ETSI TS 101 563 V2.1.1 (2024-07)

Speech and multimedia Transmission Quality (STQ);
IMS/PES/VoLTE/VoNR exchange performance requirements

Reference
RTS/STQ-00242m

Keywords
5G, CST, IMS, performance, PES, QoS, VoLTE, VoNR

ETSI

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

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

Siret N° 348 623 562 00017 - APE 7112B
Association à but non lucratif enregistrée à la
Sous-Préfecture de Grasse (06) N° w061004871

Contents

Intellectual Property Rights

Essential patents

IPRs essential or potentially essential to normative deliverables may have been declared to ETSI. The declarations pertaining to these essential IPRs, if any, are publicly available for **ETSI members and non-members**, and can be found in ETSI SR 000 314: *"Intellectual Property Rights (IPRs); Essential, or potentially Essential, IPRs notified to ETSI in respect of ETSI standards"*, which is available from the ETSI Secretariat. Latest updates are available on the ETSI Web server (https://ipr.etsi.org/).

Pursuant to the ETSI Directives including the ETSI IPR Policy, no investigation regarding the essentiality of IPRs, including IPR searches, has been carried out by ETSI. No guarantee can be given as to the existence of other IPRs not referenced in ETSI SR 000 314 (or the updates on the ETSI Web server) which are, or may be, or may become, essential to the present document.

Trademarks

The present document may include trademarks and/or tradenames which are asserted and/or registered by their owners. ETSI claims no ownership of these except for any which are indicated as being the property of ETSI, and conveys no right to use or reproduce any trademark and/or tradename. Mention of those trademarks in the present document does not constitute an endorsement by ETSI of products, services or organizations associated with those trademarks.

DECT™, **PLUGTESTS™**, **UMTS™** and the ETSI logo are trademarks of ETSI registered for the benefit of its Members. **3GPP™** and **LTE™** are trademarks of ETSI registered for the benefit of its Members and of the 3GPP Organizational Partners. **oneM2M™** logo is a trademark of ETSI registered for the benefit of its Members and of the oneM2M Partners. **GSM**® and the GSM logo are trademarks registered and owned by the GSM Association.

Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Speech and multimedia Transmission Quality (STQ).

Modal verbs terminology

In the present document "**shall**", "**shall not**", "**should**", "**should not**", "**may**", "**need not**", "**will**", "**will not**", "**can**" and "**cannot**" are to be interpreted as described in clause 3.2 of the ETSI Drafting Rules (Verbal forms for the expression of provisions).

"**must**" and "**must not**" are **NOT** allowed in ETSI deliverables except when used in direct citation.

Introduction

As a general principle, the definitions of IMS/PES/VoLTE/VoNR design objectives should be based on "best practice" performance of signalling. The values contained in the present document are "best practice" performance values measured on IMS and NGN implementations.

1 Scope

The present document contains design requirements applicable to IMS/PES/VoLTE/VoNR exchange implementations based on Recommendation ITU-T Q.543 [1]. The definitions of IMS/PES/VoLTE/VoNR design objectives are based on "best practice" performance of legacy PSTN; LTE and VoNR signalling. The values contained in the present document are "best practice" performance values measured on IMS and NGN implementations.

2 References

2.1 Normative 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 referenced document (including any amendments) applies.

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

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

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

[1] Recommendation ITU-T Q.543 (03/1993): "Digital exchange performance design objectives".

[2] Void.

[3] ETSI TS 124 229: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; 5G; IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3 (3GPP TS 24.229)".

[4] ETSI TS 183 043 (V3.4.1): "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS-based PSTN/ISDN Emulation; Stage 3 specification".

[5] Recommendation ITU-T Q.541 (03/1993): "Digital exchange design objectives - General".

[6] Recommendation ITU-T G.812 (06/2004): "Timing requirements of slave clocks suitable for use as node clocks in synchronization networks".

[7] Recommendation ITU-T G.823: "The control of jitter and wander within digital networks which are based on the 2048 kbit/s hierarchy".

[8] Recommendation ITU-T G.826 (12/2002): "End-to-end error performance parameters and objectives for international, constant bit-rate digital paths and connections".

[9] ETSI TS 102 928 (V1.1.3): "Speech and multimedia Transmission Quality (STQ); End-to-End Transmission Planning Requirements for Real Time Services in an NGN context".

[10] Recommendation ITU-T P.863 (03/2018): "Perceptual objective listening quality prediction".

[11] Recommendation ITU-T P.863.1 (06/2019): "Application guide for Recommendation ITU-T P.863".

[12] Recommendation ITU-T G.107 (06/2015): "The E-model: a computational model for use in transmission planning".

[13] Recommendation ITU-T G.107.1 (06/2019): "Wideband E-model".

[14] Recommendation ITU-T G.107.2 (03/2023): "Fullband E-model".

[15] Recommendation ITU-T G.113 (11/2007) and its Amendments: "Transmission impairments due to speech processing".

[16] Recommendation ITU-T G.722 (09/2012): "7 kHz audio-coding within 64 kbit/s".

[17] Recommendation ITU-T P.565.1 (11/2021): "Machine learning model for the assessment of transmission network impact on speech quality for mobile packet-switched voice services".

2.2 Informative 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 referenced document (including any amendments) applies.

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

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.

Not applicable.

3 Definition of terms, symbols and abbreviations

3.1 Terms

Void.

3.2 Symbols

Void.

3.3 Abbreviations

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

5GC	5G Core
AGCF	Access Gateway Control Function
AMR/G	Adaptive Multirate
ASBC	Access Session Border Controller
BSBC	Border Session Border Controller
CST	Call Setup Time
EPC	Evolved Packet Core
FB	Fullband
IAD	Integrated Access Device
IMS	IP Multimedia Subsystem
ISDN	Integrated Service Digital Network
ITU-T	ITU Telecommunication Standardization Sector
LTE	Long Term Evolution
MGW	Media Gateway
MO	Mobile Originating
MOS	Mean Opinion Score
MOS-LQE	MOS Listening Quality Estimated
MOS-LQEF	MOS Listening Quality Estimated in FB context
MOS-LQO	MOS Listening Quality Objective
MOS-LQOF	MOS Listening Quality Objective in FB context
MOS-LQxF	MOS Listening Quality Fullband (either from Objective, Subjective or Estimation assessment)

MSAN	Multi Service Access Node
MT	Mobile Terminating
NB	Narrowband
NGN	New Generation Network
NNI	Network Network Interface
NSA	Non-Standalone
PCM	Pulse Code Modulation
P-CSCF	Proxy Call Server Control Function
PES	PSTN/ISDN Emulation Subsystem
PSTN	Public Switched Telephone Network
QoE	Quality of Experience
RRC	Radio Resource Control
RTP	Real Time Protocol
SBC	Session Border Controller
SDP	Session Description Protocol
SIP	Session Initiation Protocol
TIE	Time Interval Error
UE	User Equipment
UNI	User Network Interface
VGW	Voice GateWay
VoLTE	Voice over LTE
VoNR	Voice over New Radio
WB	Wideband

4 Reference loads and parameter requirements

4.1 PSTN reference loads definitions

The PSTN reference load definitions and values described in Tables 1 to 6 are the reference load definitions described in Recommendation ITU-T Q.543 [1]. The derived PES procedures are based on the IMS/PES Emulation specification [4] and the derived SIP procedures are based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) [3].

4.2 IMS and LTE/VoNR reference loads definitions

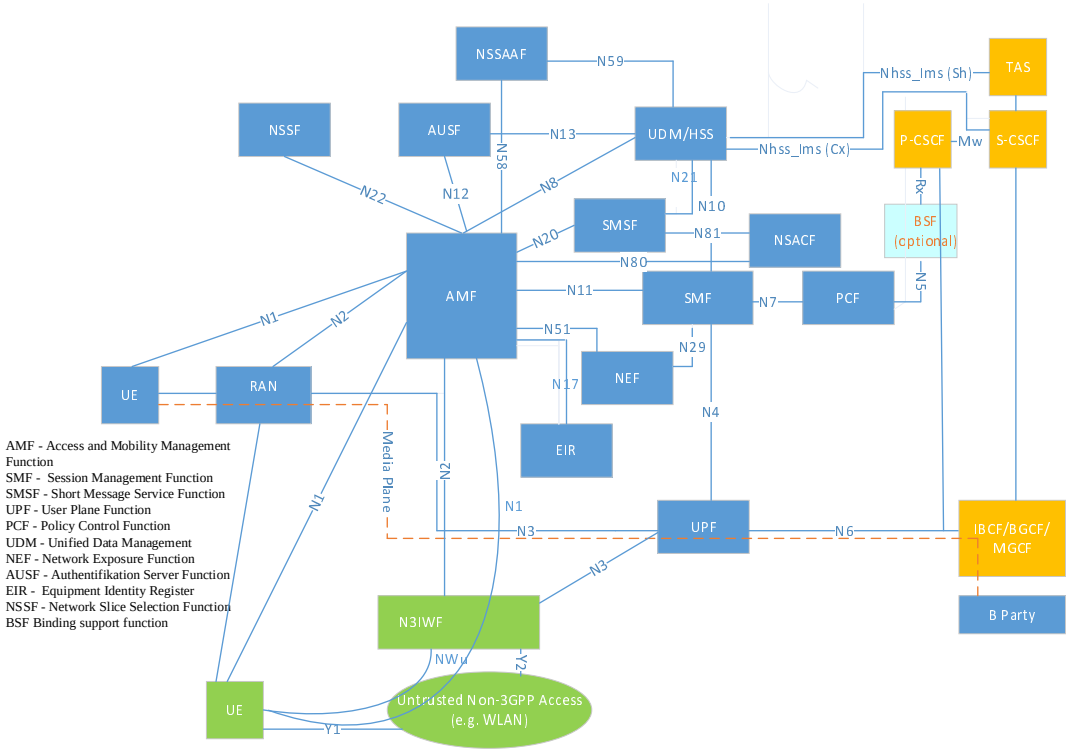

Figure 1: 5G/IMS Basic Configuration

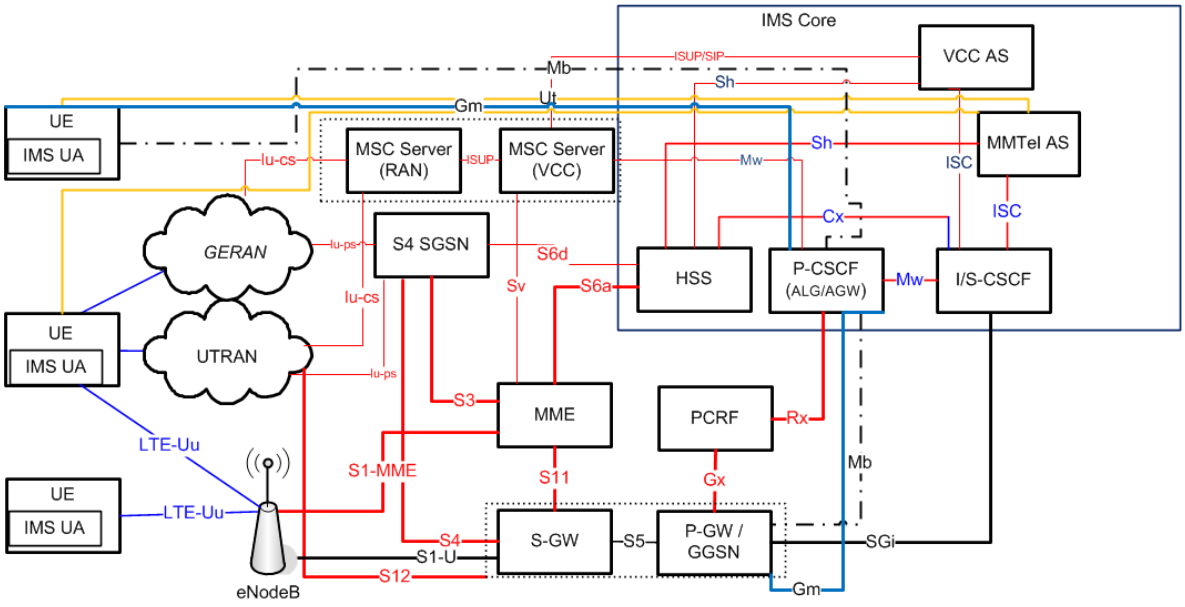

Figure 2: IMS/LTE Basic Configuration

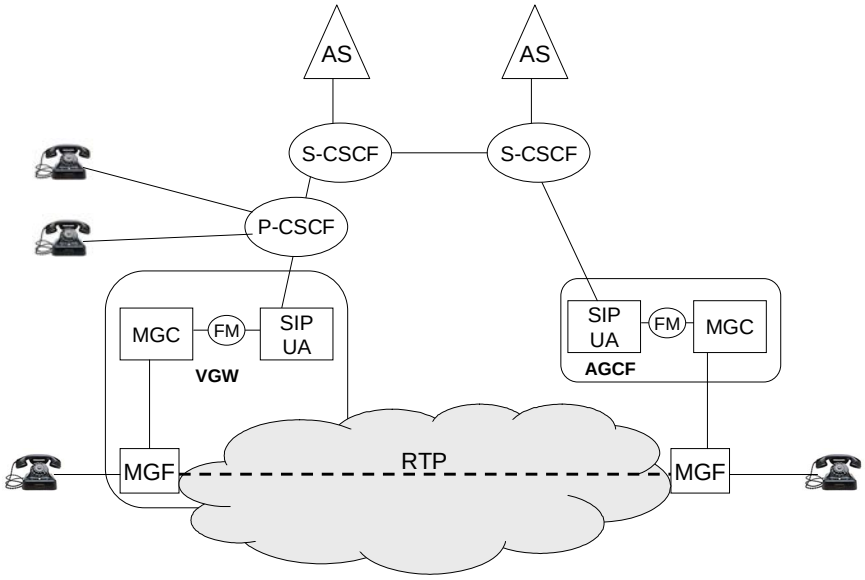

Figure 3: AGCF/VGW session processing models

4.3 Parameter requirements

IMS systems shall comply with the requirements given in Table 1 to Table 7.

Table 1

Meaning of timers	Parameter Recommendation ITU-T Q.543 [1] Detailed description	IMS, PES equivalent	Reference Load A		Reference Load B
			Mean Value	95 % probability of not exceeding	95 % probability of not exceeding
Local exchange call request delay - originating outgoing and internal traffic connections					
ANALOGUE SUBSCRIBER LINES Local exchange call request delay - originating outgoing and internal traffic connections.	Clause 2.3.2.1 of Recommendation ITU-T Q.543 [1] 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 [4] 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.	≤ 400 ms	≤ 600 ms	≤ 1 000 ms
ANALOGUE SUBSCRIBER with IAD (VGW) Local exchange call request delay - originating outgoing and internal traffic connections.		PES [4] 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.	≤ 400 ms	≤ 600 ms	≤ 1 000 ms
IMS local exchange call request delay					
IMS SUBSCRIBER Local exchange call request delay.		IMS [3] Call request delay is defined as the interval from the instant at which the INVITE message has been received from the SIP subscriber until the 100 Trying from the SBC/P-CSCF is passed back to the subscriber.	≤ 11 ms	≤ 98 ms	≤ 98 ms

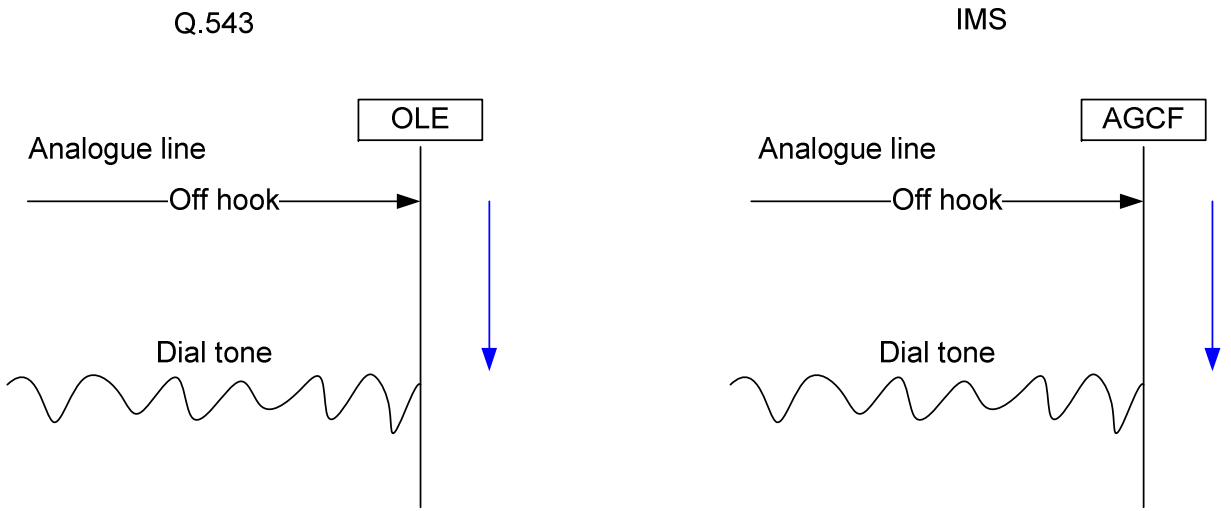

Figure 4: Local exchange analogue subscriber call request delay: overlap sending

Table 2

Meaning of timers	Parameter Recommendation ITU-T Q.543 [1] Detailed description	IMS, PES equivalent	Reference Load A		Reference Load B
			Mean Value	95 % probability of not exceeding	95 % probability of not exceeding
ANALOGUE SUBSCRIBER LINES Alerting sending Delay for terminating traffic.	Clause 2.3.6.1.1 of Recommendation ITU-T Q.543 [1] 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 [4] 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.	≤ 300 ms	≤ 450 ms	≤ 750 ms

Table 3

Meaning of timers	Parameter Recommendation ITU-T Q.543 [1] Detailed description	IMS, PES equivalent	Reference Load A		Reference Load B
			Mean Value	95 % probability of not exceeding	95 % probability of not exceeding
Alerting sending delay for internal traffic (the users 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 of Recommendation ITU-T Q.543 [1] 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 [4] 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.	≤ 300 ms	≤ 450 ms	≤ 750 ms
ANALOGUE SUBSCRIBER LINES Alerting sending Delay for internal traffic.		PES [4] 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 MGW/VGW until Ringing tone is sent towards the calling subscriber.	≤ 550 ms	≤ 800 ms	≤ 1 100 ms
VoLTE/VoNR measured at MO ASBC and MT ASBC NNI: 180 sending		VoLTE/VoNR For calls terminating sending delay is defined as the interval from the instant that a 180 message at the VoLTE - UE interface has received and 180 is sent on the VoLTE - UE towards the calling subscriber.	≤ 46 ms	≤ 68 ms	≤ 68 ms
VoLTE measured at MO and MT UNI: 180 sending (NSA)		VoLTE For calls terminating sending delay is defined as the interval from the instant that a 180 message at the VoLTE - UE interface has received and 180 is sent on the VoLTE - UE towards the calling subscriber.	≤ 60 ms	≤ 82 ms	≤ 82 ms
VoNR measured at MO and MT UNI: 180 sending (NSA)		VoNR For calls terminating sending delay is defined as the interval from the instant that a 180 message at the VoNR - UE interface has received and 180 is sent on the VoNR - UE towards the calling subscriber.	≤ 60 ms	≤ 82 ms	≤ 82 ms

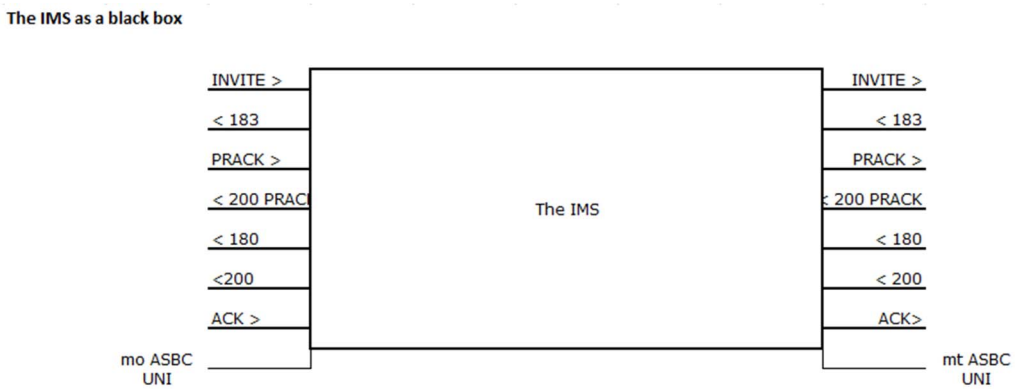

Figure 5: IMS Transit Time measured at MO ASBC and MT ASBC NNI

Table 4

Meaning of timers	Parameter Recommendation ITU-T Q.543 [1] Detailed description	IMS, PES equivalent	Reference Load A		Reference Load B
			Mean Value	95 % probability of not exceeding	95 % probability of not exceeding
Call set up delay:					
IMS SUBSCRIBER Call set up delay for Landline networks.		IMS [3] 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.	≤ 124 ms	≤ 362 ms	≤ 380 ms
Call set up delay mobile subscriber		IMS [3] 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 ASBC until the instant when the corresponding INVITE signalling information is passed on the terminating BSBC interface to the called user.	≤ 356 ms	≤ 385	≤ 422

Table 5

Meaning of timers	Parameter Recommendation ITU-T Q.543 [1] Detailed description	IMS, PES equivalent	Reference Load A		Reference Load B
			Mean Value	95 % probability of not exceeding	95 % probability of not exceeding
Call setup time (CST):					
IMS SUBSCRIBER CST		Call Setup Time (CST) The duration from when a call is made to the time of receiving a 180 ringing on the originating **Gm** interface.	≤ 192 ms	≤ 190 ms	≤ 412 ms
Call setup time measured at MO ASBC and MT ASBC NNI:		VoLTE/VoNR Call Setup Time (CST) The duration from when a call is made to the time of receiving a 180 ringing.	≤ 1 099 ms	≤ 1 234 ms	≤ 1 456ms
VoLTE Call setup time measured at MO UNI and MT UNI		VoLTE Call Setup Time (CST) The duration from when a call is made to the time of receiving a 180 ringing. MT UE being RRC Connected (no paging needed)	≤ 1 113 ms	≤ 1 248 ms	≤ 1 470 ms
VoNR Call setup time measured at MO UNI and MT UNI		VoNR Call Setup Time (CST) The duration from when a call is made to the time of receiving a 180 ringing. MT UE being RRC Connected (no paging needed)	≤ 1 113 ms	≤ 1 248 ms	≤ 1 470 ms

Table 6

Meaning of timers	Parameter Recommendation ITU-T Q.543 [1] Detailed description	IMS, PES equivalent	Reference Load A		Reference Load B
			Mean Value	95 % probability of not exceeding	95 % probability of not exceeding
Through-connection delay:					
IMS Through-connection delay		IMS [3] 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.	≤ 46 ms	≤ 52ms	≤ 52 ms
IMS Through-connection delay measured at MO ASBC and MT ASBC NNI		IMS [3] 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 BSBC 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 SBC Interface.	≤ 46 ms	≤ 52ms	≤ 52 ms
VoLTE		VoLTE 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 VoLTE - 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 VoLTE - UE interface.	≤ 60 ms	≤ 66 ms	≤ 66 ms
VoNR		VoNR 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 VoNR - 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 VoLTE - UE interface.	≤ 60 ms	≤ 66 ms	≤ 66 ms

Table 7

Meaning of timers	Parameter Recommendation ITU-T Q.543 [1] Detailed description	IMS, PES equivalent	Reference Load A		Reference Load B
			Mean Value	95 % probability of not exceeding	95 % probability of not exceeding
Connection release delay:					
IMS SUBSCRIBER Connection release delay Delay for Internal traffic, Landline		IMS [3] 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.	≤ 44 ms	≤ 49 ms	≤ 52 ms
VoLTE Connection release delay measured at MO ASBC and MT ASBC NNI		IMS [3] Connection release delay is defined as the interval from the instant when a BYE message is received at the originating or terminating VoLTE - UE interface until the instant when 200OK is sent and a corresponding BYE message is sent at the terminating or originating SBC interface respectively.	≤ 47 ms	≤ 57 ms	≤ 57 ms
VoLTE Connection release delay measured at MO UNI and MT UNI		VoLTE Connection release delay is defined as the interval from the instant when a BYE message is received at the originating or terminating VoLTE - UE interface until the instant when 200OK is sent and a corresponding BYE message is sent at the terminating or originating VoLTE - UE interface respectively.	≤ 61 ms	≤ 71 ms	≤ 71 ms
VoNR Connection release delay measured at MO UNI and MT UNI		VoNR Connection release delay is defined as the interval from the instant when a BYE message is received at the originating or terminating VoLTE - UE interface until the instant when 200OK is sent and a corresponding BYE message is sent at the terminating or originating VoNR - UE interface respectively.	≤ 61 ms	≤ 71 ms	≤ 71 ms

4.4 Call processing performance objectives

4.4.1 Premature release

The probability that an exchange malfunction will result in the premature release of an established connection in any one minute interval shall be:

$$\mathbf{P \leq 2 \times 10^{-5}}$$

4.4.2 Release failure

The probability that an exchange malfunction will prevent the required release of a connection shall be:

$$\mathbf{P \leq 2 \times 10^{-5}}$$

4.4.3 Incorrect charging or accounting

The probability of a call attempt receiving incorrect charging or accounting treatment due to an exchange malfunction shall be:

$$\mathbf{P \leq 10^{-4}}$$

4.4.4 Misrouting

The probability of a call attempt misrouted following receipt by the exchange of a valid address shall be:

$$\mathbf{P \leq 10^{-4}}$$

4.4.5 No tone

The probability of a call attempt encountering no tone following receipt of a valid address by the exchange shall be:

$$\mathbf{P \leq 10^{-4}}$$

4.4.6 Other failures

The probability of the exchange causing a call failure for any other reason not identified specifically above shall be:

$$\mathbf{P \leq 10^{-4}}$$

4.5 Transmission performance

4.5.1 64 kbit/s switched connections

The probability of a connection being established with an unacceptable transmission quality across the exchange shall be:

$$\mathbf{P \leq 10^{-5}}$$

The transmission quality across the exchange is said to be unacceptable when the bit error ratio is above the alarm condition.

NOTE: In Recommendation ITU-T G.826 [8], budgets of 18,5 % of $1,5 \times 10^{-6}$ were allocated to each national network, so the packet loss for a national connection should be no more than $2,75 \times 10^{-7}$ [9].

4.6 Slip rate

4.6.1 Normal conditions

The slip rate under normal conditions is covered in Recommendation ITU-T Q.541 [5].

4.6.2 Temporary loss of timing control

The case of temporary loss of timing control corresponds to the "holdover operation" defined and recommended in Recommendation ITU-T G.812 [6]. The allowable slip rate will correspond to the maximum relative TIE also recommended therein.

4.6.3 Abnormal conditions at the exchange input

The slip rate in case of abnormal conditions (wide phase deviations, etc.) at the exchange input is the subject of further study taking into account the requirements of Recommendation ITU-T G.823 [7].

4.7 RTP Delay of EPC/5GC from ASBC to BSBC

In a 5G Core network, ASBC and BSBC play crucial roles in managing and routing multimedia sessions between users and external networks. When transcoding is involved, it implies the conversion of media streams from one codec format to another, typically to ensure interoperability or accommodate network requirements.

Real-time Transport Protocol (RTP) Delay refers to the time taken for a packet containing audio or video data to travel from the Access Session Border Controller (ASBC) to the Border Session Border Controller (BSBC) within a 5G Core (5GC) network, with transcoding occurring at the User Network Interface (UNI) side.

Factors influencing RTP Delay in this scenario include network congestion, processing time for transcoding, queuing delays at network elements, and propagation time across network links. Minimizing RTP Delay is crucial for ensuring optimal Quality of Experience (QoE) for real-time multimedia applications such as voice calls or video conferencing within the 5G Core network architecture.

Table 8

Time delays incurred during transcoding and RTP packet transfer between SBCs	**Reference Load A Mean Value [ms]**	**Reference Load B 95 % probability of not exceeding [ms]**
Transcoding delay at the SBC between the UNI side and NNI (AMR/G.711)	80	100 See note
RTP Delay from ASBC to BSBC without transcoding on UNI side	3	3
NOTE: T(plc) < 20 ms, T(jb) < 20.		

4.8 End-to-End Listening Quality MOS-LQxF

4.8.1 Introduction to Listening Quality Requirements

The requirements for Listening Quality apply to both directions of transmission.

Since IMS/PES/VoLTE/VoNR exchange implementations can safely be assumed to be capable of delivering end-to-end fullband voice services, the requirements are assessed on both:

- the fullband scale for the objective prediction of Listening Quality (MOS-LQOF); and
- the fullband scale for the estimation of Listening Quality (MOS-LQEF).

MOS-LQO shall be assessed in accordance with Recommendation ITU-T P.863 [10] (Perceptual objective listening quality prediction) in conjunction with its application guide Recommendation ITU-T P.863.1 [11]. The algorithm of Recommendation ITU-T P.863 [10] shall be used in the fullband mode.

In cases which are compliant with the scope of Recommendation ITU-T P.565.1 [17] (e.g. that there is pure IMS calling (e.g. VoLTE) or connections are used without transcoding or re-packaging, etc.), MOS-LQOF can alternatively be obtained by following Recommendation ITU-T P.565.1 [17]. This has to be stated in the test report.

NOTE 1: Recommendation ITU-T P.863 [10] in fullband mode takes into account bandwidth limitations by detecting the absence of any speech energy above 3,8 kHz to indicate a narrowband degraded file and the absence of any speech energy above 7 kHz to indicate a wideband degraded file. With a narrowband signal the maximum achievable score is 3,8. The scale for MOS therefore is always from 1 to 5.

MOS-LQE shall be assessed in accordance with Recommendation ITU-T G.107.2 [14] (Fullband E-Model) in conjunction with equipment impairment factors Ie published in the Appendices I, IV and V of Recommendation ITU-T G.113 [15]. The requirements for the E-model derived quality estimates will be given on the Transmission Rating scale of the fullband E-Model which extends from R = 0 to R = 148 (This avoids re-transformation to the MOS scale).

NOTE 2: Ie values for wideband codecs need to be scale-transformed according to the formula provided in Appendix V to Recommendation ITU-T G.113 [15]. Ie values for narrowband codecs need to be scale-transformed by combining the formulae provided in Appendix IV and Appendix V to Recommendation ITU-T G.113 [15].

NOTE 3: For the Narrowband (NB) case described in Recommendation ITU-T G.107 [12], the transmission rating scale ranges from R = 0 (lowest possible quality) to R = 100 (optimum quality). On this scale, a default NB transmission channel including logarithmic PCM coding obtains a rating of R = 93,2. For a Wideband (WB) speech transmission channel, the quality is generally judged better than that for an NB channel. Thus, this scale range was extended in Recommendation ITU-T G.107.1 [13] to a maximum value of R = 129 for a clean wideband (50 Hz to 7 000 Hz) channel, as it is defined in Recommendation ITU-T G.722 [16]. In the fullband (FB) case, this scale was further extended to R = 148 to reflect the even higher quality of the fullband (20 Hz to 20 000 Hz) channel.

The requirements in the following clause are subdivided into requirements for:

- end-to-end narrowband service;
- end-to-end wideband services;
- end-to-end fullband services (includes super-wideband).

4.8.2 Requirements for an end-to-end narrowband service

- NB Sce. MOS-LQOF shall be > 3,5.
- NB Sce. Transmission Rating R_{FB} shall be > 90.

4.8.3 Requirements for an end-to-end wideband service

- WB Sce. MOS-LQOF shall be > 3,9.
- WB Sce. Transmission Rating R_{FB} shall be > 110.

4.8.4 Requirements for an end-to-end fullband service

- FB Sce. MOS-LQOF shall be > 4,2.
- FB Sce. Transmission Rating R_{FB} shall be > 130.

Annex A (informative): Bibliography

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

History

Document history		
V1.1.1	August 2011	Publication
V1.2.1	August 2012	Publication
V1.3.1	November 2014	Publication
V1.4.1	October 2015	Publication
V2.1.1	July 2024	Publication