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Part 2: Subsystem Configurations and Benchmarks

Reference

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ETSI

650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
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Contents

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Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee IMS Network Testing (INT).

The present document is part 2 of a multi-part deliverable covering the IMS/NGN Performance Benchmark, as identified below:

Part 1: "Core Concepts";

Part 2: "Subsystem Configurations and Benchmarks";

Part 3: "Traffic Sets and Traffic Profiles";

Part 4: "Reference Load network quality parameters".

1 Scope

The present document describes the performance benchmark methology for the IMS based services MMTel, Video Telephony and IMS/ PES. The terminology and concepts are described in TR 101 577 [i.11]. The present document is the second part of the multi-part deliverable which consists of four parts.

TS 186 008-1 [i.1] defines the overall benchmark descriptions, architectures, processes, and information models that are common to all specific benchmarking scenarios.

The present document contains the specific benchmarking use-cases and scenarios, along with scenario specific metrics and design objectives. It also defines the SUT configuration parameters. The present document also contains any required extensions to the overall descriptions present in the present document, if necessary for the specific scenario.

TS 186 008-3 [i.2] defines an initial benchmark test through the specification of a traffic set, traffic-time profile and benchmark test procedure.

TS 186 008-4 [i.3] defines Reference Load network quality parameters for the use cases defined in TS 186 008-2 [i.1].

2 References

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

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

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

2.1 Normative references

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

[1] Void.

[2] ETSI TS 123 002 (V11.4.0): "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Network architecture (3GPP TS 23.002 version 11.4.0 Release 11)".

2.2 Informative references

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

[i.1] ETSI TS 186 008-1: "IMS Network Testing (INT); IMS/NGN Performance Benchmark; Part 1: Core Concepts ".

[i.2] ETSI TS 186 008-3: "IMS Network Testing (INT); IMS/NGN Performance Benchmark; Part 3: Traffic Sets and Traffic Profiles".

[i.3] ETSI TS 186 008-4: "IMS Network Testing (INT); IMS/NGN Performance Benchmark; Part 4: Reference Load network quality parameters".

[i.4] Void.

[i.5] Recommendation ITU-T P.862.1: "Mapping function for transforming P.862 raw result scores to MOS-LQO".

[i.6] Void.

[i.7] ETSI TR 121 905: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Vocabulary for 3GPP Specifications (3GPP TR 21.905)".

[i.8] Recommendation ITU-T P.863 (01-2011): "Perceptual objective listening quality assessment".

[i.9] Void.

[i.10] Void.

[i.11] ETSI TR 101 577 (V1.1.1): "Methods for Testing and Specifications (MTS); Performance Testing of Distributed Systems; Concepts and Terminology".

[i.12] Void.

[i.13] IETF RFC 3840 Indicating User Agent Capabilities in the Session Initiation Protocol (SIP).

[i.14] Recommendation ITU-T Q.543: "Digital exchange performance design objective".

[i.15] ETSI TS 183 043: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS-based PSTN/ISDN Emulation; Stage 3 specification".

[i.16] ETSI TS 183 036: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); ISDN/SIP interworking; Protocol specification".

[i.17] ETSI TS 124 229: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3 (3GPP TS 24.229)".

[i.18] IETF RFC 3261 June 2002: SIP: "Session Initiation Protocol".

[i.19] Void.

[i.20] 3GPP TS 36.300 E-UTRA and E-UTRAN Overall Description; Stage 2.

[i.21] ETSI TS 186 025-2: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS/PES Performance Benchmark; Part 2: Subsystem Configurations and Benchmarks".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the IMS benchmarking terms and definitions given in TR 101 577 [i.11] apply.

3.2 Abbreviations

For the purposes of the present document, the abbreviations given in TR 121 905 [i.7] and the following apply:

PES IMS- based PSTN/ISDN Emulation Sub-system

4 System Under Test (SUT) subsystems

4.1 IMS/MMtel

An IMS/NGN benchmark is required to allow not only a complete IMS network but also subsystems of a network corresponding to discrete products that may be available from a supplier. To address this requirement in this multi-part deliverable, a series of subsystems are defined, which will serve as a System Under Test (SUT) for a benchmark test. IMS/NGN elements that do not appear in a subsystem are regarded as part of the test environment, which is present for a subsystem to function, but which is not itself subject to benchmarking.

Figure 1 depicts the IMS Reference Architecture. The components of the architecture are the primary building blocks, which are either defined by the IMS standard, or defined by external standards and referenced by IMS. The links between the primary building block represent reference points over which the building blocks communicate with each other.

The reference architecture is a logical architecture; no mapping of functional elements to hardware or software component is mandated.

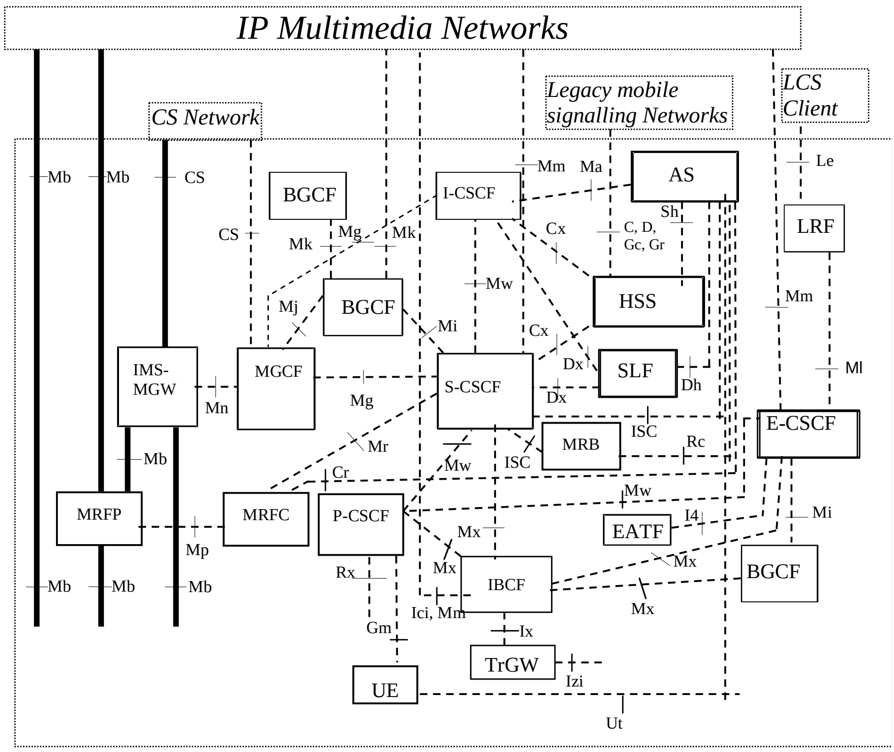


Figure 1: Overview of IMS Functional Entities [TS 123 002]

For the purposes of benchmarking, however, certain rules concerning subsystem configurations are required. These rules that benchmark measurements taken from equivalent subsystems of different vendors are comparable with one another.

The general guidelines for defining an SUT configuration are:

- All of the functional elements of the subsystem shall be present in the SUT configuration.
- All hardware elements used in the implementation of the SUT configuration shall be completely enumerated.
- All the QoS spec measurements defined at the interfaces to the SUT shall be collected as specified in the benchmark test.

- All hardware-specific measurements (e.g. CPU utilization, memory utilization, fabric bandwidth) specified in the benchmark test should be collected for all hardware elements used in the implementation of the SUT configuration.
- SUT interface characteristics shall be specified so that they can be emulated by the test system, including:
  - Security (e.g. IPSec, TLS, DTLS, etc.).
  - Interface network characteristics (e.g. up and down bandwidth, up and down latency).

4.1.1 Session Control Subsystem (SCS)

The Session Control Subsystem (SCS) consists of the P-CSCF, I-CSCF, and S-CSCF, and HSS components, as depicted in figure 2.

A valid SCS configuration consists of the set of x-CSCF building blocks, as well as the database functions HSS and SLF that support their functionality. The reference points for the SCS are the G_m reference point between the UE traffic generator and the home and visited P-CSCFs, the M_r reference point between the S-CSCF and the Simulated MRFC, the M_j reference point between the S-CSCF and the Simulated BGCF, and the test system management interface to the HSS and SLF databases.

A SUT for this subsystem may consist of either one or two SCS configurations,to allow benchmark tests to use a combination of local and roaming simulated subscribers.

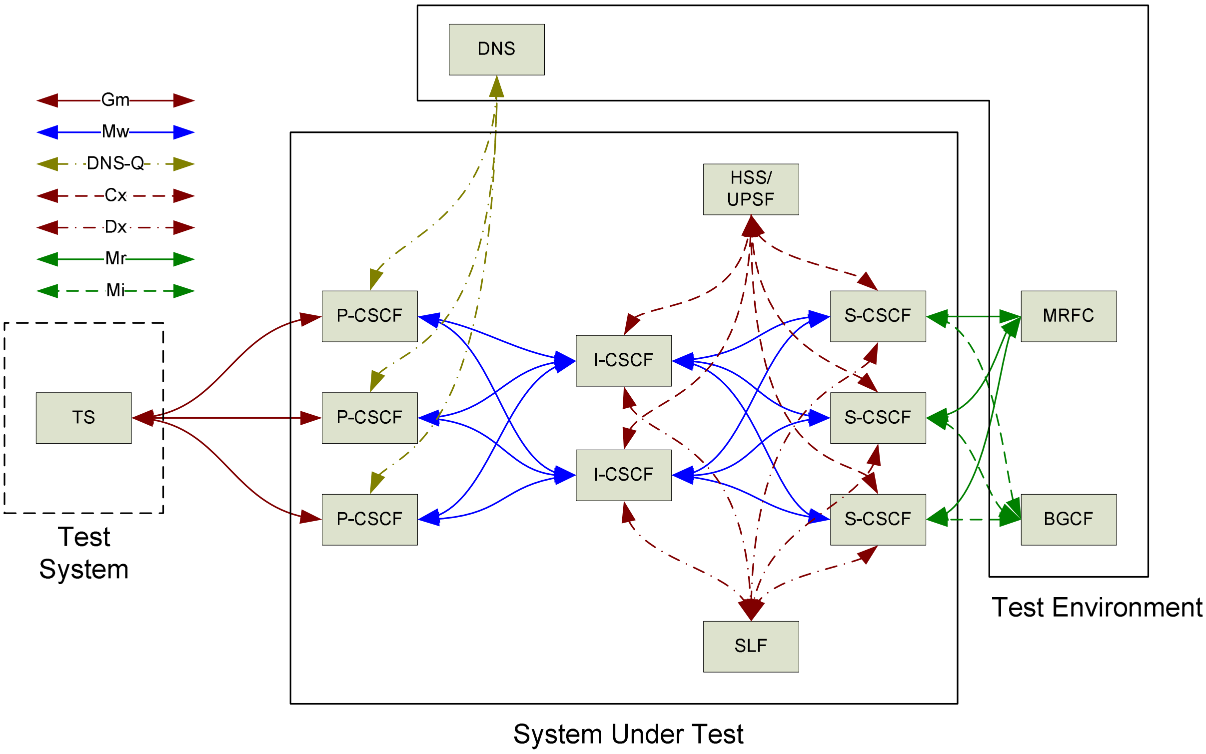


Figure 2: SUT topology for SCS

4.1.2 HSS subsystem

This subsystem refers to the HSS.

4.1.3 P-CSCF subsystem

The P-CSCF consists of the P-CSCF component, as depicted in figure 3.

A valid P-CSCF configuration consists of the set of P-CSCF building blocks. The reference points for the subsystem are the G_m reference point between the UE traffic generator and the home and visited P-CSCFs, and the M_w reference points to the other simulated components (which are part of the TS in in this configuration).

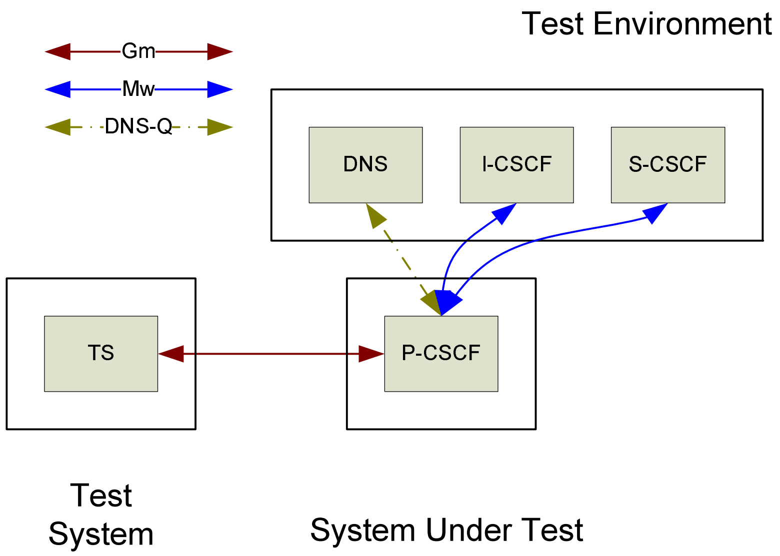


Figure 3: P-CSCF Subsystem

4.1.4 S/I-CSCF subsystem

The S/I-CSCF Subsystem consists of the S-CSCF, I-CSCF, HSS, and SLF components, as depicted in figure 4.

A valid S/I-CSCF configuration consists of the set of S-CSCF and I-CSCF building blocks, as well as the database functions HSS and SLF that support their functionality. The reference points for the subsystem are the M_w reference point between the simulated P-CSCF and the S/I-CSCFs.

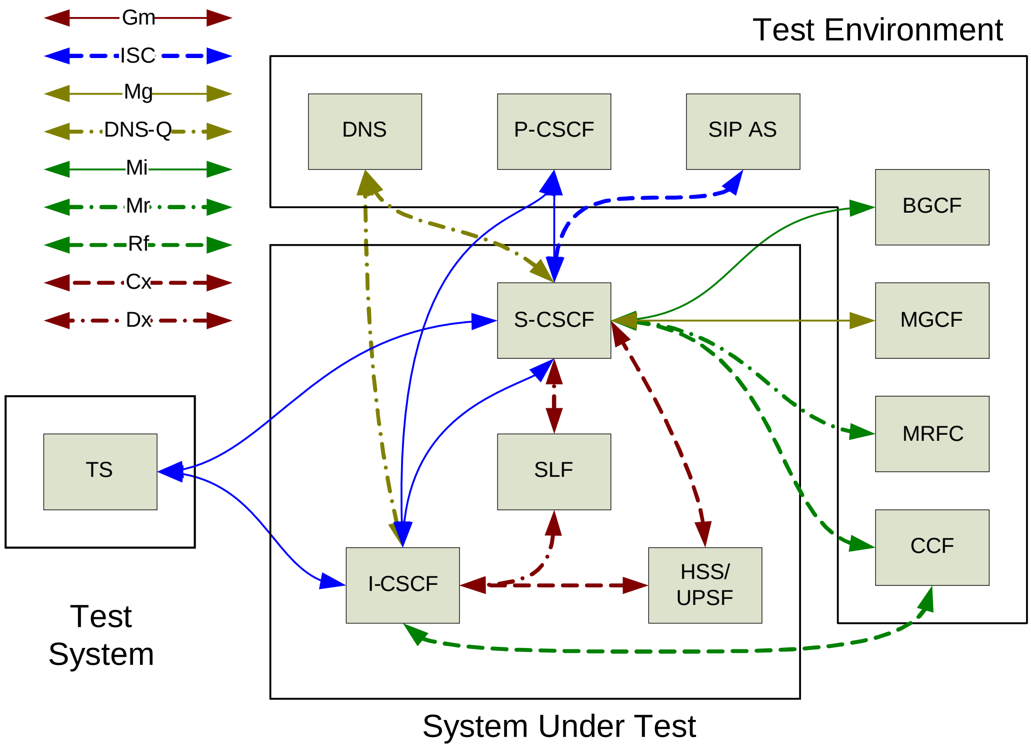


Figure 4: S/I-CSCF subsystem

4.2 IMS/PES

The System Under Test (SUT) subsystems for IMS/PES is described in TS 186 025-2 [i.21].

4.3 IMS to IMS/PES

The IMS and IMS/PES configuration are depicted in figure 5.

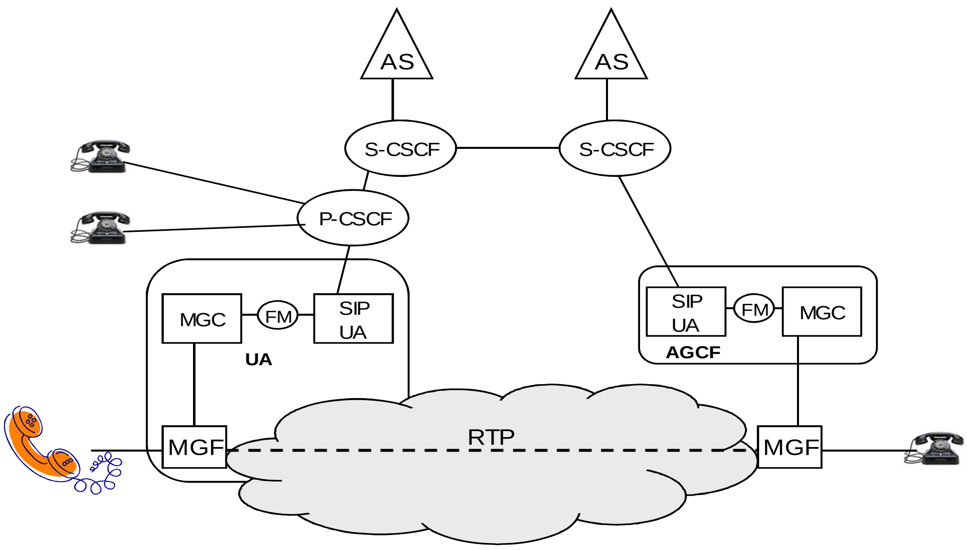


Figure 5: MMtel and IMS/PES configuration

4.4 IMS/LTE Basic Configuration

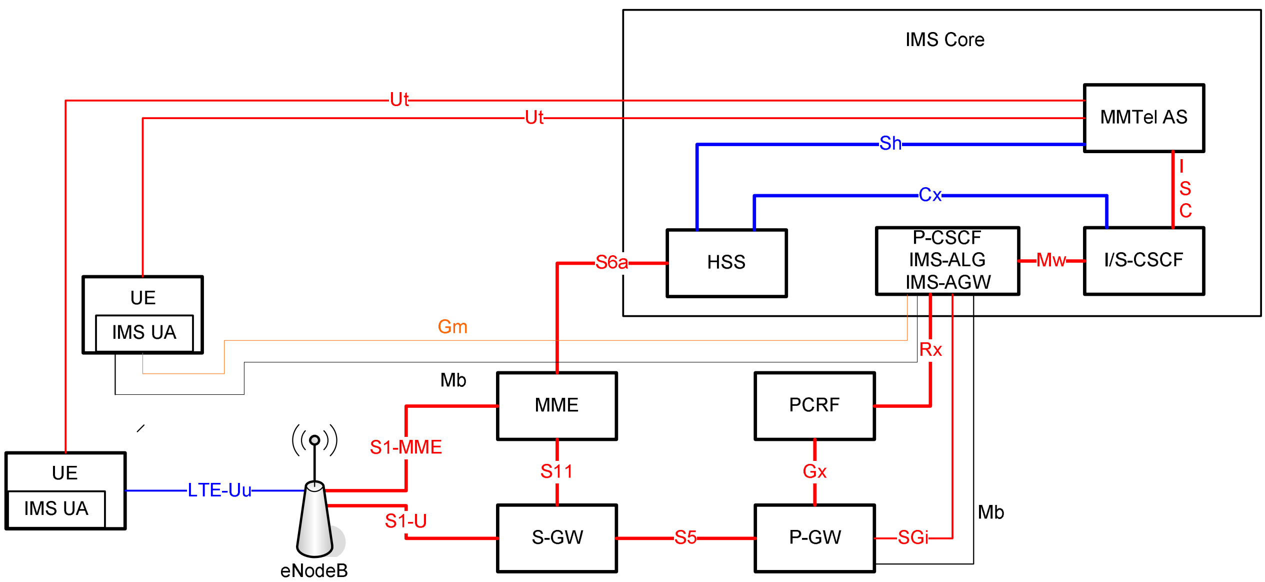


Figure 6: IMS/LTE Basic Configuration

4.5 VoLTE

The VoLTE components, are depicted in figure 7.

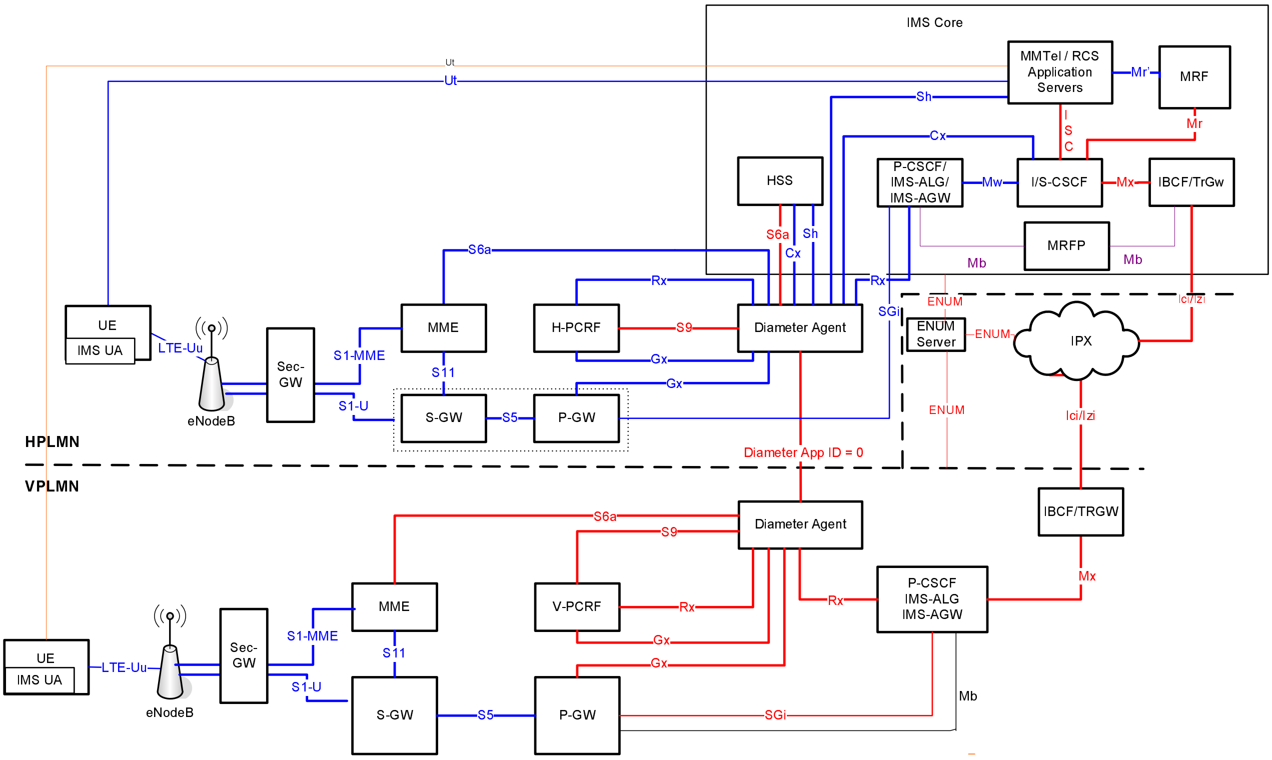


Figure 7: VoLTE Basic Configuration

5 Use cases

The following use cases, and corresponding tests, are currently defined. This clause attempts to define a set of basic use-cases and further ones can be defined similarly. These newly defined use-cases, or modifications to the ones presented here, will have to be described in a similar manner in the test report.

5.1 IMS

5.1.1 Registration/de-registration use-case 1

Registration is the first use-case that is employed when using an IMS network. During this operation the UE announces its contact location to the home domain registrar in order for the home network to route terminating messages towards it. It is performed by an UE when it is turned on. De-registration is the last operation that an UE performs before it is turned off and it is used to invalidate the registered contact information.

Because of security concerns, this operation has to be authenticated and the assigned S-CSCF challenges the UE using authentication vectors obtained from the HSS.

During the initial registration the P-CSCF also negotiates a set of security associations with the UE. Future registration/deregistration operations performed over these secure channels do not need to further be authenticated as the integrity of the messages is protected.

The registration has an attached expiration timer. Depending on the type of the network (fixed or mobile) and the usage patterns, this timer can vary from a few minutes up to one week, and it is negotiated between the UE and the home network. Before this timer expires, or when roaming to a new visited network, the UE has to start re-registration scenarios.

As part of this use-case, after registering, the UE will subscribe to its own registration status at the assigned S-CSCF. This subscription will need to be refreshed periodically, similarly to the registration. Unsubscription is not required, as the S-CSCF will automatically terminate it on de-registration. To avoid congestion, the notification timing is not strictly coupled to events that triggered them, and can have a delay in the order of seconds. The first one is sent shortly after subscription, and as a rule, the UE should be ready to respond to notification at any moment during the subscription period.

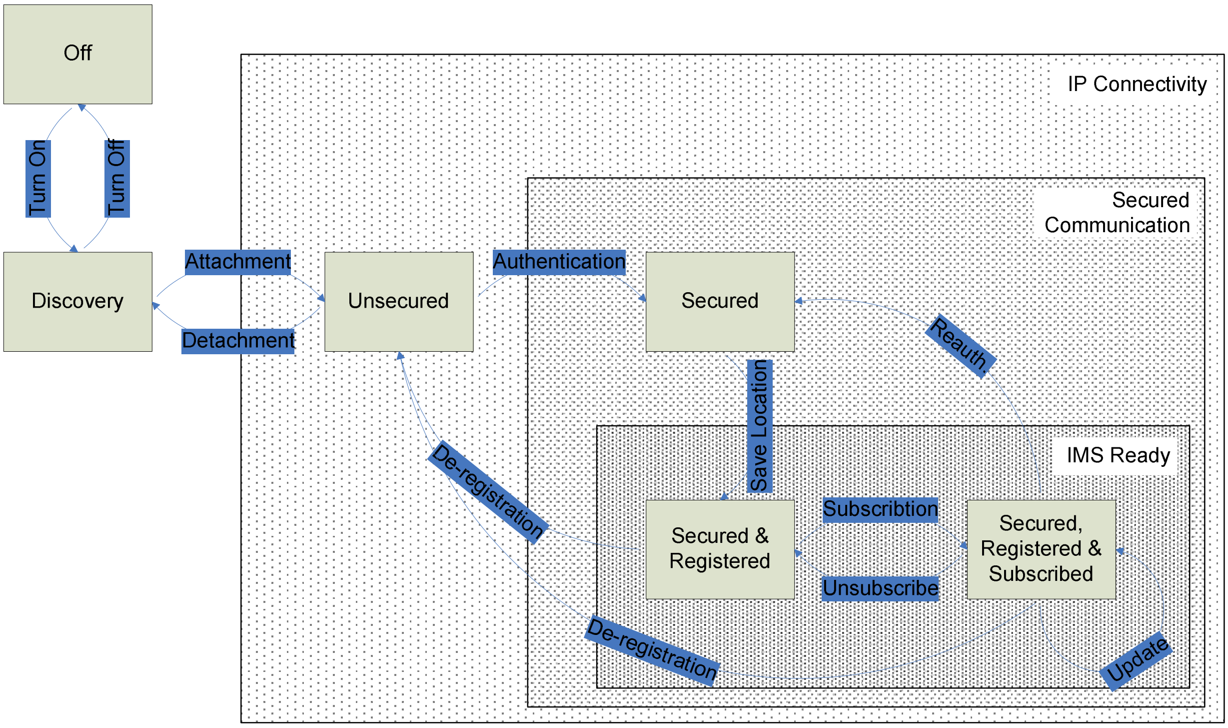


Figure 8: Registration/de-registration state machine

a) Off - In this state the UE does not have any interaction with the environment.

b) Discovery - The UE is acquiring an IP address and finds the address of the outbound Proxy-CSCF.

c) Unsecured - In this state the UE is completely attached to the IP layer and it can fully act at the signaling level. This state is mainly used to send initial registration intentions. No traffic is to be trusted by the network until the UE is authenticated. The UE should not trust incoming signaling until it will attach to a correctly authenticated network.

d) Secured - the UE begun authentication by requesting an authentication challenge. The UE can verify the authenticity of this challenge and it creates a Security Association (SA) with the Proxy-CSCF.

e) Secured and Registered - the UE sends the authentication challenge response to the network, indicating the location information that it wishes to save. While in this state, as communication is secured, updates can be performed without re-authentication.

f) Secured, Registered and Subscribed - to react to network initiated events regarding the registration status, the UE subscribes to its own registration event package. The UE will then receive notifications on changes and it can act accordingly.

For simplification, it is considered that the initial state is unsecured: the UE simulated by the test system already has IP connectivity and the Proxy-CSCF addresses are considered as input configuration for the test system.

5.1.1.1 Definition

The registration/de-registration is the process by which a UE announces, updates or deletes its location information to the home domain"s registrar. This operation is authenticated and a secure communication channel for subsequent signaling is set-up between the UE and the network.

5.1.1.2 Scenarios

While the scenario describe actions on the part of the UE, the portion of the signaling path between the UE and the complete IMS system outside the System Under Test configuration are simulated by the test environment with test system characteristics fixed with stated values.

5.1.1.2.1 Use Case 1- Scenario 1 - Successful initial registration with unprotected REGISTER requests on the SIP default port values as specified in RFC 326

The P-CSCF shall be prepared to receive the unprotected REGISTER requests on the SIP default port values as specified in RFC 3261 [i.18]. The P-CSCF shall also be prepared to receive the unprotected REGISTER requests on the port advertised to the UE during the P-CSCF discovery procedure.

5.1.1.2.2 Use Case 1- Scenario 2 - Successful initial registration with IMS AKA as a security mechanism

The P-CSCF supports the registration with IMS AKA as a security mechanism described in TS 124 229, clause 5.2.2.2 [i.17].

5.1.1.2.3 Use Case 1- Scenario 3 - Successful initial registration with SIP digest without TLS as a security mechanism

The P-CSCF supports the registration with SIP digest without TLS as a security mechanism described in TS 124 229, clause 5.2.2.3 [i.17].

5.1.1.2.4 Use Case 1- Scenario 4 - Successful initial registration with SIP digest with TLS as a security mechanism

The P-CSCF supports the registration with SIP digest with TLS as a security mechanism described in TS 124 229, clause 5.2.2.4 [i.17]

5.1.1.2.5 Use Case 1- Scenario 5 - Successful initial registration with NASS-IMS bundled authentication as a security mechanism

The P-CSCF supports the registration with with NASS-IMS bundled authentication as a security mechanism described in TS 124 229, clause 5.2.2.5 [i.17].

5.1.1.2.6 Use Case 1- Scenario 6 - Successful initial registration with GPRS-IMS-Bundled authentication as a security mechanism

The P-CSCF supports the registration with GPRS-IMS-Bundled authentication as a security mechanism described in TS 124 229, clause 5.2.2.6 [i.17].

5.1.1.2.7 Use Case 1- Scenario 7 - Re-registration - user currently registered

To refresh the registration timer the UE sends a re-registration request before the expiration time expires. This should be sent either 600 seconds before the expiration time if the registration time was greater than 1 200 seconds, or when half the registration time has passed when the registration time was under equal or less than 1 200 seconds. This scenario can also be employed at any time during the registration period when the UE intends to update its capabilities according to RFC 3840 [i.13].

If a secure channel has been set-up and it is used during this procedure, the S-CSCF does not need to authenticate the user and the signalling is presented in figure 9. If the request is not sent through the secure channel then the signalling flow is similar to that of the Initial Registration Scenario.

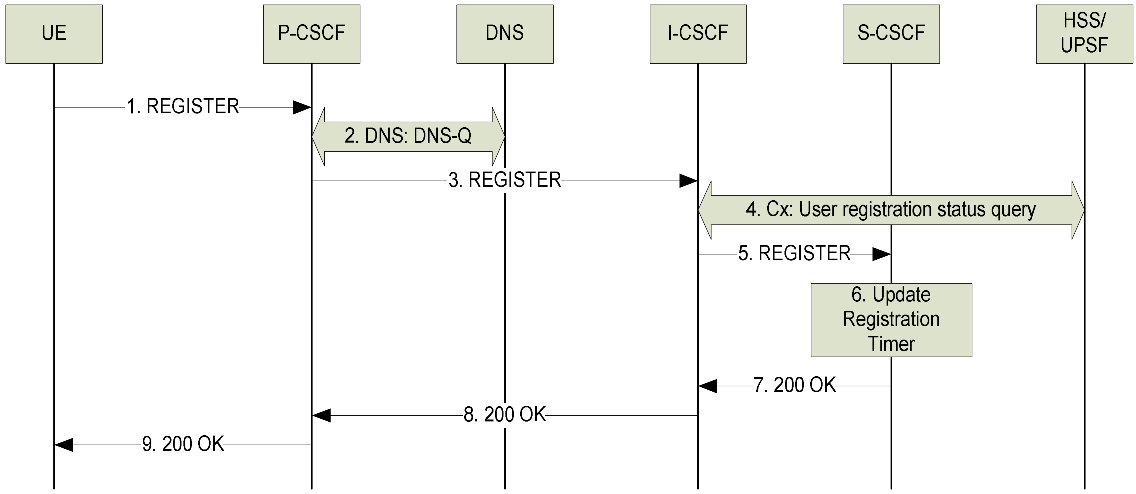


Figure 9: Re-Registration - user currently registered signalling flow

5.1.1.2.8 Use Case 1- Scenario 8 - Re-subscription - user currently registered

As the subscription might have a different expiration timer as the registration, re-subscription is not necessarily linked to re-registration. The process is detailed in figure 10.

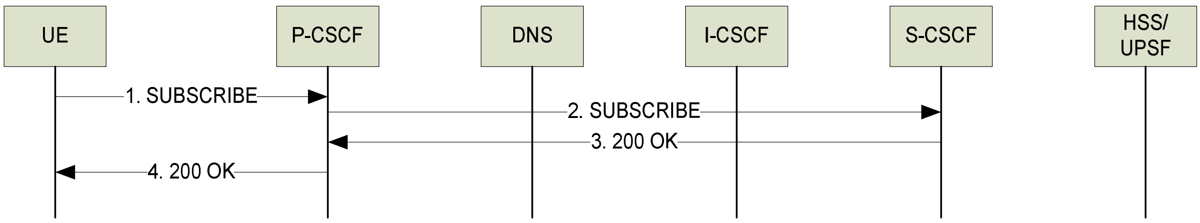


Figure 10: Re-subscription - user currently registered signaling flow

5.1.1.2.9 Use Case 1- Scenario 9 - Re-registration - user roaming

When the UE is roaming to another visited network, the procedures are similar to those of initial registration. As there are no security association set-up with the new P-CSCF, the initial REGISTER request will be authenticated. The network will internally take care of the old registration, without any UE interaction.

5.1.1.2.10 Use Case 1- Scenario 10 - UE initiated de-registration

When the UE requires terminating immediately the registration it can do so by issueing a REGISTER request with an expiration timer set to "0".

6 Session set-up/tear-down scenarios

This use-case corresponds to a normal 2-party call. The "session set-up" part refers to the establishment of the call and the "session tear-down" to its destroyal. Before this scenario is performed, the respective User Endpoints which belong to the particular SUT domain and involved in the communication have to be successfully registered/re-registered. The registration period should not during execution of this use-case (it is acceptable to do a re-registration refreshment during a call scenario).

This version of the document covers several call scenarios that are encountered the most in a real life deployments.

Then several situations for the User Endpoints can been considered, based on the IP-CAN resource reservation status on the two participating sides. For example the originating and/or terminating party might require resource reservation or they could already have the resources preallocated, before the start of the scenario.

In case that the Access Network is not part of the SUT, the IP-CAN reservation steps should be simulated by fixed delays in the Test System and the Test Report shall contain this values.

In all successful call scenarios defined in the next clauses, the signaling flow of the tear-down part is depicted as initiated on the terminating side of the call. When generating traffic, the Test System should also simulate the symmetric case when the originating user initiates the tear-down and the Test System should maintain a 1:1 ratio between the two cases.

During these scenario there are several waiting times during which the Test System should pause, like the ringing time or the call hold time. Distribution of these delays can follow a constant or a Poisson distribution.

6.1 MMTel to MMTel

MMTEL to MMTEL Use case 2				
Scenario 1	Successful call - This scenario represents the case when the call establishment is performed correctly. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.			
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side			
Message flow a) Successful call - resource resevation on both sides				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE (SDP1)	→	
	←	100 Trying		
	←	183 Session Progress		
		PRACK	→	
	←	200 OK PRACK		
		UPDATE	→	
	←	200 OK UPDATE		
	←	180 Ringing		
	←	200 OK		
		ACK	→	
		BYE	→	
	←	200 OK BYE		
		Apply post test routine		
Message flow: b), c)				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE (SDP1)	→	
	←	100 Trying		
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		BYE	→	
	←	200 OK BYE		
		Apply post test routine		

MMTEL to MMTEL Use case 2				
Scenario 2	Successful call - This scenario represents the case when the call establishment is performed correctly . Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the called user.			
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side			
Message flow a) Successful call - resource resevation on both sides				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE (SDP1)	→	
	←	100 Trying		
	←	183 Session Progress		
		PRACK	→	
	←	200 OK PRACK		
		UPDATE	→	
	←	200 OK UPDATE		
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
	←	BYE		
		200 OK BYE	→	
		Apply post test routine		
Message flow: b), c)				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE (SDP1)	→	
	←	100 Trying		
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
	←	BYE		
		200 OK BYE	→	
		Apply post test routine		

MMTEL to MMTEL Use case 2	
Scenario 3	Basic call with Fax with 33,6 kBit/s (Super G3 Fax) This scenario represents the case when in the active call state the Fax transfer on the media is performed correctly and the echo cancellers in the GW are not activated.. The call is released from the calling user. Ensure that in the active call state the data transfer is performed correctly.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8/0 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP 8
Message flow	

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE	→	
	←	100 Trying		
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		Communication		
		BYE	→	
	←	200 OK BYE		

MMTEL to MMTEL Use case 2	
Scenario 4	Basic call with Fax with 14,4 kBit/s; This scenario represents the case when in the active call state the Fax transfer on the media is performed correctly and the echo cancellers in the GW are activated. . The call is released from the calling user. Ensure that in the active call state the data transfer is performed correctly.
Options	a) a resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8/0 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP 8
Message flow	

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE	→	
	←	100 Trying		
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		Communication		
		BYE	→	
	←	200 OK BYE		

MMTEL to MMTEL Use case 2	
Scenario 5	Basic call - Fax with 14,4 kbit/s with V.152 This scenario represents the case when in the active call state the Fax transfer on the media and B-channels is performed correctly and the echo cancellers in the GW are not activated. The call is released from the calling user. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters).
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8 <dynamic-PT> a=rtpmap <dynamic-PT> PCMA/8000 a=gpmd; vbd=yes 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP <dynamic-PT> a=rtpmap <dynamic-PT> PCMA/8000 a=gpmd; vbd=yes
Message flow **SIP (Test System A)**	**Core Network** **SIP (Test System B)** INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → **Apply post test routine**

MMTEL to MMTEL Use case 2	
Scenario 6	Basic call - Fax with 14,4 kbit/s with using the T.38 in an audio m-line codec This scenario represents the case when in the active call state the Fax transfer on the media and B-channels is performed correctly and the echo cancellers in the GW are not activated. The call is released from the called user. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters).
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
SIP Parameter	INVITE: SDP m=image <Port> udptl t38 180/200 OK INVITE: SDP m=image <Port> udptl t38
Message flow **SIP (Test System A)**	**Core Network** **SIP (Test System B)** INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → **Apply post test routine**

MMTEL to MMTEL Use case 2	
Scenario 7	Called user is user busy This scenario represents the case, when the called user is user determined user busy the network initiate call clearing to the calling user.
Options	
Message flow **SIP (Test System A)**	**Core Network** **SIP (Test System B)** INVITE → ← 486 Busy Here ACK →

MMTEL to MMTEL Use case 2			
Scenario 8	CFU Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)		**Core Network**	**SIP (Test System B)**
		INVITE(Call-ID A-B)	→
		CFU is performed	
	←	INVITE(Call-ID B-C)	
		180 Ringing(Call-ID C-B)	→
	←	180 Ringing(Call-ID B-A)	
		200 OK INVITE(Call-ID C-B)	→
	←	ACK(Call-ID B-C)	
	←	200 OK INVITE(Call-ID B-A)	
		ACK(Call-ID A-B)	→
		Communication	
		Apply post test routine	

MMTEL to MMTEL Use case 2			
Scenario 9	CFB Ensure that when user A calls user B which is user determined user busy (UDUB), the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)		**Core Network**	**SIP (Test System B)**
		INVITE(Call-ID A-B)	→
		CFB is performed	
	←	INVITE(Call-ID B-C)	
		180 Ringing(Call-ID C-B)	→
	←	180 Ringing(Call-ID B-A)	
		200 OK INVITE(Call-ID C-B)	→
	←	ACK(Call-ID B-C)	
	←	200 OK INVITE(Call-ID B-A)	
		ACK(Call-ID A-B)	→
		Communication	
		Apply post test routine	

MMTEL to MMTEL Use case 2			
Scenario 10	CFNR Ensure that when user A calls user B which does not answer, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)		**Core Network**	**SIP (Test System B)**
		INVITE(Call-ID A-B)	→
	←	180 Ringing(Call-ID B-A)	
		CFNR is performed	
	←	INVITE(Call-ID B-C)	
		180 Ringing(Call-ID C-B)	→
	←	180 Ringing(Call-ID B-A)	
		200 OK INVITE(Call-ID C-B)	→
	←	ACK(Call-ID B-C)	
	←	200 OK INVITE(Call-ID B-A)	
		ACK(Call-ID A-B)	→
		Communication	
		Apply post test routine	

MMTEL to MMTEL Use case 2			
Scenario 11	CCBS User A is located in network A and user B is located in network B. User A has successfully invoked a CCBS request.Ensure that the recall from user A to user B is successful. The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)		**Core Network**	**SIP (Test System B)**
		A CCBS request was already successful	
	←	NOTIFY	
		200 OK NOTIFY	→
		INVITE	→
	←	180 Ringing	
	←	NOTIFY	
		200 OK NOTIFY	→
	←	200 OK INVITE	
		ACK	→
		Apply post test routine	

MMTEL to MMTEL Use case 2			
Scenario 12	CCNR User A is located in network A and user B is located in network B. User A has successfully invoked a CCNR request. Ensure that the recall from user A to user B is successful. The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)		**Core Network**	**SIP (Test System B)**
		A CCNR request was already successful	
	←	NOTIFY	
		200 OK NOTIFY	→
		INVITE	→
	←	180 Ringing	
	←	NOTIFY	
		200 OK NOTIFY	→
	←	200 OK INVITE	
		ACK	→
		Apply post test routine	

6.2 IMS/PES to IMS/PES

The Use Cases for IMS/PES are described in TS 186 025-2 [i.21].

6.3 MMTel - IMS/PES

6.3.1 ISDN to MMTel Use case 3

<table>
<tr><th colspan="3">ISDN to MMTel Use case 3</th></tr>
<tr><td>Scenario 1</td><td colspan="2">Basic call with BC= ITC_value - enblock sending
This scenario represents the case when the call establishment using en-bloc sending is performed correctly. The call is released from the calling user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).</td></tr>
<tr><td>Options</td><td colspan="2">a) resource reservation is on both sides
b) no resource reservation on terminating side
c) no resource reservation on either side</td></tr>
<tr><td colspan="3">Message flow</td></tr>
<tr><td>SIP (Test System A)</td><td>Core Network
INVITE →
← 100 Trying
← 180 Ringing
← 200 OK INVITE
ACK →
Communication
BYE →
← 200 OK BYE</td><td>SIP (Test System B)</td></tr>
<tr><td>SIP Parameter
PSTN XML BearerCapability element in the INVITE</td><td colspan="2">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>ITC_value<
< BCoctet4
TransferMode>00<
InformationTransferRate>10000<
BCoctet5
Layer1Identification>01<
UserInfoLayer1Protocol>00011<</td></tr>
</table>

ISDN to MMTel Use case 3			
Scenario 2	Basic call with BC= ITC_value - enblock sending This scenario represents the case when the call establishment using en-bloc sending is performed correctly. The call is released from the called user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)		**Core Network**	**SIP (Test System B)**
		INVITE ➔	
	🡐	100 Trying	
	🡐	180 Ringing	
	🡐	200 OK INVITE	
		ACK ➔	
		Communication	
	🡐	BYE	
		200 OK BYE ➔	
SIP Parameter **PSTN XML BearerCapability element in the INVITE**	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>ITC_value< < BCoctet4 TransferMode>00< InformationTransferRate>10000< BCoctet5 Layer1Identification>01< UserInfoLayer1Protocol>00011<		

Table 1: PSTN XML BearerCapability

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

ISDN to MMTel Use case 3	
Scenario 3	Basic call - overlap sending with BC= speech This scenario represents the case when the call establishment using overlap sending. The call is released from the calling user. The call is released from the calling user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side

Message flow Multiple INVITE method is used

SIP (Test System A)		Core Network		SIP (Test System 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)		
		Apply post test routine		

Message flow Overlap sending, the in-Dialogue method is used

SIP (Test System A)		Core Network		SIP (Test System 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		

ISDN to MMTel Use case 3		
Scenario 4	Basic call with BC= 3,1 KHz audio - Fax with 33,6 kbit/s This scenario represents the case when in the active call state (N10) the Fax transfer on the media and B-channels is performed correctly and the echo cancellers in the GW are not activated. The call is released from the calling user. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters).	
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side	
SIP Parameter		INVITE: SDP m=audio <Port> RTP/AVP 8/0 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP 8
Message flow **SIP (Test System A)**	**Core Network** INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → **Apply post test routine**	**SIP (Test System B)**

ISDN to MMTel Use case 3		
Scenario 5	Basic call with BC= 3,1 KHz audio - Fax with 14,4 kbit/s This scenario represents the case when in the active call state (N10) the Fax transfer on the media and B-channels is performed correctly and the echo cancellers in the GW are activated. The call is released from the calling user. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters).	
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side	
SIP Parameter		INVITE: SDP m=audio <Port> RTP/AVP 8/0 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP 8
Message flow **SIP (Test System A)**	**Core Network** INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → **Apply post test routine**	**SIP (Test System B)**

ISDN to MMTel Use case 3	
Scenario 6	Basic call with BC= 3,1 KHz audio - Fax with 14,4 kbit/s with V.152 This scenario represents the case when in the active call state (N10) the Fax transfer on the media and B-channels is performed correctly and the echo cancellers in the GW are not activated. The call is released from the calling user. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters).
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8 <dynamic-PT> a=rtpmap <dynamic-PT> PCMA/8000 a=gpmd; vbd=yes 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP <dynamic-PT> a=rtpmap <dynamic-PT> PCMA/8000 a=gpmd; vbd=yes
Message flow	

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE (SDP1)	→	
	←	180 Ringing		
	←	200 OK INVITE (SDP2)		
		ACK	→	
		Apply post test routine		

ISDN to MMTel Use case 3	
Scenario 7	Basic call with BC= 3,1 KHz audio - Fax with 14,4 kbit/s with using the T.38 in an audio m-line codec This scenario represents the case when in the active call state (N10) the Fax transfer on the media and B-channels is performed correctly and the echo cancellers in the GW are not activated. The call is released from the calling user. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters).
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
SIP Parameter	INVITE: SDP m=image <Port> udptl t38 180/200 OK INVITE: SDP m=image <Port> udptl t38
Message flow	

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE (SDP1)	→	
	←	180 Ringing		
	←	200 OK INVITE (SDP2)		
		ACK	→	
		Apply post test routine		

ISDN to MMTel Use case 3	
Scenario 8	Basic call with BC= 3,1 kHz - Modem V.32 bis (4,8 kbit/s, 9,6 kbit/s 14,4 kbit/s) This scenario represents the case when in the active call state (N10) the 3,1 kHz transfer is performed correctly The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8/0 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP 8
Message flow **SIP (Test System A)**	**Core Network** **SIP (Test System B)** INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → **Apply post test routine**

ISDN to MMTel Use case 3	
Scenario 9	*Basic call with BC= 3,1 kHz - Modem V.34 (up to 33,6 kbit/s)* *This scenario represents the case when in the active call state (N10) the 3,1 kHz transfer is performed correctly.* The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8/0 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP 8
Message flow **SIP (Test System A)**	**Core Network** **SIP (Test System B)** INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → **Apply post test routine**

ISDN to MMTel Use case 3	
Scenario 10	Called user is user determined user busy This scenario represents the case, when the called user is user determined user busy. the network initiates call clearing to the calling user.
Options	
Message flow **SIP (Test System A)**	**Core Network** **SIP (Test System B)** INVITE → ← 486 Busy Here ACK →

ISDN to MMTel Use case 3	
Scenario 11	CFU Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.
Options	resource reservation is on both sides no resource reservation on terminating side no resource reservation on either side

Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE(Call-ID A-B)	→	
	CFU is performed		
	← INVITE(Call-ID B-C)		
	180 Ringing(Call-ID C-B)	→	
	← 180 Ringing(Call-ID B-A)		
	200 OK INVITE(Call-ID C-B)	→	
	← ACK(Call-ID B-C)		
	← 200 OK INVITE(Call-ID B-A)		
	ACK(Call-ID A-B)	→	
	Communication		
Apply post test routine			

ISDN to MMTel Use case 3	
Scenario 12	CFB Ensure that when user A calls user B which is user determined user busy (UDUB), the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.
Options	resource reservation is on both sides no resource reservation on terminating side no resource reservation on either side

Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE(Call-ID A-B)	→	
	CFB is performed		
	← INVITE(Call-ID B-C)		
	180 Ringing(Call-ID C-B)	→	
	← 180 Ringing(Call-ID B-A)		
	200 OK INVITE(Call-ID C-B)	→	
	← ACK(Call-ID B-C)		
	← 200 OK INVITE(Call-ID B-A)		
	ACK(Call-ID A-B)	→	
	Communication		
Apply post test routine			

ISDN to MMTel Use case 3	
Scenario 13	CFNR Ensure that when user A calls user B which does not answer, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.
Options	resource reservation is on both sides no resource reservation on terminating side no resource reservation on either side

Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE(Call-ID A-B)	→	
	← 180 Ringing(Call-ID B-A)		
	CFNR is performed		
	← INVITE(Call-ID B-C)		
	180 Ringing(Call-ID C-B)	→	
	← 180 Ringing(Call-ID B-A)		
	200 OK INVITE(Call-ID C-B)	→	
	← ACK(Call-ID B-C)		
	← 200 OK INVITE(Call-ID B-A)		
	ACK(Call-ID A-B)	→	
	Communication		
Apply post test routine			

ISDN to MMTel Use case 3			
Scenario 14	CCBS User A is located in network A and user B is located in network B. User A has successfully invoked a CCBS request. Ensure that the recall from user A to user B is successful. The call is released from the calling user.		
Options	resource reservation is on both sides no resource reservation on terminating side no resource reservation on either side		
Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
A CCBS request was already successful			
	←	NOTIFY	
		200 OK NOTIFY	→
		INVITE	→
	←	180 Ringing	
	←	NOTIFY	
		200 OK NOTIFY	→
	←	200 OK INVITE	
		ACK	→
Apply post test routine			

ISDN to MMTel Use case 3			
Scenario 15	CCNR User A is located in network A and user B is located in network B. User A has successfully invoked a CCNR request. Ensure that the recall from user A to user B is successful. The call is released from the calling user.		
Options	resource reservation is on both sides no resource reservation on terminating side no resource reservation on either side		
Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
A CCNR request was already successful			
	←	NOTIFY	
		200 OK NOTIFY	→
		INVITE	→
	←	180 Ringing	
	←	NOTIFY	
		200 OK NOTIFY	→
	←	200 OK INVITE	
		ACK	→
Apply post test routine			

6.3.2 MMTel to ISDN Use case 4

MMTel to ISDN Use case 4			
Scenario 1	Basic call. The call is released from the calling user This scenario represents the case when the call establishment is performed correctly. The call is released from the calling user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow **SIP (Test System A)**		**Core Network**	**SIP (Test System B)**
		INVITE →	
	←	100 Trying	
	←	180 Ringing	
	←	200 OK INVITE	
		ACK →	
		Communication	
		BYE →	
	←	200 OK BYE	

MMTel to ISDN Use case 4			
Scenario 2	Basic call The call is released from the called user This scenario represents the case when the call establishment is performed correctly. The call is released from the called user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow **SIP (Test System A)**		**Core Network**	**SIP (Test System B)**
		INVITE →	
	←	100 Trying	
	←	180 Ringing	
	←	200 OK INVITE	
		ACK →	
		Communication	
		BYE ←	
	→	200 OK BYE	

MMTel to ISDN Use case 4			
Scenario 3	Basic call with BC= 3,1 KHz audio - Fax with 33,6 kbit/s This scenario represents the case when in the active call state (N10) the Fax transfer on the media and B-channels is performed correctly and the echo cancellers in the GW are not activated. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8/0 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP 8		
Message flow **SIP (Test System A)**		**Core Network**	**SIP (Test System B)**
		INVITE (SDP1) →	
	←	180 Ringing	
	←	200 OK INVITE (SDP2)	
		ACK →	
		Apply post test routine	

<table>
<tr><th colspan="3">MMTel to ISDN Use case 4</th></tr>
<tr><td>Scenario 4</td><td colspan="2">Basic call with BC= 3,1 KHz audio - Fax with 14,4 kbit/s
This scenario represents the case when in the active call state (N10) the Fax transfer on the media and B-channels is performed correctly and the echo cancellers in the GW are activated. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user</td></tr>
<tr><td>Options</td><td colspan="2">a) resource reservation is on both sides
b) no resource reservation on terminating side
c) no resource reservation on either side</td></tr>
<tr><td colspan="2">SIP Parameter</td><td>INVITE: SDP
m=audio <Port> RTP/AVP 8/0
180/200 OK INVITE: SDP
m=audio <Port> RTP/AVP 8</td></tr>
<tr><td colspan="3">Message flow
SIP (Test System A) Core Network SIP (Test System B)
INVITE (SDP1) →
← 180 Ringing
← 200 OK INVITE (SDP2)
ACK →
Apply post test routine</td></tr>
</table>

<table>
<tr><th colspan="3">MMTel to ISDN Use case 4</th></tr>
<tr><td>Scenario 5</td><td colspan="2">Basic call with BC= 3,1 KHz audio - Fax with 14,4 kbit/s with V.152
This scenario represents the case when in the active call state (N10) the Fax transfer is performed correctly and the echo cancellers in the GW are activated. The call is released from the calling user. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters).</td></tr>
<tr><td>Options</td><td colspan="2">a) resource reservation is on both sides
b) no resource reservation on terminating side
c) no resource reservation on either side</td></tr>
<tr><td colspan="2">SIP Parameter</td><td>INVITE: SDP
m=audio <Port> RTP/AVP 8 <dynamic-PT>
a=rtpmap <dynamic-PT> PCMA/8000
a=gpmd; vbd=yes

180/200 OK INVITE: SDP
m=audio <Port> RTP/AVP <dynamic-PT>
a=rtpmap <dynamic-PT> PCMA/8000
a=gpmd; vbd=yes</td></tr>
<tr><td colspan="3">Message flow
SIP (Test System A) Core Network SIP (Test System B)
INVITE (SDP1) →
← 180 Ringing
← 200 OK INVITE (SDP2)
ACK →
Apply post test routine</td></tr>
</table>

MMTel to ISDN Use case 4	
Scenario 6	Basic call with BC= 3,1 KHz audio - Fax with 14,4 kbit/s with using the T.38 in an audio m-line codec This scenario represents the case when in the active call state (N10) the Fax transfer on the media and B-channels is performed correctly and the echo cancellers in the GW are activated. The call is released from the calling user. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters).
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
SIP Parameter	INVITE: SDP m=image <Port> udptl t38 180/200 OK INVITE: SDP m=image <Port> udptl t38
Message flow **SIP (Test System A)**	**Core Network** **SIP (Test System B)** INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → **Apply post test routine**

MMTel to ISDN Use case 4	
Scenario 7	Basic call with BC= 3,1 kHz - Modem V.32 bis (4,8 kbit/s, 9,6 kbit/s, 14,4 kbit/s) This scenario represents the case when in the active call state (N10) the 3,1 kHz transfer is performed correctly The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8/0 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP 8
Message flow **SIP (Test System A)**	**Core Network** **SIP (Test System B)** INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → **Apply post test routine**

MMTel to ISDN Use case 4	
Scenario 8	Basic call with BC= 3,1 kHz - Modem V.34 (up to 33,6 kbit/s) This scenario represents the case when in the active call state (N10) the 3,1 kHz transfer is performed correctly. The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8/0 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP 8
Message flow **SIP (Test System A)**	**Core Network** **SIP (Test System B)** INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → **Apply post test routine**

MMTel to ISDN Use case 4				
Scenario 9	Called user is user determined user busy This scenario represents the case, when the called user is user determined user busy the network initiate call clearing to the calling user.			
Options				
Message flow				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE	→	
	←	486 Busy Here		
		ACK	→	

MMTel to ISDN Use case 4				
Scenario 10	CFU Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.			
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side			
Message flow				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE(Call-ID A-B)	→	
		CFU is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		

MMTel to ISDN Use case 4				
Scenario 11	CFB Ensure that when user A calls user B which is user determined user busy (UDUB), the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.			
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side			
Message flow				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE(Call-ID A-B)	→	
		CFB is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		

MMTel to ISDN Use case 4			
Scenario 12	CFNR Ensure that when user A calls user B which does not answer, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)		**Core Network**	**SIP (Test System B)**
		INVITE(Call-ID A-B)	→
	←	180 Ringing(Call-ID B-A)	
		CFNR is performed	
	←	INVITE(Call-ID B-C)	
		180 Ringing(Call-ID C-B)	→
	←	180 Ringing(Call-ID B-A)	
		200 OK INVITE(Call-ID C-B)	→
	←	ACK(Call-ID B-C)	
	←	200 OK INVITE(Call-ID B-A)	
		ACK(Call-ID A-B)	→
		Communication	
		Apply post test routine	

MMTel to ISDN Use case 4			
Scenario 13	CCBS User A is located in network A and user B is located in network B. User A has successfully invoked a CCBS request. Ensure that the recall from user A to user B is successful. The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)		**Core Network**	**SIP (Test System B)**
		A CCBS or CCNR request was already successful	
	←	NOTIFY	
		200 OK NOTIFY	→
		INVITE	→
	←	180 Ringing	
	←	NOTIFY	
		200 OK NOTIFY	→
	←	200 OK INVITE	
		ACK	→
		Apply post test routine	

MMTel to ISDN Use case 4	
Scenario 14	CCNR User A is located in network A and user B is located in network B. User A has successfully invoked a CCNR request. Ensure that the recall from user A to user B is successful. The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
Message flow	

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		A CCBS or CCNR request was already successful		
	←	NOTIFY		
		200 OK NOTIFY	→	
		INVITE	→	
	←	180 Ringing		
	←	NOTIFY		
		200 OK NOTIFY	→	
	←	200 OK INVITE		
		ACK	→	
		Apply post test routine		

6.3.3 MMTel to PSTN Use case 5

MMTel to PSTN Use case 5	
Scenario 1	Basic call. The call is released from the called user. This scenario represents the case when the call establishment is performed correctly. The call is released from the called user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
Message flow	

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE	→	
	←	100 Trying		
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		Communication		
	←	**BYE**		
		200 OK BYE	→	

MMTel to PSTN Use case 5			
Scenario 2	Basic call. The call is released from the calling user This scenario represents the case when the call establishment is performed correctly. The call is released from the calling user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE	→	
←	100 Trying		
←	180 Ringing		
←	200 OK INVITE		
	ACK	→	
	Communication		
	BYE	→	
←	200 OK BYE		

MMTel to PSTN Use case 5			
Scenario 3	Basic call with Fax with 33,6 kBit/s (Super G3 Fax) This scenario represents the case when in the active call state (N10) the Fax transfer on the media and B-channels is performed correctly and the echo cancellers in the GW are deactivated. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8/0 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP 8		
Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE (SDP1)	→	
←	180 Ringing		
←	200 OK INVITE (SDP2)		
	ACK	→	
	Apply post test routine		

MMTel to PSTN Use case 5			
Scenario 4	Basic call with Fax with 14,4 kBit/s This scenario represents the case when in the active call state (N10) the Fax transfer on the media and B-channels is performed correctly. The echo cancellers in the GW are activated. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8/0 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP 8		
Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE (SDP1)	→	
←	180 Ringing		
←	200 OK INVITE (SDP2)		
	ACK	→	
	Apply post test routine		

MMTel to PSTN Use case 5			
Scenario 5	Basic call - Fax with 14,4 kbit/s with V.152 This scenario represents the case when in the active call state (N10) the Fax transfer on the media and B-channels is performed correctly and the echo cancellers in the GW are not activated. The call is released from the called user. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters).		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8 <dynamic-PT> a=rtpmap <dynamic-PT> PCMA/8000 a=gpmd; vbd=yes 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP <dynamic-PT> a=rtpmap <dynamic-PT> PCMA/8000 a=gpmd; vbd=yes		
Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE (SDP1)	→	
←	180 Ringing		
←	200 OK INVITE (SDP2)		
	ACK	→	
	Apply post test routine		

MMTel to PSTN Use case 5			
Scenario 6	Basic call with BC= 3,1 kHz - Modem V.32 bis (4,8 kbit/s, 9,6 kbit/s 14,4 kbit/s) This scenario represents the case when in the active call state (N10) the 3,1 kHz transfer is performed correctly The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8/0 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP 8		
Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE (SDP1)	→	
←	180 Ringing		
←	200 OK INVITE (SDP2)		
	ACK	→	
	Apply post test routine		

MMTel to PSTN Use case 5			
Scenario 7	Basic call with BC= 3,1 kHz - Modem V.34 (up to 33,6 kbit/s) This scenario represents the case when in the active call state (N10) the 3,1 kHz transfer is performed correctly The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8/0 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP 8		
Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE (SDP1)	→	
←	180 Ringing		
←	200 OK INVITE (SDP2)		
	ACK	→	
	Apply post test routine		

MMTel to PSTN Use case 5				
Scenario 8	Basic call - Fax with 14,4 kbit/s with using the T.38 in an audio m-line codec This scenario represents the case when in the active call state (N10) the Fax transfer is performed correctly and the echo cancellers in the GW are not activated. The call is released from the calling user. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters).			
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side			
SIP Parameter	INVITE: SDP m=image <Port> udptl t38 180/200 OK INVITE: SDP m=image <Port> udptl t38			
Message flow				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE (SDP1)	→	
	←	180 Ringing		
	←	200 OK INVITE (SDP2)		
		ACK	→	
		Apply post test routine		

MMTel to PSTN Use case 5				
Scenario 9	Called user is user busy This scenario represents the case, when the called user is user determined user busy the network initiate call clearing to the calling user.			
Options				
Message flow				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE	→	
	←	486 Busy Here		
		ACK	→	

MMTel to PSTN Use case 5				
Scenario 10	CFU Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.			
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side			
Message flow				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE(Call-ID A-B)	→	
		CFU is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		

MMTel to PSTN Use case 5	
Scenario 11	CFB Ensure that when user A calls user B which is user determined user busy (UDUB), the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side

Message flow

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE(Call-ID A-B)	→	
		CFB is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		

MMTel to PSTN Use case 4	
Scenario 12	CFNR Ensure that when user A calls user B which does not answer, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side

Message flow

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE(Call-ID A-B)	→	
	←	180 Ringing(Call-ID B-A)		
		CFNR is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		

<table>
<tr><th colspan="4">MMTel to PSTN Use case 4</th></tr>
<tr><td>Scenario 13</td><td colspan="3">CCBS
User A is located in network A and user B is located in network B. User A has successfully invoked a CCBS request.
Ensure when the user B becomes available for CC recall, the CC recall procedure is started. Ensure that the recall from user A to user B is successful. The call is released from the calling user.</td></tr>
<tr><td>Options</td><td colspan="3">a) resource reservation is on both sides
b) no resource reservation on terminating side
c) no resource reservation on either side</td></tr>
<tr><td colspan="4">Message flow</td></tr>
<tr><td>SIP (Test System A)</td><td></td><td>Core Network</td><td>SIP (Test System B)</td></tr>
<tr><td></td><td></td><td>A CCBS request was already successful</td><td></td></tr>
<tr><td></td><td>←</td><td>NOTIFY</td><td></td></tr>
<tr><td></td><td></td><td>200 OK NOTIFY</td><td>→</td></tr>
<tr><td></td><td></td><td>INVITE</td><td>→</td></tr>
<tr><td></td><td>←</td><td>180 Ringing</td><td></td></tr>
<tr><td></td><td>←</td><td>NOTIFY</td><td></td></tr>
<tr><td></td><td></td><td>200 OK NOTIFY</td><td>→</td></tr>
<tr><td></td><td>←</td><td>200 OK INVITE</td><td></td></tr>
<tr><td></td><td></td><td>ACK</td><td>→</td></tr>
<tr><td colspan="4">Apply post test routine</td></tr>
</table>

<table>
<tr><th colspan="4">MMTel to PSTN Use case 4</th></tr>
<tr><td>Scenario 14</td><td colspan="3">CCNR
User A is located in network A and user B is located in network B. User A has successfully invoked a CCNR request.
Ensure when the user B becomes available for CC recall, the CC recall procedure is started. Ensure that the recall from user A to user B is successful. The call is released from the calling user.</td></tr>
<tr><td>Options</td><td colspan="3">a) resource reservation is on both sides
b) no resource reservation on terminating side
c) no resource reservation on either side</td></tr>
<tr><td colspan="4">Message flow</td></tr>
<tr><td>SIP (Test System A)</td><td></td><td>Core Network</td><td>SIP (Test System B)</td></tr>
<tr><td></td><td></td><td>A CCNR request was already successful</td><td></td></tr>
<tr><td></td><td>←</td><td>NOTIFY</td><td></td></tr>
<tr><td></td><td></td><td>200 OK NOTIFY</td><td>→</td></tr>
<tr><td></td><td></td><td>INVITE</td><td>→</td></tr>
<tr><td></td><td>←</td><td>180 Ringing</td><td></td></tr>
<tr><td></td><td>←</td><td>NOTIFY</td><td></td></tr>
<tr><td></td><td></td><td>200 OK NOTIFY</td><td>→</td></tr>
<tr><td></td><td>←</td><td>200 OK INVITE</td><td></td></tr>
<tr><td></td><td></td><td>ACK</td><td>→</td></tr>
<tr><td colspan="4">Apply post test routine</td></tr>
</table>

6.3.4 PSTN to MMTEL Use case 6

PSTN to MMTel Use case 6				
Scenario 1	Basic call. The call is released from the calling user This scenario represents the case when the call establishment is performed correctly. The call is released from the calling user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).			
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side			
Message flow **SIP (Test System A)**		**Core Network**		**SIP (Test System B)**
		INVITE	→	
	←	100 Trying		
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		Communication		
		BYE	→	
	←	200 OK BYE		

PSTN to MMTel - Use case 6				
Scenario 2	Basic call The call is released from the called user. This scenario represents the case when the call establishment is performed correctly. The call is released from the called user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).			
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side			
Message flow **SIP (Test System A)**		**Core Network**		**SIP (Test System B)**
		INVITE	→	
	←	100 Trying		
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		Communication		
	←	**BYE**		
		200 OK BYE	→	

PSTN to MMTel Use case 6	
Scenario 3	Basic call - overlap sending This scenario represents the case when the call establishment using overlap sending. The call is released from the calling user. The call is released from the calling user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side

Message flow Multiple INVITE method is used

SIP (Test System A)		Core Network		SIP (Test System 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)		
		Apply post test routine		

Message flow Overlap sending, the in-Dialogue method is used

SIP (Test System A)		Core Network		SIP (Test System 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		

PSTN to MMTel Use case 6		
Scenario 4	Basic call with Fax with 33,6 kBit/s (Super G3 Fax) This scenario represents the case when in the active call state (N10) the Fax transfer on the media and B-channels is performed correctly and the echo cancellers in the GW are deactivated. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.	
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side	
SIP Parameter		INVITE: SDP m=audio <Port> RTP/AVP 8/0 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP 8
Message flow **SIP (Test System A)**	**Core Network** INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → **Apply post test routine**	**SIP (Test System B)**

PSTN to MMTel Use case 6		
Scenario 5	Basic call with Fax with 14,4 kBit/s This scenario represents the case when in the active call state (N10) the Fax transfer is performed correctly. The echo cancellers in the GW are activated. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.	
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side	
SIP Parameter		INVITE: SDP m=audio <Port> RTP/AVP 8/0 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP 8
Message flow **SIP (Test System A)**	**Core Network** INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → **Apply post test routine**	**SIP (Test System B)**

PSTN to MMTel Use case 6		
Scenario 6	Basic call - Fax with 14,4 kbit/s with V.152 codec This scenario represents the case when in the active call state (N10) the Fax transfer on the media and B-channels is performed correctly and the echo cancellers in the GW are not activated. The call is released from the calling user. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters).	
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side	
SIP Parameter		INVITE: SDP m=audio <Port> RTP/AVP 8 <dynamic-PT> a=rtpmap <dynamic-PT> PCMA/8000 a=gpmd; vbd=yes 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP <dynamic-PT> a=rtpmap <dynamic-PT> PCMA/8000 a=gpmd; vbd=yes
Message flow **SIP (Test System A)**	**Core Network** INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → **Apply post test routine**	**SIP (Test System B)**

PSTN to MMTel Use case 6		
Scenario 7	Basic call - Fax with 14,4 kbit/s with using the T.38 in an audio m-line codec This scenario represents the case when in the active call state the Fax transfer is performed correctly The call is released from the calling user. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters).	
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side	
SIP Parameter		INVITE: SDP m=image <Port> udptl t38 180/200 OK INVITE: SDP m=image <Port> udptl t38
Message flow **SIP (Test System A)**	**Core Network** INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → **Apply post test routine**	**SIP (Test System B)**

PSTN to MMTel Use case 6	
Scenario 8	Basic call with BC= 3,1 kHz - Modem V.32 bis (4,8 kbit/s, 9,6 kbit/s 14,4 kbit/s) This scenario represents the case when in the active call state (N10) the 3,1 kHz transfer is performed correctly The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8/0 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP 8
Message flow	
SIP (Test System A)	**Core Network** — **SIP (Test System B)** INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → **Apply post test routine**

PSTN to MMTel Use case 6	
Scenario 9	*Basic call with BC= 3,1 kHz - Modem V.34 (up to 33,6 kbit/s)* *This scenario represents the case when in the active call state (N10) the 3,1 kHz transfer is performed correctly.* The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8/0 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP 8
Message flow	
SIP (Test System A)	**Core Network** — **SIP (Test System B)** INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → **Apply post test routine**

PSTN to MMTel - Use case 6	
Scenario 10	Called user is user busy This scenario represents the case, when the called user is user determined user busy the network initiate call clearing to the calling user.
Options	
Message flow	
SIP (Test System A)	**Core Network** — **SIP (Test System B)** INVITE → ← 486 Busy Here ACK →

PSTN to MMTel Use case 6	
Scenario 11	CFU Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.
Options	resource reservation is on both sides no resource reservation on terminating side no resource reservation on either side

Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE(Call-ID A-B)	→	
	CFU is performed		
	← INVITE(Call-ID B-C)		
	180 Ringing(Call-ID C-B)	→	
	← 180 Ringing(Call-ID B-A)		
	200 OK INVITE(Call-ID C-B)	→	
	← ACK(Call-ID B-C)		
	← 200 OK INVITE(Call-ID B-A)		
	ACK(Call-ID A-B)	→	
	Communication		
	Apply post test routine		

PSTN to MMTel Use case 6	
Scenario 12	CFB Ensure that when user A calls user B which is user determined user busy (UDUB), the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.
Options	resource reservation is on both sides no resource reservation on terminating side no resource reservation on either side

Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE(Call-ID A-B)	→	
	CFB is performed		
	← INVITE(Call-ID B-C)		
	180 Ringing(Call-ID C-B)	→	
	← 180 Ringing(Call-ID B-A)		
	200 OK INVITE(Call-ID C-B)	→	
	← ACK(Call-ID B-C)		
	← 200 OK INVITE(Call-ID B-A)		
	ACK(Call-ID A-B)	→	
	Communication		
	Apply post test routine		

PSTN to MMTel Use case 6	
Scenario 13	CFNR Ensure that when user A calls user B which does not answer, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.
Options	resource reservation is on both sides no resource reservation on terminating side no resource reservation on either side
Message flow **SIP (Test System A)** **Core Network** **SIP (Test System B)** INVITE(Call-ID A-B) → ← 180 Ringing(Call-ID B-A) **CFNR is performed** ← INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) 200 OK INVITE(Call-ID C-B) → ← ACK(Call-ID B-C) ← 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → **Communication** **Apply post test routine**	

PSTN to MMTel Use case 6	
Scenario 14	CCBS User A is located in network A and user B is located in network B. User A has successfully invoked a CCBS request. Ensure when the user B becomes available for CC recall, the CC recall procedure is started. Ensure that the recall from user A to user B is successful. The call is released from the calling user.
Options	resource reservation is on both sides no resource reservation on terminating side no resource reservation on either side
Message flow **SIP (Test System A)** **Core Network** **SIP (Test System B)** **A CCBS request was already successful** ← NOTIFY 200 OK NOTIFY → INVITE → ← 180 Ringing ← NOTIFY 200 OK NOTIFY → ← 200 OK INVITE ACK → **Apply post test routine**	

PSTN to MMTel Use case 6	
Scenario 15	CCNR User A is located in network A and user B is located in network B. User A has successfully invoked a CCNR request.Ensure that the recall from user A to user B is successful. The call is released from the calling user.
Options	resource reservation is on both sides no resource reservation on terminating side no resource reservation on either side
Message flow **SIP (Test System A)** **Core Network** **SIP (Test System B)** **A CCNR request was already successful** ← NOTIFY 200 OK NOTIFY → INVITE → ← 180 Ringing ← NOTIFY 200 OK NOTIFY → ← 200 OK INVITE ACK → **Apply post test routine**	

6.4 VoLTE to IMS/PES

6.4.1 ISDN to VoLTE Use case 7

ISDN to VoLTE Use Case 7	
Scenario 1	Basic call with BC= ITC_value - enblock sending This scenario represents the case when the call establishment using en-bloc sending is performed correctly. The call is released from the calling user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
Message flow **SIP (Test System A)** **Core Network** **SIP (Test System B)** INVITE → ← 100 Trying ← 180 Ringing ← 200 OK INVITE ACK → Communication BYE → ← 200 OK BYE	
SIP Parameter **PSTN XML BearerCapability element in the INVITE**	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>ITC_value< < BCoctet4 TransferMode>00< InformationTransferRate>10000< BCoctet5 Layer1Identification>01< UserInfoLayer1Protocol>00011<

<table>
<tr><th colspan="3">ISDN to VoLTE Use Case 7</th></tr>
<tr><td>Scenario 2</td><td colspan="2">Basic call with BC= ITC_value - enblock sending
This scenario represents the case when the call establishment using en-bloc sending is performed correctly. The call is released from the called user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).</td></tr>
<tr><td>Options</td><td colspan="2">a) resource reservation is on both sides
b) no resource reservation on terminating side
c) no resource reservation on either side</td></tr>
<tr><td colspan="3">Message flow
SIP (Test System A) | Core Network | SIP (Test System B)
INVITE →
← 100 Trying
← 180 Ringing
← 200 OK INVITE
ACK →
Communication
← BYE
200 OK BYE →</td></tr>
<tr><td colspan="2">SIP Parameter
PSTN XML BearerCapability element in the INVITE</td><td>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>ITC_value<
< BCoctet4
TransferMode>00<
InformationTransferRate>10000<
BCoctet5
Layer1Identification>01<
UserInfoLayer1Protocol>00011<</td></tr>
</table>

Table 2: PSTN XML BearerCapability

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

ISDN to VoLTE Use Case 7	
Scenario 3	Basic call - overlap sending with BC= speech This scenario represents the case when the call establishment using overlap sending. The call is released from the calling user. The call is released from the calling user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side

Message flow Multiple INVITE method is used

SIP (Test System A)		Core Network		SIP (Test System 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)		
		Apply post test routine		

Message flow Overlap sending, the in-Dialogue method is used

SIP (Test System A)		Core Network		SIP (Test System 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		

ISDN to VoLTE Use Case 7	
Scenario 4	Called user is user determined user busy This scenario represents the case, when the called user is user determined user busy. the network initiates call clearing to the calling user .
Options	

Message flow

SIP (Test System A)		Core Network		SIP (Test System B)
		INVITE	→	
	←	486 Busy Here		
		ACK	→	

ISDN to VoLTE Use case 7	
Scenario 11	CFU Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side

Message flow

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE(Call-ID A-B)	→	
		CFU is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		

ISDN to VoLTE Use Case 7	
Scenario 5	CFB Ensure that when user A calls user B which is user determined user busy (UDUB), the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side

Message flow

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE(Call-ID A-B)	→	
		CFB is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		

ISDN to VoLTE Use Case 7	
Scenario 6	CFNR Ensure that when user A calls user B which does not answer, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side

Message flow

SIP (Test System A)		Core Network		SIP (Test System B)
		INVITE(Call-ID A-B)	→	
	←	180 Ringing(Call-ID B-A)		
		CFNR is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		

ISDN to VoLTE Use Case 7	
Scenario 7	CCBS User A is located in network A and user B is located in network B. User A has successfully invoked a CCBS request. Ensure that the recall from user A to user B is successful. The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side

Message flow

SIP (Test System A)		Core Network		SIP (Test System B)
		A CCBS request was already successful		
	←	NOTIFY		
		200 OK NOTIFY	→	
		INVITE	→	
	←	180 Ringing		
	←	NOTIFY		
		200 OK NOTIFY	→	
	←	200 OK INVITE		
		ACK	→	
		Apply post test routine		

ISDN to VoLTE Use case 7	
Scenario 8	CCNR User A is located in network A and user B is located in network B. User A has successfully invoked a CCNR request. Ensure that the recall from user A to user B is successful. The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side

Message flow

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		A CCNR request was already successful		
	←	NOTIFY		
		200 OK NOTIFY	→	
		INVITE	→	
	←	180 Ringing		
	←	NOTIFY		
		200 OK NOTIFY	→	
	←	200 OK INVITE		
		ACK	→	
		Apply post test routine		

6.4.2 VoLTE to ISDN Use case 8

VoLTE to ISDN Use Case 8	
Scenario 1	Basic call. The call is released from the calling user This scenario represents the case when the call establishment is performed correctly. The call is released from the calling user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side

Message flow

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE	→	
	←	100 Trying		
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		Communication		
		BYE	→	
	←	200 OK BYE		

VoLTE to ISDN Use Case 8	
Scenario 2	Basic call The call is released from the called user This scenario represents the case when the call establishment is performed correctly. The call is released from the called user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side

Message flow

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE	→	
	←	100 Trying		
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		Communication		
		BYE	←	
	→	200 OK BYE		

VoLTE to ISDN Use Case 8	
Scenario 3	Called user is user determined user busy This scenario represents the case, when the called user is user determined user busy the network initiate call clearing to the calling user.
Options	

Message flow

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE	→	
	←	486 Busy Here		
		ACK	→	

VoLTE to ISDN Use Case 8	
Scenario 4	CFU Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side

Message flow

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE(Call-ID A-B)	→	
		CFU is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		

VoLTE to ISDN Use Case 8			
Scenario 5	CFB Ensure that when user A calls user B which is user determined user busy (UDUB), the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)		**Core Network**	**SIP (Test System B)**
		INVITE(Call-ID A-B)	→
		CFB is performed	
	←	INVITE(Call-ID B-C)	
		180 Ringing(Call-ID C-B)	→
	←	180 Ringing(Call-ID B-A)	
		200 OK INVITE(Call-ID C-B)	→
	←	ACK(Call-ID B-C)	
	←	200 OK INVITE(Call-ID B-A)	
		ACK(Call-ID A-B)	→
		Communication	
		Apply post test routine	

VoLTE to ISDN Use Case 8			
Scenario 6	CFNR Ensure that when user A calls user B which does not answer, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)		**Core Network**	**SIP (Test System B)**
		INVITE(Call-ID A-B)	→
	←	180 Ringing(Call-ID B-A)	
		CFNR is performed	
	←	INVITE(Call-ID B-C)	
		180 Ringing(Call-ID C-B)	→
	←	180 Ringing(Call-ID B-A)	
		200 OK INVITE(Call-ID C-B)	→
	←	ACK(Call-ID B-C)	
	←	200 OK INVITE(Call-ID B-A)	
		ACK(Call-ID A-B)	→
		Communication	
		Apply post test routine	

VoLTE to ISDN Use Case 8			
Scenario 7	CCBS User A is located in network A and user B is located in network B. User A has successfully invoked a CCBS request. Ensure that the recall from user A to user B is successful. The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)		**Core Network**	**SIP (Test System B)**
		A CCBS request was already successful	
	←	NOTIFY	
		200 OK NOTIFY	→
		INVITE	→
	←	180 Ringing	
	←	NOTIFY	
		200 OK NOTIFY	→
	←	200 OK INVITE	
		ACK	→
		Apply post test routine	

VoLTE to ISDN Use case 8			
Scenario 8	CCNR User A is located in network A and user B is located in network B. User A has successfully invoked a CCNR request.Ensure that the recall from user A to user B is successful. The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)		**Core Network**	**SIP (Test System B)**
		A CCNR request was already successful	
	←	NOTIFY	
		200 OK NOTIFY	→
		INVITE	→
	←	180 Ringing	
	←	NOTIFY	
		200 OK NOTIFY	→
	←	200 OK INVITE	
		ACK	→
		Apply post test routine	

6.4.3 VoLTE - PSTN Use case 9

VoLTE to PSTN Use Case 9					
Scenario 1	Basic call. The call is released from the called user. This scenario represents the case when the call establishment is performed correctly. The call is released from the called user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).				
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side				
Message flow					
SIP (Test System A)		**Core Network**		**SIP (Test System B)**	
		INVITE	→		
	←	100 Trying			
	←	180 Ringing			
	←	200 OK INVITE			
		ACK	→		
		Communication			
	←	**BYE**			
		200 OK BYE	→		

VoLTE to PSTN Use Case 9					
Scenario 2	Basic call. The call is released from the calling user This scenario represents the case when the call establishment is performed correctly. The call is released from the calling user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).				
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side				
Message flow					
SIP (Test System A)		**Core Network**		**SIP (Test System B)**	
		INVITE	→		
	←	100 Trying			
	←	180 Ringing			
	←	200 OK INVITE			
		ACK	→		
		Communication			
		BYE	→		
	←	200 OK BYE			

VoLTE to PSTN Use Case 9					
Scenario 3	Called user is user busy This scenario represents the case, when the called user is user determined user busy the network initiate call clearing to the calling user.				
Options					
Message flow					
SIP (Test System A)		**Core Network**		**SIP (Test System B)**	
		INVITE	→		
	←	486 Busy Here			
		ACK	→		

VoLTE to PSTN Use Case 9	
Scenario 4	CFU Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side

Message flow

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE(Call-ID A-B)	→	
		CFU is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		

VoLTE to PSTN Use Case 9	
Scenario 5	CFB Ensure that when user A calls user B which is user determined user busy (UDUB), the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.
Options	

Message flow

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE(Call-ID A-B)	→	
		CFB is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		

VoLTE to PSTN Use Case 9	
Scenario 6	CFNR Ensure that when user A calls user B which does not answer, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side

Message flow

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE(Call-ID A-B)	→	
	←	180 Ringing(Call-ID B-A)		
		CFNR is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		

VoLTE to PSTN Use Case 9			
Scenario 7	CCBS User A is located in network A and user B is located in network B. User A has successfully invoked a CCBS request. Ensure when the user B becomes available for CC recall, the CC recall procedure is started. Ensure that the recall from user A to user B is successful. The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)		**Core Network**	**SIP (Test System B)**
		A CCBS request was already successful	
	←	NOTIFY	
		200 OK NOTIFY	→
		INVITE	→
	←	180 Ringing	
	←	NOTIFY	
		200 OK NOTIFY	→
	←	200 OK INVITE	
		ACK	→
		Apply post test routine	

VoLTE to PSTN Use Case 9			
Scenario 8	CCNR User A is located in network A and user B is located in network B. User A has successfully invoked a CCNR request. Ensure that the recall from user A to user B is successful. The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)		**Core Network**	**SIP (Test System B)**
		A CCNR request was already successful	
	←	NOTIFY	
		200 OK NOTIFY	→
		INVITE	→
	←	180 Ringing	
	←	NOTIFY	
		200 OK NOTIFY	→
	←	200 OK INVITE	
		ACK	→
		Apply post test routine	

6.4.4 PSTN to VoLTE Use case 10

PSTN to VoLTE Use case 10				
Scenario 1	Basic call. The call is released from the calling user This scenario represents the case when the call establishment is performed correctly. The call is released from the calling user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).			
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side			
Message flow				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE	→	
	←	100 Trying		
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		Communication		
		BYE	→	
	←	200 OK BYE		

PSTN to VoLTE Use case 10				
Scenario 2	Basic call The call is released from the called user. This scenario represents the case when the call establishment is performed correctly. The call is released from the called user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).			
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side			
Message flow				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE	→	
	←	100 Trying		
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		Communication		
	←	**BYE**		
		200 OK BYE	→	

PSTN to VoLTE Use case 10	
Scenario 3	Basic call - overlap sending This scenario represents the case when the call establishment using overlap sending. The call is released from the calling user. The call is released from the calling user. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters).
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side

Message flow Multiple INVITE method is used

SIP (Test System A)		Core Network		SIP (Test System 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)		
		Apply post test routine		

Message flow Overlap sending, the in-Dialogue method is used

SIP (Test System A)		Core Network		SIP (Test System 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		

PSTN to VoLTE Use case 10	
Scenario 4	Called user is user busy This scenario represents the case, when the called user is user determined user busy the network initiate call clearing to the calling user.
Options	

Message flow

SIP (Test System A)		Core Network		SIP (Test System B)
		INVITE	→	
	←	486 Busy Here		
		ACK	→	

PSTN to VoLTE Use case 10			
Scenario 5	CFU Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)		**Core Network**	**SIP (Test System B)**
		INVITE(Call-ID A-B)	→
		CFU is performed	
	←	INVITE(Call-ID B-C)	
		180 Ringing(Call-ID C-B)	→
	←	180 Ringing(Call-ID B-A)	
		200 OK INVITE(Call-ID C-B)	→
	←	ACK(Call-ID B-C)	
	←	200 OK INVITE(Call-ID B-A)	
		ACK(Call-ID A-B)	→
		Communication	
		Apply post test routine	

PSTN to VoLTE Use case 10			
Scenario 6	CFB Ensure that when user A calls user B which is user determined user busy (UDUB), the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)		**Core Network**	**SIP (Test System B)**
		INVITE(Call-ID A-B)	→
		CFB is performed	
	←	INVITE(Call-ID B-C)	
		180 Ringing(Call-ID C-B)	→
	←	180 Ringing(Call-ID B-A)	
		200 OK INVITE(Call-ID C-B)	→
	←	ACK(Call-ID B-C)	
	←	200 OK INVITE(Call-ID B-A)	
		ACK(Call-ID A-B)	→
		Communication	
		Apply post test routine	

PSTN to VoLTE Use case 10			
Scenario 7	CFNR Ensure that when user A calls user B which does not answer, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)		**Core Network**	**SIP (Test System B)**
		INVITE(Call-ID A-B)	→
	←	180 Ringing(Call-ID B-A)	
		CFNR is performed	
	←	INVITE(Call-ID B-C)	
		180 Ringing(Call-ID C-B)	→
	←	180 Ringing(Call-ID B-A)	
		200 OK INVITE(Call-ID C-B)	→
	←	ACK(Call-ID B-C)	
	←	200 OK INVITE(Call-ID B-A)	
		ACK(Call-ID A-B)	→
		Communication	
		Apply post test routine	

PSTN to MMTel Use case 6			
Scenario 8	CCBS User A is located in network A and user B is located in network B. User A has successfully invoked a CCBS request. Ensure that the recall from user A to user B is successful. The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)		**Core Network**	**SIP (Test System B)**
		A CCBS request was already successful	
	←	NOTIFY	
		200 OK NOTIFY	→
		INVITE	→
	←	180 Ringing	
	←	NOTIFY	
		200 OK NOTIFY	→
	←	200 OK INVITE	
		ACK	→
		Apply post test routine	

PSTN to VoLTE Use case 10				
Scenario 9	CCNR User A is located in network A and user B is located in network B. User A has successfully invoked a CCNR request.Ensure that the recall from user A to user B is successful. The call is released from the calling user.			
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side			
Message flow				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		A CCNR request was already successful		
	←	NOTIFY		
		200 OK NOTIFY	→	
		INVITE	→	
	←	180 Ringing		
	←	NOTIFY		
		200 OK NOTIFY	→	
	←	200 OK INVITE		
		ACK	→	
		Apply post test routine		

6.5 VoLTE to VoLTE Use case 11

VoLTE to VoLTE Use case 11				
Scenario 1	Successful call - This scenario represents the case when the call establishment is performed correctly. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.			
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side			
Message flow a) Successful call - resource resevation on both sides				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE (SDP1)	→	
	←	100 Trying		
	←	183 Session Progress		
		PRACK	→	
	←	200 OK PRACK		
		UPDATE	→	
	←	200 OK UPDATE		
	←	180 Ringing		
	←	200 OK		
		ACK	→	
		BYE	→	
	←	200 OK BYE		
		Apply post test routine		
Message flow: b), c)				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE (SDP1)	→	
	←	100 Trying		
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		BYE	→	
	←	200 OK BYE		
		Apply post test routine		

VoLTE to VoLTE Use case 11			
Scenario 2	Successful call - This scenario represents the case when the call establishment is performed correctly. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the called user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow a) Successful call - resource resevation on both sides			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE (SDP1)	→	
←	100 Trying		
←	183 Session Progress		
	PRACK	→	
←	200 OK PRACK		
	UPDATE	→	
←	200 OK UPDATE		
←	180 Ringing		
←	200 OK		
	ACK	→	
←	BYE		
	200 OK BYE	→	
	Apply post test routine		
Message flow: b), c)			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE (SDP1)	→	
←	100 Trying		
←	180 Ringing		
←	200 OK INVITE		
	ACK	→	
←	BYE		
	200 OK BYE	→	
	Apply post test routine		

VoLTE to VoLTE Use case 11			
Scenario 3	Basic call - Fax with 14,4 kbit/s with using the T.38 in an audio m-line codec This scenario represents the case when in the active call state the Fax transfer on the media and B-channels is performed correctly and the echo cancellers in the GW are not activated. The call is released from the called user. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters).		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
SIP Parameter	INVITE: SDP m=image <Port> udptl t38 180/200 OK INVITE: SDP m=image <Port> udptl t38		
Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE (SDP1)	→	
←	180 Ringing		
←	200 OK INVITE (SDP2)		
	ACK	→	
	Apply post test routine		

VoLTE to VoLTE Use case 11			
Scenario 4	Called user is user busy This scenario represents the case, when the called user is user determined user busy the network initiate call clearing to the calling user.		
Options			
Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE	→	
←	486 Busy Here		
	ACK	→	

VoLTE to VoLTE Use case 11			
Scenario 5	CFU Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE(Call-ID A-B)	→	
	CFU is performed		
←	INVITE(Call-ID B-C)		
	180 Ringing(Call-ID C-B)	→	
←	180 Ringing(Call-ID B-A)		
	200 OK INVITE(Call-ID C-B)	→	
←	ACK(Call-ID B-C)		
←	200 OK INVITE(Call-ID B-A)		
	ACK(Call-ID A-B)	→	
	Communication		
	Apply post test routine		

VoLTE to VoLTE Use case 11			
Scenario 6	CFB Ensure that when user A calls user B which is user determined user busy (UDUB), the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE(Call-ID A-B)	→	
	CFB is performed		
←	INVITE(Call-ID B-C)		
	180 Ringing(Call-ID C-B)	→	
←	180 Ringing(Call-ID B-A)		
	200 OK INVITE(Call-ID C-B)	→	
←	ACK(Call-ID B-C)		
←	200 OK INVITE(Call-ID B-A)		
	ACK(Call-ID A-B)	→	
	Communication		
	Apply post test routine		

VoLTE to VoLTE Use case 11			
Scenario 7	CFNR Ensure that when user A calls user B which does not answer, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE(Call-ID A-B)	→	
←	180 Ringing(Call-ID B-A)		
	CFNR is performed		
←	INVITE(Call-ID B-C)		
	180 Ringing(Call-ID C-B)	→	
←	180 Ringing(Call-ID B-A)		
	200 OK INVITE(Call-ID C-B)	→	
←	ACK(Call-ID B-C)		
←	200 OK INVITE(Call-ID B-A)		
	ACK(Call-ID A-B)	→	
	Communication		
	Apply post test routine		

VoLTE to VoLTE Use case 11			
Scenario 8	CCBS User A is located in network A and user B is located in network B. User A has successfully invoked a CCBS request. Ensure that the recall from user A to user B is successful. The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	A CCBS or CCNR request was already successful		
←	NOTIFY		
	200 OK NOTIFY	→	
	INVITE	→	
←	180 Ringing		
←	NOTIFY		
	200 OK NOTIFY	→	
←	200 OK INVITE		
	ACK	→	
	Apply post test routine		

VoLTE to VoLTE Use case 11				
Scenario 9	CCNR User A is located in network A and user B is located in network B. User A has successfully invoked a CCNR request. Ensure that the recall from user A to user B is successful. The call is released from the calling user.			
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side			
Message flow				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		A CCBS or CCNR request was already successful		
	←	NOTIFY		
		200 OK NOTIFY	→	
		INVITE	→	
	←	180 Ringing		
	←	NOTIFY		
		200 OK NOTIFY	→	
	←	200 OK INVITE		
		ACK	→	
		Apply post test routine		

6.6 MMTel -VoLTE

6.6.1 VoLTE to MMTel Use case 12

VoLTE to MMTel Use case 12				
Scenario 1	Successful call - This scenario represents the case when the call establishment is performed correctly. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.			
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side			
Message flow a) Successful call - resource resevation on both sides				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE (SDP1)	→	
	←	100 Trying		
	←	183 Session Progress		
		PRACK	→	
	←	200 OK PRACK		
		UPDATE	→	
	←	200 OK UPDATE		
	←	180 Ringing		
	←	200 OK		
		ACK	→	
		BYE	→	
	←	200 OK BYE		
		Apply post test routine		
Message flow: b), c)				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE (SDP1)	→	
	←	100 Trying		
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		BYE	→	
	←	200 OK BYE		
		Apply post test routine		

VoLTE to MMTel Use case 12			
Scenario 2	Successful call - This scenario represents the case when the call establishment is performed correctly . Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the called user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow a) Successful call - resource resevation on both sides			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE (SDP1)	→	
←	100 Trying		
←	183 Session Progress		
	PRACK	→	
←	200 OK PRACK		
	UPDATE	→	
←	200 OK UPDATE		
←	180 Ringing		
←	200 OK		
	ACK	→	
←	BYE		
	200 OK BYE	→	
	Apply post test routine		
Message flow: b), c)			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE (SDP1)	→	
←	100 Trying		
←	180 Ringing		
←	200 OK INVITE		
	ACK	→	
←	BYE		
	200 OK BYE	→	
	Apply post test routine		

VoLTE to MMTel Use case 12			
Scenario 3	Basic call - Fax with 14,4 kbit/s with using the T.38 in an audio m-line codec This scenario represents the case when in the active call state the Fax transfer on the media and B-channels is performed correctly and the echo cancellers in the GW are not activated. The call is released from the called user. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters).		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
SIP Parameter	INVITE: SDP m=image <Port> udptl t38 180/200 OK INVITE: SDP m=image <Port> udptl t38		
Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE (SDP1)	→	
←	180 Ringing		
←	200 OK INVITE (SDP2)		
	ACK	→	
	Apply post test routine		

VoLTE to MMTel Use case 12	
Scenario 4	Called user is user busy This scenario represents the case, when the called user is user determined user busy the network initiate call clearing to the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
Message flow **SIP (Test System A)** **Core Network** **SIP (Test System B)** INVITE → ← 486 Busy Here ACK →	

VoLTE to MMTel Use case 12	
Scenario 5	CFU Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
Message flow **SIP (Test System A)** **Core Network** **SIP (Test System B)** INVITE(Call-ID A-B) → **CFU is performed** ← INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) 200 OK INVITE(Call-ID C-B) → ← ACK(Call-ID B-C) ← 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → **Communication** **Apply post test routine**	

VoLTE to MMTel Use case 12	
Scenario 6	CFB Ensure that when user A calls user B which is user determined user busy (UDUB), the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
Message flow **SIP (Test System A)** **Core Network** **SIP (Test System B)** INVITE(Call-ID A-B) → **CFB is performed** ← INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) 200 OK INVITE(Call-ID C-B) → ← ACK(Call-ID B-C) ← 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → **Communication** **Apply post test routine**	

VoLTE to MMTel Use case 12				
Scenario 7	CFNR Ensure that when user A calls user B which does not answer, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.			
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side			
Message flow				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE(Call-ID A-B)	→	
	←	180 Ringing(Call-ID B-A)		
		CFNR is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		

VoLTE to MMTel Use case 12				
Scenario 8	CCBS User A is located in network A and user B is located in network B. User A has successfully invoked a CCBS request. Ensure that the recall from user A to user B is successful. The call is released from the calling user.			
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side			
Message flow				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		A CCNR request was already successful		
	←	NOTIFY		
		200 OK NOTIFY	→	
		INVITE	→	
	←	180 Ringing		
	←	NOTIFY		
		200 OK NOTIFY	→	
	←	200 OK INVITE		
		ACK	→	
		Apply post test routine		

VoLTE to MMTel Use case 12			
Scenario 9	CCNR User A is located in network A and user B is located in network B. User A has successfully invoked a CCNR request. Ensure that the recall from user A to user B is successful. The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	A CCNR request was already successful		
←	NOTIFY		
	200 OK NOTIFY	→	
	INVITE	→	
←	180 Ringing		
←	NOTIFY		
	200 OK NOTIFY	→	
←	200 OK INVITE		
	ACK	→	
	Apply post test routine		

6.6.2 MMTel to VoLTE Use case 13

MMTel to VoLTE Use case 13			
Scenario 1	Successful call - This scenario represents the case when the call establishment is performed correctly. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow a) Successful call - resource resevation on both sides			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE (SDP1)	→	
←	100 Trying		
←	183 Session Progress		
	PRACK	→	
←	200 OK PRACK		
	UPDATE	→	
←	200 OK UPDATE		
←	180 Ringing		
←	200 OK		
	ACK	→	
	BYE	→	
←	200 OK BYE		
	Apply post test routine		
Message flow: b), c)			
SIP (Test System A)	**Core Network**		**SIP (Test System B)**
	INVITE (SDP1)	→	
←	100 Trying		
←	180 Ringing		
←	200 OK INVITE		
	ACK	→	
	BYE	→	
←	200 OK BYE		
	Apply post test routine		

MMTel to VoLTE Use case 13	
Scenario 2	Successful call - This scenario represents the case when the call establishment is performed correctly . Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the called user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side

Message flow a) Successful call - resource resevation on both sides

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE (SDP1)	→	
	←	100 Trying		
	←	183 Session Progress		
		PRACK	→	
	←	200 OK PRACK		
		UPDATE	→	
	←	200 OK UPDATE		
	←	180 Ringing		
	←	200 OK		
		ACK	→	
	←	BYE		
		200 OK BYE	→	
		Apply post test routine		

Message flow: b), c)

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE (SDP1)	→	
	←	100 Trying		
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
	←	BYE		
		200 OK BYE	→	
		Apply post test routine		

MMTel to VoLTE Use case 13	
Scenario 3	Basic call - Fax with 14,4 kbit/s with using the T.38 in an audio m-line codec This scenario represents the case when in the active call state the Fax transfer on the media and B-channels is performed correctly and the echo cancellers in the GW are not activated. The call is released from the called user. Ensure that in the active call state the data transfer is performed correctly (e.g. testing QoS parameters).
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
SIP Parameter	INVITE: SDP m=image <Port> udptl t38 180/200 OK INVITE: SDP m=image <Port> udptl t38

Message flow

SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE (SDP1)	→	
	←	180 Ringing		
	←	200 OK INVITE (SDP2)		
		ACK	→	
		Apply post test routine		

MMTel to VoLTE Use case 13	
Scenario 4	Called user is user busy This scenario represents the case, when the called user is user determined user busy the network initiate call clearing to the calling user.
Options	
Message flow **SIP (Test System A)** **Core Network** **SIP (Test System B)** INVITE → ← 486 Busy Here ACK →	

MMTel to VoLTE Use case 13	
Scenario 5	CFU Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
Message flow **SIP (Test System A)** **Core Network** **SIP (Test System B)** INVITE(Call-ID A-B) → **CFU is performed** ← INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) 200 OK INVITE(Call-ID C-B) → ← ACK(Call-ID B-C) ← 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → **Communication** **Apply post test routine**	

MMTel to VoLTE Use case 13	
Scenario 6	CFB Ensure that when user A calls user B which is user determined user busy (UDUB), the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side
Message flow **SIP (Test System A)** **Core Network** **SIP (Test System B)** INVITE(Call-ID A-B) → **CFB is performed** ← INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) 200 OK INVITE(Call-ID C-B) → ← ACK(Call-ID B-C) ← 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → **Communication** **Apply post test routine**	

MMTel to VoLTE Use case 13				
Scenario 7	CFNR Ensure that when user A calls user B which does not answer, the call is forwarded to user C. Ensure that in the active call state the voice transfer is performed correctly (e.g. testing QoS parameters). The call is released from the calling user.			
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side			
Message flow				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		INVITE(Call-ID A-B)	→	
	←	180 Ringing(Call-ID B-A)		
		CFNR is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		

MMTel to VoLTE Use case 13				
Scenario 8	CCBS User A is located in network A and user B is located in network B. User A has successfully invoked a CCBS request.Ensure that the recall from user A to user B is successful. The call is released from the calling user.			
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side			
Message flow				
SIP (Test System A)		**Core Network**		**SIP (Test System B)**
		A CCBS request was already successful		
	←	NOTIFY		
		200 OK NOTIFY	→	
		INVITE	→	
	←	180 Ringing		
	←	NOTIFY		
		200 OK NOTIFY	→	
	←	200 OK INVITE		
		ACK	→	
		Apply post test routine		

MMTel to VoLTE Use case 13			
Scenario 9	CCNR User A is located in network A and user B is located in network B. User A has successfully invoked a CCNR request. Ensure that the recall from user A to user B is successful. The call is released from the calling user.		
Options	a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side		
Message flow			
SIP (Test System A)		**Core Network**	**SIP (Test System B)**
		A CCNR request was already successful	
	←	NOTIFY	
		200 OK NOTIFY	→
		INVITE	→
	←	180 Ringing	
	←	NOTIFY	
		200 OK NOTIFY	→
	←	200 OK INVITE	
		ACK	→
		Apply post test routine	

7 Metrics and design objectives

7.1 Delay probability

This clause defines the delay parameters, the corresponding values are defined in TS 186 008-4 [i.3].

IMS systems shall comply with the requirements given in the following tables.

Table 3

Meaning of timers	Parameter Q.543 [i.14] Detailed description	IMS/PES equivalent
Local exchange call request delay - originating outgoing and internal traffic connections		
ANALOGUE SUBSCRIBER LINES Local exchange call request delay - originating outgoing and internal traffic connections.	clause 2.3.2.1 [i.14] For ANALOGUE SUBSCRIBER LINES, call request delay is defined as the interval from the instant when the off-hook condition is recognizable at the subscriber line interface of the exchange until the exchange begins to apply dial tone to the line. The call request delay interval is assumed to correspond to the period at the beginning of a call attempt during which the exchange is unable to receive any call address information from the subscriber.	PES [i.15] For ANALOGUE SUBSCRIBER LINES connected to the AGCF/MSAN . Call request delay is defined as the interval from the instant when the off-hook condition is recognizable at the subscriber line interface of the AGCF/MSAN until the AGCF/MSAN begins to apply dial tone to the line.
ANALOGUE SUBSCRIBER with IAD (VGW) Local exchange call request delay - originating outgoing and internal traffic connections.		PES [i.15] For ANALOGUE SUBSCRIBER LINES connected to the VGW. Call request delay is defined as the interval from the instant when the off-hook condition is recognizable at the subscriber line interface of the VGW until the VGW begins to apply dial tone to the line.

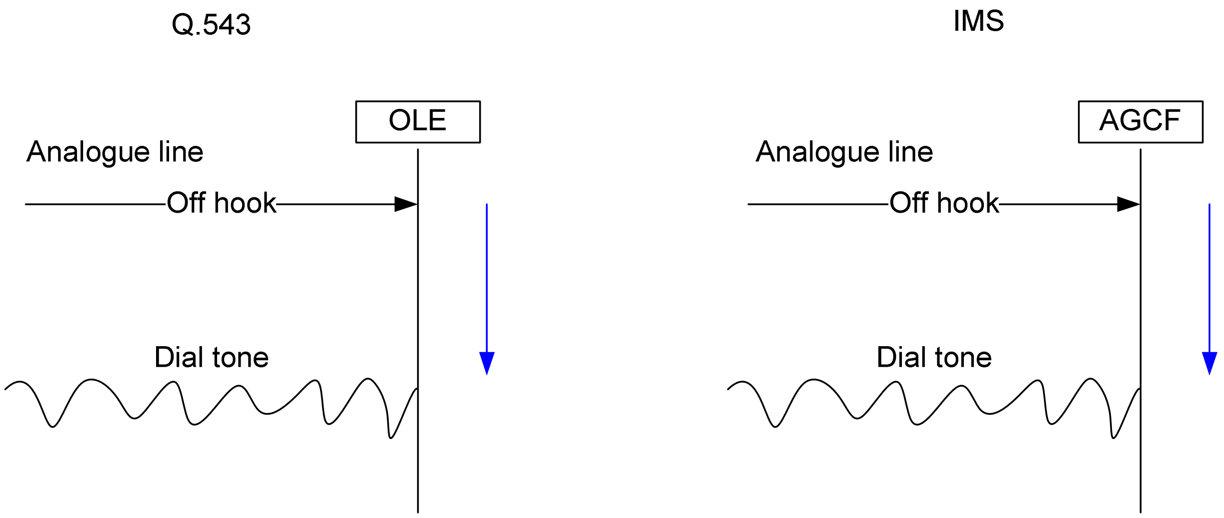


Figure 11: Local exchange analogue subscriber call request delay: overlap sending

Table 4

Meaning of timers	Parameter Q.543 [i.14]	IMS/PES equivalent
	Detailed description	
Local exchange ISDN subscriber call request delay: overlap sending		
ISDN SUBSCRIBER LINES Local exchange call request delay - Overlap sending.	clause 2.3.2.2 [i.14] Local exchange call request delay. Call request delay is defined as the interval from the instant at which the SETUP message has been received from the subscriber signalling system until the SETUP ACKNOWLEDGE message is passed back to the subscriber signalling system.	ISDN [i.16] Call request delay is defined as the interval from the instant at which the SETUP message has been received from the subscriber signalling system until the SETUP ACKNOWLEDGE message is passed back to the subscriber signalling system.
IMS SUBSCRIBER Local exchange call request delay.		IMS [i.17] Call request delay is defined as the interval from the instant at which the INVITE message has been received from the SIP subscriber until the 100 Trying from the SBC/P-CSCF is passed back to the subscriber.

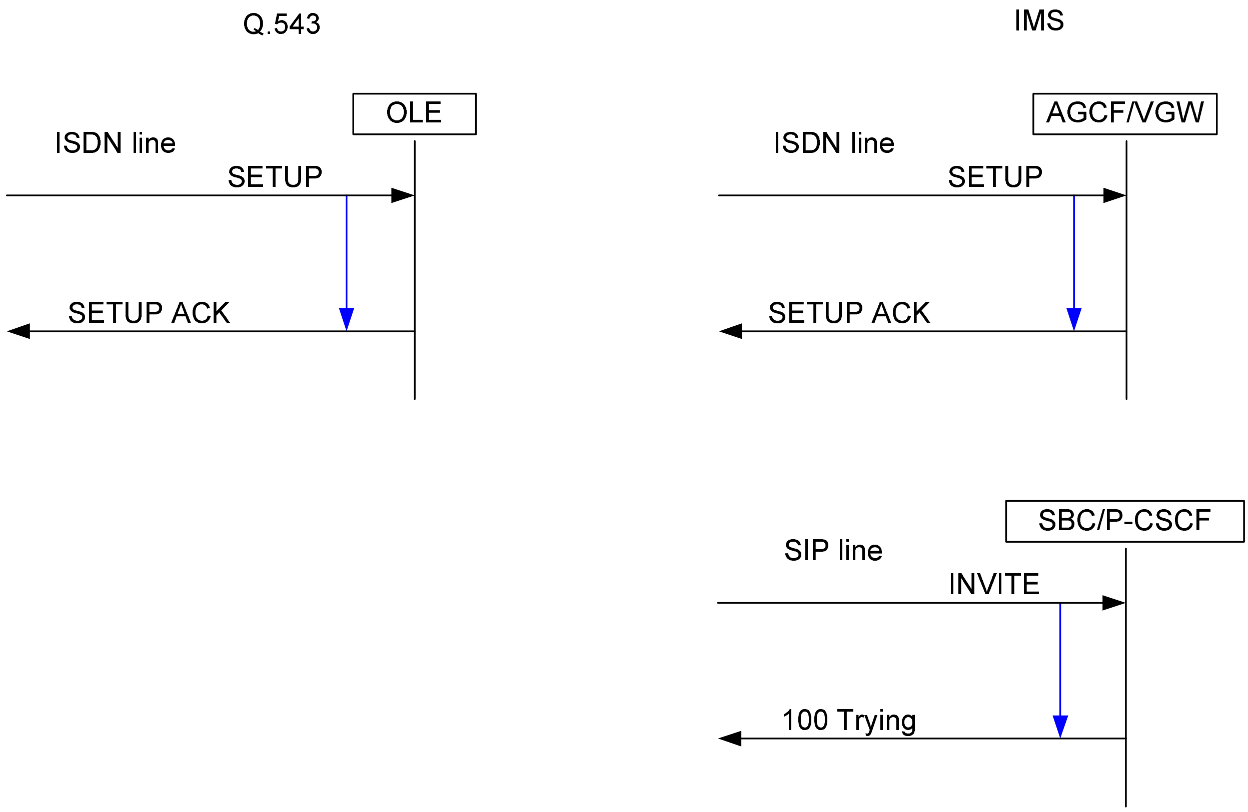


Figure 12: Local exchange ISDN subscriber call request delay: overlap sending

Table 5

Meaning of timers	Parameter Q.543 [i.14]	IMS/PES equivalent
	Detailed description	
Local exchange ISDN subscriber call request delay: en Block sending		
ISDN SUBSCRIBER LINES Local exchange call request delay en - block sending.	clause 2.3.2.3 [i.14] For DIGITAL SUBSCRIBER LINES using en-bloc sending, call request delay is defined as the interval from the instant at which the SETUP message is received from the subscriber signalling system until the call proceeding message is passed back to the subscriber signalling system.	ISDN [i.16] For ISDN using en-bloc sending, call request delay is defined as the interval from the instant at which the SETUP message is received from the subscriber signalling system until the CALL PROCCEDING message is passed back to the subscriber signalling system.

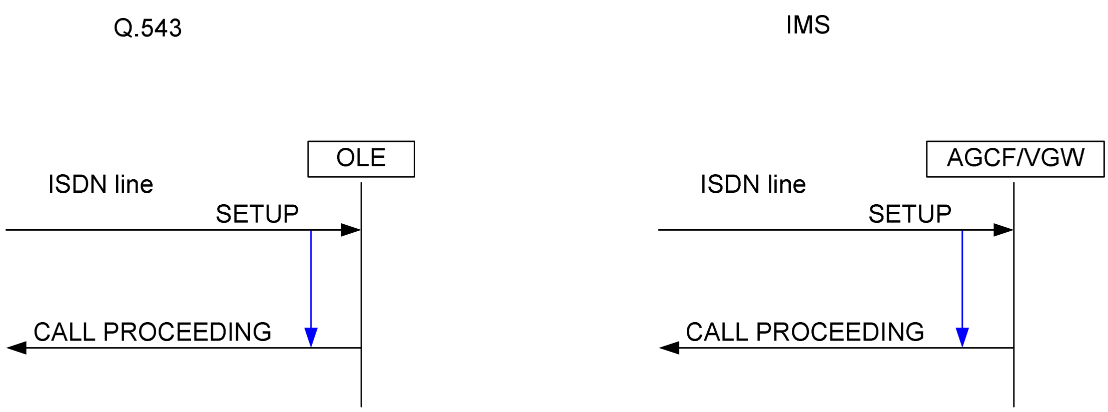


Figure 13: Local exchange ISDN subscriber call request delay: en Block sending

Table 6

Meaning of timers	Parameter Q.543 [i.14] Detailed description	IMS/PES equivalent
Alerting sending delay for terminating traffic (the users are in different locations, controlled by different S-CSCF/P-CSCF)		
ANALOGUE SUBSCRIBER LINES Alerting sending Delay for terminating traffic.	clause 2.3.6.1.1 [i.14] For calls terminating on ANALOGUE SUBSCRIBER LINES, alerting sending delay is defined as the interval from the instant when the last digit is available for processing in the exchange until the ringing tone is sent backwards toward the calling user.	PES [i.15] For calls terminating on ANALOGUE SUBSCRIBER LINES, **alerting sending** delay is defined as the interval from the instant when the last digit is available for processing in the AGCF/MSAN until the **ringing tone** is sent toward the calling user.
ISDN SUBSCRIBER LINES Alerting sending Delay for terminating traffic.	clause 2.3.6.1.2 [i.14] For calls terminating on DIGITAL SUBSCRIBER LINES, the alerting sending delay is defined as the interval from the instant that an ALERTING message is received from the digital subscriber line signalling system to the instant at which an ADDRESS COMPLETE message is passed to the interexchange signalling system or ringing tone is sent backward toward the calling user.	ISDN [i.16] For calls terminating on ISDN, the alerting sending delay is defined as the interval from the instant that an ALERTING message is received from the digital subscriber line signalling to the instant at which an AGCF/MSAN sends the 180 Ringing backward toward the calling user.

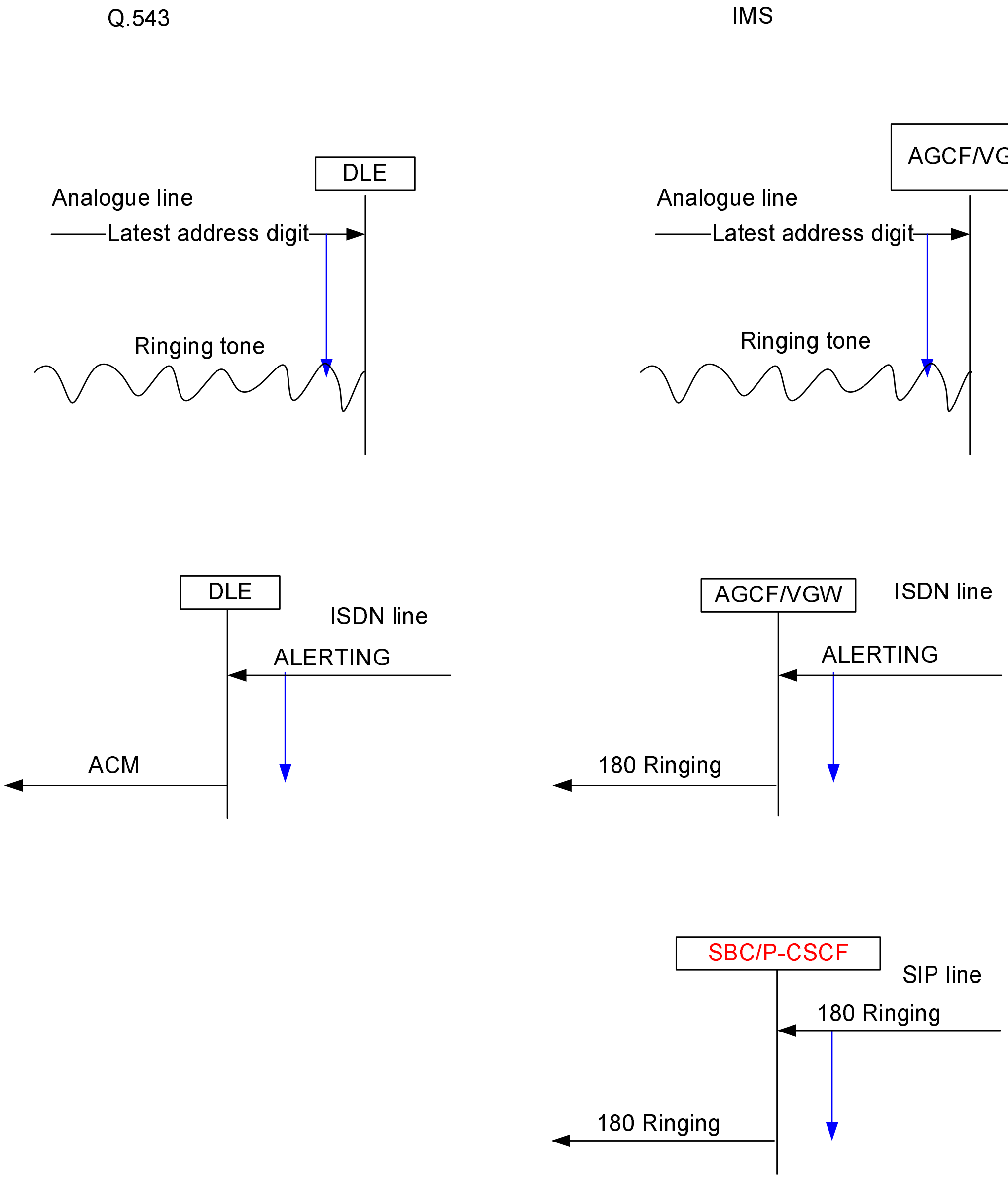


Figure 14: Local exchange Alerting sending delay for terminating traffic (in different locations)

Table 7

Meaning of timers	Parameter Q.543 [i.14]	IMS/PES equivalent
	Detailed description	
Alerting sending delay for internal traffic (the user are in same locations, controlled by same AGCF/VGW or P-CSCF)		
ANALOGUE SUBSCRIBER LINES Alerting sending Delay for internal traffic.	clause 2.3.6.2.1 [i.14] For calls terminating on ANALOGUE SUBSCRIBER LINES, alerting sending delay is defined as the interval from the instant that the signalling information is available for processing in the exchange until ringing tone is applied to an ANALOGUE calling subscriber.	PES [i.15] For calls terminating on ANALOGUE SUBSCRIBER LINES, alerting sending delay is defined as the interval from the instant that the signalling information is available for processing in the AGCF/MSAN until Ringing tone is sent towards the calling subscriber.
ANALOGUE SUBSCRIBER LINES VGW Alerting sending Delay for internal traffic.		PES [i.15] For calls terminating on ANALOGUE SUBSCRIBER LINES, alerting sending delay is defined as the interval from the instant that the signalling information is available for processing in the VGW until Ringing tone is sent towards the calling subscriber.
ISDN SUBSCRIBER LINES Alerting sending Delay for Internal traffic.	clause 2.3.6.2.2 [i.14] For internal calls terminating on DIGITAL SUBSCRIBER LINES originating from DIGITAL SUBSCRIBER LINES, alerting sending delay is defined as the interval from the instant that an ALERTING message is received from the signalling system of the called subscriber's line until the ALERTING message is applied to the calling subscriber line.	ISDN [i.16] For calls terminating on ISDN, alerting sending delay is defined as the interval from the instant that an ALERTING message is received and ALERTING is sent towards the calling subscriber.
IMS SUBSCRIBER LINES 180 sending Delay for Internal traffic.		IMS [i.17] For calls terminating sending delay is defined as the interval from the instant that an 180 message at the Gm interface has received and 180 is sent on the Gm towards the calling subscriber.
LTE SUBSCRIBER LINES 180 sending Delay for Internal traffic.		LTE For calls terminating sending delay is defined as the interval from the instant that an 180 message at the LTE UE interface has received and 180 is sent on the LTE UE towards the calling subscriber.

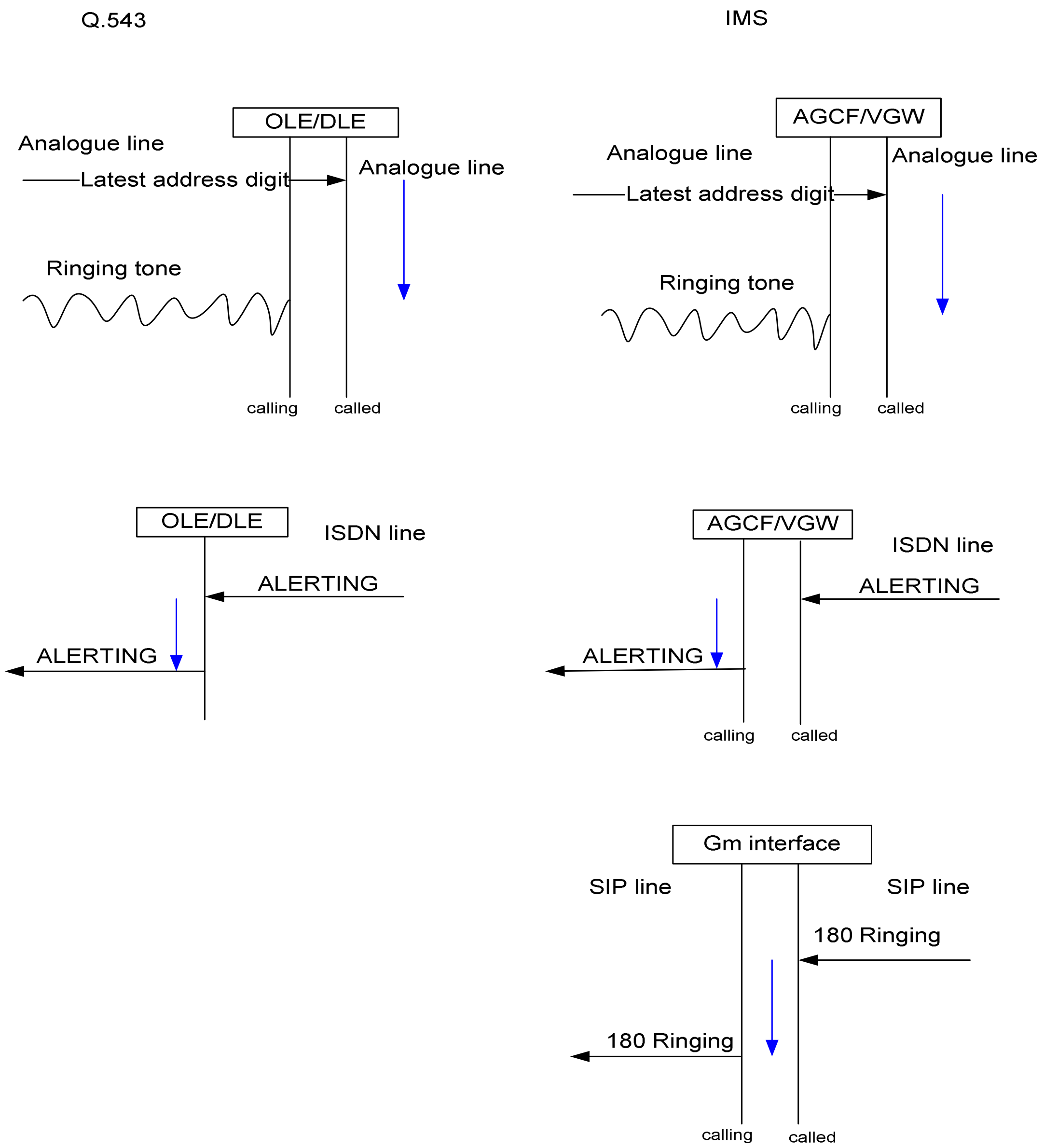


Figure 15: Alerting sending delay for internal traffic
(the user are in same locations, controlled by same AGCF/VGW or P-CSCF)

Table 8

Meaning of timers	Parameter Q.543 [i.14]	IMS/PES equivalent
	Detailed description	
Call set up delay		
ISDN SUBSCRIBER LINES Call set up delay using overlap signalling.	clause 2.4.3.1 [i.14] Call set-up delay is defined as the interval from the instant when the signalling information required for routing is received from the incoming signalling system until the instant when the corresponding signalling information is passed to the outgoing signalling system. Exchange call setup delay for originating outgoing traffic connections, digital subscriber lines. The time interval starts when the INFORMATION message received contains a "sending complete indication" or when the address information necessary for call set-up is complete and ends when the corresponding signalling information is passed to the outgoing signalling system.	ISDN [i.16] Sending, the time interval starts when the INFORMATION message received contains a "sending complete indication" and ends when the INVITE message on the Ic interface has been sent.
		ISDN [i.16] Sending, the time interval starts when the INFORMATION message received contains a "sending complete indication" and ends when the INVITE message on terminating Gm interface has been sent.
		IMS [i.17] Session initiation delay is defined as the interval from the instant when the INVITE signalling information is received from the calling user on the originating Gm interface until the instant when the corresponding INVITE signalling information is passed on the terminating Gm interface to the called user.
		IMS [i.17] Session initiation delay is defined as the interval from the instant when the INVITE signalling information is received from the calling user on the originating Gm interface until the instant when the corresponding INVITE signalling information is passed on the terminating Ic interface to the called user.(without preconditions)

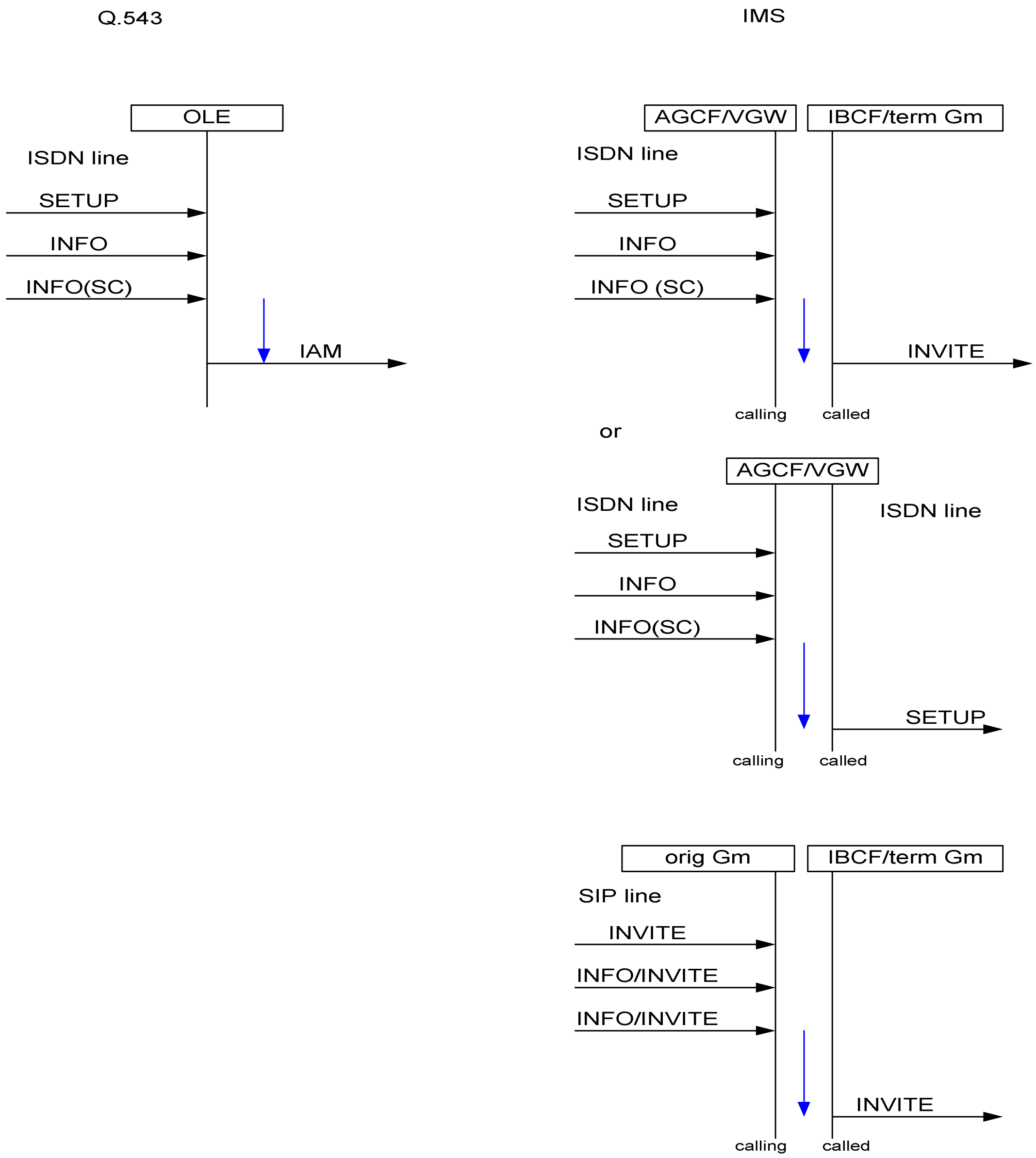


Figure 16: Call set up delay: Overlap sending is used

Table 9

Meaning of timers	Parameter Q.543 [i.14] Detailed description	IMS/PES equivalent
Call set up delay: en Block sending is used		
ISDN SUBSCRIBER LINES Call set up delay using en-block signalling.	clause 2.4.3.1 [i.14] Exchange call setup delay for originating outgoing traffic connections. For call attempts using en-bloc sending Call set-up delay is defined as the interval from the instant when the signalling information required for routing is received from the incoming signalling system until the instant when the corresponding signalling information is passed to the outgoing signalling system. The time interval starts when the SETUP message received contains a "sending complete indication" or when the address information necessary for call set-up is complete and ends when the call setup is sent on the outgoing signalling system.	ISDN [i.16] Call set-up delay is defined as the interval from the instant when the signalling information including Sending Complete (#) is received from the incoming signalling system until the instant when the corresponding INVITE signalling information is passed to the Ic interface.
		ISDN [i.16] Call set-up delay is defined as the interval from the instant when the signalling information including Sending Complete (#) is received from the incoming signalling system until the instant when the corresponding INVITE signalling information is passed to the terminating Gm interface.
		ISDN [i.16] Call set-up delay for Internal traffic is defined as the interval from the instant when the SETUP including Sending Complete (#) is received from the incoming signalling system until the instant when the corresponding SETUP signalling information is passed to the called line signalling system (see note).
IMS SUBSCRIBER Call set up delay using for Internal traffic.		IMS [i.17] Session initiation delay is defined as the interval from the instant when the INVITE signalling information is received from the calling user on the originating Gm interface until the instant when the corresponding INVITE signalling information is passed on the terminating Gm interface to the called user.
		IMS [i.17] Session initiation delay is defined as the interval from the instant when the INVITE signalling information is received from the calling user on the originating Gm interface until the instant when the corresponding INVITE signalling information is passed on the terminating Ic interface to the called user (without preconditions).
		IMS [i.17] Session initiation delay is defined as the interval from the instant when the INVITE signalling information is received from the calling user on the originating Gm interface until the instant when the corresponding INVITE signalling information is passed on the terminating Ic interface to the called user (with preconditions).

Meaning of timers	Parameter Q.543 [i.14]	IMS/PES equivalent
	Detailed description	
LTE		LTE [i.20] Session initiation delay is defined as the interval from the instant when the INVITE signalling information is received from the calling user on the originating LTE _UE (ECM Connected) interface until the instant when the corresponding INVITE signalling information is passed on the terminating LTE _UE (ECM Connected) interface to the called user with QCI 5 (see note).
		LTE [i.20] Session initiation delay is defined as the interval from the instant when the INVITE signalling information is received from the calling user on the originating LTE _UE (ECM Connected) interface until the instant when the corresponding INVITE signalling information is passed on the terminating LTE _UE (ECM Connected) interface to the called user with QCI 1.
NOTE: If SC (#) is not included the setup delay may increase up to the digit collection timer (15 s).		

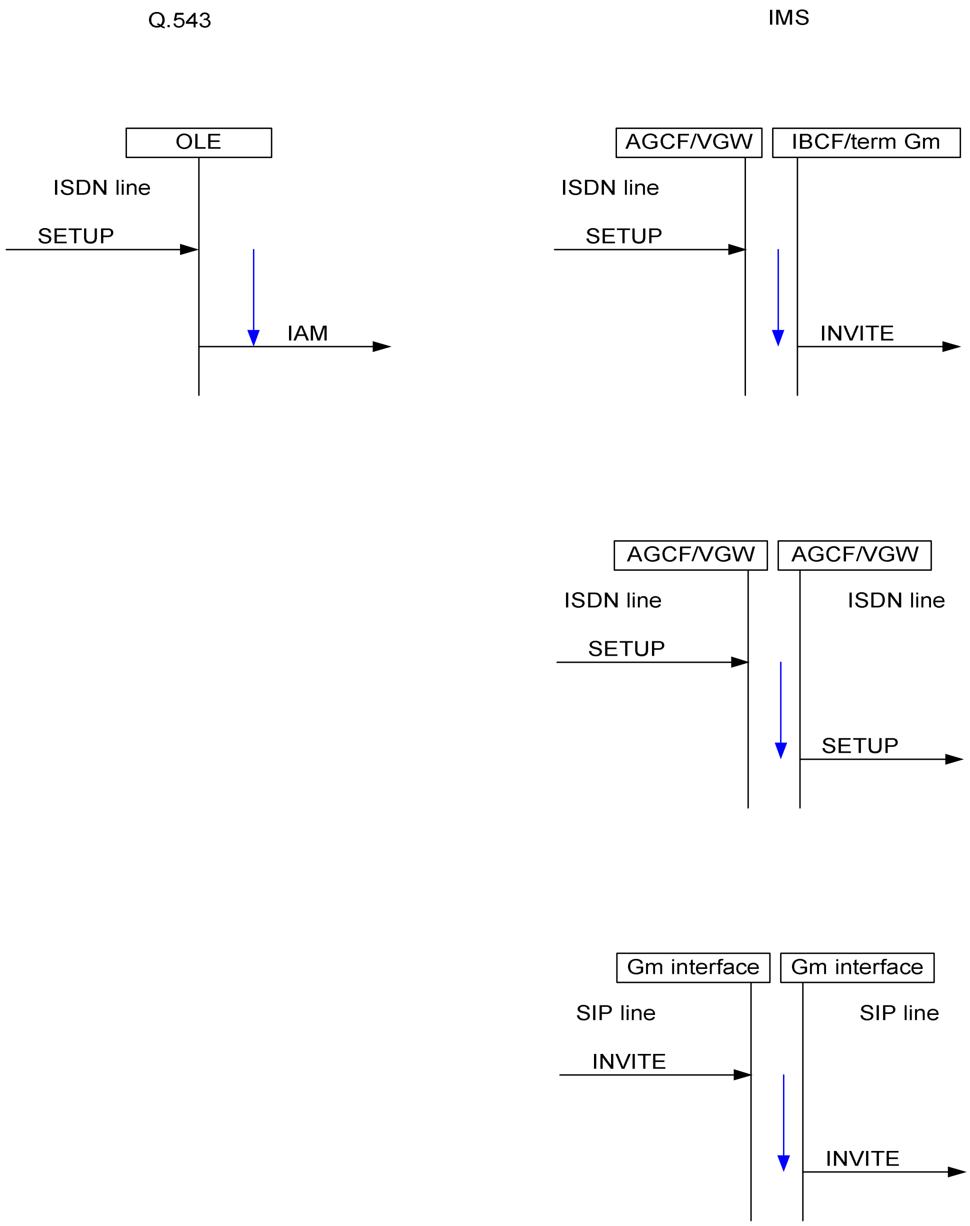


Figure 17: Call set up delay: en Block sending is used

Table 10

Meaning of timers	Parameter Q.543 [i.14]	IMS/PES equivalent
	Detailed description	
Through-connection delay		
ISDN SUBSCRIBER LINES Through-connection delay.	clause 2.4.4.2 [i.14] Through-connection delay. The through connection delay is defined as the interval from the instant that the CONNECT message is received from the called line signalling system until the through connection is established and available for carrying traffic and the ANSWER and CONNECT ACKNOWLEDGEMENT messages have been passed to the appropriate signalling systems.	ISDN [i.16] The through connection delay is defined as the interval from the instant that the CONNECT message is received from the called line signalling system until the through connection is established and available for carrying traffic and the CONNECT message has been sent to the calling user signalling system (see note).
IMS Through-connection delay Delay for Internal traffic.		IMS [i.17] The through connection delay is defined as the interval from the instant that the 200 OK message is received from the called user at the terminating Gm interface until the through connection is established and available for carrying traffic and the 200 OK message has been sent to the calling user on the originating Gm interface.
LTE		LTE [i.20] The through connection delay is defined as the interval from the instant that the 200 OK message is received from the called user at the terminating LTE _UE interface until the through connection is established and available for carrying traffic and the 200 OK message has been sent to the calling user on the originating LTE _UE interface.

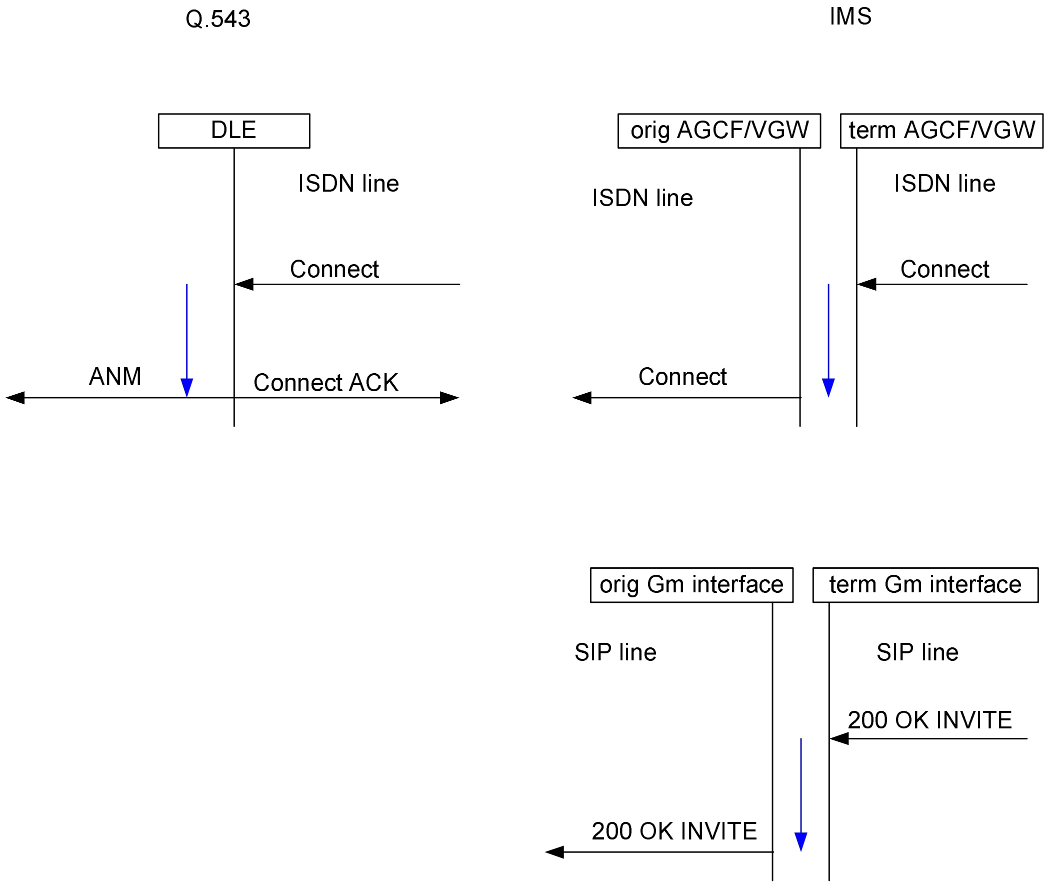


Figure 18: Through-connection delay

Table 11

Meaning of timers	**Parameter Q.543 [i.14]**	**IMS/PES equivalent**
	Detailed description	
Connection release delay		
ISDN SUBSCRIBER LINES Connection call release delay.	clause 2.4.6 [i.14] Connection release delay is defined as the interval from the instant when DISCONNECT or RELEASE message is received from a signalling system until the instant when the connection is no longer available for use on the call (and is available for use on another call) and a corresponding RELEASE or DISCONNECT message is passed to the other signalling system involved in the connection.	ISDN [i.16] Connection release delay is defined as the interval from the instant when DISCONNECT or RELEASE message is received from a signalling system until the instant when RELEASE COMPLETE is sent and a corresponding RELEASE or DISCONNECT message is sent, or vice versa.
IMS SUBSCRIBER Connection call release delay Delay for Internal traffic.		IMS [i.17] Connection release delay is defined as the interval from the instant when a BYE message is received at the originating or terminating Gm interface until the instant when 200OK is sent and a corresponding BYE message is sent at the terminating or originating Gm interface respectively.
LTE - IMS SUBSCRIBER Connection call release delay Delay for Internal traffic.		LTE [i.20] Connection release delay is defined as the interval from the instant when a BYE message is received at the originating or terminating LTE UE interface until the instant when 200OK is sent and a corresponding BYE message is sent at the terminating or originating LTE UE interface respectively.

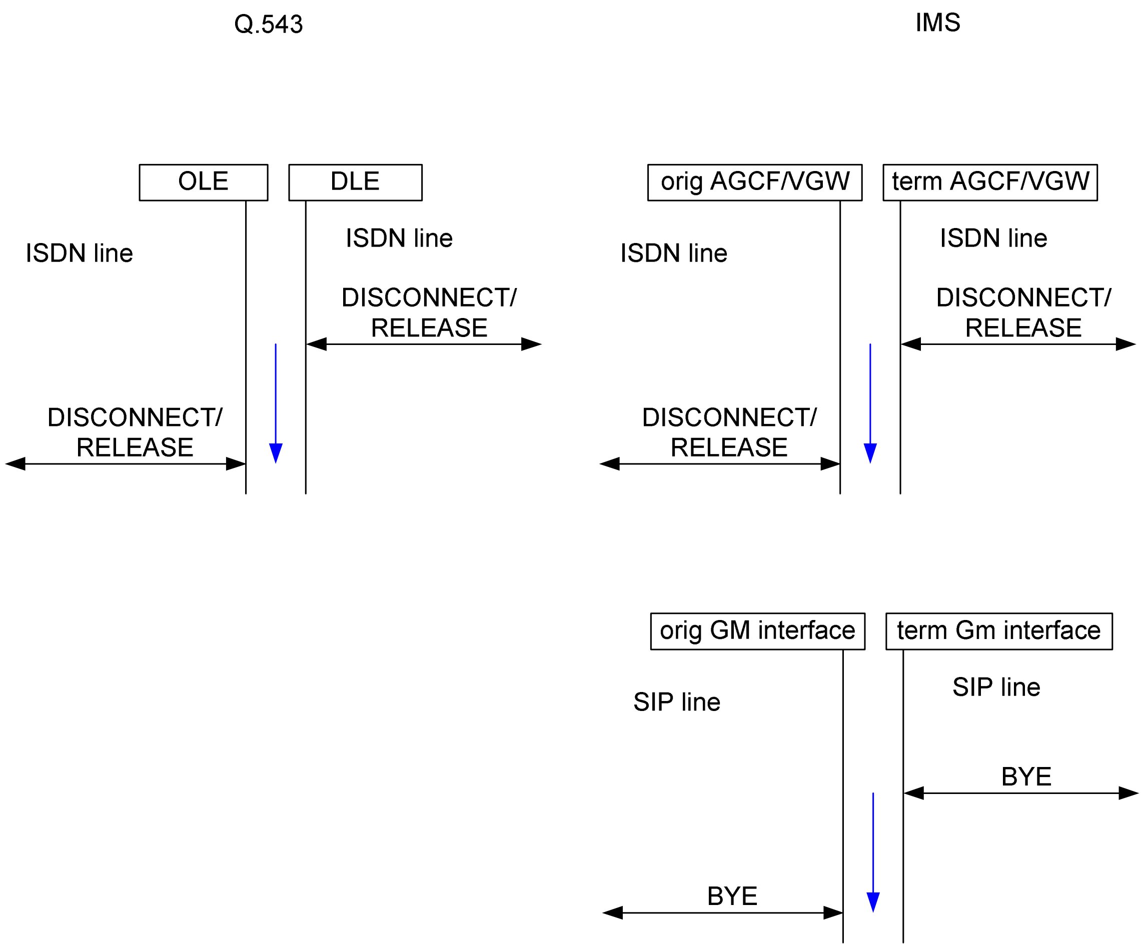


Figure 19: Connection call release delay

7.2 Speech quality analysis

This clause defines a set of parameters which enables the speech quality analysis of the system under test. They are divided in three parts: speech quality, speech level and PESQ offset.

Table12 shows the speech quality parameters based on PESQ/POLQA.

Table 13 shows the speech level parameters.

Table 14 shows the PESQ/POLQA offset parameters.

Table 12: Speech Quality parameters based on PESQ/POLQA

Speech Quality Summary	
	P.862.1/P.863
Min	
Max	
Mean	
Std-Dev	

Table 13: Speech level parameters

Speech Level Summary (Optional)				
	Active Level	**Peak**	**Noise**	**Signal to Interval Noise**
Min				
Max				
Mean				
Std-Dev				

Table 14: PESQ/POLQA offset parameters

Delay Summary - Delay (PESQ/POLQA Time Offset)	
Min	
Max	
Mean	
Std-Dev	
Range	

7.3 Call Profiler Traffic Patterns

This clause defines call profiles which are nowadays implemented in benchmark test systems.

7.3.1 Saw Tooth

The Saw Tooth ramps up to a peak number of calls and then ramps down from peak.

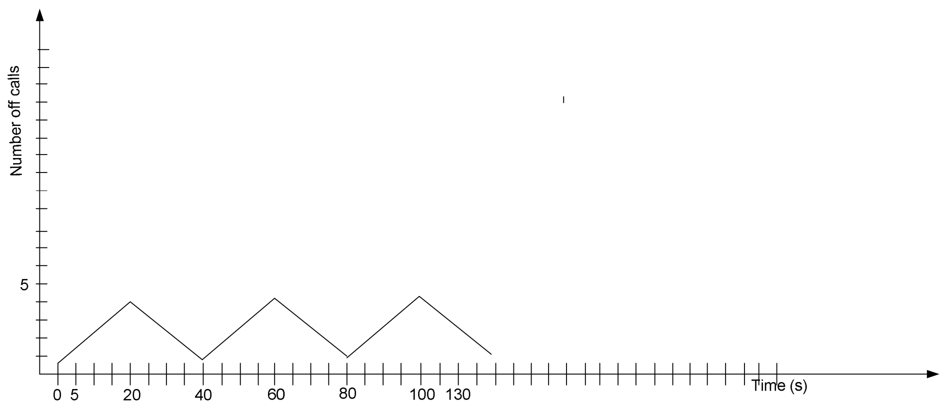


Figure 20: Example of saw tooth call profile

7.3.2 Blast

Blast - all calls go off-hook simultaneously, are connected for a specified time, and then disconnected.

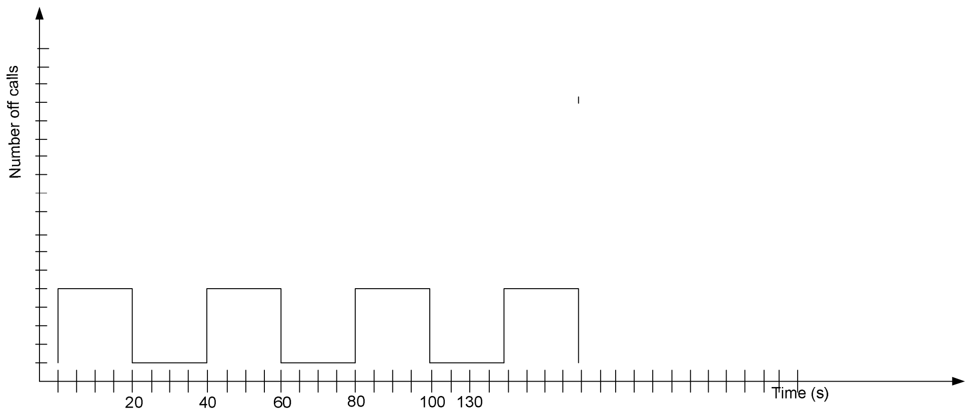


Figure 21: Example of saw tooth call profile

7.3.3 Rolling Blast

Rolling Blast - a defined set of channels go off-hook at once, and the pattern is repeated for all assigned channels.

7.3.4 Ramp

Ramp - gradually increases connected calls to a specified number and then maintains those number of calls.

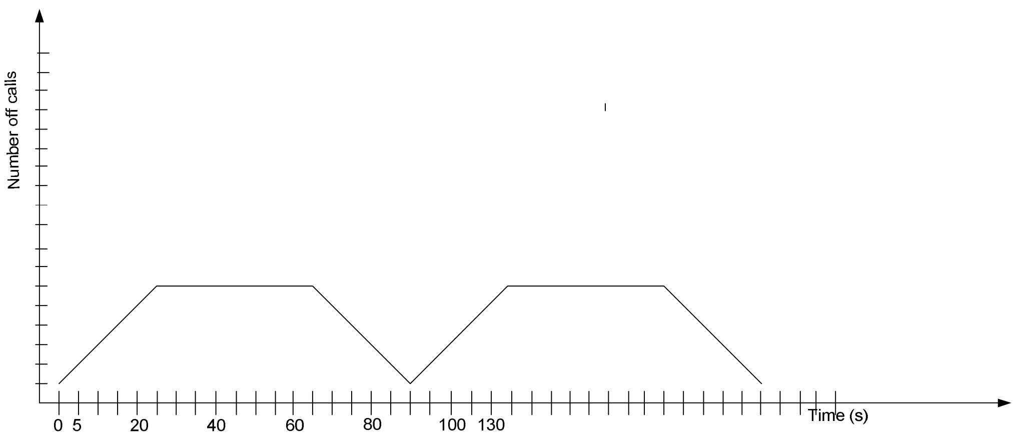


Figure 22

7.3.5 Steady Call Rate

Steady Call Rate - delivers a fixed, regulated call rate into the system under test.

7.3.6 Poisson Distribution

Poisson Distribution - defines call arrival rate by a statistical distribution.

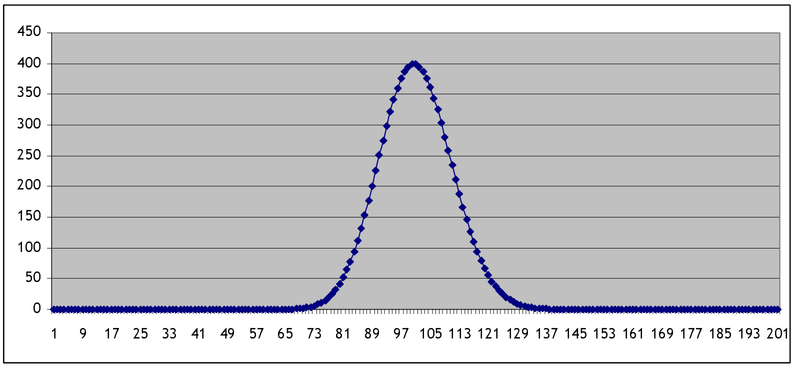

450
400
350
300
250
200
150
100
50
0
1 9 17 25 33 41 49 57 65 73 81 89 97 105 113 121 129 137 145 153 161 169 177 185 193 201

Figure 23

Annex A (informative): Calls flows

This annex defines the calls flows which should be implemented to simulate ISDN - non-ISDN environment.

Figure A.1 presents the call flow for the – IMS/PES environment calling side.

Figure A.2 presents the call flow for the – IMS/PES environment called side.

Figure A.3 presents the call flow for the ISDN environment for voice calls calling side - overlap.

Figure A.4 presents the call flow for the ISDN environment for voice calls calling side - enblock.

Figure A.5 presents the call flow for the ISDN environment for voice calls called side.

Figure A.6 presents the call flow for the ISDN environment for data calls calling side.

Figure A.7 presents the call flow for the ISDN environment for data calls called side.

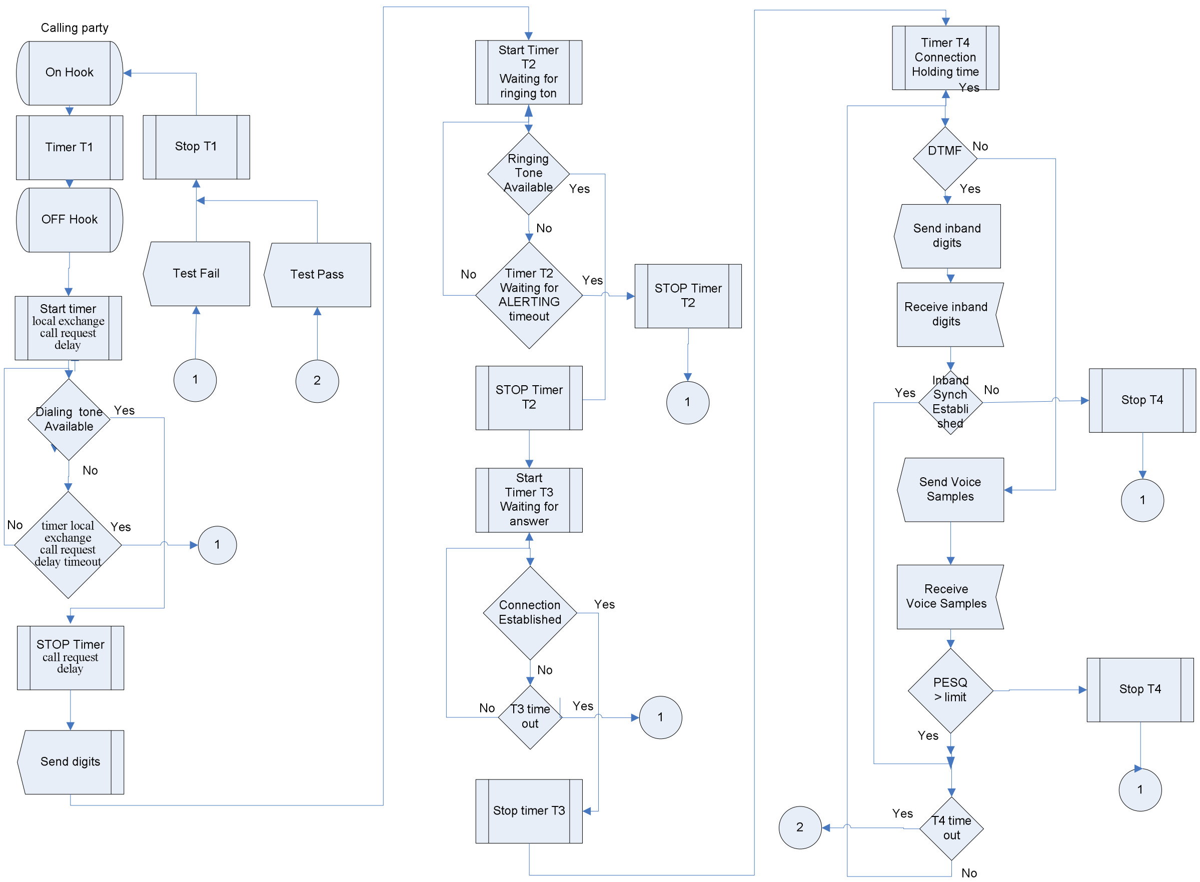


Figure A.1

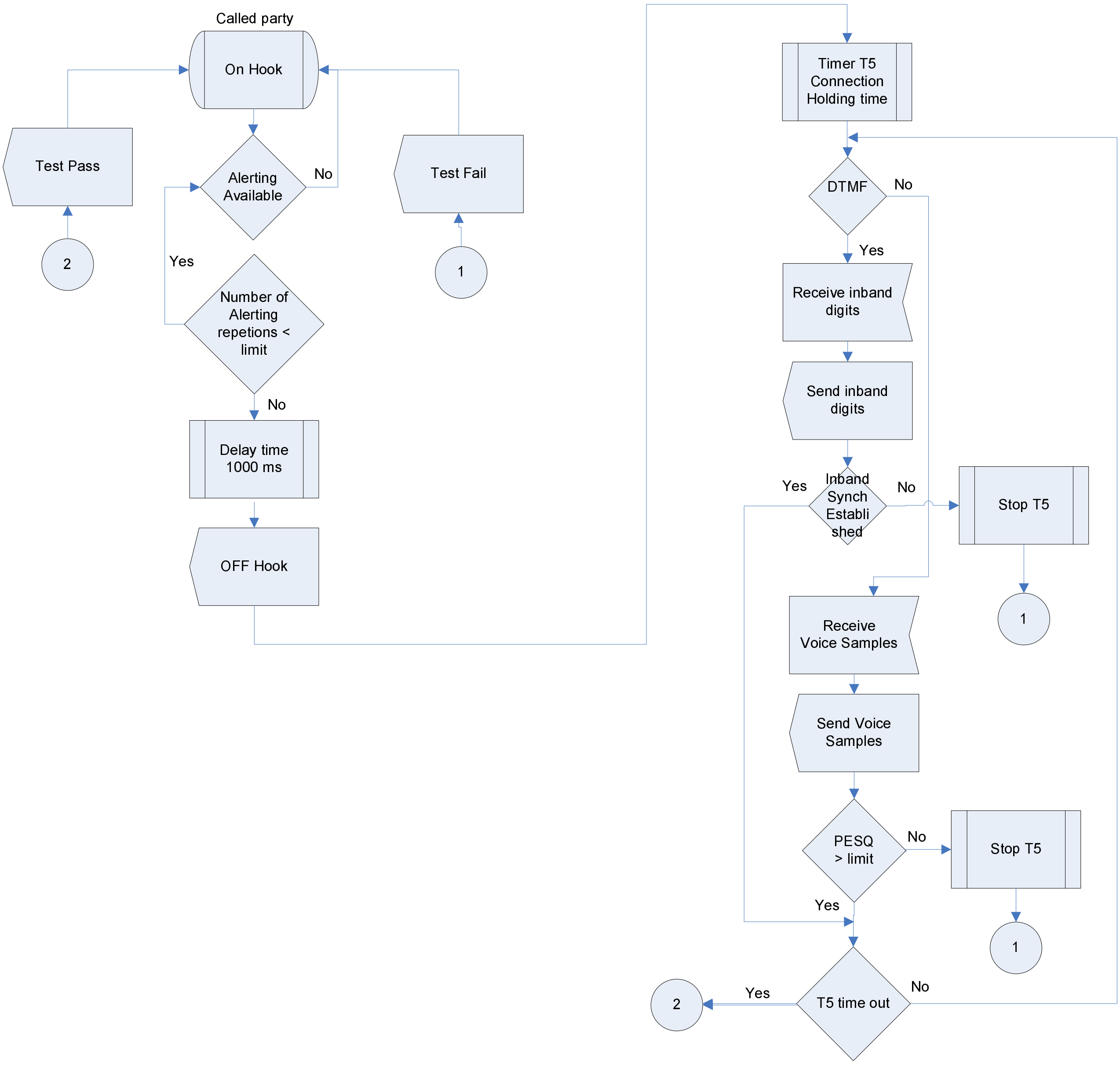
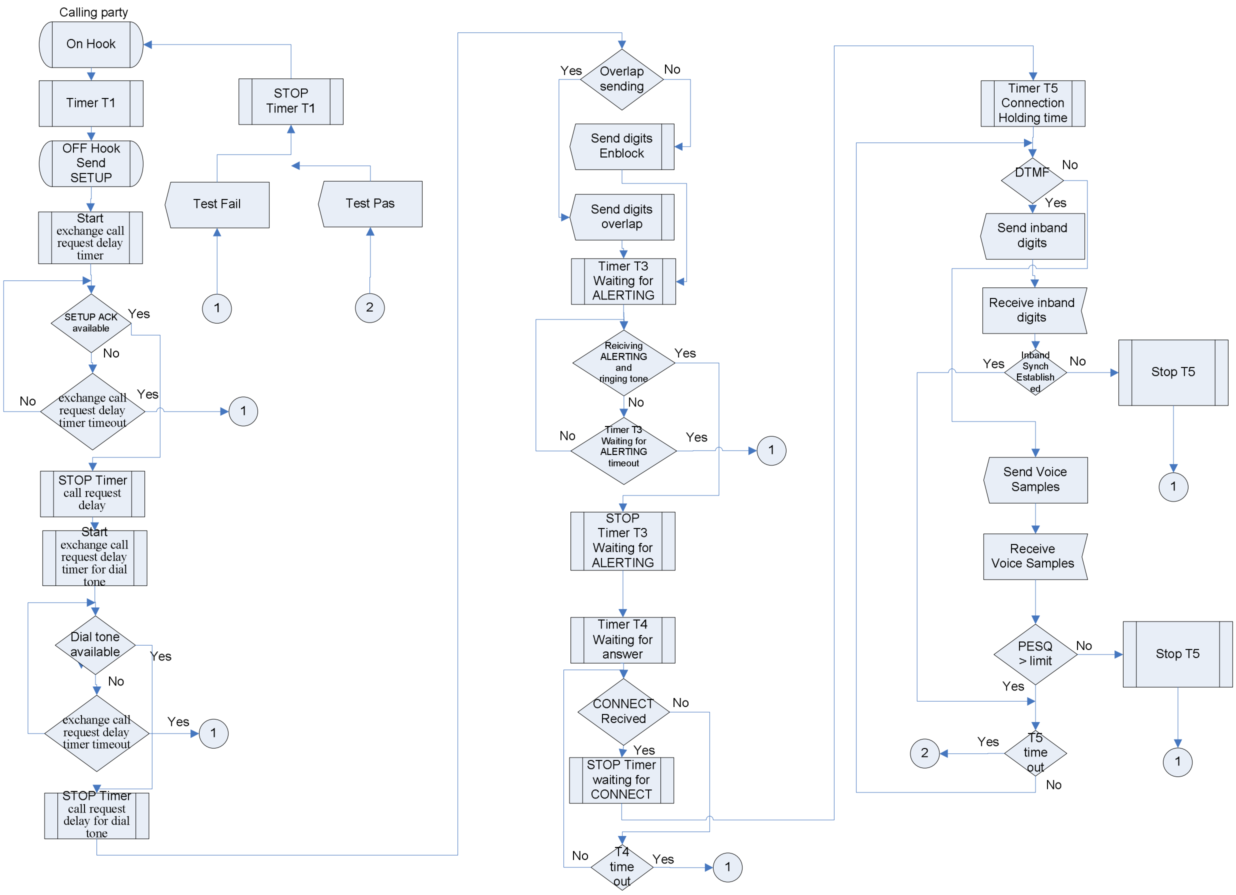
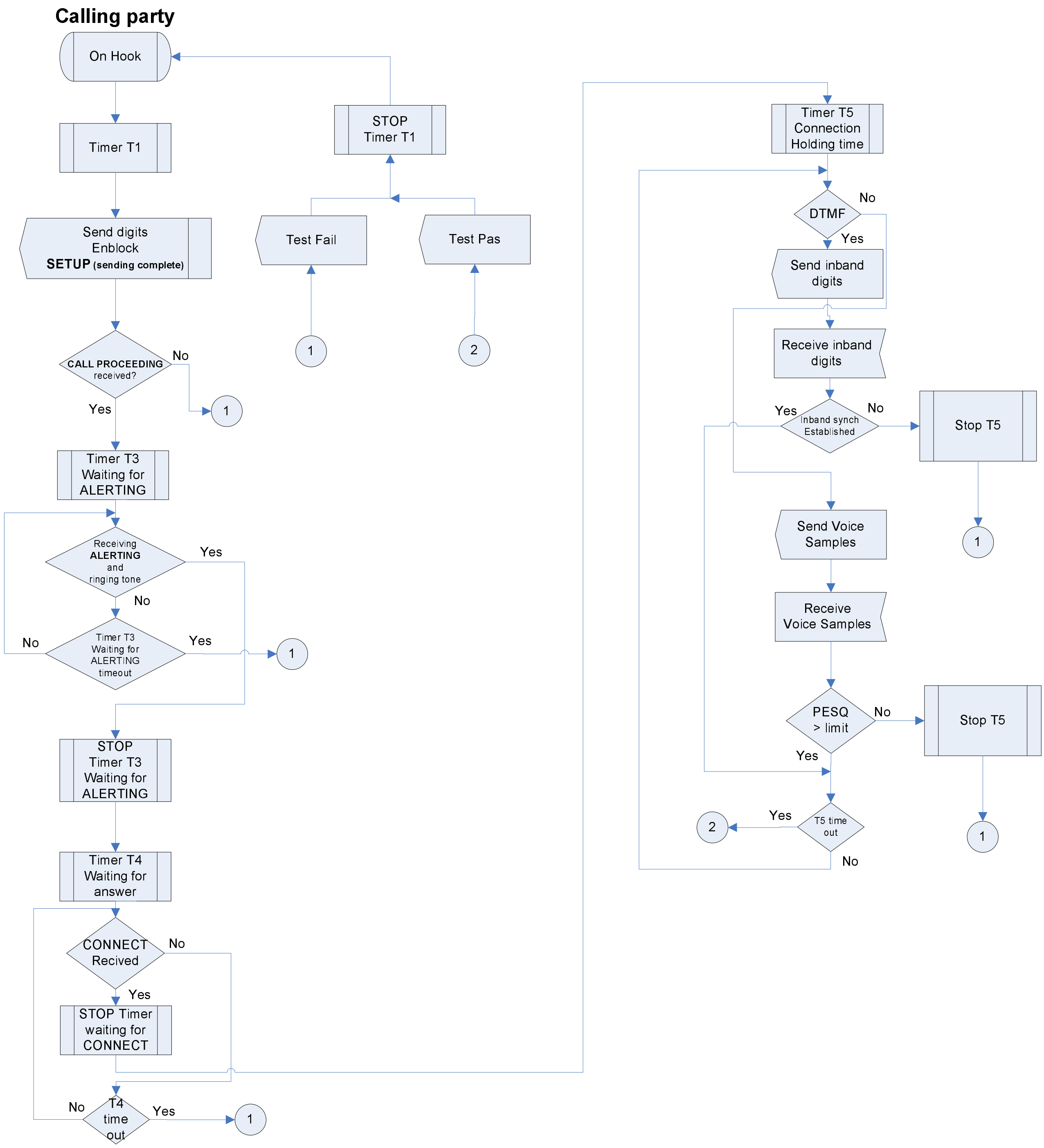

Called party
On Hook
Test Pass
Alerting Available
No
Test Fail
2
1
Yes
Number of Alerting repetions < limit
No
Delay time 1000 ms
OFF Hook
Timer T5 Connection Holding time
DTMF
No
Yes
Receive inband digits
Send inband digits
Yes
Inband Synch Established
No
Stop T5
1
Receive Voice Samples
Send Voice Samples
PESQ > limit
No
Stop T5
Yes
1
2
Yes
T5 time out
No

Figure A.2



Calling party
On Hook
Timer T1
OFF Hook Send SETUP
Start exchange call request delay timer
SETUP ACK available
exchange call request delay timer timeout
STOP Timer call request delay
Start exchange call request delay timer for dial tone
Dial tone available
exchange call request delay timer timeout
STOP Timer call request delay for dial tone
STOP Timer T1
Test Fail
Test Pas
Overlap sending
Send digits Enblock
Send digits overlap
Timer T3 Waiting for ALERTING
Reiciving ALERTING and ringing tone
Timer T3 Waiting for ALERTING timeout
STOP Timer T3 Waiting for ALERTING
Timer T4 Waiting for answer
CONNECT Recived
STOP Timer waiting for CONNECT
T4 time out
Timer T5 Connection Holding time
DTMF
Send inband digits
Receive inband digits
Inband Synch Establish ed
Stop T5
Send Voice Samples
Receive Voice Samples
PESQ > limit
Stop T5
T5 time out
Yes
No

Figure A.3



Calling party
On Hook
Timer T1
Send digits
Enblock
SETUP (sending complete)
CALL PROCEEDING
received?
No
Yes
1
Timer T3
Waiting for
ALERTING
Receiving
ALERTING
and
ringing tone
Yes
No
Timer T3
Waiting for
ALERTING
timeout
No
Yes
1
STOP
Timer T3
Waiting for
ALERTING
Timer T4
Waiting for
answer
CONNECT
Recived
No
Yes
STOP Timer
waiting for
CONNECT
T4
time
out
No
Yes
1
STOP
Timer T1
Test Fail
Test Pas
1
2
Timer T5
Connection
Holding time
DTMF
No
Yes
Send inband
digits
Receive inband
digits
Inband synch
Established
Yes
No
Stop T5
1
Send Voice
Samples
Receive
Voice Samples
PESQ
> limit
No
Yes
Stop T5
1
T5 time
out
Yes
No
2

Figure A.4

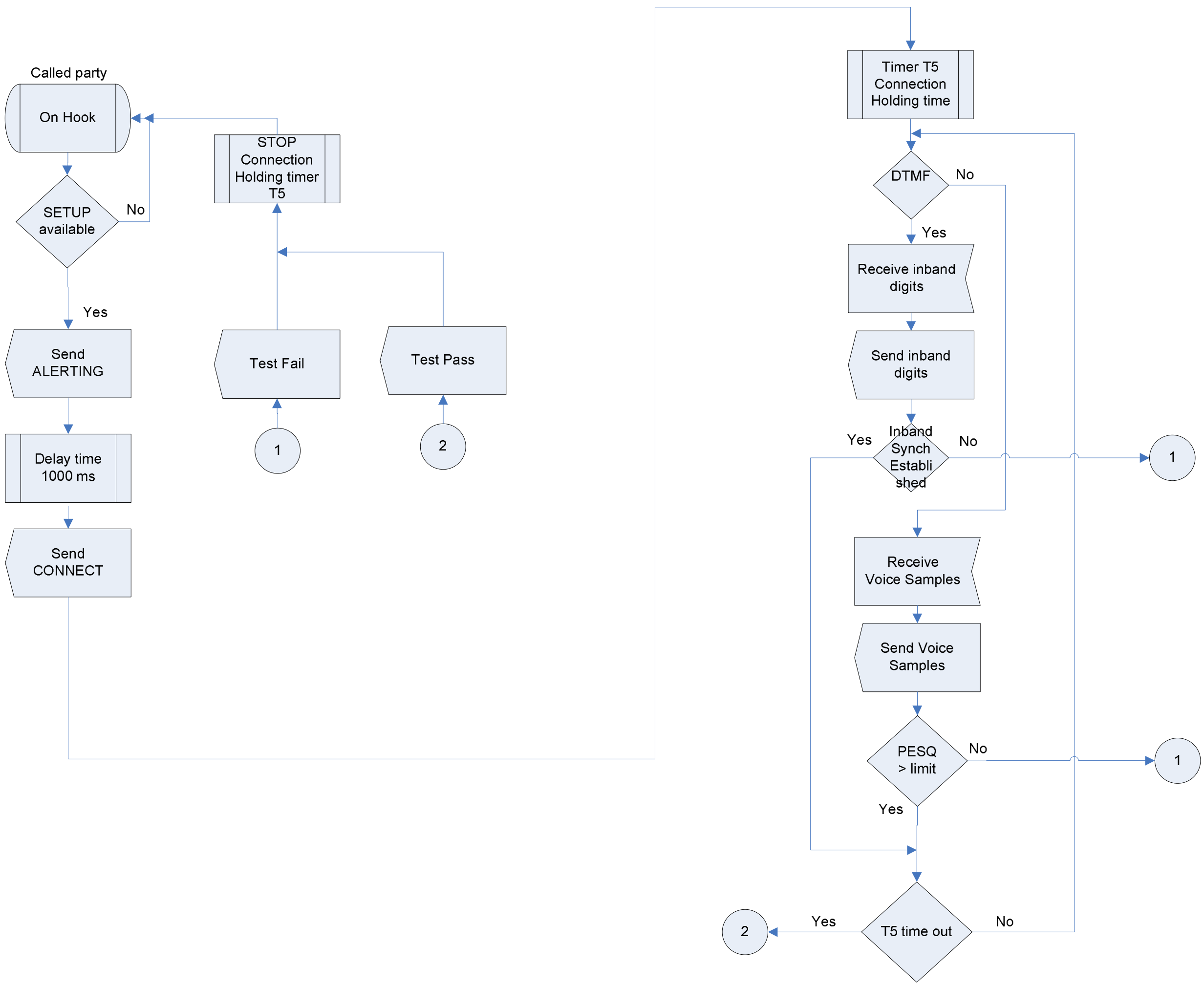

Called party
On Hook
SETUP available
No
Yes
Send ALERTING
Delay time 1000 ms
Send CONNECT
STOP Connection Holding timer T5
Test Fail
Test Pass
1
2
Timer T5 Connection Holding time
DTMF
No
Yes
Receive inband digits
Send inband digits
Inband Synch Established
Yes
No
1
Receive Voice Samples
Send Voice Samples
PESQ > limit
No
1
Yes
2
Yes
T5 time out
No

Figure A.5

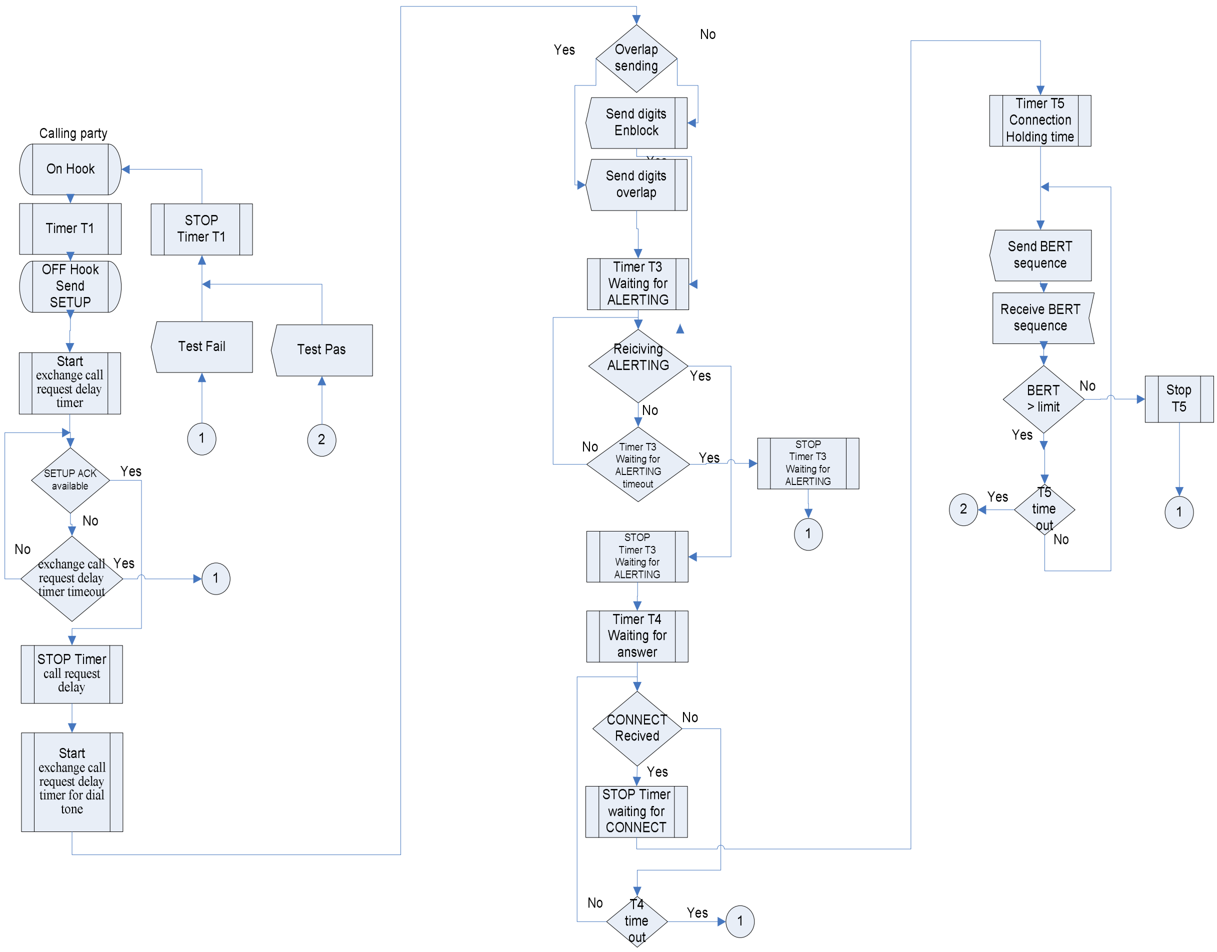

Calling party
On Hook
Timer T1
OFF Hook Send SETUP
Start exchange call request delay timer
SETUP ACK available
Yes
No
exchange call request delay timer timeout
No
Yes
1
STOP Timer call request delay
Start exchange call request delay timer for dial tone
STOP Timer T1
Test Fail
Test Pas
1
2
Overlap sending
Yes
No
Send digits Enblock
Send digits overlap
Timer T3 Waiting for ALERTING
Reiciving ALERTING
Yes
No
Timer T3 Waiting for ALERTING timeout
No
Yes
STOP Timer T3 Waiting for ALERTING
1
STOP Timer T3 Waiting for ALERTING
Timer T4 Waiting for answer
CONNECT Recived
No
Yes
STOP Timer waiting for CONNECT
T4 time out
No
Yes
1
Timer T5 Connection Holding time
Send BERT sequence
Receive BERT sequence
BERT > limit
No
Yes
Stop T5
1
T5 time out
Yes
No
2

Figure A.6

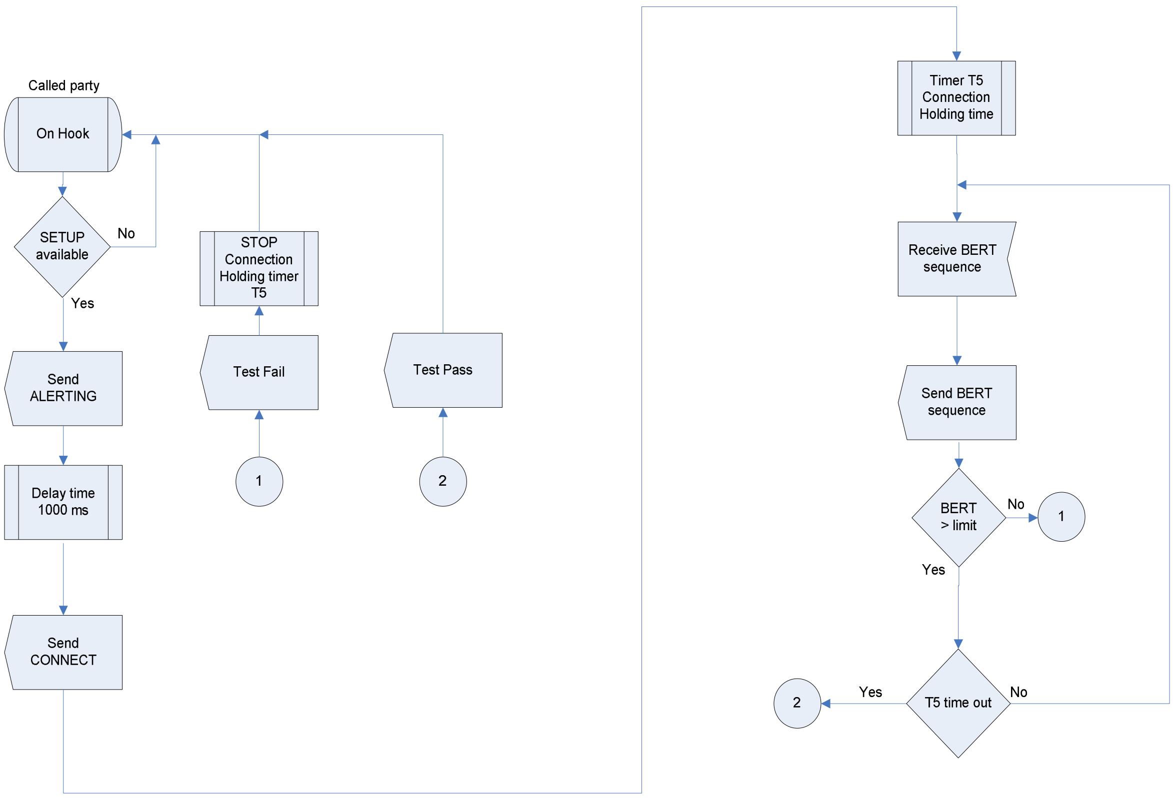

Called party
On Hook
SETUP available
No
Yes
Send ALERTING
Delay time 1000 ms
Send CONNECT
STOP Connection Holding timer T5
Test Fail
1
Test Pass
2
Timer T5 Connection Holding time
Receive BERT sequence
Send BERT sequence
BERT > limit
No
1
Yes
T5 time out
Yes
2
No

Figure A.7

Annex B (informative): Load profiles examples

This annex defines the load profiles to simulate ISDN - non-ISDN environments.

Figure B.1: the load simulates 2,0 CAPS, call duration 100 s, number of simulated users 200. The number of calls increases each 500 ms. After the call duration of 100 s the calls will be released. The call setup phase is marked orange, the call release phase blue.

Figure B.2: the load simulates 2,66 CAPS, call duration 15 s, number of simulated users 30. The number of calls increases each 500 ms. After a call duration of 15 s the calls will be released. In the time interval of 5 s are tested simultaneous ISDN call setups using five channels. In order to simulate a load of 2,0 CAPS, the increase of number of calls is changed to 1,5 per second.

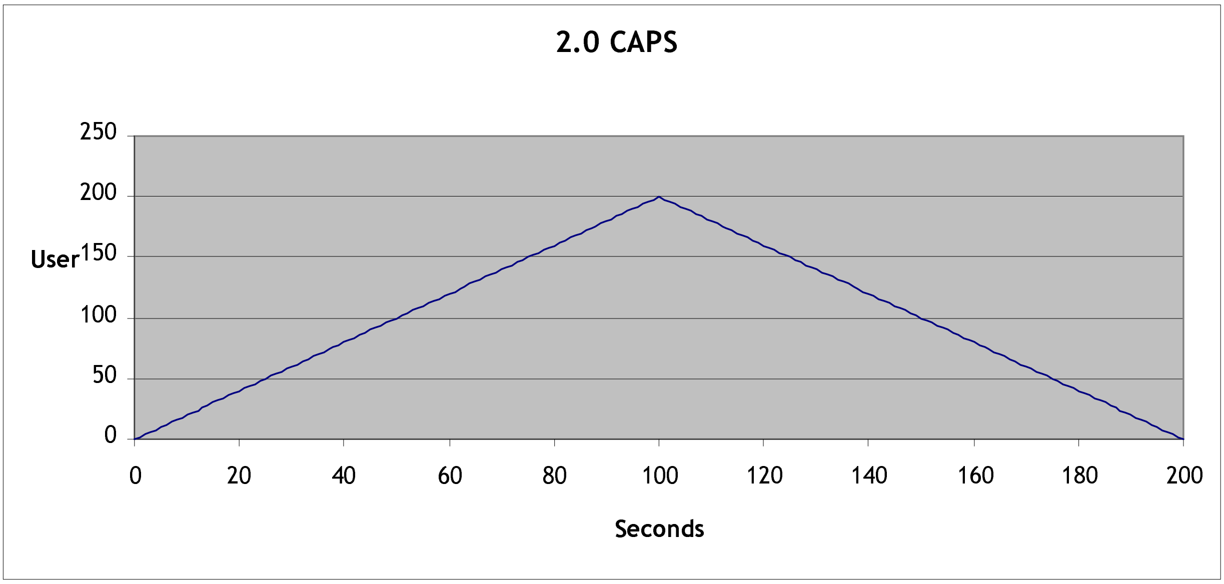


Figure B.1

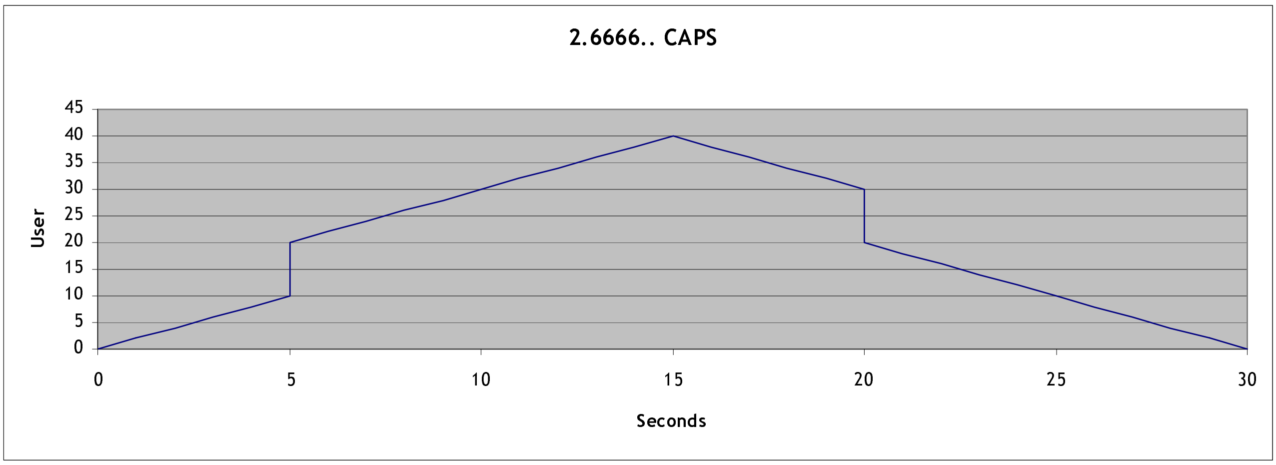


Figure B.2

Annex C (informative): Examples for Test Reports

C.1 Example of a Call Detail report

CALL DETAIL REPORT

Test Name: Basic Call

Start Time:

Stop Time:

Date	Time	Call ID	Server	Chan	Status	Called Number	Len	Lat ms	T1	T2	T3	T4

AVERAGE									
Date	Time	Calls Successful	Calls Failed:	Call Length	Latency ms	T1	T2	T3	T4

C.2 Example of a call summary report

CALL SUMMARY REPORT

Test Name: Basic Call

Start Time:

Stop Time:

Server	Channel	Attempts	Successful	Failure	Call Length (s)	Connect Latency (ms)

C.3 Example of a voice summary report

Test Name: VQ TEST

Start Time:

Stop Time:

SPEECH LATENCY REPORT

Server	Channel	Number of Tests	Average Speech Latency

Total Number of Tests: **Total Average Speech Latency:**

DTMF REPORT

Server	Channel	Number of Failures

Total Number of Tests:

PESQ/POLQA REPORT

Server	Channel	Number of Tests	Average PESQ/POLQA Score	Average Offset
Total Averages:				

C.4 Example of a voice quality detail report

Test Name: VQ TEST

Packetsphere Test:

Start Time:

Stop Time:

SPEECH LATENCY REPORT

TimeStamp	Call ID	Server	Channel	Speech Lat

Number of Speech Latency Tests:

Average: **(ms)**

Minimum: **(ms)**

Maximum: **(ms)**

DTMF REPORT

TimeStamp	Call ID	Server	Channel	Expected Digits	Recieved Digits

Number of DTMF Test Failures:

PESQ REPORT

TimeStamp	Call ID	Server	Channel	PESQ/POLQA Value	Offset time	Prompt Name

	Value	Offset time
Total Average:		
Minimum		
Maximum		

Number of PESQ/POLQA Tests:

PESQ/POLQA Score Above Threshold:

Annex D (informative): Bibliography

ETSI TS 123 228: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); IP Multimedia Subsystem (IMS); Stage 2 (3GPP TS 23.228 Release 6)".

ETSI TS 124 247: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Messaging service using the IP Multimedia (IM) Core Network (CN) subsystem; Stage 3 (3GPP TS 24.247)".

ETSI ES 282 002 (V1.1.1): "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS/PES/ISDN Emulation Sub-system (PES); Functional architecture".

ETSI TS 101 563 (V1.2.1): "Speech and multimedia Transmission Quality (STQ); IMS/PES exchange performance requirements".

Recommendation ITU-T P.862: "Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs".

Recommendation ITU-T P.56: "Objective measurement of active speech level".

IEEE 802.16: "A Technical Overview of the WirelessMAN™Air Interface for Broadband Wireless Access".

IEEE 802.1, 802.1Q-2011: "IEEE Standard for Local and metropolitan area networks--Media Access Control (MAC) Bridges and Virtual Bridged Local Area Networks".

IEEE 802.3: "Ethernet Working Group".

History

Document history		
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