine												
Comments	<p>Check: Is a BICC/ISUP IAM encapsulated in the in the INVITE request?</p> <p>Check: Is the Calling party number present in the encapsulated IAM and the screening indicator is set to 'Network provided' or 'user provided, verified and passed' and the Presentation restriction indicator is set to 'allowed'?</p> <p>Check: Is the P-Asserted-Identity header field derived from the Calling party number in the encapsulated IAM?</p> <p>Check: Is the value 'id' not present in the Privacy header field (if included)?</p> <p>Repeat this test in reverse direction.</p>												

Test case number	SS_oir_SIP-I_002												
Test case group	SIP-SIP/Service/OIR												
Reference	7.1.3/[ITU-T Q.1912.5]												
SELECTION EXPRESSION	[Network A] SE 17 AND [Network A] SE 47 AND SE 52												
Test purpose	<p>SIP-I support. ISUP Additional calling party number presentation restricted in the encapsulated IAM.</p> <p>Ensure when BICC/ISUP – SIP-I interworking applies in the originating network the BICC/ISUP IAM is encapsulated in the INVITE request. The From field is derived from the Additional calling party number in the encapsulated IAM. The 'Presentation restriction' indicator in the Generic number parameter is set to 'allowed' no Privacy value 'id' is present in the INVITE request.</p>												
Configuration	The originating user in the PSTN/PLMN part of Network A is subscribed to the 'no screening option'												
SIP parameter	<p>INVITE</p> <p>P-Asserted-Identity=[derived from the ISUP calling party number]</p> <p>From=[derived from the ISUP Additional calling party number]</p> <p>Privacy: id</p> <p>Content-Type: multipart/mixed;boundary=[any boundary name]</p> <p>--[any boundary name]</p> <p>Content-Type: application/isup;version=itu-t92</p> <p>Content-Disposition: signal;handling=required</p> <p>IAM</p> <p>Calling party number</p> <p>Screening indicator</p> <p>Network Provided</p> <p>Presentation restriction</p> <p>restricted</p> <p>Address signal</p> <p>Generic number</p> <p>Number Qualifier Indicator</p> <p>Additional calling party number</p> <p>Screening indicator</p> <p>user provided, not verified</p> <p>Presentation restriction</p> <p>restricted</p> <p>Address signal</p> <p>--[any boundary name]--</p>												
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SIP (Network A)	Interconnection interface	SIP (Network B)											
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Comments	<p>Check: Is a BICC/ISUP IAM encapsulated in the in the INVITE request?</p> <p>Check: Is the Calling party number present in the encapsulated IAM and the screening indicator is set to 'Network Provided' and the Presentation restriction indicator is set to 'restricted'?</p> <p>Check: Is the P-Asserted-Identity header field derived from the Calling party number in the encapsulated IAM?</p> <p>Check: Is a Generic number parameter, Number Qualifier Indicator set to Additional calling party number present and the screening indicator is set to 'user provided, not verified' and the Presentation restriction indicator is set to 'restricted'?</p> <p>Check: Is the From header field derived from the Additional calling party number in the encapsulated IAM?</p> <p>Check: Is the value 'id' present in the Privacy header field?</p> <p>Repeat this test in reverse direction.</p>												

9.2 Video call

9.2.1 Testing of SIP protocol requirements

9.2.1.1 Test purposes for Basic call, successful

Test case number	SS_bcall_video_001																																																
Test case group	BCALL/successful																																																
Reference	[ETSI ES 202 667]; [ITU-T P.910]; [ITU-T P.911]																																																
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Test purpose	<p>Calling user adds one-way video call to an ongoing VoLTE call</p> <p>Ensure that the VoLTE and video device which is used includes its video capability in the signaling.</p> <p>Calling user adds video to an ongoing VoLTE call, the device sends a new invitation message with information about the additional video media component, which is treated by the IMS and EPC domains resulting in the addition of a dedicated bearer for the video stream.</p> <p>Ensure that the calling user can drop video and continue just with voice.</p> <p>Ensure that call remains in intelligible/high quality conversation for video and voice.</p> <p>The acceptability thresholds for lip synchronization are 185 ms when sound is delayed with respect to the video, and 90 ms when sound is advanced with respect to the video. [ETSI ES 202 667]</p> <p>The subjective video quality assessment methods for multimedia applications which predict a mean opinion score (MOS) on a five-point ACR scale are defined in [ITU-T P.910]. The global audiovisual MOS score is defined in [ITU-T P.911].</p> <p>The voice call is released from the called user.</p>																																																
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Test case number	SS_bcall_video_002																																																
Test case group	BCALL/successful																																																
Reference	[ETSI ES 202 667]; [ITU-T P.910]; [ITU-T P.911]																																																
SELECTION EXPRESSION																																																	
Test purpose	<p>Calling user adds two-way video call to an ongoing VoLTE call</p> <p>Ensure that the VoLTE and video device which is used includes its video capability in the signaling.</p> <p>Calling user adds video to an ongoing VoLTE call, the device sends a new invitation message with information about the additional video media component, which is treated by the IMS and EPC domains resulting in the addition of a dedicated bearer for the video stream.</p> <p>Ensure that the calling user can drop video and continue just with voice.</p> <p>Ensure that call remains in intelligible/high quality conversation for video and voice.</p> <p>The acceptability thresholds for lip synchronization are 185 ms when sound is delayed with respect to the video, and 90 ms when sound is advanced with respect to the video. [ETSI ES 202 667]</p> <p>The subjective video quality assessment methods for multimedia applications which predict a MOS on a five-point ACR scale are defined in [ITU-T P.910]. The global audiovisual MOS score is defined in [ITU-T P.911].</p> <p>The voice call is released from the called user.</p>																																																
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Test case number	SS_bcall_video_003	
Test case group	BCALL/successful	
Reference	[ETSI ES 202 667]; [ITU-T P.910]; [ITU-T P.911]	
SELECTION EXPRESSION		
Test purpose	<p>Calling user adds one-way video call to an ongoing VoLTE call</p> <p>Ensure that the VoLTE and video device which is used includes its video capability in the signaling.</p> <p>Calling user adds video to an ongoing VoLTE call, the device sends a new invitation message with information about the additional video media component, which is treated by the IMS and EPC domains resulting in the addition of a dedicated bearer for the video stream.</p> <p>Ensure that the calling user can drop video and continue just with voice.</p> <p>Ensure that call remains in intelligible/high quality conversation for video and voice. The acceptability thresholds for lip synchronization are 185 ms when sound is delayed with respect to the video, and 90 ms when sound is advanced with respect to the video. [ETSI ES 202 667]</p> <p>The subjective video quality assessment methods for multimedia applications which predict a MOS on a five-point ACR scale are defined in [ITU-T P.910]. The global audiovisual MOS score is defined in [ITU-T P.911].</p> <p>The voice call is released from the calling user.</p>	
Configuration		
SIP parameter		
Message flow		
SIP (Network A)	Interconnection interface	SIP (Network B)
	INVITE →	
	← 180 Ringing	
	← 200 OK INVITE	
	ACK →	
	Communication audio	
	Re-INVITE video →	
	← 200 OK INVITE	
	ACK →	
	Communication video	
	Re-INVITE audio →	
	← 200 OK INVITE	
	ACK →	
	Communication audio	
	BYE →	
	← 200 OK BYE	
Comments		

Test case number	SS_bcall_video_004																																																
Test case group	BCALL/successful																																																
Reference	[ETSI ES 202 667]; [ITU-T P.910]; [ITU-T P.911]																																																
SELECTION EXPRESSION																																																	
Test purpose	<p>Calling user adds two-way video call to an ongoing VoLTE call</p> <p>Ensure that the VoLTE and video device which is used includes its video capability in the signaling.</p> <p>Calling user adds video to an ongoing VoLTE call, the device sends a new invitation message with information about the additional video media component, which is treated by the IMS and EPC domains resulting in the addition of a dedicated bearer for the video stream.</p> <p>Ensure that the calling user can drop video and continue just with voice.</p> <p>Ensure that call remains in intelligible/high quality conversation for video and voice.</p> <p>The acceptability thresholds for lip synchronization are 185 ms when sound is delayed with respect to the video, and 90 ms when sound is advanced with respect to the video. [ETSI ES 202 667]</p> <p>The subjective video quality assessment methods for multimedia applications which predict a MOS on a five-point ACR scale are defined in [ITU-T P.910]. The global audiovisual MOS score is defined in [ITU-T P.911].</p> <p>The voice call is released from the calling user</p>																																																
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SIP parameter																																																	
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Comments																																																	

Test case number	SS_bcall_video_005																														
Test case group	BCALL/successful																														
Reference	[ETSI ES 202 667]; [ITU-T P.910]; [ITU-T P.911]																														
SELECTION EXPRESSION																															
Test purpose	<p>Calling user is establishing a two-way video call Ensure that the VoLTE and video device which is used includes its video capability in the signaling. Calling user is establishing a video call with information about video media component, which is treated by the IMS and EPC domains resulting in the addition of a dedicated bearer for the video stream.</p> <p>Ensure that call remains in intelligible/high quality conversation for video and voice. The acceptability thresholds for lip synchronization are 185 ms when sound is delayed with respect to the video, and 90 ms when sound is advanced with respect to the video. [ETSI ES 202 667]</p> <p>The subjective video quality assessment methods for multimedia applications which predict MOS on a five-point ACR scale are defined in [ITU-T P.910]. The global audiovisual MOS score is defined in [ITU-T P.911]. The voice call is released from the called user.</p>																														
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	ACK	→																													
	Communication video and audio																														
	Communication																														
	← BYE																														
	200 OK BYE	→																													
Comments																															

Test case number	SS_bcall_video_006	
Test case group	BCALL/successful	
Reference	[ETSI ES 202 667]; [ITU-T P.910]; [ITU-T P.911]	
SELECTION EXPRESSION		
Test purpose	<p>Calling user is establishing a two-way video call</p> <p>Ensure that the VoLTE and video device which is used includes its video capability in the signaling.</p> <p>Calling user is establishing a video call with information about video media component, which is treated by the IMS and EPC domains resulting in the addition of a dedicated bearer for the video stream.</p> <p>Ensure that call remains in intelligible/high quality conversation for video and voice.</p> <p>The acceptability thresholds for lip synchronization are 185 ms when sound is delayed with respect to the video, and 90 ms when sound is advanced with respect to the video. [ETSI ES 202 667]</p> <p>The subjective video quality assessment methods for multimedia applications which predict a MOS on a five-point ACR scale are defined in [ITU-T P.910]. The global audiovisual MOS score is defined in [ITU-T P.911].</p> <p>The voice call is released from the calling user.</p>	
Configuration		
SIP parameter		
Message flow		
SIP (Network A)	Interconnection interface	SIP (Network B)
	INVITE →	
	← 180 Ringing	
	← 200 OK INVITE	
	video and audio	
	ACK →	
	Communication video and audio	
	Communication	
	← BYE →	
	200 OK BYE	
Comments		

10 QoS/QoE/ test requirements

The conduction of voice quality measurements follows the descriptions found in [ETSI EG 202 057-2], [ITU-T Q.543], [ETSI TS 101 563] and clauses 6.6.3.1 and 6.6.3.2 of [ETSI TS 102 250-2].

The access points of the test equipment, which are used for inserting or retrieving signals needed for determining the speech quality parameters, shall conform to the reference characteristics as provided in the following relevant standards:

- [ETSI EG 202 425] for VoIP access;
- [ETSI TBR 21] for analogue access.

The properties of the test equipment shall be known, and values measured for each parameter shall be corrected accordingly by the impairments introduced by the test equipment. Specifically, delay introduced by the test equipment shall be known and measurement results shall be corrected by the delay introduced by the test equipment.

The simultaneous transmission of voice and data through uploads, downloads or IPTV use is an additional user-related scenario. For this reason, voice quality measurements have been included where in parallel to the voice connection, active upload and download of data is simulated. This provides information about any potential prioritization of voice data when the entire bandwidth is being utilized.

The key performance indicators (KPIs) listed in Table 10-1 are recorded as part of voice quality measurements.

Table 10-1 – Overview of KPI for voice quality measurements

1.	Call setup delay [ITU-T Q.543] and session initiation call setup delay [ETSI TS 101 563]
2.	Call setup time (PDD) [ETSI ES 202 765-2]
3.	Premature release probability (call failure rate), see clause 7.4 in [ITU-T Q.3933]
4.	Telephony cut-off call ratio [%] (call drop rate), see clause 7.5 in [ITU-T Q.3933]
5.	Media establishment delay, see clause 7.7 in [ITU-T Q.3933]
6.	Level of active speech signal, see clause 7.8 in [ITU-T Q.3933]
7.	Noise level, see clause 7.9 in [ITU-T Q.3933]
8.	Signal-to-noise ratio, see clause 7.10 in [ITU-T Q.3933]
9.	Speech signal attenuation, see clause 7.11 in [ITU-T Q.3933]
10.	Talker echo delay, see clause 7.12 in [ITU-T Q.3933]
11.	Double talk, see clause 7.13 in [ITU-T Q.3933]
12.	Interrupted voice transmission, see clause 7.14 in [ITU-T Q.3933]
13.	Listening speech quality, see clause 7.15 in [ITU-T Q.3933]
14.	Listening speech quality stability, see clause 7.16 in [ITU-T Q.3933]
15.	End-to-end audio delay, see clause 7.17 in [ITU-T Q.3933]
16.	End-to-end audio delay variation, see clause 7.18 in [ITU-T Q.3933]
17.	Frequency response, see clause 7.19 in [ITU-T Q.3933]
18.	Fax transmission T.30 (fax, bit rate \leq 14,4 kbit/s and fax, bit rate \geq 14,4 kbit/s) see clause 7.20 in [ITU-T Q.3933]
19.	Early media, see clause 7.21 in [ITU-T Q.3933]
20.	Jitter Buffer and IP periodization response time, see clause 7.22 in [ITU-T Q.3933]

For VoLTE interconnect and roaming with QoE and QoS tests are following KPI mandatory:

1.	call setup time (PDD)
2.	Listening speech quality
3.	End-to-end audio delay, see clause 7.17 in [ITU-T Q.3933]
4.	End-to-end audio delay variation, see clause 7.18 in [ITU-T Q.3933]
5.	Early media, see clause 7.21 in [ITU-T Q.3933]

10.1 Call setup time (PDD)

See [ETSI ES 202 765-2].

To determine the call setup time in a VoIP implementation, the time in seconds from the sending of the INVITE signal through the "A" side, until the receipt of the 200 OK signal is measured on the "A" side is measured, or the time in seconds from the sending of the INVITE signal through the "A" side until the receipt of the 180 Ringing signal on the "A" side is recorded.

10.2 Listening speech quality

10.2.1 Connections with one voice channel

See [ETSI TS 103 222-3].

For the **single voice channel test**, a test call consisting of the following three parts should be used:

- 1) Channel convergence quality test;
- 2) Listening speech quality test;
- 3) DTMF test.

Table 10-2 gives an overview of the connection options without parallel data transfer.

Table 10-2 – Configurations options for connections without parallel data transfer

Connections without parallel data transfer	Voice from	Voice to
	VoLTE	Multimedia telephony service (MMTel) (IMS) fixed access
	VoLTE	VoLTE
	LTE mobile network with CSFB	IMS PSTN emulation subsystem (PES) with access gateway control function (AGCF) (or PSTN or ISDN access)
	Universal mobile telecommunications system (UMTS)	IMS PES with AGCF (or PSTN or ISDN access)
	LTE mobile network with CSFB	IMS PES with voice gateway (VGW)
	UMTS	IMS PES with VGW
	UMTS	UMTS

Figures 10-1 and 10-2 depict the scenarios VoLTE to VoLTE and VoLTE to MMTel for the measurement of voice quality.

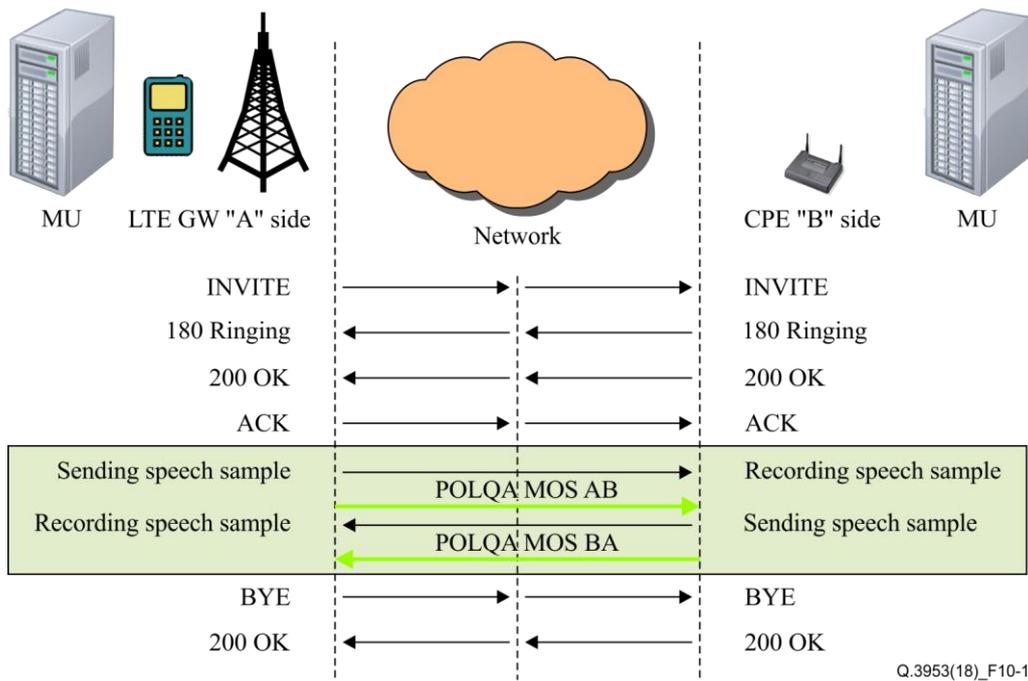


Figure 10-1 – VoLTE voice quality measurement for a mobile – Fixed network connection

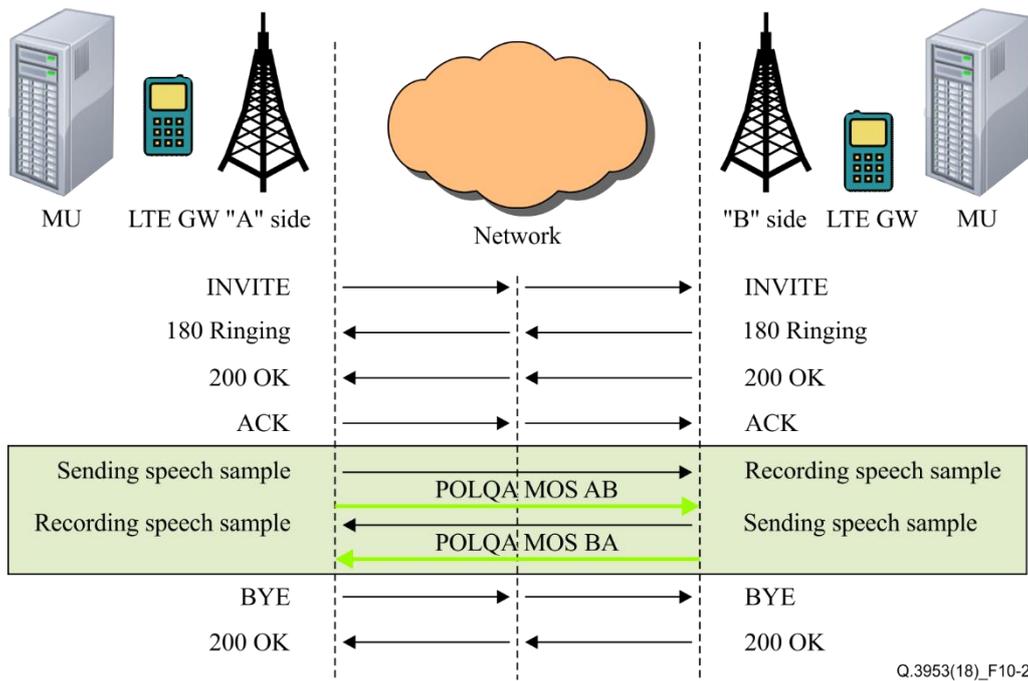


Figure 10-2 – VoLTE voice quality measurement for a mobile – Mobile connection

10.2.1 General aspects of listening speech quality

Listening speech quality represents the intrinsic quality of a speech signal as perceived by a user at the receiving end. This indicator takes into account the impairments introduced by the transmission system. The mean opinion score-listening quality objective (MOS-LQO) is obtained by comparing speech samples:

- The original undistorted reference speech signal;
- The degraded signal received at the local end, where the measurement is applied.

[ITU-T P.863] recommends two samples from each of two male and two female speakers, i.e., eight sentence pairs. Some applications may only permit shorter test durations. Typically, test sentence material in subjective tests has a 0.5 second silence lead in, two sentences, and then a 0.5 second silence at the end of the signal. Further information can be found in [ETSI TR 103 138] and [ITU-T P.863.1].

To ensure comparable voice quality results it shall be ensured that the test equipment uses the codec described in the first line of the m line in the SDP part which is the preconfigured codec by the network operator.

10.2.2 General aspects of voice channel test calls

For the all voice channel tests, an aligned structure of the voice call shall be used. In this call sentence pairs (male / female) fulfilling the requirements of [ITU-T P.863.1] shall be transmitted from A to B and from B to A. Speech files especially tested for the use with [ITU-T P.863] are published in Annex C of [ITU-T P.501], where samples in different languages are covered.

In principle all voice channel tests consist of three parts:

- 1) Channel convergence quality test;
- 2) Listening speech quality test;
- 3) DTMF test.

Which parts are actually used and how they are structured is defined for the individual test cases in the clauses below.

The channel convergence quality test starts with a listening speech quality test from B to A after the connection is established. This initial test provides information about the listening quality during convergence of the channel.

For the analysis of the initial listening speech quality during convergence of the channel the method according to [ITU-T P.863] in superwideband (SWB) mode based on only two sentences (one female voice and one male voice) is used. For this purpose, a male voice (e.g., "Four hours of steady work faced us") and a female voice (e.g., "The hogs were fed chopped corn and garbage") can be selected from the test sentences provided in Annex C of [ITU-T P.501].

After convergence of the channel the regular listening speech quality test starts using [ITU-T P.863] in SWB mode based on eight sentences (two male and two female voices, two sentences each).

Usually, the listening speech quality tests should start 10 seconds after the connection is established. This 10-second pause is recommended for converging the speech processing components and building up the IP-buffer at the receiving side and can be used for the channel convergence quality test as described above. It is assumed that the convergence has finished after 10 seconds. In the event of a proven shorter convergence, the pause can be shorter.

In case the channel can be assumed as converged from the beginning, and/or the separation of the channel convergence quality is not of interest, the listening quality test can start at any time after the connection is established.

Within the listening speech quality test, for example the following English samples can be selected from the test sentences provided in Annex C of [ITU-T P.501]:

- *Female 1:*
 - These days a chicken leg is a rare dish.
 - The hogs were fed with chopped corn and garbage.
- *Female 2:*
 - Rice is often served in round bowls.
 - A large size in stockings is hard to sell.
- *Male 1:*
 - The juice of lemons makes fine punch.
 - Four hours of steady work faced us.
- *Male 2:*
 - The birch canoe slid on smooth planks.
 - Glue the sheet to the dark blue background.

If a global application is of interest, optionally the male and female tests sentences of other languages provided in Annex C of [ITU-T P.501] can be used.

After conducting all evaluations, the derived MOS scores for each sample in the listening test are averaged over all received and scored samples separately for each direction A-B and B-A.

DTMF test: DTMF tones are often used for remote controlling equipment and need to be tested in an established voice channel too for correct transmission. It is recommended to test DTMF before or after the listening speech quality test but in each case after the channel has converged. The DTMF test should consist of DTMF tones (70 ms signal, 100 ms pause) and shall contain the tones 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, *, #.

Technical comments:

- If the interrupted voice transmission time is > 1 second and the call connection is maintained, the call is rated as interrupted (see clause 5.13 in [ITU-T Q.3933]).
- If all eight sentences (four samples) are sent within one file, the score calculation according to [ITU-T P.863] shall be performed separately for each sample (two sentences per sample).
- In case the sampling frequency at the input measuring interface is 8 kHz, as it usual for ISDN and narrowband applications, the input speech signal used shall be band limited at 3 800 kHz (see [ETSI TR 103 138]).
- When the sampling frequency at the input measuring interface is 16 kHz as required for wideband telephony the input speech signal used shall be band limited between 100 Hz and 7 600 kHz with a band pass filter providing a minimum of 24 dB/Oct. filter roll off, when feeding into the receive direction (see [ETSI TR 103 138]).
- The input test signal levels are referred to the average level of the (band limited in receive direction) test signal, averaged over the complete test sequence unless specified otherwise. It is recommended to adjust the active speech level to –26 dB OVL as specified in [ETSI TR 103 138].

Table 10-3 depicts a single voice channel test.

Table 10-3 – Single voice channel test

Relative time	Test equipment A		NETWORK		Test equipment B
CALL A to B					
T0 - 2	INVITE	→		→	INVITE
	180 Ringing	←		←	80 Ringing
T0	200 OK	←		←	200 OK
	ACK	→		→	ACK
Start Convergence Quality test					
T0	Start Audio Receive BA_1 (female and male)	←		←	Start Audio Send BA_1 (female and male)
	End Audio Receive BA_1	←		←	End Audio Send BA_1
Stop Convergence Quality test					
Listening Speech Quality test					
T0 + 10 s	Start Audio Send AB_1 (female 1)	□		□	Start Audio Receive AB_1 (female 1)
	End Audio Send AB_1 (female 1)	□		□	End Audio Receive AB_1 (female 1)
	Start Audio Send AB_2 (female 2)	□		□	Start Audio Receive AB_2 (female 2)
	End Audio Send AB_2 (female 2)	□		□	End Audio Receive AB_2 (female 2)
	Start Audio Send AB_3 (male 1)	□		□	Start Audio Receive AB_3 (male 1)
	End Audio Send AB_3 (male 1)	□		□	End Audio Receive AB_3 (male 1)
	Start Audio Send AB_4 (male 2)	□		□	Start Audio Receive AB_4 (male 2)
	End Audio Send AB_4 (male 2)	□		□	End Audio Receive AB_4 (male 2)
1 s	Pause				
	Start Audio Receive BA_1 (female 1)	□		□	Start Audio Send BA_1 (female 1)
	End Audio Receive BA_1 (female 1)	□		□	End Audio Send BA_1 (female 1)
	Start Audio Receive BA_2 (female 2)	□		□	Start Audio Send BA_2 (female 2)
	End Audio Receive BA_2 (female 2)	□		□	End Audio Send BA_2 (female 2)
	Start Audio Receive BA_3 (male 1)	□		□	Start Audio Send BA_3 (male 1)
	End Audio Receive BA_3 (male 1)	□		□	End Audio Send BA_3 (male 1)
	Start Audio Receive BA_4 (male 2)	□		□	Start Audio Send BA_4 (mal 2)
	End Audio Receive BA_4 (male 2)	□		□	End Audio Send BA_4 (male 2)
1 s	Pause				
	Start DTMF Send AB_1	□		□	Start DTMF Receive AB_1
	End DTMF Send AB_1	□		□	End DTMF Receive AB_1
	Start DTMF Receive BA_1	□		□	Start DTMF Send BA_1
	End DTMF Receive BA_1	□		□	End DTMF Send BA_1
	BYE	→		→	BYE
	200 OK	□		□	200 OK

10.3 End-to-end audio delay

See [ETSI TS 103 222-3].

This parameter represents the global delay from one user to the other. This indicator takes into account the transmission delay of networks but also the processing delay in sending and receiving terminals. The end-to-end delay can be measured acoustically from mouth to ear, from one access point to the other. The delay can be calculated based on cross correlation between the signal at the MRP (at one access) and the signal at the ERP (at the other access) using the test methods as described e.g., in [ETSI ES 202 737] and [ETSI ES 202 739].

Electrically the end-to-end delay can be measured based on cross correlation between the signal at the electrical measurement point at one access and the signal at the electrical measurement point at the other access.

The test signal consists of a series of composite source signals (CSSs) using a nominal network level of -16 dBm0 as described in [ITU-T P.501]. The test signal consists of the voiced part as described in [ITU-T P.501] followed by a pseudo random noise sequence with a periodicity of minimum 500 ms (described also in [ETSI ES 202 737] and [ETSI ES 202 739]).

NOTE – If the expected delay is higher than 500 ms a pseudo random sequence with a higher periodicity should be used.

10.4 End -to-end audio delay variation

See [ETSI TS 103 222-3].

The test signal consists of a series of CSSs using a nominal network level of -16 dBm0 with a total duration of 120 seconds. The pause of the CSS-sequence should be 150 ms. The delay of every CSS should be measured.

The delay variation for each CSS $D(i)$ compared to the first CSS (as described in [ITU-T P.501]) of the analysis period is calculated:

$$D(i) = T1 - Ti$$

With:

T1: Delay of the first CSS
 Ti: Delay CSS number i

10.5 Early media call flow options and listening speech quality

10.5.1 Introduction

Early media refers to media (e.g., audio and video) which are exchanged before a particular session is accepted by the called user (in terms of the signalling). Within a dialogue, early media occurs from the moment the initial INVITE is sent until the UAS generates a final response. It may be unidirectional or bidirectional, and can be generated by the caller, the called party, or both. Typical examples of early media generated by the called party are ringing tone and announcements (e.g., queuing status). Early media generated by the caller typically consists of voice commands or DTMF tones to drive interactive voice response (IVR) systems. See Figure 10-3.

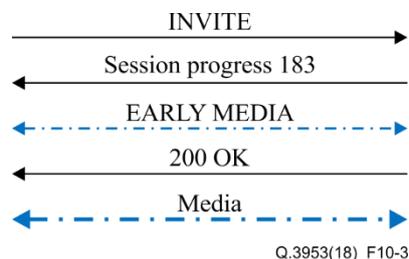


Figure 10-3 – Early media SIP overview

10.5.1.1 Call flow 1

The call flow below requires that the following be configured in the incoming SIP profile:

Precondition support: Supported/Required

P-Early-Media: Enabled

Table 10-4 – Early media with preconditions, option 1

Test equipment A		NETWORK		Test equipment B
CALL A to B				
INVITE	→	Supported 100rel, precondition SDP: a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv	→	INVITE
100 Trying	←		←	100 Trying
183 Session Progress	←	Supported 100rel, precondition SDP: a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv a=conf:qos remote sendrecv	←	183 Session Progress
PRACK	→		→	PRACK
200 PRACK	←		←	200 PRACK
UPDATE	→	SDP a=curr:qos local sendrecv a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv	→	UPDATE
200 OK UPDATE	←	SDP a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv	←	200 OK UPDATE
Start early media quality test				
Start Audio Receive BA_1 (female and male)	←		←	Start Audio Send BA_1 (female and male)
End Audio Receive BA_1	←		←	End Audio Send BA_1
Stop early media quality test				
180 Ringing	←		←	180 Ringing
200 OK	←	Supported 100rel SDP: a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv	←	200 OK
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

10.5.1.2 Call flow 2

The call flow below requires that the following be configured in the incoming SIP profile:

- Precondition support: Supported/Required
- P-Early-Media: Enabled.

Table 10-5 – Early media with preconditions, option 2

Test equipment A		NETWORK		Test equipment B
CALL A to B				
INVITE	→	Supported 100rel, precondition SDP: a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv	→	INVITE
100 Trying	←		←	100 Trying
183 Session Progress	←	Supported 100rel, precondition SDP: a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv a=conf:qos remote sendrecv	←	183 Session Progress
PRACK	→		→	PRACK
200 PRACK	←		←	200 PRACK
UPDATE	→	SDP a=curr:qos local sendrecv a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv	→	UPDATE
200 OK UPDATE	←	SDP a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv	←	200 OK UPDATE
Start early media quality test				
Start Audio Receive BA_1 (female and male)	←		←	Start Audio Send BA_1 (female and male)
End Audio Receive BA_1	←		←	End Audio Send BA_1
Stop early media quality test				
183 Session Progress	←	Supported 100rel SDP: a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv	←	183 Session Progress
PRACK	→		→	PRACK
200 PRACK	←		←	200 PRACK
180 Ringing	←		←	180 Ringing
200 OK	←	Supported 100rel SDP: a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv	←	200 OK
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

Table 10-6 – Precondition options

Require Header = <i>Precondition</i>	Supported Header = <i>Precondition</i>	Supported Header = <i>100rel</i>	Comments
YES	n/a	Yes	Precondition applied to 183 Session Progress sent to Network A
		No	
	NO	Yes	
		No	Call will be rejected
NO	YES	Yes	Precondition applied to 183 Session Progress sent to Network A
		No	
	NO	Yes	
		No	Precondition not applied. Call proceeds

10.5.2 Early media listening speech quality convergence quality test

See [ETSI TS 103 222-3].

For the synchronization of the voice samples a 700 Hz tone (100 ms signal) as trigger event can be used.

The principle of testing 'early media' is the same as defined for the convergence quality test according to clause 5.15.2 of [ITU-T Q.3933] reference benchmarking, background traffic profiles and KPIs for VoIP and FoIP in fixed networks. However, only one speech sample (male/female voice) has to be transmitted from the called party to the calling party emulating the 'early media' transfer. The general technical aspects for this speech samples are the same as defined in clause 5.15.2 of [ITU-T Q.3933]

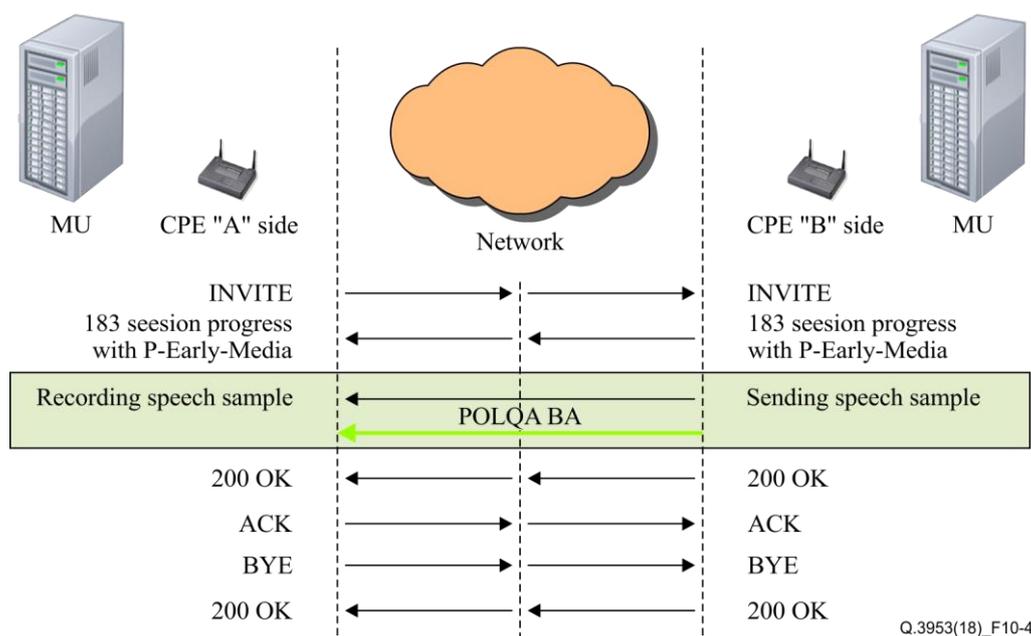


Figure 10-4 – Early media generated by the called party – general overview

10.6 Fax transmission

[ITU-T Q.4016] contains the testing specification of call establishment procedures based on SIP/SDP and [ITU-T H.248] for a real-time fax over IP service. The listed tests have to be interpreted as "minimal requirements" for fax support between SIP-enabled devices for real-time fax over IP.

10.7 ViLTE KPI

See [ETSI TS 126 114]

For the following video codecs the following KPI applies:

- Bi-directional video (ITU-T H.264 advanced video coding (AVC) level 1.1, IPv6, RTCP and MBR>GBR bearer).
- The video bandwidths used for defining maximum bit rate (MBR) and guaranteed bit rate (GBR) are assumed to be 192 kbit/s and 64 kbit/s, respectively.
- Bi-directional video (ITU-T H.264 AVC level 1.2, 384 kbit/s, IPv4, RTCP and MBR=GBR bearer).
- The video bandwidth is assumed to be 384 kbit/s and the IPv4 overhead 20 kbit/s (assuming 15 fps and 4 IP packets per frame), resulting in 404 kbit/s.
- Bi-directional video (ITU-T H.264 AVC level 1.2, 384 kbit/s, IPv6, RTCP and MBR=GBR bearer).
- The video bandwidth is assumed to be 384 kbit/s and the IPv6 overhead 32 kbit/s (assuming 15fps and 4 IP packets per frame), resulting in 416 kbit/s.
- Bi-directional video (ITU-T H.264 AVC level 1.2, IPv4, RTCP and MBR>GBR bearer).
- The video bandwidths used for defining MBR and GBR are assumed to be 384 kbit/s and 192 kbit/s, respectively. The IPv4 overhead is 20 kbit/s (assuming 15 fps and 4 IP packets per frame) for MBR and 10 kbit/s (assuming 15 fps and 2 IP packets per frame) for GBR, resulting in 404 kbit/s and 202 kbit/s, respectively.
- Bi-directional video (ITU-T H.264 AVC level 1.2, IPv6, RTCP and MBR>GBR bearer).
- The video bandwidths used for defining MBR and GBR are assumed to be 384 kbit/s and 192 kbit/s, respectively. The IPv6 overhead is 32 kbit/s (assuming 15 fps and 4 IP packets per frame) for MBR and 16 kbit/s (assuming 15 fps and 2 IP packets per frame) for GBR, resulting in 416 kbit/s and 208 kbit/s, respectively.
- Bi-directional video (H.265 (HEVC) main profile, main tier, level 3.1, 500 kbit/s, IPv6, RTCP and MBR=GBR bearer).
- The video bandwidth is assumed to be 500 kbit/s and the IPv6 overhead 36 kbit/s (assuming 25 fps, 3 IP packets per frame and IPv6), resulting in 540 kbit/s. Adding 5 percent for RTCP increases the bandwidth by 27 kbit/s. However, the RTCP bandwidth is limited to max 14 kbit/s, see clause 7.3.1 of [ETSI TS 126 114]. Rounding up to the next higher integer multiple of 8 kbit/s gives 560 kbit/s.
- Bi-directional video (H.265 (HEVC) main profile, main tier, level 3.1, 500/40 kbit/s, IPv6, RTCP and MBR>GBR bearer).
- The video bandwidths used for defining MBR and GBR are assumed to be 500 kbit/s and 40 kbit/s, respectively. The IPv6 overhead is 36 kbit/s (assuming 25 fps and 3 IP packets per frame) for MBR and 2.4 kbit/s (assuming QCIF, 5 fps and 1 IP packets per frame) for GBR, resulting in 540 kbit/s and 45 kbit/s, respectively.
- Bi-directional video (H.265 (HEVC) main profile, main tier, level 3.1, 600 kbit/s, IPv6, RTCP and MBR=GBR bearer).
- The video bandwidth is assumed to be 600 kbit/s and the IPv6 overhead 36 kbit/s (assuming 25 fps, 3 IP packets per frame and IPv6), resulting in 640 kbit/s.
- Bi-directional video (H.265 (HEVC) main profile, main tier, level 3.1, 600/40 kbit/s, IPv6, RTCP and MBR>GBR bearer).
- The video bandwidths used for defining MBR and GBR are assumed to be 600 kbit/s and 40 kbit/s, respectively. The IPv6 overhead is 36 kbit/s (assuming 25 fps and 3 IP packets per

- frame) for MBR and 2.4 kbit/s (assuming QCIF, 5 fps and 1 IP packets per frame) for GBR, resulting in 640 kbit/s and 45 kbit/s, respectively.
- Bi-directional video (H.265 (HEVC) main profile, main tier, level 3.1, 650 kbit/s, IPv6, RTCP and MBR=GBR bearer).
 - The video bandwidth is assumed to be 650 kbit/s and the IPv6 overhead 36 kbit/s (assuming 25 fps, 3 IP packets per frame and IPv6), resulting in 690 kbit/s. Adding 5% for RTCP increases the bandwidth by 34,5 kbit/s. However, the RTCP bandwidth is limited to max 14 kbit/s, see clause 7.3.1 of [ETSI TS 126 114]. Rounding up to the next higher integer multiple of 8 kbit/s gives 704 kbit/s.
 - Bi-directional video (H.265 (HEVC) main profile, main tier, level 3.1, 650/40 kbit/s, IPv6, RTCP and MBR>GBR bearer).
 - The video bandwidths used for defining MBR and GBR are assumed to be 650 kbit/s and 40 kbit/s, respectively. The IPv6 overhead is 36 kbit/s (assuming 25 fps and 3 IP packets per frame) for MBR and 2.4 kbit/s (assuming QCIF, 5 fps and 1 IP packets per frame) for GBR, resulting in 690 kbit/s and 45 kbit/s, respectively.
 - Bi-directional video (H.265 (HEVC) main profile, main tier, level 3.1, 750 kbit/s, IPv6, RTCP and MBR=GBR bearer).
 - The video bandwidth is assumed to be 750 kbit/s and the IPv6 overhead 48 kbit/s (assuming 25 fps, 4 IP packets per frame and IPv6), resulting in 800 kbit/s.
 - Bi-directional video (H.265 (HEVC) main profile, main tier, level 3.1, 750/40 kbit/s, IPv6, RTCP and MBR>GBR bearer).
 - The video bandwidths used for defining MBR and GBR are assumed to be 750 kbit/s and 40 kbit/s, respectively. The IPv6 overhead is 48 kbit/s (assuming 25 fps and 4 IP packets per frame) for MBR and 2.4 kbit/s (assuming QCIF, 5 fps and 1 IP packets per frame) for GBR, resulting in 800 kbit/s and 45 kbit/s, respectively.

Table 10-7 – QoS requirements for bi-directional video

Traffic class	Conversational class	Notes
Maximum SDU size (octets)	1 400	Maximum size of IP packets.
Residual BER	10^{-5}	Reflects the desire to have a medium level of protection to achieve an acceptable compromise between packet loss rate and speech transport delay and delay variation.
SDU error ratio	$7 \cdot 10^{-3}$	A packet loss rate of 0.7% per wireless link is in general sufficient for video services.
Transfer delay (ms)	170 ms	Indicates maximum delay for 95th percentile of the distribution of delay for all delivered SDUs between the UE and the PS domain during the lifetime of a bearer service. Permits the derivation of the radio access network (RAN) part of the total transfer delay for the radio access bearer. This attribute allows RAN to set transport formats and H-ARQ/ARQ parameters such as the discard timer.
Lip synchronization	180/90 ms	The acceptability thresholds for lip synchronization are 185 ms when sound is delayed with respect to the video, and 90 ms when sound is advanced with respect to the video.

Annex A

Excel test list

(This annex forms an integral part of this Recommendation.)

The Excel test list is a normative part of this Recommendation. The interconnection and roaming scenarios should be selected depending on the network infrastructure (Table 6-1) and company strategy.

Please see the attachment to this Recommendation.

Appendix I

Examples of codecs which can be used in the case of interworking/roaming

(This appendix does not form an integral part of this Recommendation.)

Mostly used codecs:

- 1) AMR-WB
- 2) FR-AMR-WB
- 3) AMR-NB V2 Modeset 4.75/5.90/7.40/12.20
- 4) AMR-NB Modeset 4.75/5.90/7.40/12.20
- 5) FR-AMR
- 6) GSM-EFR
- 7) GSM-FR
- 8) ITU-T G.729
- 9) ITU-T G.711 a-Law
- 10) ITU-T T.38
- 11) IETF Clearmode

Table I.1 – Codecs used in case of interworking/roaming from 4G to 3G

VARIABLE	Calling mobile carrier 4G	Terminating mobile carrier UMTS 2G/3G codec
VA_01	VOLTE (4G), AMR-WB TRFO	UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6,60, 8,85, 12,65 kbit/s) without BICC
VA_02		UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6,60, 8,85, 12,65 kbit/s) with BICC
VA_03		UMTS (3G) – UMTS-AMR-NB V2 Config Set 1 (4,75, 5,90, 7,40, 12,2 kbit/s) without BICC
VA_04		UMTS (3G) – UMTS- AMR-NB V2 Config Set 1 (4,75, 5,90, 7,40, 12,2 kbit/s) with BICC
VA_05		GSM (2G) – GSM-EFR/GSM-FR on UE without BICC
VA_06		GSM (2G) – GSM-EFR/GSM-FR on UE with BICC
VA_07		GSM (2G) – GSM-FR/GSM-FR on UE without BICC
VA_08		GSM (2G) – GSM-FR/GSM-FR on UE with BICC
VA_09	VOLTE (4G), AMR-WB noTRFO	UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6,60, 8,85, 12,65 kbit/s) without BICC
VA_10		UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6,60, 8,85, 12,65 kbit/s) with BICC
VA_11		UMTS (3G) – UMTS-AMR2 Config Set 1 (4,75, 5,90, 7,40, 12,2 kbit/s) without BICC
VA_12		UMTS (3G) - UMTS-AMR2 Config Set 1 (4,75, 5,90, 7,40, 12,2 kbit/s) with BICC
VA_13		GSM (2G) – GSM-EFR/GSM-FR on UE without BICC
VA_14		GSM (2G) – GSM-EFR/GSM-FR on UE with BICC
VA_15		GSM (2G) – GSM-FR/GSM-FR on UE without BICC
VA_16		GSM (2G) – GSM-FR/GSM-FR on UE with BICC

Table I.2 – Codecs used in case of interworking/roaming 3G to 4G

VARIABLE	Calling mobile carrier 3G	Terminating mobile carrier 4G
VA_1	UMTS (3G), TRFO UMTS-FR-AMR-WB Config Set 0 (6,60, 8,85, 12,65 kbit/s) UMTS-AMR2 Config Set 1 (4,75, 5,90, 7,40, 12,2 kbit/s)	<ul style="list-style-type: none"> • VoLTE (4G) <ul style="list-style-type: none"> – AMR-WB mode-set 0,1,2 (6,60, 8,85, 12,65 kbit/s); mode-change-period=2; mode-change-neighbor=1; 20 ms; preconditions used; tel. events – AMR NB mode-set 0,2,4,7 (4,75, 5,90, 7,40, 12,2 kbit/s); mode-change-period=2; mode-change-neighbor=1; 20 ms, preconditions used, tel. events
VA_2	UMTS (3G), noTRFO UMTS-FR-AMR-WB Config Set 0 (6,60, 8,85, 12,65 kbit/s) UMTS-AMR2 Config Set 1 (4,75, 5,90, 7,40, 12,2 kbit/s)	<ul style="list-style-type: none"> • VoLTE (4G) <ul style="list-style-type: none"> – AMR-WB mode-set 0,1,2 (6,60, 8,85, 12,65 kbit/s); mode-change-period=2; mode-change-neighbor=1; 20 ms; preconditions used; tel. events – AMR NB mode-set 0,2,4,7 (4,75, 5,90, 7,40, 12,2 kbit/s); mode-change-period=2; mode-change-neighbor=1; 20 ms, preconditions used, tel. events
VA_3	GSM (2G), TRFO GSM-FR-AMR-WB	<ul style="list-style-type: none"> • VoLTE (4G) <ul style="list-style-type: none"> – AMR-WB mode-set 0,1,2 (6,60, 8,85, 12,65 kbit/s); mode-change-period=2; mode-change-neighbor=1; 20 ms; preconditions used; tel. events – AMR NB mode-set 0,2,4,7 (4,75, 5,90, 7,40, 12,2 kbit/s); mode-change-period=2; mode-change-neighbor=1; 20 ms, preconditions used, tel. events
VA_4	GSM (2G), noTRFO GSM-FR-AMR-WB	<ul style="list-style-type: none"> • VoLTE (4G) <ul style="list-style-type: none"> – AMR-WB mode-set 0,1,2 (6,60, 8,85, 12,65 kbit/s); mode-change-period=2; mode-change-neighbor=1; 20 ms; preconditions used; tel. events – AMR NB mode-set 0,2,4,7 (4,75, 5,90, 7,40, 12,2 kbit/s); mode-change-period=2; mode-change-neighbor=1; 20 ms, preconditions used, tel. events
VA_5	GSM (2G), TRFO GSM-FR-AMR	<ul style="list-style-type: none"> • VoLTE (4G) <ul style="list-style-type: none"> – AMR-WB mode-set 0,1,2 (6,60, 8,85, 12,65 kbit/s); mode-change-period=2; mode-change-neighbor=1; 20 ms; preconditions used; tel. events – AMR NB mode-set 0,2,4,7 (4,75, 5,90, 7,40, 12,2 kbit/s); mode-change-period=2; mode-change-neighbor=1; 20 ms, preconditions used, tel. events
VA_6	GSM (2G), noTRFO GSM-FR-AMR	<ul style="list-style-type: none"> • VoLTE (4G) <ul style="list-style-type: none"> – AMR-WB mode-set 0,1,2 (6,60, 8,85, 12,65 kbit/s); mode-change-period=2; mode-change-neighbor=1; 20 ms; preconditions used; tel. events – AMR NB mode-set 0,2,4,7 (4,75, 5,90, 7,40, 12,2 kbit/s); mode-change-period=2; mode-change-neighbor=1; 20 ms, preconditions used, tel. events
VA_7	GSM (2G), TRFO GSM-HR-AMR	<ul style="list-style-type: none"> • VoLTE (4G) <ul style="list-style-type: none"> – AMR-WB mode-set 0,1,2 (6,60, 8,85, 12,65 kbit/s); mode-change-period=2; mode-change-neighbor=1; 20 ms; preconditions used; tel. events – AMR NB mode-set 0,2,4,7 (4,75, 5,90, 7,40, 12,2 kbit/s); mode-change-period=2; mode-change-neighbor=1; 20 ms, preconditions used, tel. events
VA_8	GSM (2G), noTRFO GSM-HR-AMR	<ul style="list-style-type: none"> • VoLTE (4G) <ul style="list-style-type: none"> – AMR-WB mode-set 0,1,2 (6,60, 8,85, 12,65 kbit/s); mode-change-period=2; mode-change-neighbor=1; 20 ms; preconditions used; tel. events – AMR NB mode-set 0,2,4,7 (4,75, 5,90, 7,40, 12,2 kbit/s); mode-change-period=2; mode-change-neighbor=1; 20 ms, preconditions used, tel. events

Table I.3 – Codecs used in case of interworking/roaming 3G to 3G

VARIABLE	Calling mobile carrier 3G	Terminating mobile carrier 3G
VA_1	UMTS (3G), TRFO UMTS-FR-AMR-WB Config Set 0 (6.60, 8.85, 12.65 kbit/s) UMTS-AMR2 Config Set 1 (4.75, 5.90, 7.40, 12.2 kbit/s)	<ul style="list-style-type: none"> • UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6.60, 8.85, 12.65 kbit/s) without BICC • UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6.60, 8.85, 12.65 kbit/s) with BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (12.2 kbit/s) without BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (4.75, 5.90, 7.40, 12.2 kbit/s) without BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (12.2 kbit/s) with BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (4.75, 5.90, 7.40, 12.2 kbit/s) with BICC • GSM (2G) – GSM-EFR/GSM-FR on UE without BICC • GSM (2G) – GSM-EFR/GSM-FR on UE with BICC
VA_2	UMTS (3G), noTRFO UMTS-FR-AMR-WB Config Set 0 (6.60, 8.85, 12.65 kbit/s) UMTS-AMR2 Config Set 1 (4.75, 5.90, 7.40, 12.2 kbit/s)	<ul style="list-style-type: none"> • UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6.60, 8.85, 12.65 kbit/s) without BICC • UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6.60, 8.85, 12.65 kbit/s) with BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (12.2 kbit/s) without BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (4.75, 5.90, 7.40, 12.2 kbit/s) without BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (12.2 kbit/s) with BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (4.75, 5.90, 7.40, 12.2 kbit/s) with BICC • GSM (2G) – GSM-EFR/GSM-FR on UE without BICC • GSM (2G) – GSM-EFR/GSM-FR on UE with BICC
VA_3	GSM (2G), TRFO GSM-FR-AMR-WB	<ul style="list-style-type: none"> • UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6.60, 8.85, 12.65 kbit/s) without BICC • UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6.60, 8.85, 12.65 kbit/s) with BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (12.2 kbit/s) without BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (4.75, 5.90, 7.40, 12.2 kbit/s) without BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (12.2 kbit/s) with BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (4.75, 5.90, 7.40, 12.2 kbit/s) with BICC • GSM (2G) – GSM-EFR/GSM-FR on UE without BICC • GSM (2G) – GSM-EFR/GSM-FR on UE with BICC
VA_4	GSM (2G), noTRFO GSM-FR-AMR-WB	<ul style="list-style-type: none"> • UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6.60, 8.85, 12.65 kbit/s) without BICC • UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6.60, 8.85, 12.65 kbit/s) with BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (12.2 kbit/s) without BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (4.75, 5.90, 7.40, 12.2 kbit/s) without BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (12.2 kbit/s) with BICC

Table I.3 – Codecs used in case of interworking/roaming 3G to 3G

VARIABLE	Calling mobile carrier 3G	Terminating mobile carrier 3G
		<ul style="list-style-type: none"> • UMTS (3G) – UMTS-AMR2 Config Set 1 (4.75, 5.90, 7.40, 12.2 kbit/s) with BICC • GSM (2G) – GSM-EFR/GSM-FR on UE without BICC • GSM (2G) – GSM-EFR/GSM-FR on UE with BICC
VA_5	GSM (2G), TRFO GSM-FR-AMR	<ul style="list-style-type: none"> • UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6.60, 8.85, 12.65 kbit/s) without BICC • UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6.60, 8.85, 12.65 kbit/s) with BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (12.2 kbit/s) without BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (4.75, 5.90, 7.40, 12.2 kbit/s) without BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (12.2 kbit/s) with BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (4.75, 5.90, 7.40, 12.2 kbit/s) with BICC • GSM (2G) – GSM-EFR/GSM-FR on UE without BICC • GSM (2G) – GSM-EFR/GSM-FR on UE with BICC
VA_6	GSM (2G), noTRFO GSM-FR-AMR	<ul style="list-style-type: none"> • UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6.60, 8.85, 12.65 kbit/s) without BICC • UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6.60, 8.85, 12.65 kbit/s) with BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (12.2 kbit/s) without BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (4.75, 5.90, 7.40, 12.2 kbit/s) without BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (12.2 kbit/s) with BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (4.75, 5.90, 7.40, 12.2 kbit/s) with BICC • GSM (2G) – GSM-EFR/GSM-FR on UE without BICC • GSM (2G) – GSM-EFR/GSM-FR on UE with BICC
VA_7	GSM (2G), TRFO GSM-HR-AMR	<ul style="list-style-type: none"> • UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6.60, 8.85, 12.65 kbit/s) without BICC • UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6.60, 8.85, 12.65 kbit/s) with BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (12.2 kbit/s) without BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (4.75, 5.90, 7.40, 12.2 kbit/s) without BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (12.2 kbit/s) with BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (4.75, 5.90, 7.40, 12.2 kbit/s) with BICC • GSM (2G) – GSM-EFR/GSM-FR on UE without BICC • GSM (2G) – GSM-EFR/GSM-FR on UE with BICC
VA_8	GSM (2G), noTRFO GSM-HR-AMR	<ul style="list-style-type: none"> • UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6.60, 8.85, 12.65 kbit/s) without BICC • UMTS (3G) – UMTS-FR-AMR-WB Config Set 0 (6.60, 8.85, 12.65 kbit/s) with BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (12.2 kbit/s) without BICC • UMTS (3G) – UMTS-AMR2 Config Set 1 (4.75, 5.90, 7.40, 12.2 kbit/s) without BICC

Table I.3 – Codecs used in case of interworking/roaming 3G to 3G

VARIABLE	Calling mobile carrier 3G	Terminating mobile carrier 3G
		<ul style="list-style-type: none">• UMTS (3G) – UMTS-AMR2 Config Set 1 (12.2 kbit/s) with BICC• UMTS (3G) – UMTS-AMR2 Config Set 1 (4.75, 5.90, 7.40, 12.2 kbit/s) with BICC• GSM (2G) – GSM-EFR/GSM-FR on UE without BICC• GSM (2G) – GSM-EFR/GSM-FR on UE with BICC

Appendix II

Execution of grouped tests

(This appendix does not form an integral part of this Recommendation.)

Execution of grouped tests 2G/3G – VoLTE (SIP-I)		
	NNI Req.	
Test call	TS 101 585	Description
1)	SS_bcall_001	Ensure that call establishment using overlap sending is performed correctly.
	SS_bcall_003	
	SS_bcall_010	Ensure that the call clearing procedure is performed correctly when the called user clears after answer.
	SS_bcall_011	
	SS_bcall_012	The test call is successful in the case if the call setup time does not exceed the values listed in Table 9-1 and call is stable in unanswered and answered phases, the call remains in intelligible/high quality conversation phase for 80 seconds. The voice quality test procedures are described clause 8.2.
	SS_bcall_013	
	SS_bcall_014	
	SS_bcall_015	
	SS_bcall_016	
	SS_bcall_017	The test scenarios are listed in Table 6-5.
SS_DTMF 01	Ensure that the Request line in the INVITE contains in the user part the telephone number of the destination user equipment formatted as a 'tel' URI in the global number format and the host portion is set to the host name of the interconnected network. The user URI parameter is present and set to 'phone'.	
SIP-I		
SS_bcall_SIP-I_01	Ensure that if the Record-Route header is present in the INVITE establishes a communication between a user of Network A and a user of Network B the topmost header is set to the IBCF of Network A.	
SS_bcall_SIP-I_02		
SS_bcall_SIP-I_03	Ensure that the Via header is present in the INVITE establishes a communication between a user of Network A and a user of Network B and the topmost header is set to the IBCF of Network A and contains a branch parameter.	
SS_bcall_SIP-I_06		
SS_bcall_SIP-I_08	Ensure if a Record-Route header was present in the initial INVITE that the Record-Route header is present in the 180 Ringing provisional response as the first response from Network B upon a connection establish setup from Network A.	
	Ensure that if a Record-Route header was present in the initial INVITE the Route header may be present in the BYE request sent from the originating user equipment in Network A the topmost Route header or entry is set to the IBCF of Network B.	
	Ensure that if a Record-Route header was present in the initial INVITE the Route header may be present in the BYE request sent from the terminating user equipment in Network B the topmost Route header or entry is set to the IBCF of Network A.	
	Ensure that if a Record-Route header was present in the initial INVITE the Route header may be present in ACK from Network A upon a connection establishment from Network A is completed the topmost Route header or entry is set to the IBCF of Network B.	

Execution of grouped tests 2G/3G – VoLTE (SIP-I)		
	NNI Req.	
Test call	TS 101 585	Description
		<p>Ensure that call establishment and the correct handling of the SDP parameters of the INVITE message defined as: TYPE_SDP is performed correctly. Ensure that in the active call state the voice/data transfer on the media channels is performed correctly (e.g., testing QoS parameters). In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in Table 9-1 applies.</p>
		<p>SIP-I support, Basic call, IAM present in the INVITE request. Ensure that when a call initiated in the PSTN or the PLMN and the ISUP – SIP-I interworking is applicable in the originating network, an ISUP IAM is encapsulated in the initial INVITE request. Ensure that all the mandatory parameters in the IAM are present and the values are valid and the Transmission medium requirement parameter is consistent with the SDP.</p>
		<p>Ensure that the original IAM is encapsulated in any INVITE request. The test call is successful in the case if the call setup time does not exceed the values listed in Table 9-1 and call is stable in unanswered and answered phases, the call remains in intelligible/high quality conversation phase for 80 seconds. The voice quality test procedures are described clause 8.2.</p>
		<p>SIP-I support, Basic call, ACM present in the 180 response. Ensure that on receipt of a 180 Ringing provisional response and an SIP-I – ISUP interworking is applicable in the terminating network the Backward call indicators parameter in the encapsulated ACM is present and the values are valid. Ensure that the values of the optional parameters in the encapsulated ACM are valid.</p>
		<p>SIP-I support. Basic call, ANM present in a 200 OK INVITE response. Ensure that on receipt of a 200 OK INVITE final response and an SIP-I – ISUP interworking is applicable in the terminating network the ISUP ANM is encapsulated in the 200 OK. Ensure that the values of the optional parameters in the encapsulated ANM are valid.</p>
		<p>SIP-I support. Basic call, REL present in a BYE request sent from the terminating network. Ensure that a ISUP REL message is encapsulated in a BYE request sent in the release procedure initiated from the terminating user when SIP-I – ISUP interworking is applicable in the terminating network. Ensure the validity of the cause indicator in the encapsulated REL. Ensure that the ISUP RLC is encapsulated in the 200 OK BYE.</p>

Execution of grouped tests 2G/3G – VoLTE (SIP-I)		
	NNI Req.	
Test call	TS 101 585	Description
2)	SS_bcall_SIP-I_04 SS_bcall_SIP-I_05	<p>SIP-I support. Basic call, early ACM present in the 183 response. Ensure that on receipt of a 183 Session Progress provisional response and an SIP-I – ISUP interworking is applicable in the terminating network the Backward call indicators parameter in the encapsulated ACM is present and the value of the Called party's status indicator is set to 'no indication'. Ensure that the values of the optional parameters in the encapsulated ACM are valid.</p> <p>SIP-I support. Basic call, CPG present in a 180 response. Ensure that on receipt of a 180 Ringing provisional response and an SIP-I – ISUP interworking is applicable in the terminating network the Event indicator in the encapsulated CPG is present and set to 'ALERTING'. Ensure that the values of the optional parameters in the encapsulated CPG are valid.</p> <p>Transmission of DTMF Ensure that the ability of transmission of DTMF can be performed by the originating and destination user. The transmission can be done by:</p> <ul style="list-style-type: none"> • DTMF in the RTP stream • Either by indicating in the SDP offer in the RTP stream • Or by the SIP INFO/NOTIFY method for DTMF tone generation <p>DTMF test: The DTMF test should consist DTMF tones (70 ms signal, 100 ms pause) and shall contain the tones 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, *, #. The transmission shall be tested in both directions.</p>
3)	SS_bcall_002 SS_bcall_SIP-I_07	<p>Ensure that the call clearing procedure is performed correctly when the calling user clears after answer;</p> <p>SIP-I support. Basic call, REL present in a BYE request sent from the originating network. Ensure that an ISUP REL message is encapsulated in a BYE request sent in the release procedure initiated from the originating user when ISUP – SIP-I interworking is applicable in the originating network. Ensure the validity of the cause indicator in the encapsulated REL. Ensure that the ISUP RLC is encapsulated in the 200 OK BYE.</p>
4)	SS_bcall_018	<p>First response 200 OK INVITE. Ensure that call establishment and the correctly if the called user answers with a 200 OK message.</p>

Bibliography

- [b-GSMA IR.65] GSM Association, *IMS Roaming, Interconnection and Interworking Guidelines Version 27.0*, August 2017.
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