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Speech and multimedia Transmission Quality (STQ);
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Part 1: Reference benchmarking, background traffic profiles and KPIs for VoIP and FoIP in fixed networks

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

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

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

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

The present document is part 1 of a multi-part deliverable covering the Reference benchmarking, background traffic profiles and KPIs, as identified below:

Part 1: "Reference benchmarking, background traffic profiles and KPIs for VoIP and FoIP in fixed networks";

Part 2: "Reference benchmarking and KPIs for High speed internet";

Part 3: "Reference benchmarking, background traffic profiles and KPIs for UMTS and VoLTE";

Part 4: "Reference benchmarking for IPTV, Web TV and RCS-e Video Share".

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Introduction

The present document describes possible key performance indicators for VoIP and FoIP as well as framework requirements for reference benchmarking particularly with regard to background traffic. The latest version replaces clause 6 with load generation methods that are appropriate for the agreed scope of the present document.

1 Scope

The present document:

- identifies and defines possible key performance indicators for voice and fax telephony services;
- defines benchmarking methods for the spectrum of potential applications.

The offer of new NGN services requires new KPIs, QoS measurement and benchmarking methods which are needed to ensure the quality of new services. To ensure the comparability of test results, reference benchmarking methods and background traffic load profiles are needed.

The scope of the defined testing procedures is the evaluation of the network access by VoIP and FoIP fixed-network services. The measurements are conducted stationary between a subscriber access-point to a measurement point emulating an idealized termination point in the core network. All access technologies offered by the operator under test are considered. In this context the measurements and key performance indicators determinations are performed by analysing signals accessible on the network.

The present document is the first part of the multi-part deliverable which consists of four parts.

The present document contains possible KPIs for VoIP and FoIP as well as framework requirements for reference benchmarking particularly with regard to background traffic profiles.

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 E.800 (09-2008): "Definitions of terms related to quality of service".

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

[3] ETSI TS 101 563 (V1.3.1): "Speech and multimedia Transmission Quality (STQ); IMS/PES/VoLTE exchange performance requirements".

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

[5] ETSI ES 202 765-2 (V1.2.1): "Speech and multimedia Transmission Quality (STQ); QoS and network performance metrics and measurement methods; Part 2: Transmission Quality Indicator combining Voice Quality Metrics".

[6] Recommendation ITU-T G.131 (11-2003): "Talker echo and its control".

[7] ETSI ES 203 021-3 (V2.1.2): "Access and Terminals (AT); Harmonized basic attachment requirements for Terminals for connection to analogue interfaces of the Telephone Networks; Update of the technical contents of TBR 021, EN 301 437, TBR 015, TBR 017; Part 3: Basic Interworking with the Public Telephone Networks".

[8] ETSI TBR 003 (Edition 1) (11-1995): "Integrated Services Digital Network (ISDN); Attachment requirements for terminal equipment to connect to an ISDN using ISDN basic access".

[9] ETSI TBR 004 (Edition 1) (11-1995): "Integrated Services Digital Network (ISDN); Attachment requirements for terminal equipment to connect to an ISDN using ISDN primary rate access".

[10] ETSI EN 300 175-8: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 8: Speech and audio coding and transmission".

[11] Recommendation ITU-T O.41 (10-1994): "Psophometer for use on telephone-type circuits".

[12] Recommendation ITU-T P.56 (12-2011): "Objective measurement of active speech level".

[13] Recommendation ITU-T P.501 (03-2017): "Test signals for use in telephonometry".

[14] ETSI ES 202 737 (V1.4.1): "Speech and multimedia Transmission Quality (STQ); Transmission requirements for narrowband VoIP terminals (handset and headset) from a QoS perspective as perceived by the user".

[15] ETSI ES 202 739 (V1.4.1): "Speech and multimedia Transmission Quality (STQ); Transmission requirements for wideband VoIP terminals (handset and headset) from a QoS perspective as perceived by the user".

[16] Recommendation ITU-T P.340 (05-2000): "Transmission characteristics and speech quality parameters of hands-free terminals".

[17] Recommendation ITU-T P.502 (05-2000): "Objective test methods for speech communication systems using complex test signals".

[18] Recommendation ITU-T P.863.1 (09-2014): "Application guide for Recommendation ITU-T P.863".

[19] Recommendation ITU-T E.458 (02-1996): "Figure of merit for facsimile transmission performance".

[20] Recommendation ITU-T E.453 (08-1994): "Facsimile image quality as corrupted by transmission-induced scan line errors".

[21] ETSI TS 102 250-2 (V2.3.1): "Speech and multimedia Transmission Quality (STQ); QoS aspects for popular services in mobile networks; Part 2: Definition of Quality of Service parameters and their computation".

[22] Recommendation ITU-T Y.1541: "Network performance objectives for IP-based services".

[23] Recommendation ITU-T G.711: "Pulse code modulation (PCM) of voice frequencies".

[24] Recommendation ITU-T V.34: "A modem operating at data signalling rates of up to 33 600 bit/s for use on the general switched telephone network and on leased point-to-point 2-wire telephone-type circuits".

[25] Recommendation ITU-T V.17: "A 2-wire modem for facsimile applications with rates up to 14 400 bit/s".

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.

[i.1] Void.

[i.2] ETSI ETR 138 (12-1997): "Network Aspects (NA); Quality of service indicators for Open Network Provision (ONP) of voice telephony and Integrated Services Digital Network (ISDN)".

[i.3] ETSI EG 202 425 (V1.1.1): "Speech Processing, Transmission and Quality Aspects (STQ); Definition and implementation of VoIP reference point".

[i.4] ETSI EG 202 057-2: "Speech and multimedia Transmission Quality (STQ); User related QoS parameter definitions and measurements; Part 2: Voice telephony, Group 3 fax, modem data services and SMS".

[i.5] ETSI TR 103 138: "Speech and multimedia Transmission Quality (STQ); Speech samples and their use for QoS testing".

[i.6] IEC 61260:1995: "Electroacoustics - Octave-band and fractional-octave-band filters".

[i.7] Recommendation ITU-T T.30 (09-2005): "Procedures for document facsimile transmission in the general switched telephone network".

[i.8] Recommendation ITU-T T.38 (09-2010): "Procedures for real-time Group 3 facsimile communication over IP networks".

[i.9] Recommendation ITU-T T.24: "Standardized digitized image set".

[i.10] Recommendation ITU-T G.168 (04-2015): "Digital network echo cancellers".

[i.11] Recommendation ITU-T V.25: "Automatic answering equipment and general procedures for automatic calling equipment on the general switched telephone network including procedures for disabling of echo control devices for both manually and automatically established calls".

[i.12] IETF RFC 8337 (March 2018): "Model-Based Metrics for Bulk Transport Capacity", M.Mathis and A.Morton.

NOTE: Available from https://tools.ietf.org/html/rfc8337.

[i.13] IETF RFC 4122 (July 2005): "A Universally Unique IDentifier (UUID) URN Namespace".

3 Definition of terms, symbols and abbreviations

3.1 Terms

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

benchmark: evaluation of performance value/s of a parameter or set of parameters for the purpose of establishing value/s as the norm against which future performance achievements may be compared or assessed

NOTE: The definition is taken from Recommendation ITU-T E.800 [1].

3.2 Symbols

Void.

3.3 Abbreviations

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

AB	Direction of call establishment User B to user A
ACK	ACKnowledgement
AGCF	Access Gateway Control Function
AGW	Access GateWay
AIMD	Additive Increase, Multiplicative Decrease

ANS	ANswer Tone
AS	Application Server
BA	Direction of call establishment User A to user B
BRI	Basic Rate Interface
BST	Broadband Speed Test
CED	CallED station identification tone
CFR	Call Failure Rate
CI	Common Interface
CLI	Calling Line Identification
CNG	CalliNG tone
CPE	Customer Premises Equipment
CSCF	Call Session Control Function
CSS	Composite Source Signal
DL	DownLink
DNS	Domain Name System
DSS1	Digital subscriber Signalling System No. 1
DTMF	Dual-Tone Multi-Frequency signalling
ECM	Error Correction Mode
ERP	Ear Reference Point
FE	Functional Entity
FM	Feature Manager
FoIP	Fax over IP
FOM	Figure Of Merit
FTP	File Transfer Protocol
HTTP	Hypertext Transfer Protocol
HTTPS	Hypertext Transfer Protocol Secure
IAD	Integrated Access Device
IEC	International Electrotechnical Commission
IETF	Internet Engineering Task Force
IMAP	Internet Message Access Protocol
IMAPS	Internet Message Access Protocol Secure
IMS	Internet Multimedia Subsystem
IP	Internet Protocol
IPTV	Internet Protocol TeleVision
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunication Union - Telecommunication standardization sector
IVR	Interactive Voice Response
KPI	Key Performance Indicator
LQO	Listening Quality Objective
MGC	Media Gateway Controller
MGW	Media GateWay
MMTel	MultiMedia Telephony service
MOS	Mean Opinion Score
MRP	Mouth Reference Point
MSAN	Multi-Service Access Nodes
NGN	New Generation Network
NTP	Network Time Protocol
OVL	OVerLoad point
PCMA	Pulse Code Modulation A-law
P-CSCF	Proxy - Call Session Control Function
PES	PSTN Emulation Subsystem
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
PT	ProTocol
QoS	Quality of Service
RDP	Remote Desktop Protocol
RFC	Request For Comments
RTCP	Real Time Control Protocol
RTP	Real Time Protocol
RTSP	Real Time Streaming Protocol
S-CSCF	Service - Call Session Control Function
SDP	Session Description Protocol

SIP	Session Initiation Protocol
SMTP	Simple Mail Transfer Protocol
SMTPS	Simple Mail Transfer Protocol Secure
SNR	Speech signal level/Noise level
SSH	Secure SHell
SSL	Secure Sockets Layer
SWB	Super Wide Band
TCP	Transmission Control Protocol
TELR	Talker Echo Loudness Rating
TLS	Transport Layer Security
TOR	Terminal Owning Region
TR	Technical Report
TV	TeleVision
UA	User Agent
UAS	User Agent Server
UDP	User Datagram Protocol
UE	User Equipment
UL	UpLink
UMTS	Universal Mobile Telecommunications System
UNiA	User Network interface A
UNiB	User Network interface B
UUID	Universally Unique IDentifier

NOTE: As described in IETF RFC 4122 [i.13].

VBD	Voice Band Data
VGW	Voice GateWay
VoIP	Voice over IP
VoLTE	Voice over Long Term Evolution

4 Management Summary

4.1 Introduction

The spectrum of potential applications of a benchmarking platform requires measurements including but not limited to the following: analogue (a/b), ISDN, VoIP (including SIP trunking) and high-speed internet.

The performance data which are collected will be relevant for a real-world environment encompassing a mix of technologies. The scope of the defined testing procedures is the evaluation of the network access by VoIP and FoIP fix-network services. The measurements are conducted stationary between a subscriber access-point to a measurement point emulating an idealized termination point in the core network.

4.2 Scope of functionality

A benchmarking platform can be distributed across a larger region or an entire country. In this case several server systems should be also part of the setup, including: a business intelligence platform; a data warehouse, a management system and a system for evaluating of media (e.g. video, audio and voice) quality.

The measurement systems at the user premises are connected electrically to ISDN ports via a VGW (IAD) or directly to a CPE or Ethernet port (e.g. MMTel fixed access).

The test system (QoS control and data server) is connected through ISDN connections (via IMS PES with AGCF (or PSTN or ISDN Access) or IMS PES with VGW) or MMTel (IMS) fixed access lines for voice quality measurements.

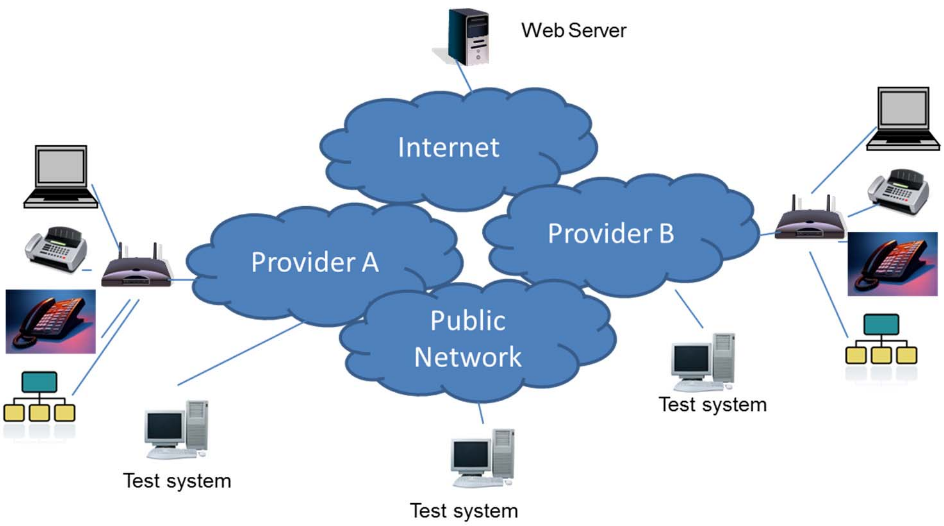

Figure 1: Setup of the benchmarking platform

5 Technical concept

5.1 Voice over IP

The conduction of voice quality measurements is following the descriptions that can be found in ETSI EG 202 057-2 [i.4], Recommendation ITU-T Q.543 [4], ETSI TS 101 563 [3] and ETSI TS 102 250-2 [21], clauses 6.6.3.1 and 6.6.3.2.

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

- ETSI EG 202 425 [i.3] for VoIP access;
- ETSI ES 203 021-3 [7] for analogue access;
- ETSI TBR 003 [8] for ISDN BRI access;
- ETSI TBR 004 [9] for ISDN PRI access;
- ETSI EN 300 175-8 [10].

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

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

The KPI listed in table 1 are recorded as part of the voice quality measurements.

Table 1: Overview of KPI for voice quality measurements

1.	Call set-up delay [4] and session initiation call set-up delay [3]
2.	Call set-up time (Post Dialling Delay) [5]
3.	Premature release probability (Call Failure Rate), see clause 5.4
4.	Telephony Cut-off Call Ratio [%] (Call drop rate), see clause 5.5
5.	Media establishment delay, see clause 5.6
6.	Level of active speech signal, see clause 5.7
7.	Noise level, see clause 5.8
8.	Signal to Noise ratio, see clause 5.9
9.	Speech signal attenuation, see clause 5.10
10.	Talker echo delay, see clause 5.11
11.	Double talk, see clause 5.12
12.	Interrupted voice transmission, see clause 5.13
13.	Listening speech quality, see clause 5.14
14.	Listening speech quality stability, see clause 5.15
15.	End-to-end audio delay, see clause 5.16
16.	End-to-end audio delay variation, see clause 5.17
17.	Frequency response, see clause 5.18
18.	Fax transmission T.30 (Fax, bit rate ≤ 14,4 kbit/s and Fax, bit rate ≥ 14,4 kbit/s), see clause 5.19
19.	Early media, see clause 5.20
20.	Jitter Buffer and IP periodization response time, see clause 5.21

5.2 Call set-up delay and Session initiation call set-up delay

The testing methodology for the call set-up delay is described in ETSI TS 101 563 [3].

Call set-up delay is defined as the interval from the instant when the signalling information required for outgoing circuit selection is received from the incoming signalling system until the instant when the corresponding signalling information is passed to the outgoing signalling system.

For SIP (e.g. SIP Trunking, IMS) Session initiation call set-up 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.

Figure 2 depicts some of the call set up measurement options between AGCF/VGW and the Gm Interface.

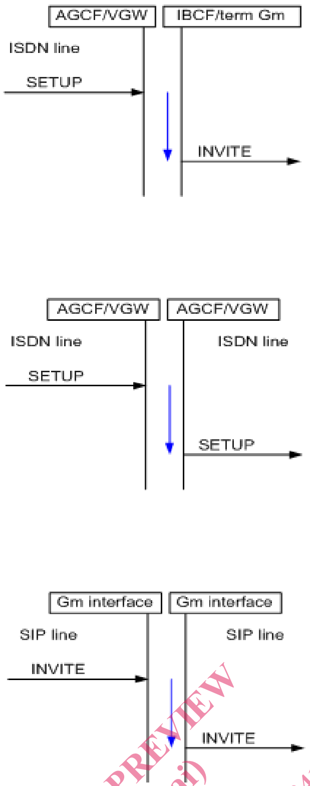

Figure 2: Call set up delay and Session initiation call set-up delay: en-bloc sending is used

Table 2 gives an overview of the call set-up delay configuration options.

Table 2: Call set-up delay configurations

	From	To
Call set up delay and Session initiation call set-up delay	MMTel (IMS) fixed access	MMTel (IMS) fixed access
	MMTel (IMS) fixed access	IMS PES with AGW (PSTN or ISDN Access)
	MMTel (IMS) fixed access	IMS PES with VGW
	IMS PES with AGW (PSTN or ISDN Access)	MMTel (IMS) fixed access
	IMS PES with AGW (PSTN or ISDN Access)	IMS PES with AGW(PSTN or ISDN Access)
	IMS PES with AGW (PSTN or ISDN Access)	IMS PES with VGW
	IMS PES with VGW	IMS PES with VGW
	IMS PES with VGW	IMS PES with AGW (PSTN or ISDN Access)
	IMS PES with VGW	IMS PES with VGW
NOTE: The Call set-up delay values are specified in ETSI TS 101 563 [3].		

Figure 3 illustrates the session processing model used by the AGCF and VGW functional entities.

An AGCF is modelled as comprising H.248 Media Gateway Controller (MGC), Feature Manager (FM), and SIP UA functionality. An AGCF interfaces to a Media Gateway (MGW) and also to the S-CSCF (via P1 and Mw reference points respectively).

A functional modelling of the VGW contains an entity similar to H.248 Media Gateway Controller, a Feature Manager, a SIP UA, and MGW functionality. The VGW interfaces to the P-CSCF using the Gm reference point.

The SIP UA functionality provides the interface to the other components of the IMS-based architecture. It is involved in registration and session processing as well as in event subscription/notification procedures with application servers.

The MGC functionality enables the session processing functionality to interface with existing line signalling such as analogue signalling or DSS1.