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IMS Network Testing (INT); IMS/NGN Performance Benchmark; Part 2: Subsystem Configurations and Benchmarks Reference

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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";
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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.

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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.



System Under Test

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).



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.

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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					
Options	a) resource reservation	is on both sides				
	b) no resource reservation	tion on terminating side				
1	c) no resource reservation	tion on either side				
Message flow a)	Successful call - resour	ce resevation on both sides				
SIP (Test Systen	n 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 rout	ine			
Message flow: b), c)					
SIP (Test Systen	n A)	Core Network	_	SIP (Test System B)		
		INVITE (SDP1)	→			
	÷	100 Trying				
	÷	180 Ringing				
	÷	200 OK INVITE				
		ACK	→			
	_	BYE	→			
	+	200 OK BYE				
		Apply post test rout	ine			

		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 reservat	tion on terminating side				
1	c) no resource reservat	tion on either side				
Message flow a)	Successful call - resour	ce resevation on both sides				
SIP (Test Systen	n 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 rout	ine			
Message flow: b), c)					
SIP (Test Systen	n A)	Core Network		SIP (Test System B)		
		INVITE (SDP1)	→			
	+	100 Trying				
	+	180 Ringing				
	+	200 OK INVITE				
		ACK	→			
	+	BYE				
		200 OK BYE	→			
		Apply post test rout	ine			

	Μ	IMTEL to MMTEL Us	e case 2		
Scenario 3	Basic call with Fax with 33.6 kBit/s (Super G3 Fax)				
	This scenario represents the	case when in the act	, ive 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 th	he active call state the	data transfe	er is performed correctly.	
Options	a) resource reservation is or	n both sides			
	b) no resource reservation of	on terminating side			
	c) no resource reservation of	on either side			
SIP Parameter	INVITE: SDP				
	m=audio <por< td=""><td>t> RTP/AVP 8/0</td><td></td><td></td></por<>	t> RTP/AVP 8/0			
	180/200 OK INVITE: SDP				
	m=audio <por< td=""><td>t> RTP/AVP 8</td><td></td><td></td></por<>	t> RTP/AVP 8			
Message flow					
SIP (Test Syste	em A)	Core Network		SIP (Test System B)	
	-	INVITE	→		
	+	100 Trying			
	+	180 Ringing			
	+	200 OK IŇVIŤE			
		ACK	→		
		Communication			
		BYE	<mark>→</mark>		
	÷	200 OK BYE			

	N	IMTEL to MMTEL Us	e case 2	
Scenario 4	Basic call with Fax with 14,4 kBit/s;			
	This scenario represents the	case when in the act	ive call state	the Fax transfer on the media is
	performed correctly and the	echo cancellers in the	GW are act	ivated The call is released from the
	calling user. Ensure that in th	ne active call state the	e data transfe	er is performed correctly.
Options	a) a resource reservation is	on both sides		
	b) no resource reservation of	on terminating side		
	c) no resource reservation of	on either side		
SIP Parameter	INVITE: SDP			
	m=audio <por< td=""><td>t> RTP/AVP 8/0</td><td></td><td></td></por<>	t> RTP/AVP 8/0		
	180/200 OK INVITE: SDP			
	m=audio <por< td=""><td>t> RTP/AVP 8</td><td></td><td></td></por<>	t> RTP/AVP 8		
Message flow				
SIP (Test Syste	m 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	cenario 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></dynamic-pt></port>				
	a=rtpmap <dynamic-pt> PCMA/8000</dynamic-pt>				
	a=gpmd; vbd=yes				
	180/200 OK INVITE: SDP				
	m=audio <port> RTP/AVP <dynamic-pt></dynamic-pt></port>				
	a=rtpmap <dynamic-pt> PCMA/8000</dynamic-pt>				
	a=gpmd; vbd=yes				
Message flow					
SIP (Test Syste	m 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					n-line codec
This scenario represents the case when in the active call state the Fax transfer on the media and B-					he Fax transfer on the media and B-
channels is performed correctly and the echo cancellers in the GW are not activated. The call is					GW are not activated. The call is
	released from t	he called	user. Ensure that in the act	tive call state	e the data transfer is performed correctly
	(e.g. testing Qo	S param	eters).		
Options	 a) resource res 	servation	is on both sides		
	b) no resource	reservat	ion on terminating side		
	c) no resource	reservat	ion on either side		
SIP Parameter		INVITE:	SDP		
		m	i=image <port> udptl t38</port>		
		180/200	OK INVITE: SDP		
		r	i=image <port> udptl t38</port>		
Message flow					
SIP (Test Syste	m A)		Core Network		SIP (Test System B)
			INVITE (SDP1)	→	
← 180 Ringing					
		÷	200 OK INVITE (SDP2)	_	
			ACK	→	
			Apply post test rout	tine	

	MMTEL to MMTEL Use case 2				
Scenario 7	cenario 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 Syste	m A) (Core Network INVITE ← 486 Busy Here ACK	SIP (Test System B) →		

		MMTEL to MMTEL Use case 2	2				
Scenario 8	CFU Ensure that when user A state the voice transfer the calling user.	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 flov SIP (Test Sy	w stem A) C C C C	Core Network 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	 + + + + + 	SIP (Test System B)			

			MMTEL to MMTEL Use case t	2				
0				2				
Scenario 9	CFB							
	Ensure th	nat when user	A calls user B which is user determ	ined us	er 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							
Outions			ation is an both sides					
Options		source reserv						
	b) no	o resource res	ervation on terminating side					
	c) no	o resource res	ervation on either side					
Message flow								
SIP (Test Syste	em A)		Core Network		SIP (Test System B)			
				د				
			CEP is performed					
		←	INVITE(Call-ID B-C)	_				
			180 Ringing(Call-ID C-B)	\rightarrow				
		←	180 Ringing(Call-ID B-A)					
			200 OK INVITE(Call-ID C-B)	→				
		4	ACK(Call-ID B-C)	-				
		Ļ						
		T						
			ACK(Call-ID A-B)					
			Communication					
			Apply post test routine					

	MI	MTEL 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 flov SIP (Test Sys	v stem A) ← 1 ← 1 ← 1 200 ← 200	Core Network INVITE(Call-ID A-B) 80 Ringing(Call-ID B-A) CFNR is performed INVITE(Call-ID B-C) 80 Ringing(Call-ID C-B) 80 Ringing(Call-ID B-A) 0 OK INVITE(Call-ID C-B) ACK(Call-ID B-C) 0 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) Communication Apply post test routine	SIP (Test System B) → → →				

MMTEL to MMTEL Use case 2							
Scenario 11	CCBS						
	User A is located in netwo	ork A and user B is located in n	etwork B. User A has successfu	ully invoked a			
	CCBS request.Ensure that	at the recall from user A to use	B is successful.				
	The call is released from	the calling user.					
Options	a) resource reservation is on both sides						
	b) no resource reservation	on on terminating side					
	c) no resource reservation	on on either side					
Message flow							
SIP (Test Syst	em A)	Core Network	SIP (Test System B	3)			
	A C	CBS request was already su	cessful				
	+	NOTIFY					
		200 OK NOTIFY	\rightarrow				
	-	INVITE	7				
	~	180 Ringing					
	L	NOTIEV					
	•		-				
	←	200 OK INVITE					
	•	ACK	→				
	Apply post test routine						

		MMTEL to MMTEL Use cas	ə 2					
Scenario 12	CCNR							
	User A is located in netwo	ork A and user B is located in n	etwork 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 on terminating side						
	c) no resource reservation	on on either side						
Message flow	1							
SIP (Test Sys	tem A)	Core Network	SIP (Test System B)					
	A C	CNR request was already su	cessful					
	+	NOTIFY						
		200 OK NOTIFY	→					
	-	INVITE	→					
	+	180 Ringing						
	4	NOTIEY						
	×		→					
		200 011101111	-					
	+	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

			ISDN to MMTel Use	case 3			
Scenario 1	Basic call with	BC= ITC value - enblock sending					
	This scenario	represents the	case when the call e	stablishmen	t using en-bloc sending is performed		
	correctly. The	call is released	from the calling use	r. Ensure tha	at in the active call state the voice transfer		
	is performed c	orrectly (e.g. te	sting QoS paramete	rs).			
Options	a) resource i	reservation is o	n both sides				
	b) no resour	ce reservation of	on terminating side				
	c) no resour	ce reservation	on either side				
Message flow							
SIP (Te	est System A)		Core Network	_	SIP (Test System B)		
			INVITE	→			
		+	100 Trying				
		+	180 Ringing				
		÷	200 OK INVITE				
			ACK	→			
			Communication				
			<mark>BYE</mark>	→			
		+	200 OK BYE				
SIP Parameter		INVITE:					
PSTN XML Be	earerCapability	Content-T	ype: application/vnd.	etsi.pstn+xn	าไ		
element in the	INVITE	Content-D	isposition: signal;har	ndling=optio	nal		
		xml version</th <th>i="1.0" encoding="ut</th> <th>f-8"?></th> <th></th>	i="1.0" encoding="ut	f-8"?>			
		PSTN					
		BearerCa	pability				
		BCocte	et3				
		Co	dingStandard>00<				
		Info	ormationTransferCat	ability> <mark>ITC</mark>	value<		
		< BCocte	et4				
		Tra	ansferMode>00<				
		Info	ormationTransferRat	e>10000<			
		BCocte	et5				
		Lay	ver1Identification>01	<			
		Us	erInfoLayer1Protoco	l>00011<			

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				IS	DN to MMTel Us	e case 3		
Scenario 2	Basi	c call with	BC= ITC	value	- enblock sending			
	This	scenario i	represents	the ca	se when the call	establishme	nt using en-bloc sending is performed	
	correctly. The call is released from the called user. Ensure that in the active call state the voice transf							
	is pe	rformed c	orrectly (e.	.a. test	ing QoS paramet	ers).		
Options	a)	resource	reservatio	on is or	hoth sides	,		
optiono	b) no resou			ation c	n terminating side	ć		
	()	no resou		ation c	n either side			
Message flow	<u> </u>	10 16300						
SIP (Test Sv	stem Δ)				Core Network		SIP (Test System B)	
	stem Aj					→		
			4	_				
					100 Trying			
					200 OK INVITE	•		
					ACK	7		
				-	Communication			
					BYE			
					200 OK BYE	→		
SIP Paramete	er		INVITE:					
PSTN XML B	BearerCa	pability	Conte	ent-Typ	e: application/vno	d.etsi.pstn+x	ml	
element in th	ne INVIT	E	Content-Disposition: signal;handling=optional					
						•		
			xml version="1.0" encoding="utf-8"?					
			PSTN		5			
			BearerCapability					
			BCoctet3					
				Infor	mationTransferCa	hability> <mark>ITC</mark>		
			BCoctet/					
			TransforMode>00<					
			Industriation Transfor Potos 10000 -					
			BLOCIEIS					
				Laye	Indentification>0	1<		
			User	inioLayer1Protoc	>11000/11<			

Table 1: PSTN XML	BearerCapability
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ITC_value	BC Information transfer capability	XML InformationTransferCabability		
ITC_VA_1	Speech	'00000'		
ITC_VA_2	3,1 kHz audio	<mark>'10000'</mark>		
ITC_VA_3	unrestricted digital information	<mark>'01000'</mark>		

		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 ca	Iling user. The call is released from the	e calling use	r. 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 res	ervation on either side						
Message flow N	Aultiple INVITE meth	od is used						
SIP (Test Syste	m A)	Core Network		SIP (Test System B)				
		INVITE(CSq 1)	→					
			``					
	L	INVITE(CSQ 2)	7					
	T		_					
		ACK						
			→					
	+	484 Address Incomplete(CSg 2)	-					
	_	ACK	→					
		INVITE(CSq 4)	→					
	+	484 Address Incomplete(CSq 3)						
		ACK	→					
	+	180 Ringing(CSq 4)						
		Apply post test routine						
Message flow C	Overlap sending, the	in-Dialogue method is used						
SIP (Test Syste	m A)			SIP (Test System B)				
		INVITE(CSq 1) 1	→					
	7	484 Address Incomplete(USq 1)	-					
		ACK	7					
			د					
	4	183 Session Progress(CSg 2)						
	Ľ	PRACK	→					
	+	200 OK PRACK	-					
	-							
		INFO	→					
	+	200 OK INFO						
		INFO	→					
	+	200 OK INFO						
	+	180 Ringing(CSq 2)						
		Apply post test routine						

			ISDN to MMTel Use c	ase 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 b) no resou c) no resou	source reservation is on both sides resource reservation on terminating side resource reservation on either side					
SIP Parameter		INVITE: 180/200	SDP m=audio <port> RTP/ OK INVITE: SDP m=audio <port> RTP/</port></port>	'AVP 8/0 'AVP 8			
Message flow SIP (Test Syste	m A)	4	Core Network INVITE (SDP1) 180 Ringing 200 OK INVITE (SDP2) ACK	→ →	SIP (Test System B)		

ACK Apply post test routine

				ISDN to MMTel Use	2003	
0 · 5	- ·		20 04			
Scenario 5	Basi	c call with I	3C = 3,1	KHZ audio - Fax with 14,4 P	(DIT/S	
	This	scenario re	epresent	s the case when in the activ	e call state	(N10) the Fax transfer on the media and
	B-ch	annels is p	erforme	d correctly and the echo car	ncellers in th	ne GW are activated. The call is released
	from	the calling	user. Er	nsure that in the active call	state the dat	ta transfer is performed correctly (e.g.
	testir	ng QoS pa	rameters	5).		
Options	a)	resource	reservati	ion is on both sides		
	b)	no resour	ce reser	vation on terminating side		
	c)	no resour	ce reser	vation on either side		
SIP Parameter			INVITE:	SDP		
				m=audio <port> RTF</port>	/AVP 8/0	
			180/200	OK INVITE: SDP		
				m=audio <port> RTF</port>	/AVP 8	
Message flow						
SIP (Test Syste	mA)			Core Network		SIP (Test System B)
	-			INVITE (SDP1)	→	
			←	180 Ringing		
			←	200 OK INVITE (SDP2)		
				ACK	→	
				Apply post test rou	ıtine	

			ISDN to MMTel Use of	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 reb) no resourcec) no resource	e reservation is on both sides urce reservation on terminating side				
SIP Parameter	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=rtpmap <dynamic-pt> PCMA/8000 a=rtpmap <dynamic-pt> PCMA/8000 a=rtpmap <dynamic-pt> PCMA/8000</dynamic-pt></dynamic-pt></dynamic-pt></dynamic-pt></dynamic-pt></port></dynamic-pt></dynamic-pt></port>					
Message flow SIP (Test	System A)	4	Core Network INVITE (SDP1) 180 Ringing 200 OK INVITE (SDP2) ACK Apply post test rou	→ → utine	SIP (Test System B)	

Apply post test routine

[
	ISDN to MMTELUSE 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 OoS parameters)					
Ontions	a) resource reservation is on both sides					
Options	a) resource reservation on terminating side					
	b) no resource reservation on terminating side					
-	c) no resource reservation on either side					
SIP Parameter	ilP Parameter INVITE: SDP					
	m=image <port> udptl t38</port>					
	180/200 OK INVITE: SDP					
	m=image <port> udptl t38</port>					
Message flow						
SIP (Test	System A) Core Network SIP (Test System B)					
← 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	r INVITE: SDP				
	m=audio <port> RTP/AVP 8/0</port>				
	180/200 OK INVITE: SDP				
	m=audio <port> RTP/AVP 8</port>				
Message flow					
SIP (Test S	System A) Core Network SIP (Test System B)				
, ,	INVITE (SDP1) →				
	← 180 Ringing				
	← 200 OK INVITĚ (ŠDP2)				
	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				
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</port></port>				
Message flow SIP (Test	v st System A) Core Network SIP (Test Sy INVITE (SDP1) → € 180 Ringing € 200 OK INVITE (SDP2) ACK → Apply post test routine	ystem B)			

	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 INVITE 486 Busy Here ACK	→ →	SIP (Test System B)	

		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 Sys	v stem A) Col CF C C C Col	re Network INVITE(Call-ID A-B) U 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) mmunication	SIP (T → → →	est System B)
Apply post te	est 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 Syst	tem A) Cor CFI C C C Cor	e Network INVITE(Call-ID A-B) 3 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) nmunication	SIP (Test S → → →	ystem B)
Apply post tes	st routine			

		ISDN to MMTel Use case	3		
Scenario 13	CFNR Ensure that when that in the active call is released fr	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 reserva no resource rese no resource rese	resource reservation is on both sides no resource reservation on terminating side no resource reservation on either side			
Message flow SIP (Test Sys	tem A) C	Core Network INVITE(Call-ID A-B) 180 Ringing(Call-ID B-A) FNR 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	SIP (Test Sys → → →	item B)	
Apply post te	st routine				

		ISDN to MMTel Us	e case 3			
Scenario 14	CCBS					
	User A is locate	d in network A and user B is loca	ted in network B. User A has successfully inv	/oked a		
	CCBS request.					
	Ensure that the	recall from user A to user B is su	ccessful. The call is released from the calling	j user.		
Options	resource reserv	ation is on both sides				
	no resource res	ervation on terminating side				
	no resource res	ervation on either side				
Message flow	1					
SIP (Test System A) Core Network SIP (Test System B)						
A CCBS requ	est was already s	uccessful				
	•	- NOTIFY				
		200 OK NOTIFY	\rightarrow			
			7			
	•	180 Ringing				
	T		د			
		200 OK NOTIF	7			
	4					
	•	ACK	→			
Apply post te	st routine		-			

		ISDN to MMTel U	Jse 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 Syste A CCNR reques	m A) Core st was already succe ←	e Network essful NOTIFY 200 OK NOTIFY	SIP (Test System B) →			
INVITE → € 180 Ringing						
← NOTIFY →						
Apply post test	← 200 OK INVITE ACK →					

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

		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 e	stablishmen	t is performed correctly. The call is	
	released from the called user	r. Ensure that in the a	ctive call sta	ate the voice transfer is performed	
	correctly (e.g. testing QoS pa	arameters).			
Options	a) resource reservation is o	n both sides			
	b) no resource reservation of	on terminating side			
	c) no resource reservation of	on either side			
Message flow					
SIP (Test Syste	em A)	Core Network		SIP (Test System B)	
		INVITE	→		
	÷	100 Trying			
	+	180 Ringing			
	+	200 OK INVITE			
		ACK	→		
		Communication			
		BYE	÷		
	<mark>→</mark>	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 o	correctly (e.g.	testing QoS parameters). The call is
	released fr	om the calling	g user.		
Options	a) resour	ce reservatio	n is on both sides		
	b) no res	ource reserva	ation on terminating side		
	c) no res	ource reserva	ation on either side		
SIP Parameter	Parameter INVITE: SDP				
			m=audio <port> RTF</port>	P/AVP 8/0	
		180/200	OK INVITE: SDP		
			m=audio <port> RTF</port>	P/AVP 8	
Message flow					
SIP (Test Syste	m A)		Core Network		SIP (Test System B)
			INVITE (SDP1)	→	
		+	180 Ringing		
		+	200 OK INVITE (SDP2)		
			ACK	→	
			Apply post test rou	utine	

				MMTel to ISDN Use	case 4	
Scenario 4	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					
Options	a) r b) r C) r	resource reservation is on both sides no resource reservation on terminating side no resource reservation on either side				
SIP Parameter		1	NVITE: 80/200	SDP m=audio <port> RTF OK INVITE: SDP m=audio <port> RTF</port></port>	P/AVP 8/0 P/AVP 8	
Message flow SIP (Test Syste	m A)		÷	Core Network INVITE (SDP1) 180 Ringing 200 OK INVITE (SDP2) ACK Apply post test ro	→ → utine	SIP (Test System B)

		MMTel to ISDN Use c	ase 4		
Connoria 5 Pasia call with PC-21 KHz audia. Eav with 14.4 khita with 1/152					
Dasic call with $DC = 3$, 1×12 and $0 - 7$ as with $1 + 4$, $4 \times 10^{\circ}$ s with 1×132				turn of a single strategies of	
I his 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 t	the active	e call state the data transfer	is performed correctly (e.	g. testing QoS	
parameters).					
a) resource	e reserva	ation is on both sides			
b) no reso	urce rese	ervation on terminating side			
c) no reso	urce rese	ervation on either side			
	INVITE:	SDP			
	m=audio <port> RTP/AVP 8 <dvnamic-pt></dvnamic-pt></port>				
	a=rtpmap <dvnamic-pt> PCMA/8000</dvnamic-pt>				
	a=gpmd: vbd=ves				
		a-gpma, voa-yee			
	180/200	OK INVITE: SDP			
	r	m=audio <port> RTP/AVP <</port>	dvnamic-PT>		
	6	=rtpmap <dvnamic-pt> PC</dvnamic-pt>	MA/8000		
		a=apmd: vbd=ves			
		gp			
m A)		Core Network	SIP (Test System B)	
		INVITE (SDP1)	→		
	4	180 Ringing	2		
	È				
	•	ACK	→		
		Apply post test rou	tine		
	Basic call with This scenario r correctly and th Ensure that in t parameters). a) resource b) no resou c) no resou c) no resou	Basic call with BC= 3,1 This scenario represent correctly and the echo of Ensure that in the active parameters). a) resource reserva b) no resource reserva c) no resource reserva c) no resource reserva no resource reserva no resource reserva no resource reserva a 180/200 r a a m A) €	MMTel to ISDN Use of Basic call with BC= 3,1 KHz audio - Fax with 14,4 k This scenario represents the case when in the activ correctly and the echo cancellers in the GW are act Ensure that in the active call state the data transfer parameters). a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side INVITE: SDP m=audio <port> RTP/AVP 8 a=rtpmap <dynamic-pt> PC a=gpmd; vbd=yes 180/200 OK INVITE: SDP m=audio <port> RTP/AVP < a=gpmd; vbd=yes 180/200 OK INVITE (SDP) fmachine core Network INVITE (SDP1) € 200 OK INVITE (SDP2) ACK Apply post test rou</port></dynamic-pt></port>	MMTel to ISDN Use case 4 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 correctly and the echo cancellers in the GW are activated. The call is release Ensure that in the active call state the data transfer is performed correctly (e.) parameters). a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side 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 m=audio <port> RTP/AVP a=rtpmap <dynamic-pt> PCMA/8000 a=gpmd; vbd=yes 180/200 OK INVITE: SDP m=audio <port> RTP/AVP meaudio <port> RTP/AVP meaudio <port> RTP/AVP a=rtpmap <dynamic-pt> PCMA/8000 a=gpmd; vbd=yes maintering Iso Ringing MA) Core Network SIP (Iso Ringing 200 OK INVITE (SDP2) ACK Apply post test routine </dynamic-pt></port></port></port></dynamic-pt></port></port></dynamic-pt></dynamic-pt></port>	

	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</port></port>				
Message flow SIP (Test Syste	em 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	ter INVITE: SDP					
	m=audio <port> RTP/AVP 8/0</port>					
	180/200 OK INVITE: SDP					
	m=audio <port> RTP/AVP 8</port>					
Message flow						
SIP (Test Syste	em 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</port>				
	180/200 OK INVITE: SDP				
	m=audio <port> RTP/AVP 8</port>				
Message flow					
SIP (Test Syste	em A) Core Network SIP (Test System B)				
	INVITE (SDP1) ->				
← 180 Ringing					
	← 200 OK INVITE (SDP2)				
	ACK 🗲				
	Apply post test routine				

			MMTel to ISDN U	se case 4	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	N						
SIP (Test Sys	stem 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 Sys	r tem A) ← ← ← ←	Core Network 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	SIP (Test System B) → → →		

		MMTel to ISDN Use case 4			
Scenario 11	CFB				
	Ensure that when user	A calls user B which is user determ	ined use	er busy (UDUB), the call is forwarded	
	to user C. Ensure that	in the active call state the voice trans	sfer is p	erformed correctly (e.g. testing QoS	
	parameters). The call is	s released from the calling user.			
Options	a) resource reservation	on is on both sides			
	b) no resource reserv	ation on terminating side			
	c) no resource reserv	ation on either side			
Message flow					
SIP (Test Syst	em 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 IS	SDN 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 flov SIP (Test Sys	tem A) Core N INVITE(C ← 180 Ringing CFNR is p ← INVITE(C 180 Ringing ← 180 Ringing 200 OK INVIT ← ACK(Ca ← 200 OK INVIT ACK(Ca Commu Apply po	etwork all-ID A-B) → (Call-ID B-A) berformed all-ID B-C) (Call-ID C-B) → (Call-ID B-A) E(Call-ID C-B) → II-ID B-C) E(Call-ID B-A) II-ID A-B) → nication st test routine	SIP (Test System B)			

		MMTel to ISDN Use case	4			
Scenario 13	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 th	at the recall from user A to use	er B is successful. The call is released from the			
	calling user.					
Options	a) resource reservation i	s on both sides				
	 b) no resource reservation 	on on terminating side				
	c) no resource reservation	on on either side				
Message flow	/					
SIP (Test System A) Core Network SIP (Test System B)						
	A CCBS	or CCNR request was alread	dy successful			
	+	NOTIFY				
		200 OK NOTIFY	→			
		INVITE	→			
	÷	180 Ringing				
		NOTIEN				
	T		د			
		200 OK NOTIFT	7			
	F	200 OK INVITE				
	•	ACK	→			
	Apply post test routine					
		MMTel to ISDN Use case	4			
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Scenario 14	CCNR User A is located in netw CCNR request. Ensure that the recall from The call is released from	vork A and user B is located in n om user A to user B is successfu n the calling user.	etwork B. User A has successfully invoked a I.			
Options	 a) resource reservation b) no resource reserva c) no resource reserva 	n is on both sides tion on terminating side tion on either side				
Message flow SIP (Test Sys	tem A) A CCBS C	Core Network S or CCNR request was alread NOTIFY 200 OK NOTIFY	SIP (Test System B) y successful ➔			
	÷	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 e	stablishmen	t is performed correctly. The call is	
	released from the called user	r. Ensure that in the a	ctive call sta	ate the voice transfer is performed	
	correctly (e.g. testing QoS pa	arameters).			
Options	a) resource reservation is c	on both sides			
	b) no resource reservation on terminating side				
	c) no resource reservation	on either side			
Message flow					
SIP (Test Syste	em A)	Core Network		SIP (Test System B)	
	-	INVITE	→		
	+	100 Trying			
	+	180 Ringing			
	÷	200 OK INVITE			
		ACK	→		
		Communication			
	<mark>←</mark>	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 Syst	tem A) ← ←	Core Network INVITE 100 Trying 180 Ringing 200 OK INVITE ACK Communication BYE	→ → →	SIP (Test System B)	

				5	
			WINITEL to PSIN Use	case 5	
Scenario 3	Basic call with	Fax with 3	33,6 kBit/s (Super G3 Fax)		
	This scenario r	epresents	the case when in the activ	/e call state (I	N10) the Fax transfer on the media and
	B-channels is p	erformed	correctly and the echo car	ncellers in the	e GW are deactivated. Ensure that in
	the active call	state the c	lata transfer is performed o	correctly (e.a.	testing QoS parameters). The call is
	released from t	he calling	user.	, , , ,	5 · · · /
Options	a) resource r	eservatior	n is on both sides		
	b) no resourc	e reserva	tion on terminating side		
	c) no resource	e reserva	tion on either side		
SIP Parameter		INVITE:	SDP .		
		m=audio <port> RTP/AVP 8/0</port>			
		180/200	OK INVITE: SDP		
			m=audio <port> RTP</port>	/AVP 8	
Message flow					
SIP (Test Syste	m A)		Core Network		SIP (Test System B)
			INVITE (SDP1)	→	
	← 180 Ringing				
		÷	200 OK INVITE (SDP2)		
			ACK	→	
			Apply post test rou	ıtine	

			MMTel to PSTN Use	case 5	
Scenario 4	Basic call with I	ax 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 p	erformed	d correctly. The echo cance	llers in the G	W are activated. Ensure that in the
	active call state	the data	transfer is performed corre	ectly (e.g. test	ting QoS parameters). The call is
	released from t	he calling	g 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</port>	/AVP 8/0	
		180/200	OK INVITE: SDP		
			m=audio <port> RTP</port>	/AVP 8	
Message flow					
SIP (Test Syste	m A)		Core Network		SIP (Test System B)
			INVITE (SDP1)	→	
		+	180 Ringing		
		÷	200 OK INVITE (SDP2)		
			ACK	→	
			Apply post test rou	ıtine	

			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 p	erformed	d correctly and the echo car	ncellers in the G	W are not activated. The call is		
	released from t	he called	l user. Ensure that in the ac	tive call state th	e data transfer is performed correctly		
	(e.g. testing QoS parameters).						
Options	a) resource r	eservatio	n is on both sides				
	b) no resourc	e reserva	ation on terminating side				
	c) no resourc	e reserva	ation on either side				
SIP Parameter		INVITE:	SDP				
		n	n=audio <port> RTP/AVP 8</port>	<dynamic-pt></dynamic-pt>			
		a=rtpmap <dynamic-pt> PCMA/8000</dynamic-pt>					
		a=gpmd; vbd=yes					
		180/200 OK INVITE: SDP					
		m=audio <port> RTP/AVP <dynamic-pt></dynamic-pt></port>					
		a=rtpmap <dynamic-pt> PCMA/8000</dynamic-pt>					
		a	=gpmd; vbd=yes				
Message flow							
SIP (Test System A)			Core Network	_	SIP (Test System B)		
			INVITE (SDP1)	→			
		÷	180 Ringing				
		÷	200 OK INVITE (SDP2)	•			
				7			
			Apply post test rou	lune			

	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</port>					
	180/200 OK INVITE: SDP					
	m=audio <port> RTP/AVP 8</port>					
Message flow						
SIP (Test Syste	m A) Core Network SIP (Test System B)					
	INVITE (SDP1) →					
	← 180 Ringing					
	← 200 OK INVITE (SDP2)					
	ACK 🗲					
	Apply post test routine					

	MMTel to PSTN Use cas	ie 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 c	all 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</port>					
	180/200 OK INVITE: SDP					
	m=audio <port> RTP/AVP 8</port>					
Message flow						
SIP (Test Syste	m A) Core Network	SIP (Test System B)				
	INVITE (SDP1)	→				
	← 180 Ringing					
	← 200 OK INVITE (SDP2)					
	ACK	→				
	Apply post test routin	e				

			MMTel to PSTN Use c	ase 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: m 180/200 m	<mark>SDP</mark> n=image <port> udptl t38 <mark>OK INVITE: SDP</mark> n=image <port> udptl t38</port></port>			
Message flow SIP (Test System A)		4	Core Network INVITE (SDP1) 180 Ringing 200 OK INVITE (SDP2) ACK Apply post test rout	→ → tine	SIP (Test System B)	

	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	1					
SIP (Test Sys	tem A)	Core ←	e Network INVITE 486 Busy Here ACK	→ →	SIP (Test System B)	

			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) re b) n c) n	 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 Syst	em A)	+ + +	Core Network 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	+ + + +	SIP (Test System B)		

		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 Sys	/ tem A) ← ← ← ←	Core Network 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	⇒ > > >	r (Test System B)		

		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 Syst	em A) ← ← ← ← ←	Core Network 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	SIP (Test System B) → → →			

		MMTel to PSTN Use case	4			
Scenario 13	CCBS					
	User A is located in netwo	ork A and user B is located in r	etwork B. User A ha	as 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 c	all is released from t	he calling user.		
Options	a) resource reservation is on both sides					
	b) no resource reservation	on on terminating side				
	c) no resource reservation	on on either side				
Message flow	1					
SIP (Test System A) Core Network SIP (Test System B)						
	A C	CBS request was already su	ccessful			
	+	NOTIFY				
		200 OK NOTIFY	→			
		INVITE	→			
	+	180 Ringing				
	+	NOTIFY				
		200 OK NOTIFY	→			
	+	200 OK INVITE				
		ACK	→			
		Apply post test routine				

		MMTel to PSTN Use case	4				
Scenario 14	4 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 b	ecomes available for CC recal	I, the CC recall proced	dure is started. Ensure			
	that the recall from user A	to user B is successful. The c	all is released from the	e calling user.			
Options	a) resource reservation is	on both sides					
	b) no resource reservatio	n on terminating side					
	c) no resource reservatio	n on either side					
Message flow							
SIP (Test Syst	SIP (Test System A) Core Network SIP (Test System B)						
	A CO	CNR request was already su	cessful				
	÷	NOTIFY					
		200 OK NOTIFY	→				
		INVITE	→				
	÷	180 Ringing					
	F	NOTIEY					
	•	200 OK NOTIFY	→				
	+	200 OK INVITE					
		ACK	→				
		Apply post test routine					

6.3.4 PSTN to MMTEL Use case 6

		PSTN to MMTel Use	case 6			
Scenario 1	Basic call. The call is release	d from the calling use	ər			
This scenario represents the case when the call establishment is performed correctly. The						
	released from the calling use	r. Ensure that in the a	active call st	ate the voice transfer is performed		
correctly (e.g. testing QoS parameters).						
Options	a) resource reservation is on both sides					
-	b) no resource reservation of	on terminating side				
	c) no resource reservation o	n either side				
Message flow						
SIP (Test Syst	tem A)	Core Network		SIP (Test System B)		
		INVITE	→			
	+	100 Trying				
	+	180 Ringing				
	+	200 OK INVITE				
		ACK	→			
		Communication				
		BYE	<mark>→</mark>			
	+	200 OK BYE				

	F	STN to MMTel - Use	e case 6		
Scenario 2	Basic call The call is released from the called user.				
	This scenario represents the	case when the call es	stablishmer	t is performed correctly. The call is	
	released from the called user	. Ensure that in the a	ctive call sta	ate the voice transfer is performed	
	correctly (e.g. testing QoS pa	rameters).			
Options	a) resource reservation is or	n both sides			
	b) no resource reservation o	n terminating side			
	c) no resource reservation o	n either side			
Message flow					
SIP (Test Syste	em A)	Core Network		SIP (Test System B)	
		INVITE	→		
	+	100 Trying			
	+	180 Ringing			
	+	200 OK INVITE			
		ACK	→		
		Communication			
	Contraction of the second sec second second sec	BYE			
		200 OK BYE	→		

		PSTN to MMTel Use case 6						
Scenario 3	Basic call - overlap	sending	oont using a	worlon conding. 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 rese	rvation is on both sides						
	 b) no resource reservation on terminating side c) no resource reservation on terminating side 							
Message flow M	(ultiple INVITE me	thod is used						
SIP (Test Syste	m A)	Core Network		SIP (Test System B)				
		INVITE(CSq 1)	→					
	-	INVITE(CSq 2)	→					
	•	 484 Address Incomplete(CSq 1) 	د					
		ACK	7					
	-	INVITE(CSq 3)	→					
	•	 484 Address Incomplete(CSq 2) 	د					
		ACK	7					
		INIVITE(CSq.4)	→					
	÷	 484 Address Incomplete(CSq 3) 	2					
		ACK	→					
	ŧ	 180 Ringing(CSq 4) Apply post test routine 						
Message flow C	Overlap sending, th	e in-Dialogue method is used						
SIP (Test Syste	m A)	Core Network	_	SIP (Test System B)				
		INVITE(CSq 1) 1	→					
	T	484 Address Incomplete(USq 1)	→					
		AGR						
		INVITE(CSq 2) 2	→					
	+	183 Session Progress(CSq 2)						
	L		→					
	T	200 OK PRACK						
		INFO	→					
	÷	200 OK INFO						
		INFO	→					
	+	200 OK INFO						
	÷	180 Ringing(CSq 2) Apply post test routine						

			PSTN to MMTel Use of	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 minutes 180/200 OK INV m			SDP m=audio <port> RTP, OK INVITE: SDP m=audio <port> RTP,</port></port>	/AVP 8/0 /AVP 8		
Message flow SIP (Test System A) Core Network SIP (Test System B) INVITE (SDP1) →						

200 OK INVITE (SDP2) ACK Apply post test routine

→

			PSTN to MMTel Use of	case 6		
Scenario 5	Basic call with Fax with 14,4 kBit/s					
	This scenario r	epresent	s the case when in the activ	e call state	(N10) the Fax transfer is performed	
	correctly. The	echo cano	cellers in the GW are activa	ted. Ensure	that in the active call state the data	
	transfer is perf	ormed co	rrectly (e.g. testing QoS par	ameters).	The call is released from the calling user.	
Options	a) resource r	eservatio	n is on both sides			
	b) no resourc	e reserva	ation on terminating side			
	c) no resource	e reserva	ation on either side			
SIP Parameter		INVITE:	SDP			
		m=audio <port> RTP/AVP 8/0</port>				
		180/200	OK INVITE: SDP			
			m=audio <port> RTP</port>	/AVP 8		
Message flow						
SIP (Test Syste	m 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 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=rtpmap <dynamic-pt> PCMA/8000 a=rtpmap <dynamic-pt> PCMA/8000 a=rtpmap <dynamic-pt> PCMA/8000</dynamic-pt></dynamic-pt></dynamic-pt></dynamic-pt></dynamic-pt></port></dynamic-pt></dynamic-pt></port>					
Message flow SIP (Test Syste	em A) Core Network SIP (Test System B) INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → Apply post test routine					

			PSTN to MMTel Use of	ase 6	
Scenario 7	Basic call - Fax with 14,4 kbit/s with using the T.38 in an audio m-line codec				
	This scenario	represent	s the case when in the activ	e call state th	e Fax transfer is performed correctly
	The call is rele	ased fron	n the calling user. Ensure the	at in the activ	e call state the data transfer is
	performed cor	rectly (e.g	. testing QoS parameters).		
Options	a) resource	reservatio	n is on both sides		
	b) no resour	ce reserva	ation on terminating side		
	c) no resour	ce reserva	ation on either side		
SIP Parameter INVITE: SDP					
		n	n=image <port> udptl t38</port>		
		180/200	OK INVITE: SDP		
		n	n=image <port> udptl t38</port>		
Message flow					
SIP (Test Syste	m A)		Core Network		SIP (Test System B)
			INVITE (SDP1)	→	
		+	180 Ringing		
		÷	200 OK INVITE (SDP2)		
			ACK	→	
			Apply post test rou	tine	

	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</port>					
Message flow SIP (Test Syste	m 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</port>					
	180/200 OK INVITE: SDP					
	m=audio <port> RTP/AVP 8</port>					
Message flow						
SIP (Test Syste	m 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 Syste	em A)	Core Network INVITE ← 486 Busy Here ACK	SIP (Test System B) ➔			

		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).				
Options	resource reservation is on both sides no resource reservation on terminating side no resource reservation on either side				
Message flov SIP (Test Sys	v stem A) Con CF C C Con	re Network INVITE(Call-ID A-B) U 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) mmunication	SIP (T → → →	'est System Β)	
		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 Sys	tem A) Core Netw INVI CFB is pe ← INVI 180 ← 180 200 ← ACK ← 200 ACK Communi	vork TE(Call-ID A-B) erformed TE(Call-ID B-C) Ringing(Call-ID C-B) Ringing(Call-ID B-A) OK INVITE(Call-ID C-B) (Call-ID B-C) OK INVITE(Call-ID B-A) ((Call-ID A-B) ication	 → → → → 	SIP (Test System B)		
		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 Syst	Core Network 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	SIP (Test System B) → → →			

		PSTN to MMTel U	Jse 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 reservatio no resource reserva no resource reserva	n is on both sides ation on terminating side ation on either side				
Message flow SIP (Test Syst A CCBS reque	tem A) Cor est was already succ ← ← ←	e Network essful NOTIFY 200 OK NOTIFY INVITE 180 Ringing NOTIFY 200 OK NOTIFY	SIP (Test System B) → →			
	+	200 OK INVITE ACK Apply post test	→ troutine			

		PSTN to MMTel U	se case 6	
Scenario 15	CCNR User A is located in CCNR request.Ens calling user.	n network A and user B is loc sure that the recall from user	cated in network B. User A has successfully invoked a A to user B is successful. The call is released from th	ı ıe
Options	resource reservation no resour	on is on both sides ation on terminating side ation on either side		
Message flow SIP (Test Sys A CCNR requ	v stem A) Co lest was already succ	re Network cessful	SIP (Test System B)	
	× ×	200 OK NOTIFY	``	
	+	INVITE 180 Ringing	→	
	÷	<mark>NOTIFY</mark> 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

			SDN to VoLTE Use	Case 7			
Scenario 1	cenario 1 Basic call with BC= ITC value - enblock sending						
	This scenario	represents the	case when the call e	stablishment	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 c	performed correctly (e.g. testing QoS parameters).					
Options	a) resource	reservation is o	n both sides	/			
- F	b) no resour	ce reservation	on terminating side				
	c) no resour	ce reservation	on either side				
Message flow							
SIP (Test Syste	m A)		Core Network		SIP (Test System B)		
	,,		INVITE	→			
		4	100 Trying	-			
		Ļ	180 Ringing				
		Ļ					
		``		→			
			Communication				
		4		-			
SIP Parameter			200 OR DIE				
PSTN YMI Boa	rorCanability	Content-T	where application/und	otci netn⊥vml			
element in the		Content-Type, application/vio.etsi.psi(1+xiii) Content-Disposition: signal:handling-ontional					
		<pre>c2vml.varcian="1.0" ancoding="utf 8"2></pre>					
		C C C C C C C C C C C C C C C C C C C					
		BoarorCo	pobility				
		PCostot2					
		Dublielo Cadia a Chan dardy 00					
		Information I ransfer Cabability>II C_value<			alue<		
			el4				
		116	anorenvioue>00<	as 10000 a			
		Information Fransfer Kate>10000<					
		BCOCE	UD Vor1Idoptification: 01				
		Layer1Identification>01<					
		Us	erintoLayer1Protoco	1>00011<			

			ISDN to VoLTE Use	Case 7				
Scenario 2	Basic call with	BC= ITC valu	BC= ITC value - enblock sending					
	This scenario	represents the	case when the call e	stablishmei	nt using en-bloc sending is performed			
	correctly. The	call is released	d from the called user	. Ensure th	at in the active call state the voice transfer			
	is performed c	rmed correctly (e.g. testing QoS parameters).						
Options	a) resource reservation is on both sides							
	b) no resour) no resource reservation on terminating side						
	c) no resour	ce reservation	on either side					
Message flov	v							
SIP (Test Sys	stem A)		Core Network		SIP (Test System B)			
			INVITE	→				
		+	100 Trying					
		÷	180 Ringing					
		÷	200 OK IŇVIŤE					
			ACK	→				
			Communication					
		←	BYE					
			200 OK BYE	→				
SIP Paramete	er	INVITE:						
PSTN XML B	earerCapability	Content-Type: application/vnd.etsi.pstn+xml						
element in th	e INVITÉ	Content-Disposition: signal;handling=optional						
				•				
		xml versio</th <th>n="1.0" encoding="utf</th> <th>-8"?></th> <th></th>	n="1.0" encoding="utf	-8"?>				
		PSTN						
		BearerCapability						
		BCoctet3						
		CodingStandard>00<						
		Information TransferCabability>ITC value<						
		< BCoctet4						
		TransferMode>00<						
		InformationTransferRate>10000<						
		BCoctet5						
		La	ver1Identification>01	<				
		Us	serInfoLayer1Protocol	>00011<				

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

		ISDN to VoLTE Use Case 7							
Scenario 3	Basic call - overlap sending with BC= speech								
	This scenario repres	sents the case when the call establishn	nent using o	overlap sending. The call is					
	released from the ca	alling user. The call is released from the	e calling us	er. Ensure that in the active call					
	state the voice trans	fer is performed correctly (e.g. testing	QoS param	eters).					
Options	a) resource reservation is on both sides								
	b) no resource reservation on terminating side								
	C) no resource rese	ervation on either side							
Message flow Multiple INVITE method is used									
SIP (Test Syste	m A)	Core Network	-	SIP (Test System B)					
		INVITE(CSq 1)	→						
			•						
	L	INVITE(CSq 2)	7						
	τ	484 Address Incomplete(CSq 1)	د						
		ACK	7						
			-						
	4	484 Address Incomplete(CSq 2)							
	×		→						
			-						
		INVITE(CSq 4)	→						
	+	484 Address Incomplete(CSq 3)							
		ACK	→						
	+	180 Ringing(CSq 4)							
		Apply post test routine							
Message flow C	Overlap sending, the	in-Dialogue method is used							
SIP (Test Syste	m 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)	_						
	,		7						
	←	ZUU UK PKAUK							
		INFO	د						
	L		7						
	T								
		INFO	→						
	4		-						
	•								
	+	180 Ringing(CSg 2)							
	-	Apply post test routine							

		ISDN to VoLTE Us	se 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 Syste	em A) Co ←	re Network INVITE 486 Busy Here ACK	SIP (Test System B) ➔			

	ISI	ON to VoLTE Use case 7				
Scenario 11	CFU Ensure that when user A calls u state the voice transfer is perfor the calling user.	ser B, the call is forwarded to us med correctly (e.g. testing QoS p	er C. Ensure that in the active call parameters). The call is released from			
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 Sys	/ tem A) ← II + 180 ← 180 ← 180 ← 200 (← ← 200 (Core Network NVITE(Call-ID A-B) CFU is performed NVITE(Call-ID B-C) Ringing(Call-ID C-B) Ringing(Call-ID B-A) DK INVITE(Call-ID B-C) DK INVITE(Call-ID B-C) DK INVITE(Call-ID B-C) Communication Communication Compose test routine	SIP (Test System B)			

	ISDN to VoLTE Use Case	7			
Scenario 5	CFB Ensure that when user A calls user B which is user dete to user C. Ensure that in the active call state the voice t parameters). The call is released from the calling user.	ermined user busy (UDUB), the call is forwarded ransfer is performed correctly (e.g. testing QoS			
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 Sys	tem A) Core Network 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	SIP (Test System B) →			

		ISDN to VoLTE Use Case 7			
Scenario 6	CFNR Ensure that when user A cal that in the active call state th call is released from the call	ls user B which does not ansv e voice transfer is performed	ver, the call is correctly (e.g.	forwarded to user C. Ensure testing QoS parameters). The	
Options	 a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side 				
Message flov SIP (Test Sy	w stem A) ← ← ↓ 20 ← 20	Core Network 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) 00 OK INVITE(Call-ID C-B) ACK(Call-ID B-C) 00 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) Communication Apply post test routine	⇒ ⇒ ÷	(Test System B)	

		ISDN to VoLTE Use Case 7	7			
Scenario 7	CCBS					
	User A is located in netw	ork A and user B is located in ne	etwork B. User A has successfully invoked	da		
	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 reservat	tion on either side				
Message flow						
SIP (Test Syste	em A)	Core Network	SIP (Test System B)			
	A C	CBS reque <u>st was a</u> lready suc	ccessful			
	+	NOTIFY				
		200 OK NOTIFY	→			
			_			
	-	INVITE	→			
	÷	180 Ringing				
	÷					
		200 OK NOTIFY	⇒			
	C		د			
		AUN Apply post tost routing	7			
		Apply post lest routine				

		ISDN to VoLTE Use case	,	
Scenario 8	CCNR User A is located in netwo CCNR request. Ensure that the recall fror The call is released from	ork A and user B is located in r n user A to user B is successfu the calling user.	etwork B. User A has successful	ly invoked a
Options	 a) resource reservation i b) no resource reservation c) no resource reservation 	s on both sides on on terminating side on on either side		
Message flov SIP (Test Sys	w stem A) A C ←	Core Network CNR request was already su NOTIFY 200 OK NOTIFY	SIP (Test System B cessful ➔)
	÷	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				
Options	correctly (e.g. testing QoS parameters). a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side				
Message flow SIP (1	v Fest System A)	+ + +	Core Network INVITE 100 Trying 180 Ringing 200 OK INVITE ACK Communication BYE	→ → →	SIP (Test System B)

Sasic call The call is released This scenario represents the eleased from the called user correctly (e.g. testing QoS pa	d from the called user case when the call e r. Ensure that in the a arameters).	stablishment ctive call sta	t is performed correctly. The call is te the voice transfer is performed
 a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side 			
System A) ← ← ←	Core Network INVITE 100 Trying 180 Ringing 200 OK INVITE ACK Communication BYE	→ → ←	SIP (Test System B)
	This scenario represents the eleased from the called use correctly (e.g. testing QoS pa a) resource reservation is on c) no resource reservation c c) no resource reservation c System A)	This scenario represents the case when the call eleased from the called user. Ensure that in the algorrectly (e.g. testing QoS parameters). a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side System A) Core Network INVITE 100 Trying € 180 Ringing € 200 OK INVITE ACK Communication BYE 200 OK BYE	This scenario represents the case when the call establishment eleased from the called user. Ensure that in the active call state correctly (e.g. testing QoS parameters). a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side System A) Core Network 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 Syste	m A)	Core ←	e Network INVITE 486 Busy Here ACK	→ →	SIP (Test System B)		

		VoLTE to ISDN Use Case 8	}		
Scenario 4	CFU Ensure that when user state the voice transfer the calling user.	A calls user B, the call is forwarded is performed correctly (e.g. testing	d to user C. Ensure that in the active call g QoS parameters). The call is released f	rom	
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 Syste	em A) ← ← ← ←	Core Network 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	SIP (Test System B) → → →		

	VoLTE to ISDN Use Case	8			
Scenario 5	CFB Ensure that when user A calls user B which is user deter to user C. Ensure that in the active call state the voice tra parameters). The call is released from the calling user.	rmined us ansfer is p	er busy (UDUB), the call is forwarded performed correctly (e.g. testing QoS		
Options	 a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side 				
Message flov SIP (Test Sy	w stem A) Core Network 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 B-A) 200 OK INVITE(Call-ID C-B) Communication Apply post test routine	→ → →	SIP (Test System B)		

		VoLTE to ISDN Use Case 8					
Scenario 6	CFNR Ensure that when user that in the active call st call is released from the	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 b) no resource reservation c) no resource reservation 	 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 Sys	v stem A) C C C C C	Core Network 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	SIP (Test System B) → → →				

		VoLTE to ISDN Use Case	≥8	
Scenario 7	CCBS			
	User A is located in netwo	ork A and user B is located in r	network B. User A has successfully invoked	а
	CCBS request.			
	Ensure that the recall from	n user A to user B is successfu	ul. The call is released from the calling user.	
Options	a) resource reservation	is on both sides		
	b) no resource reservat	ion on terminating side		
	c) no resource reservat	ion on either side		
Message flow	V			
SIP (Test Sys	stem A)	Core Network	SIP (Test System B)	
	A C	CBS request was already su	uccessful	
	+	NOTIFY		
		200 OK NOTIFY	→	
		INVITE	\rightarrow	
	+	180 Ringing		
	+	NOTIFY		
		200 OK NOTIFY	→	
	_			
	÷	200 OK INVITE	<u>.</u>	
		ACK	→	
		Apply post test routine	<u>ب</u>	

		VoLTE to ISDN Use case	28			
Scenario 8	CCNR					
	User A is located in netwo	ork A and user B is located in r	network B. User A has successfully invoked	а		
	CCNR request.Ensure the	at the recall from user A to use	er B is successful.			
	The call is released from t	he calling user.				
Options	a) resource reservation	is on both sides				
	 b) no resource reservati 	on on terminating side				
	 c) no resource reservati 	on on either side				
Message flow	1					
SIP (Test Sys	SIP (Test System A) Core Network SIP (Test System B)					
	A C	CNR request was already su	uccessful			
	F	NOTIFY	<u>.</u>			
		200 OK NOTIFY	→			
			ـ			
	L		7			
	F	Too Kinging				
	4	NOTIEY				
	*	200 OK NOTIFY	→			
	←	200 OK INVITE				
		ACK	→			
		Apply post test routine	9			

6.4.3 VoLTE - PSTN Use case 9

		VoLTE to PSTN Use	Case 9	
Scenario 1	Scenario 1 Basic call. The call is released from the called user.			
	This scenario represents the	case when the call e	stablishmer	t is performed correctly. The call is
	released from the called use	r. Ensure that in the a	ctive call sta	ate the voice transfer is performed
	correctly (e.g. testing QoS pa	arameters).		·
Options	Dotions (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 Syst	tem A)	Core Network		SIP (Test System B)
		INVITE	→	
	+	100 Trying		
	+	180 Ringing		
	+	200 OK IŇVIŤE		
АСК →				
		Communication		
	←	BYE		
	_	200 OK BYE	→	

		VoLTE to PSTN Use	Case 9		
Scenario 2	Basic call. The call is release	d from the calling use	er		
	This scenario represents the case when the call establishment is performed correctly. The call is				
	released from the calling use	r. Ensure that in the a	ctive call sta	ate the voice transfer is performed	
	correctly (e.g. testing QoS pa	rameters).			
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 Syste	em 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	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	1					
SIP (Test System A)		Core ∣ ←	Network INVITE 486 Busy Here ACK	→ →	SIP (Test System B)	

	V	OLTE 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 of b) no resource reservation of c) no resource reservation of 	n both sides on terminating side on either side			
Message flov SIP (Test Sys	w stem A) ← 1 ← 1 200 ← ← 200	Core Network INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C) 80 Ringing(Call-ID C-B) 80 Ringing(Call-ID B-A) 0 OK INVITE(Call-ID C-B) ACK(Call-ID B-C) 0 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) Communication Apply post test routine	SIP (Test System B) → → →		

		VoLTE to PSTN Use Case 9	
Scenario 5	CFB Ensure that when user to user C. Ensure that parameters). The call is	A calls user B which is user determi in the active call state the voice trans s released from the calling user.	ned user busy (UDUB), the call is forwarded sfer is performed correctly (e.g. testing QoS
Options			
Message flow			
SIP (Test Syste	•m A) ← ← ← ←	Core Network 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	SIP (Test System B) → → →

		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 reservent b) no resource res c) no resource res 	ration is on both sides ervation on terminating side ervation on either side			
Message flow SIP (Test Syster	m A) ← ← ← ←	Core Network 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	SIP (Test System B) → → →		

		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 b) no resource reservation c) no resource reservation 	is on both sides ion on terminating side ion on either side			
Message flov SIP (Test Sys	v stem A) A C C	Core Network CBS request was already su NOTIFY 200 OK NOTIFY	SIP (Test System B) ccessful ➔		
	÷	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 netw	ork A and user B is located in n	etwork B. User A has successfully invoked a			
	CCNR request.					
	Ensure that the recall from	m user A to user B is successfu	I. The call is released from the calling user.			
Options	a) resource reservation	n is on both sides				
	b) no resource reserva	tion on terminating side				
	c) no resource reserva	ition on either side				
Message flow						
SIP (Test Syste	SIP (Test System A) Core Network SIP (Test System B)					
	A	CCNR reque <u>st was a</u> lready suc	ccessful			
	+	NOTIFY				
		200 OK NOTIFY	\rightarrow			
			<u>.</u>			
	_	INVITE	\rightarrow			
	+	180 Ringing				
	-					
	+					
	L					
	•		۷.			
		AUN Apply post test routine	7			

6.4.4 PSTN to VoLTE Use case 10

		PSTN to VoLTE Use	case 10		
Scenario 1	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 us	er. Ensure that in the a	active call st	ate the voice transfer is performed	
	correctly (e.g. testing QoS p	arameters).			
Options	ions a) resource reservation is on both sides				
	b) no resource reservation	on terminating side			
	c) no resource reservation	on either side			
Message flow	N				
SIP (Test System A)	Core Network		SIP (Test System B)	
-		INVITE	→		
	+	100 Trying			
	+	180 Ringing			
	+	200 OK INVITE			
		ACK	→		
		Communication			
		BYE	→		
	+	200 OK BYE			

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		PSTN to VoLTE Use	case 10		
Scenario 2	Basic call The call is released from the called user.				
	This scenario represents th	e case when the call e	stablishmen	t is performed correctly. The call is	
	released from the called us	er. Ensure that in the a	ctive call sta	ate the voice transfer is performed	
	correctly (e.g. testing QoS p	parameters).			
Options	a) resource reservation is	on both sides			
	b) no resource reservation	n on terminating side			
	c) no resource reservation	n on either side			
Message flow	· /				
SIP (T	Fest 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 s	sending		verting The cell in			
	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 rese	ervation on either side					
Message flow N	Alessage flow Multiple INVITE method is used						
SIF (Test S	bystem Aj		→	SIF (Test System B)			
			-				
		INVITE(CSq 2)	→				
	+	484 Address Incomplete(CSq 1)					
		ACK	→				
			د				
	4	484 Address Incomplete(C.Sq.2)	7				
		ACK	→				
		-					
	L	INVITE(CSq 4)	7				
	C	ACK	→				
	F	180 Ringing(CSg 4)	2				
		Apply post test routine					
Message flow C	Overlap sending, the	in-Dialogue method is used					
SID /Test S	System A)	Coro Notwork		SID (Test System P)			
SIF (Test a	bystem Aj		→	SIF (Test System B)			
	←	484 Address Incomplete(CSg 1)					
		ACK	→				
		INVITE(CSq 2) 2	→				
	+	183 Session Progress(CSq 2)	د				
	4		7				
	x						
		INFO	→				
	+	200 OK INFO					
		INFO	→				
	←	200 OK INFO	-				
	+	180 Ringing(CSq 2) Apply post test routine					

	PSTN to VoLTE Use case 10					
Scenario 4	rio 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	ons					
Message flow						
SIP (Test Syst	em A)	Core	e Network INVITE 486 Busy Here ACK	→ →	SIP (Test System B)	

		PSTN to VoLTE Use case 10			
Scenario 5	CFU				
	Ensure that when user A c	alls user B, the call is forwarded	to user C. E	nsure that in the active call	
	state the voice transfer is p	performed correctly (e.g. testing	QoS parame	ters).	
	The call is released from the	ne calling user.	·		
Options	a) resource reservation is on both sides				
	b) no resource reservatio	n on terminating side			
	c) no resource reservatio	n on either side			
Message flo	w				
SIP (Test Sy	stem 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 Syste	em A) ← ← ← ←	Core Network 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	SIP (Test System B) → → →		

		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 b) no resource reserva c) no resource reserva 	n is on both sides tion on terminating side tion on either side				
Message flow SIP (Test Sys	v stem A) ← ← ← ←	Core Network 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	SIP (Test System B) → → →			

		PSTN to MMTel Use case	≥ 6		
Scenario 8	CCBS				
	User A is located in netw	ork A and user B is located in r	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 reservat	on on terminating side			
	c) no resource reservat	on on either side			
Message flow	/				
SIP (Test Sys	tem A)	Core Network	SIP (Test System B)		
	A C	CBS reque <u>st was a</u> lready su	Jccessful		
	+	NOTIFY			
		200 OK NOTIFY	→		
	-	INVITE	→		
	÷	180 Ringing			
	-				
	÷				
		200 OK NOTIFY	7		
	L				
	E		د		
		ADD	7		
		Apply post test fourine	,		

		PSTN to VoLTE Use case 10	
Scenario 9	CCNR User A is located in ne CCNR request.Ensure calling user.	etwork A and user B is located in network that the recall from user A to user B is s	B. User A has successfully invoked a successfull. The call is released from the
Options	 a) resource reservati b) no resource reservati c) no resource reservati 	ion is on both sides vation on terminating side vation on either side	
Message flow SIP (Test Syste	em A)	Core Network A CCNR request was already successi	SIP (Test System B) ful
	(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 ca	ase 11	
Scenario 1	Succ	essful call - This sco	enario represents the case w	hen the call	establishment is performed correctly.
	Ensu	ire that in the active	call state the voice transfer	is performed	correctly (e.g. testing QoS
	para	meters). The call is	released from the calling use	er.	
Options	a)	resource reservati	on is on both sides		
	b)	no resource reserv	ation on terminating side		
	C)	no resource reserv	ation on either side		
Message flow a	Suc	cessful call - resourd	ce resevation on both sides		
SIP (Test Syster	n 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 rout	ine	
Message flow: b), C)				
SIP (Test Syster	n A)		Core Network		SIP (Test System B)
		-	INVITE (SDP1)	→	
		÷	100 Trying		
		÷	180 Ringing		
		+	200 OK INVITE	•	
			ACK	7	
				7	
		F	200 OK BYE	·	
			Apply post test rout	ne	

		VoLTE to VoLTE Use c	ase 11	
Scenario 2	Successful call - This sc	enario 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 use	er.	
Options	a) resource reservation	n is on both sides		
	b) no resource reserva	ation on terminating side		
	c) no resource reserva	ation on either side		
Message flow a)	Successful call - resour	ce resevation on both sides		
SIP (Test Syster	n 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 rout	tine	
Message flow: b	o), c)			
SIP (Test Syster	n A)	Core Network		SIP (Test System B)
		INVITE (SDP1)	→	
	+	100 Trying		
	+	180 Ringing		
	+	200 OK INVITE	_	
		ACK	→	
	+	BYE	_	
		200 OK BYE	→	
		Apply post test rout	tine	

	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</port></port>			
Message flow SIP (Test Syste	em A) Core Network INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → Apply post test routine	SIP (Test System B)		

	VoLTE to VoLTE Use case 11					
Scenario 4	Called user is us This scenario re initiate call clear	ser busy presents th ing to the c	ne case, when the ca calling user.	lled user	is user determined user busy the network	
Options			•			
Message flow	N					
SIP (Test Sys	stem A)	Core	e Network		SIP (Test System B)	
		÷	INVITE 486 Busy Here ACK	→ →		

		VoLTE to VoLTE Use case 11	1	
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 flov SIP (Test Sys	v stem A) C C C C	Core Network 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	SIP (Test System B) → → →	

		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 flov SIP (Test Sys	v stem A) C C C C	Core Network 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	→ → → →	IP (Test System B)	

			VoLTE to VoLTE Use case 1'	1	
Scenario 7	CFNR Ensure that in call is	e that when user the active call si released from th	A calls user B which does not answ tate the voice transfer is performed of e calling user	ver, the c correctly	call is forwarded to user C. Ensure (e.g. testing QoS parameters). The
Options	a) b) c)	resource reserva no resource rese no resource rese	ation is on both sides ervation on terminating side ervation on either side		
Message flow SIP (Test Syst	tem A)	← ← ← ←	Core Network 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	 + + + + + 	SIP (Test System B)

		VoLTE to VoLTE Use case	11		
Scenario 8	CCBS User A is located in netw CCBS request.	ork A and user B is located in n	etwork B. User A has successfully invoked a		
	Ensure that the recall from The call is released from	m user A to user B is successfu the calling user.	ıl.		
Options	 a) resource reservation b) no resource reservation c) no resource reservation 	n is on both sides ation on terminating side ation on either side			
Message flow SIP (Test Syste	em A) A CCBS ←	Core Network or CCNR request was alread NOTIFY 200 OK NOTIFY	SIP (Test System B) ly successful ➔		
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 netwo	ork A and user B is located in r	network B. User A has successfully invoked a	а
	CCNR request.		· · · · · ·	
	Ensure that the recall from	n user A to user B is successfu	ıl.	
	The call is released from	the calling user.		
Options	a) resource reservation	is on both sides		
•	b) no resource reservat	ion on terminating side		
	c) no resource reservat	ion on either side		
Message flo	w			
SIP (Test Sv	stem A)	Core Network	SIP (Test System B)	
	A CCBS	or CCNR request was alread	lv 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

				4.0	
			VoLTE to MMTel Use c	case 12	
Scenario 1	Successful call - This scenario represents the case when the call establishment is performed correctly.				
	Ensure that in the a	active c	all state the voice transfer	r is performed	correctly (e.g. testing QoS
	parameters). The c	all is re	leased from the calling us	ser.	
Options	a) resource reserv	vation is	s on both sides		
	b) no resource res	servatio	on on terminating side		
	c) no resource res	servatio	on on either side		
Message flow	a) Successful call - re	esource	e resevation on both sides		
SIP (Tes	t Svstem A)		Core Network		SIP (Test System B)
- (,		INVITE (SDP1)	→	
	•	F	100 Trying		
	•	F	183 Session Progress		
			PRACK	→	
	•	F	200 OK PRACK		
			UPDATE	→	
	•	F	200 OK UPDATE		
	•	F	180 Rinaina		
	•	F	200 OK		
			ACK	→	
			BYE	→	
	•	F	200 OK BYE		
			Apply post test rou	tine	
Message flow:	: b), c)				
SIP (Tes	t System A)		Core Network		SIP (Test System B)
	, ,		INVITE (SDP1)	→	
	•	F	100 Trying		
	•	F	180 Ringing		
	•	F	200 OK INVITE		
			ACK	→	
			BYE	→	
	•	F	200 OK BYE	-	
		-	Apply post test rou	tine	
l					

		VoLTE to MMTel Use c	ase 12			
Scenario 2	2 Successful call - This scenario represents the case when the call establishment is performed					
	Ensure that in the active	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 use	er.			
Options	a) resource reservatio	n is on both sides				
	b) no resource reserva	ation on terminating side				
	c) no resource reserva	ation on either side				
Message flow a	a) Successful call - resour	ce resevation on both sides				
SIP (Test	System A)	Core Network		SIP (Test System B)		
-		INVITE (SDP1)	→			
1	+	100 Trying				
1	+	183 Session Progress				
		PRACK	→			
	←	200 OK PRACK				
		UPDATE	→			
	←	200 OK UPDATE				
	←	180 Ringing				
	+	200 OK				
		ACK	→			
	+	BYE				
		200 OK BYE	→			
		Apply post test rout	ine			
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 rout	ine			

		VoLTE to MMTel Use of	e 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 perfor	med correctly and the echo cance	cellers in the GW are not activated. The call is			
	released from the	called user. Ensure that in the ac	active call state the data transfer is performed correctly			
	(e.g. testing QoS p	parameters).				
Options	a) resource rese	rvation is on both sides				
-	b) no resource re	eservation on terminating side				
	c) no resource re	eservation on either side				
SIP Parameter	ter INVITE: SDP					
		m=image <port> udptl t38</port>				
	<mark>18</mark>	0/200 OK INVITE: SDP				
		m=image <port> udptl t38</port>				
Message flow						
SIP (Test	System A)	Core Network	SIP (Test System B)			
		INVITE (SDP1)	→			
		← 180 Ringing				
	← 200 OK INVITE (SDP2)					
	ACK →					
		Apply post test rou	outine			

		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	o the calling user.		
Options	 a) resource reserva b) no resource reserva c) no resource reserva 	resource reservation is on both sides no resource reservation on terminating side no resource reservation on either side		
Message flow SIP (Test Syster	m A)	Core Network INVITE ← 486 Busy Here ACK	SIP (→	Test System B)

VoLTE to	MMTel Us	e case 12

	VoL	TE to MMTel Use case 12	
Scenario 5	CFU Ensure that when user A calls us state the voice transfer is perform the calling user.	ser B, the call is forwarded to med correctly (e.g. testing Q	o user C. Ensure that in the active call oS parameters). The call is released from
Options	 a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side 		
Message flov SIP (Test Sys	v stem A)	Core Network NVITE(Call-ID A-B) CFU is performed NVITE(Call-ID B-C) Ringing(Call-ID C-B) Ringing(Call-ID B-A) OK INVITE(Call-ID C-B) ACK(Call-ID B-C) OK INVITE(Call-ID B-A) ACK(Call-ID A-B) Communication pply post test routine	SIP (Test System B) → → →

		VoLTE to MMTel Use case 12	2		
Scenario 6	CFB Ensure that when us to user C. Ensure that parameters).	er A calls user B which is user determ at in the active call state the voice tran	nined user busy (UDUB), the call is forv nsfer is performed correctly (e.g. testing	varded J QoS	
	The call is released f	rom 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 Syst	tem A) ← ← ← ←	Core Network 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	SIP (Test System B) → → →		
	VoLTE to N	MTel 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 flo SIP (Test Sy	w stem A) Core INVITE(← 180 Ringin CFNR is ← INVITE(180 Ringin ← 180 Ringin 200 OK INVI ← ACK(C ← 200 OK INVI ACK(C Comm Apply p	Network Call-ID A-B) → g(Call-ID B-A) performed Call-ID B-C) g(Call-ID C-B) → g(Call-ID B-A) TE(Call-ID C-B) → all-ID B-C) TE(Call-ID B-A) all-ID A-B) → unication ost test routine	SIP (Test System B)		

		VoLTE to MMTel Use case	9 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 Syste	m A) A (←	Core Network CCNR request was already su NOTIFY 200 OK NOTIFY	SIP (Test System B) Iccessful		
	INVITE → ← 180 Ringing				
 ► NOTIFY 200 OK NOTIFY → ← 200 OK INVITE 					
ACK → Apply post test routine					

			VoLTE to MMTel Use case 1	2	
Scenario 9	CCNR				
	User	A is located in ne	twork A and user B is located in ne	etwork B.	User A has successfully invoked a
	CCN	R request. Ensure	e that the recall from user A to user	B is succ	essful.
	The	call is released fro	m the calling user.		
Options	a)	resource reserva	tion is on both sides		
-	b)	no resource rese	rvation on terminating side		
	C)	no resource rese	rvation on either side		
Message flow	1				
SIP (Test System A) Core Network SIP (Test System B)			SIP (Test System B)		
	A CCNR request was already successful				
		+	NOTIFY		
			200 OK NOTIFY	→	
			INVITE	→	
		+	180 Ringing		
		-			
		+	NOTIFY		
200 OK NOTIFY →					
		•	200 OK INVITE	-	
			AUK	7	
	Apply post test routine				

6.6.2 MMTel to VoLTE Use case 13

			MMTel to VoLTE Use ca	ase 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				
	para	meters). The call is	released from the calling use	er.	
Options	a)	resource reservation	on is on both sides		
	b)	no resource reserv	ation on terminating side		
	C)	no resource reserv	ation on either side		
Message flow a) Suc	cessful call - resourd	ce resevation on both sides		
SIP (Test Syster	n 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	\rightarrow	
		←	200 OK BYE		
			Apply post test rout	ine	
Message flow: b), c)				
SIP (Test Syster	n A)		Core Network	_	SIP (Test System B)
		_	INVITE (SDP1)	\rightarrow	
		+	100 Trying		
		(180 Ringing		
		+	200 OK INVITE	_	
			ACK	→	
		-	BYE	→	
		+	200 OK BYE		
			Apply post test rout	ine	

		MMTel to VoLTE Use ca	ase 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 use	er.			
Options	a) resource reservation	on is on both sides				
	b) no resource reserv	ation on terminating side				
	c) no resource reserv	vation on either side				
Message flow a)	Successful call - resou	rce resevation on both sides				
SIP (Test Syster	n 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 rout	ine			
Message flow: b), c)					
SIP (Test Syster	n A)	Core Network		SIP (Test System B)		
		INVITE (SDP1)	→			
	+	100 Trying				
	+	180 Ringing				
	+	200 OK INVITE	_			
		ACK	→			
	+	BYE	_			
		200 OK BYE	. →			
		Apply post test rout	ine			

	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</port></port>				
Message flow SIP (Test Syste	m A) Core Network SIP (Test System B) INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → Apply post test routine				

INVITE 486 Busy Here ACK ←

→

→

		MMTel to VoLTE Use case 13	3	
Scenario 5	CFU			
	Ensure that when user	A calls user B, the call is forwarded	l to user	r C. Ensure that in the active call
	state the voice transfer	is performed correctly (e.g. testing	QoS pa	arameters). The call is released from
	the calling user.			
Options	a) resource reservation	on is on both sides		
	b) no resource reserv	ation on terminating side		
	 C) no resource reservence 	ation on either side		
SIP (Test Sys	stem A)	Core Network	→	SIP (Test System 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 Sys	, tem A) ← ← ← ← ←	Core Network 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	SIP (Test System B) → → →		

	MMTel to Vo	DLTE 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				
Options	 a) resource reservation is on both sides b) no resource reservation on terminating side c) no resource reservation on either side 				
Message flov SIP (Test Sy	w Core № stem A) Core № INVITE(C INVITE(C ← 180 Ringing ← INVITE(C 180 Ringing 180 Ringing ← 180 Ringing 200 OK INVIT 200 OK INVIT ← ACK(Ca Commu Apply po	letwork call-ID A-B) → g(Call-ID B-A) performed call-ID B-C) g(Call-ID C-B) → g(Call-ID C-B) → g(Call-ID C-B) → g(Call-ID B-A) → TE(Call-ID C-B) → ull-ID B-C) → TE(Call-ID B-A) → ull-ID A-B) → unication → st test routine →	SIP (Test System B)		

		MMTel to VoLTE Use case	: 13		
Scenario 8	ario 8 CCBS				
	User A is located in netwo	ork A and user B is located in r	network B. User A has successfully invoked a	a l	
	CCBS request.Ensure the	at the recall from user A to use	r B is successful.		
	The call is released from	the calling user.			
Options	a) resource reservation	is on both sides			
	b) no resource reservat	ion on terminating side			
	 c) no resource reservat 	ion on either side			
Message flow	N				
SIP (Test Sys	stem A)	Core Network	SIP (Test System B)		
	A C	CBS reque <u>st was a</u> lready su	ıccessful		
	÷	NOTIFY			
	200 OK NOTIFY →				
			_		
	_	INVITE	→		
	÷	180 Ringing			
200 OK NOTIFY 7					
Apply post test routine					

		MMTel to VoLTE Use case	÷ 13		
Scenario 9	CCNR				
	User A is located in netwo	ork A and user B is located in n	network B. User A has successfully invoked a		
	CCNR request. Ensure th	at the recall from user A to use	er B is successful.		
	The call is released from	the calling user.			
Options	a) resource reservation	s on both sides			
	b) no resource reservation	on on terminating side			
	c) no resource reservation	on on either side			
Message flow	N				
SIP (Test System A) Core Network SIP (Test System B)					
	A CCNR request was already successful				
	÷	NOTIFY			
		200 OK NOTIFY	→		
			7		
	← 180 Ringing				
	-	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]	IMS/PES equivalent		
-	Detailed description	-		
Local exchange call re	ocal 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.		

Q.543

IMS



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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 s	ubscriber call request delay: overlap sendin	g
ISDN SUBSCRIBER	clause 2.3.2.2 [i.14]	ISDN [i.16]
LINES	Local exchange call request delay.	Call request delay is defined as the interval
Local exchange call	Call request delay is defined as the	from the instant at which the SETUP
request delay -	interval from the instant at which the	message has been received from the
Overlap sending.	SETUP message has been received from	subscriber signalling system until the
	the subscriber signalling system until the	SETUP ACKNOWLEDGE message is
	SETUP ACKNOWLEDGE message is	passed back to the subscriber signalling
	passed back to the subscriber signalling	system.
	system.	
IMS SUBSCRIBER		IMS [i.17]
		Call request delay is defined as the interval
Local exchange call		from the instant at which the INVITE
request delay.		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 s	ubscriber call request delay: en Block sendi	ng
ISDN SUBSCRIBER	clause 2.3.2.3 [i.14]	ISDN [i.16]
LINES	For DIGITAL SUBSCRIBER LINES using	For ISDN using en-bloc sending, call
Local exchange call	en-bloc sending, call request delay is	request delay is defined as the interval
request delay en -	defined as the interval from the instant at	from the instant at which the SETUP
block sending.	which the SETUP message is received	message is received from the subscriber
	from the subscriber signalling system	signalling system until the CALL
	until the call proceeding message is	PROCCEDING message is passed back to
	passed back to the subscriber signalling	the subscriber signalling system.
	system.	

Q.543

IMS





Tabl	e 6
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Meaning of timers	Parameter Q.543 [i.14]	IMS/PES equivalent
J	Detailed description	
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
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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	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 lc interface has been sent
	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 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)



Q.543





IMS

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

Table 9

Meaning of timers	Parameter Q.543 [i.14]	IMS/PES equivalent
	Detailed description	
Call set up delay: en Blo	ock 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	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
	information necessary for call set-up is complete and ends when the call setup is sent on the outgoing signalling system.	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).
		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).

Detailed description	
	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.
t	included the setup delay may increase up



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

Table 1	0
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Meaning of timers	Parameter Q.543 [i.14]	IMS/PES equivalent
	Detailed description	•
Through-connection d	lelay	·
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
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Parameter Q.543 [i.14]	IMS/PES equivalent
Detailed description	
lav	
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 [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 [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
	Parameter Q.543 [i.14] Detailed description lay 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.



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: S	speech Quali	y para	ameters	based	on	PESQ/F	POLQA
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Speech Quality Summary				
P.862.1/P.863				
Min				
Max				
Mean				
Std-Dev				

Table 13: Speech level parameters

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.



Figure 23

Annex A (informative): Calls flows

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

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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



Figure A.2



Figure A.3



Figure A.4



Figure A.5







Figure A.6



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.









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	Т3	T4

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AVERAGE									
Date	Time	Calls Successful	Calls Failed:	Call Length	Latency ms	T1	T2	Т3	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:

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)".

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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".

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IEEE 802.16: "A Technical Overview of the WirelessMANTMAir 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".
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