SLOVENSKI STANDARD

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Kakovost prenosa govora in večpredstavnih vsebin (STQ) - Prenosne zahteve za ozkopasovne terminale VoIP (ročne in naglavne) glede na kakovost storitev (QoS), kot jih dojema uporabnik

Speech and multimedia Transmission Quality (STQ) - Transmission requirements for narrowband VoIP terminals (handset and headset) from a QoS perspective as perceived by the user

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Speech and multimedia Transmission Quality (STQ); Transmission requirements for narrowband VoIP terminals (handset and headset) from a QoS perspective as perceived by the user

Reference

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Foreword

This ETSI Standard (ES) 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).

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Introduction

Traditionally, the analogue and digital telephones were interfacing switched-circuit 64 kbit/s PCM networks. With the fast growth of IP networks, terminals directly interfacing packet-switched networks (VoIP) are being rapidly introduced. Such IP network edge devices may include gateways, specifically designed IP phones, soft phones or other devices connected to the IP based networks and providing telephony service. Since the IP networks will be in many cases interworking with the traditional PSTN and private networks, many of the basic transmission requirements have to be harmonised with specifications for traditional digital terminals. However, due to the unique characteristics of the IP networks including packet loss, delay, etc. new performance specifications, as well as appropriate measuring methods, will have to be developed. Terminals are getting increasingly complex, advanced signal processing is used to address the IP specific issues. Also, the VoIP terminals may use other than 64 kbit/s PCM (Recommendation ITU-T G.711 [7]) speech algorithms.

The advanced signal processing of terminals is targeted to speech signals. Therefore, wherever possible speech signals are used for testing in order to achieve mostly realistic test conditions and meaningful results.

The present document provides speech transmission performance for narrowband VoIP handset and headset terminals.

NOTE: Requirement limits are given in tables, the associated curve when provided is given for illustration.

1 Scope

The present document provides speech transmission performance requirements for 4 kHz narrowband VoIP handset and headset terminals; it addresses all types of IP based terminals, including wireless and soft phones.

In contrast to other standards which define minimum performance requirements it is the intention of the present document to specify terminal equipment requirements which enable manufacturers and service providers to enable good quality end-to-end speech performance as perceived by the user.

In addition to basic testing procedures, the present document describes advanced testing procedures taking into account further quality parameters as perceived by the user.

It is the intention of the present document to describe terminal performance parameters in such way that the remaining variation of parameters can be assessed purely by the E-model.

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 http://docbox.etsi.org/Reference.

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

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

[1] ETSI EN 300 726: "Digital cellular telecommunications system (Phase 2+) (GSM); Enhanced Full Rate (EFR) speech transcoding (GSM 06.60)".

[2] ETSI TS 126 171: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; General description (3GPP TS 26.171)".

[3] Recommendation ITU-T G.107: "The E-model: a computational model for use in transmission planning".

[4] Recommendation ITU-T G.108: "Application of the E-model: A planning guide".

[5] Recommendation ITU-T G.109: "Definition of categories of speech transmission quality".

[6] Recommendation ITU-T G.122: "Influence of national systems on stability and talker echo in international connections".

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

[8] Recommendation ITU-T G.723.1: "Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s".

[9] Recommendation ITU-T G.726: "40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)".

[10] Recommendation ITU-T G.729: "Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)".

[11] Recommendation ITU-T G.729.1: "G.729-based embedded variable bit-rate coder: An 8-32 kbit/s scalable wideband coder bitstream interoperable with G.729".

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

[13] Recommendation ITU-T P.57: "Artificial ears".

[14] Recommendation ITU-T P.58: "Head and torso simulator for telephonometry".

[15] Recommendation ITU-T P.64: "Determination of sensitivity/frequency characteristics of local telephone systems".

[16] Recommendation ITU-T P.79: "Calculation of loudness ratings for telephone sets".

[17] Recommendation ITU-T P.340: "Transmission characteristics and speech quality parameters of hands-free terminals".

[18] Recommendation ITU-T P.380: "Electro-acoustic measurements on headsets".

[19] Recommendation ITU-T P.501: "Test signals for use in telephonometry".

[20] Recommendation ITU-T P.502: "Objective test methods for speech communication systems using complex test signals".

[21] Recommendation ITU-T P.581: "Use of head and torso simulator for hands-free and handset terminal testing".

[22] IEC 61260-1: "Electroacoustics - Octave-band and fractional-octave-band filters - Part 1: Specifications".

[23] Recommendation ITU-T P.800.1: "Mean Opinion Score (MOS) terminology".

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

[25] ETSI TS 103 224: "Speech and multimedia Transmission Quality (STQ); A sound field reproduction method for terminal testing including a background noise database".

[26] Recommendation ITU-T P.863: "Perceptual objective listening quality assessment".

[27] Recommendation ITU-T P.863.1: "Application guide for Recommendation ITU-T P.863".

[28] Recommendation ITU-T P.1010: "Fundamental voice transmission objectives for VoIP terminals and gateways".

[29] IETF RFC 3550: "RTP: A Transport Protocol for Real-Time Applications".

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] ETSI EG 201 377-1: "Speech and multimedia Transmission Quality (STQ); Specification and measurement of speech transmission quality; Part 1: Introduction to objective comparison measurement methods for one-way speech quality across networks".

[i.2] ETSI EG 202 425: "Speech Processing, Transmission and Quality Aspects (STQ); Definition and implementation of VoIP reference point".

[i.3] ETSI EG 202 396-3: "Speech and multimedia Transmission Quality (STQ); Speech Quality performance in the presence of background noise; Part 3: Background noise transmission - Objective test methods".

[i.4] NIST Net.

NOTE: Available at https://www-x.antd.nist.gov/itg/nistnet/.

[i.5] Netem.

NOTE: Available at http://www.linuxfoundation.org/en/Net:Netem.

[i.6] Trace Control for Netem (TCN): "A. Keller, Trace Control for Netem, Semester Thesis SA-2006-15, ETH Zürich, 2006".

3 Definitions and abbreviations

3.1 Definitions

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

artificial ear: device for the calibration of earphones incorporating an acoustic coupler and a calibrated microphone for the measurement of the sound pressure and having an overall acoustic impedance similar to that of the median adult human ear over a given frequency band

codec: combination of an analogue-to-digital encoder and a digital-to-analogue decoder operating in opposite directions of transmission in the same equipment

Composite Source Signal (CSS): signal composed in time by various signal elements

diffuse field equalization: equalization of the HATS sound pick-up, equalization of the difference, in dB, between the spectrum level of the acoustic pressure at the ear Drum Reference Point (DRP) and the spectrum level of the acoustic pressure at the HATS Reference Point (HRP) in a diffuse sound field with the HATS absent using the reverse nominal curve given in table 3 of Recommendation ITU-T P.58 [14]

Ear Reference Point (ERP): virtual point for geometric reference located at the entrance to the listener's ear, traditionally used for calculating telephonometric loudness ratings

ear-Drum Reference Point (DRP): point located at the end of the ear canal, corresponding to the ear-drum position

freefield reference point: point located in the free sound field, at least in 1,5 m distance from a sound source radiating in free air

NOTE: In case of a head and torso simulator (HATS) in the centre of the artificial head with no artificial head present.

Head And Torso Simulator (HATS) for telephonometry: manikin extending downward from the top of the head to the waist, designed to simulate the sound pick-up characteristics and the acoustic diffraction produced by a median human adult and to reproduce the acoustic field generated by the human mouth

Mouth Reference Point (MRP): point located on axis and 25 mm in front of the lip plane of a mouth simulator

nominal setting of the volume control: when a receive volume control is provided, the setting which is closest to the nominal RLR of 2 dB

3.2 Abbreviations

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

AM-FM	Amplitude Modulation-Frequency Modulation
AMR	Adaptative Multi-Rate

AMR-NB	Adaptive Multi-Rate NarrowBand
CS	Composite Source
CSS	Composite Source Signal
DRP	ear Drum Reference Point
EC	Echo Canceller
EFR	Enhanced Full Rate
EL	Echo Loss
ERP	Ear Reference Point
ETH	Eidgenössische Technische Hochschule
FFT	Fast Fourrier Transform
GSM	Global System for Mobile communications
HATS	Head And Torso Simulator
IEC	International Electrotechnical Commission
IP	Internet Protocol
IPDV	IP Packet Delay Variation
ITU-T	International Telecommunication Union - Telecommunication standardization sector
MOS	Mean Opinion Score
MOS-LQOy	Mean Opinion Score - Listening Quality Objective

NOTE: y being N for narrow-band, M for mixed and S for superwideband. See Recommendation ITU-T P.800.1 [23].

MRP	Mouth Reference Point
NIST	National Institute of Standards and Technology
NLP	Non Linear Processor
PBX	Private Branch eXchange
PC	Personal Computer
PCM	Pulse Code Modulation
POLQA	Perceptual Objective Listening Quality Assessment
PLC	Packet Loss Concealment
PN	Pseudo-random Noise
POI	Point Of Interconnect
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RLR	Receive Loudness Rating
RMS	Root Mean Square
RTP	Real Time Protocol
SLR	Send Loudness Rating
STMR	SideTone Masking Rating
TCLw	Terminal Coupling Loss (weighted)
TCN	Trace Control for Netem
TDM	Time Division Multiplex
TOSQA	Telecommunication Objective Speech Quality Assessment
VAD	Voice Activity Detector

4 General considerations

4.1 Default coding algorithm

VoIP terminals shall support the coding algorithm according to Recommendation ITU-T G.711 [7] (both μ-law and A-law). VoIP terminals may support other coding algorithms.

NOTE: Associated Packet Loss Concealment (PLC) e.g. as defined in Recommendation ITU-T G.711 [7] appendix I should be used.

4.2 End-to-end considerations

In order to achieve a desired end-to-end speech transmission performance (mouth-to-ear) it is recommended that the general rules of transmission planning are carried out with the E-model of Recommendation ITU-T G.107 [3] taking into account that the E-model does not yet address headsets; this includes the a-priori determination of the desired category of speech transmission quality as defined in Recommendation ITU-T G.109 [5].

While, in general, the transmission characteristics of single circuit-oriented network elements, such as switches or terminals can be assumed to have a single input value for the planning tasks of Recommendation ITU-T G.108 [4], this approach is not applicable in packet based systems and thus there is a need for the transmission planner's specific attention.

In particular the decision as to which delay measured according to the present document should is acceptable or representative for the specific configuration is the responsibility of the individual transmission planner.

Recommendation ITU-T G.108 with its amendments [4] provides further guidance on this important issue.

The following optimum terminal parameters from a users' perspective need to be considered:

- Minimized delay in send and receive direction.
- Optimum loudness Rating (RLR, SLR).
- Compensation for network delay variation.
- Packet loss recovery performance.
- Maximized terminal coupling loss.

5 Test equipment

5.1 IP half channel measurement adaptor

The IP half channel measurement adaptor is described in ETSI EG 202 425 [i.2].

5.2 Environmental conditions for tests

The following conditions shall apply for the testing environment:

a) Ambient temperature: 15 °C to 35 °C (inclusive);

b) Relative humidity: 5 % to 85 %;

c) Air pressure: 86 kPa to 106 kPa (860 mbar to 1 060 mbar).

5.3 Accuracy of measurements and test signal generation

Unless specified otherwise, the accuracy of measurements made by test equipment shall be equal to or better than:

Table 1: Measurement accuracy

Item	Accuracy
Electrical signal level	±0,2 dB for levels ≥ -50 dBV ±0,4 dB for levels < -50 dBV
Sound pressure	±0,7 dB
Frequency	±0,2 %
Time	±0,2 %
Application force	± 2 N
Measured maximum frequency	20 kHz

Unless specified otherwise, the accuracy of the signals generated by the test equipment shall be better than:

Table 2: Accuracy of test signal generation

Quantity	Accuracy
Sound pressure level at Mouth Reference Point (MRP)	±3 dB for frequencies from 100 Hz to 200 Hz ±1 dB for frequencies from 200 Hz to 4 000 Hz ±3 dB for frequencies from 4 000 Hz to 14 000 Hz
Electrical excitation levels	±0,4 dB across the whole frequency range
Frequency generation	±2 % (see note)
Time	±0,2 %
Specified component values	±1 %
NOTE: This tolerance may be used to avoid measurements at critical frequencies, e.g. those due to sampling operations within the terminal under test.	

For terminal equipment which is directly powered from the mains supply, all tests shall be carried out within ±5 % of the rated voltage of that supply. If the equipment is powered by other means and those means are not supplied as part of the apparatus, all tests shall be carried out within the power supply limit declared by the supplier. If the power supply is a.c. the test shall be conducted within ±4 % of the rated frequency.

5.4 Network impairment simulation

At least one set of requirements is based on the assumption of an error free packet network, and at least one other set of requirements is based on a defined simulated malperformance of the packet network.

An appropriate network simulator has to be used, for example NIST net [i.4] (https://www-x.antd.nist.gov/itg/nistnet/) or Netem [i.5].

Based on the positive experience, STQ have made during the ETSI Speech Quality Test Events with "NIST Net" this will be taken as a basis to express and describe the variations of packet network parameters for the appropriate tests.

Here is a brief blurb about NIST Net:

> The NIST Net network emulator is a general-purpose tool for emulating performance dynamics in IP networks. The tool is designed to allow controlled, reproducible experiments with network performance sensitive/adaptive applications and control protocols in a simple laboratory setting. By operating at the IP level, NIST Net can emulate the critical end-to-end performance characteristics imposed by various wide area network situations (e.g. congestion loss) or by various underlying sub network technologies (e.g. asymmetric bandwidth situations of xDSL and cable modems).

NIST Net is implemented as a kernel module extension to the Linux™ operating system and an X Window System-based user interface application. In use, the tool allows an inexpensive PC-based router to emulate numerous complex performance scenarios, including: tunable packet delay distributions, congestion and background loss, bandwidth limitation, and packet reordering/duplication. The X interface allows the user to select and monitor specific traffic streams passing through the router and to apply selected performance "effects" to the IP packets of the stream. In addition to the interactive interface, NIST Net can be driven by traces produced from measurements of actual network conditions. NIST Net also provides support for user defined packet handlers to be added to the system. Examples of the use of such packet handlers include: time stamping/data collection, interception and diversion of selected flows, generation of protocol responses from emulated clients.

The key points of Netem can be summarized as follows:

- Netem is nowadays part of most Linux™ distributions, it only has to be switched on, when compiling a kernel. With Netem, there are the same possibilities as with nistnet, there can be generated loss, duplication, delay and jitter (and the distribution can be chosen during runtime). Netem can be run on a Linux™-PC running as a bridge or a router (Nistnet only runs on routers).

- With an amendment of Netem, TCN (Trace Control for Netem) [i.6] which was developed by ETH Zurich, it is even possible, to control the behaviour of single packets via a trace file. So it is for example possible to generate a single packet loss, or a specific delay pattern. This amendment is planned to be included in new Linux™ kernels, nowadays it is available as a patch to a specific kernel and to the iproute2 tool (iproute2 contains Netem).

- It is not advised to define specific distortion patterns for testing in standards, because it will be easy to adapt devices to these patterns (as it is already done for test signals). But if a pattern is unknown to a manufacturer, the same pattern can be used by a test lab for different devices and gives comparable results. It is also possible to take a trace of Nistnet distortions, generate a file out of this and playback the exact same distortions with Netem.

NOTE: NIST Net™, NETEM™, Linux™ and X Window System™ are examples of suitable products available commercially. This information is given for the convenience of users of the present document and does not constitute an endorsement by ETSI of these product(s).

5.5 Acoustic environment

Unless stated otherwise measurements shall be conducted under quiet and "anechoic" conditions. Depending on the distance of the transducers from mouth and ear a quiet office room may be sufficient e.g. for handsets where artificial mouth and artificial ear are located close to the acoustical transducers.

However, for some headsets or handset terminals with smaller dimension an anechoic room will be required.

In cases where real or simulated background noise is used as part of the testing environment, the original background noise shall not be noticeably influenced by the acoustical properties of the room.

In all cases where the performance of acoustic echo cancellers shall be tested a realistic room which represents the typical user environment for the terminal shall be used.

Standardized measurement methods for measurements with variable echo paths are for further study.

5.6 Influence of terminal delay on measurements

As delay is introduced by the terminal, care shall be taken for all measurements where exact position of the analysis window is required. It shall be checked that the test is performed on the test signal and not on any other signal.

6 Requirements and associated measurement methodologies

6.1 Notes

NOTE 1: In general the test methods as described in the present document apply. If alternative methods exist they may be used if they have been proven to give the same result as the method described in the present document. This will be indicated in the test report.

NOTE 2: Due to the time variant nature of IP connections delay variation may impair the measurements. In such cases the measurement has to be repeated until a valid measurement result is achieved.

6.2 Test setup

6.2.1 General

The preferred acoustical access to terminals is the most realistic simulation of the "average" subscriber. This can be made by using HATS (Head And Torso Simulator) with appropriate ear simulation and appropriate means to fix handset and headset terminals in a realistic and reproducible way to the HATS. HATS is described in Recommendation ITU-T P.58 [14], appropriate ears are described in Recommendation ITU-T P.57 [13] (type 3.3 and type 3.4 ear), a proper positioning of handsets under realistic conditions is to be found in Recommendation ITU-T P.64 [15].

The preferred way of testing a terminal is to connect it to a network simulator with exact defined settings and access points. The test sequences are fed in either electrically, using a reference codec or using the direct signal processing approach or acoustically using ITU-T specified devices.

When a coder with variable bit rate is used for testing terminal electro acoustical parameters, the bit rate recognized giving the best characteristics should be selected, e.g.:

- AMR-NB (ETSI TS 126 171 [2]): 12,2 kbit/s.
- Recommendation ITU-T G.729.1 [11]: 32 kbit/s.