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Kakovost prenosa govora in večpredstavnih vsebin (STQ) - Prenosne zahteve za ozkopasovne in širokopasovne domače prehode in druge medijske prehode po protokolu IP glede na kakovost storitev (QoS), kot jih dojema uporabnik

Speech and multimedia Transmission Quality (STQ) - Transmission Requirements for IP-based Narrowband and Wideband Home Gateways and Other Media Gateways from a QoS Perspective as Perceived by the User

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Foreword

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

Introduction

Traditionally, the analogue and digital telephones were interfacing switched-circuit 64 kbit/s PCM networks. With the fast growth of IP networks, packet-switched networks (VoIP) interfacing PSTN networks and mobile networks, as well as different types of IP-terminals, are being rapidly introduced. Different types of gateways are used to interconnect to such IP networks. 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 harmonized between these different types of network from an end-to-end perspective, including specifications for the edge points.

The present document covers IP-based narrowband and wideband home gateways and other media gateways. It aims to enhance the interoperability and end-to-end quality.

In contrast to other standards which define minimum performance requirements, it is the intention of the present document to specify gateway equipment requirements which enable manufacturers and service providers to enable end-to-end speech performance as perceived by the user. These requirements are absolutely necessary to ensure a good quality, but they are not sufficient. They have to be combined with requirements (and associated relevant measurement methods) for other elements in the transmission chain (core IP network, PSTN, terminals), as well as for the whole mouth-to-ear transmission path.

1 Scope

The present document provides speech transmission performance requirements for narrowband and wideband media gateways from a QoS perspective as perceived by the user. Media gateways can be network or home based, they may include a transcoding function. The present document covers the following types of IP-based media gateways:

- ATA (Analogue Terminal Adapter), home gateway IP to POTS
- ITA (ISDN Terminal Adapter), home gateway IP to ISDN
- IAD (Integrated Access device), home router including ATA or ITA
- Network based ATA and ITA
- Carrier grade media gateway, network gateway IP to TDM
- IP-to-IP media gateway, network gateway with transcoding and/or other media processing

DECT interfaces of media gateways are excluded from the present document and should be measured according to the relevant DECT standards.

Interfaces of media gateways used together with terminals as a system (i.e. connected via Ethernet or with a proprietary interface) are excluded in the present document and should be measured according to the relevant terminal standard.

If a media gateway includes more than one interface type (e.g. POTS and ISDN), each interface has to be dealt with differently.

The requirements available in the present document will ensure a high compatibility with IP-and TDM-based fixed and wireless terminals and networks, including DECT and mobile terminals.

It is the aim to optimize interoperability, the listening and talking quality and the conversational performance. Related requirements and test methods are defined in the present document.

The present document does not apply to media gateways with 4-wire analogue interfaces.

2 References

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

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

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

2.1 Normative references

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

[1] 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); AMR speech codec, wideband; General description (3GPP TS 26.171 version 6.0.0 Release 6)".

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

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

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

[6] ITU-T Recommendation G.100.1: "The use of the decibel and of relative levels in speechband telecommunications".

[7] ITU-T Recommendation G.111: "Loudness Ratings (LRs) in an international connection".

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

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

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

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

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

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

[14] ITU-T Recommendation G.1020: "Performance parameter definitions for quality of speech and other voiceband applications utilizing IP networks".

[15] ITU-T Recommendation P.50: "Artificial voices".

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

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

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

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

[20] ISO 3 (1973): "Preferred numbers - Series of preferred numbers".

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

[22] ETSI TS 102 971: "Access and Terminals (AT); Public Switched Telephone Network (PSTN); Harmonized specification of physical and electrical characteristics of a 2-wire analogue interface for short line interface".

[23] ETSI ES 201 970: "Access and Terminals (AT); Public Switched Telephone Network (PSTN); Harmonized specification of physical and electrical characteristics at a 2-wire analogue presented Network Termination Point (NTP)".

[24] ITU-T Recommendation G.168: "Digital network echo cancellers".

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

[26] ITU-T Recommendation G.722: "7 kHz audio-coding within 64 kbit/s".

[27] ITU-T Recommendation G.722.1: "Low-complexity coding at 24 and 32 kbit/s for hands-free operation in systems with low frame loss".

[28] ITU-T Recommendation G.722.2: "Wideband coding of speech at around 16 kbit/s using Adaptive Multi-Rate Wideband (AMR-WB)".

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

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 EG 202 396-1: "Speech and multimedia Transmission Quality (STQ); Speech quality performance in the presence of background noise; Part 1: Background noise simulation technique and background noise database".

[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] IETF RFC 2833: "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals".

[i.5] IETF RFC 4733: "RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals".

[i.6] ETSI TR 102 927: "Speech and multimedia Transmission Quality (STQ); Packet Loss Concealment (PLC) performance measurement setup for home networks".

3 Definitions and abbreviations

3.1 Definitions

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

0dBr point: reference point always located at the digital side of the gateway, for IP-IP gateways located at the input of the MGW under test

NOTE: See ITU-T Recommendation G.100.1 [6].

2-wire interface: in the context of the present document, the telephony analogue interface over 2-wires used in the local loop

4-wire interface: in the context of the present document, a 4-wire digital interface with separate channels for both directions, irrespective of the physical transmission technology

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

MGW with 2-wire interface: MGW with an analogue 2-wire interface (ATA)

MGW with 4-wire interface: MGW with only 4-wire interfaces, e.g. ITA, IP-to-IP and wireless access points

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

receive direction: the direction from packet switched interfaces towards a synchronous interface (e.g. ISDN, analogue) or between two packet switched interfaces (for media gateways with packet switched transport on only one side)

NOTE: For media gateways with packet switched transport on both sides (IP-to-IP-MGW), the requirements of the receive direction have to be applied in both directions.

receive interface: interface in the measurement setup, where a receive signal is injected and/or a send signal is measured.

send direction: direction from a synchronous interface (e.g. ISDN, analogue) towards a packet switched interface (for media gateways with packet switched interface on only one side)

NOTE: For media gateways with packet switched interfaces on both sides the requirements of the send direction are not relevant.

send interface: interface in the measurement setup, where a send signal is injected and/or a receive signal is measured

3.2 Abbreviations

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

ATA	Analogue Terminal Adapter
CLR	Circuit Loudness Rating
CSS	Composite Source Signal
DSL	Digital Subscriber Line
DSLAM	Digital Subscriber Line Access Multiplexer
DTMF	Dual Tone Multi Frequency
EL	Echo Loss
IAD	Integrated Access device
IP	Internet Protocol
ITA	ISDN Terminal Adapter
JLR	Junction Loudness Rating
MGW	Media GateWay
MOS-LQOy	Mean Opinion Score - Listening Quality Objective

NOTE: See ITU-T Recommendation P.800.1 [21].

NLP	Non Linear Processor
PCM	Pulse Code Modulation
PESQ™	Perceptual Evaluation of Speech Quality™
PLC	Packet Loss Concealment
PN	Pseudo-random Noise
POI	Point Of Interconnect
POTS	Plain Old Telephone Service
PSTN	Public Switched Telephone Network
QoS	Quality of Service
TCL	Terminal Coupling Loss
TCN	Trace Control for Netem™
TDM	Time Division Multiplexing
VAD	Voice Activity Detection
VoIP	Voice over Internet Protocol

4 General considerations

4.1 Default Coding Algorithm

Narrowband VoIP gateways shall support the coding algorithm according to ITU-T Recommendation G.711 [9] (both µ-law and A-law). VoIP gateways may support other coding algorithms.

Wideband VoIP gateways shall support the coding algorithm according to ITU-T Recommendation G.722 [26]. VoIP gateways may support other coding algorithms.

NOTE: Associated Packet Loss Concealment (PLC) e.g. as defined in ITU-T Recommendation G.711 [9] 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 ITU-T Recommendation G.107 [3]; this includes the a-priori determination of the desired category of speech transmission quality as defined in ITU-T Recommendation 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 ITU-T Recommendation 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 is acceptable or representative for the specific configuration is the responsibility of the individual transmission planner.

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

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

- Minimized delay in send and receive direction.
- Optimum loudness Rating (JLR).
- Compensation for network delay variation.
- Packet loss recovery performance.
- Maximized echo loss.
- Immunity to false detection of DTMF in speech signal.

4.3 Parameters to be investigated

4.3.1 Applicability of parameters to different MGWs

Table 1: Parameter applicability

	2-wire home and network MGW	4-wire MGW (excl. IP-to-IP MGW)	4-wire MGW (IP-to-IP-MGW)	wireless home MGW
6.2 Codec independent parameters				
6.2.1 Send frequency response	M	M	NA	M
6.2.2 Circuit Loudness Rating in Send	M	M	NA	M
6.2.3 Linearity Range for CLR(SND)	M	M	NA	M
6.2.4 Send Distortion	M	M	NA	M
6.2.5 Spurious Out-of-Band Signals in Send direction	M	NA	NA	NA
6.2.6 Send Noise	M	M	NA	M
6.2.7 Receive frequency response	M	M	MM	M
6.2.8 Circuit Loudness Rating in Receive	M	M	MM	M
6.2.9 Linearity Range for CLR(RCV)	M	M	MM	M
6.2.10 Receive Distortion	M	M	MM	M
6.2.11 Out-of-Band Signals in Wideband to Narrowband Transcoding	NA	M	M	M
6.2.12 Spurious Out-of-band Signals Narrowband to Wideband Transcoding	NA	M	M	M
6.2.13 Minimum activation level and sensitivity in Receive direction	FFS	FFS	FFS	FFS
6.2.14 Receive Noise	M	M	MM	M
6.2.15 Double Talk Performance				
6.2.15.1 Attenuation Range in Send Direction during Double Talk	M	M	M	M
6.2.15.2 Attenuation Range in Receive	M	M	M	M

	2-wire home and network MGW	4-wire MGW (excl. IP-to-IP MGW)	4-wire MGW (IP-to-IP-MGW)	wireless home MGW
Direction during Double Talk				
6.2.15.3 Detection of Echo Components during Double Talk	M	M	M	M
6.2.15.4 Minimum activation level and sensitivity of double talk detection	FFS	FFS	FFS	FFS
6.2.16 Switching characteristics				
6.2.16.1 Activation in Send Direction	M	M	NA	M
6.2.16.2 Activation in Receive Direction	M	M	M	M
6.2.16.3 Silence Suppression and Comfort Noise Generation	FFS	FFS	FFS	FFS
6.2.17 Background Noise Performance				
6.2.17.1 Performance in send direction in the presence of background noise	M	M	MM	M
6.2.17.2 Quality of Speech with Background Noise	M	M	MM	M
6.2.17.3 Quality of Background Noise Transmission (with Far End Speech)	M	M	MM	M
6.2.17.4 Quality of Background Noise Transmission (with Near End Speech)	M	M	MM	M
6.2.18 Quality of echo cancellation				
6.2.18.2 Echo Performance acc. To G.168	M	M	NA	M
6.2.18.3 TCLw	M	M	NA	M
6.2.18.4Temporal echo effects	M	M	NA	M
6.2.18.5 Spectral Echo Attenuation	M	M	NA	M
6.2.18.6 Occurrence of Artefacts	FFS	FFS	NA	FFS
6.2.19 Variant Impairments; Network dependant				
6.2.19.1 Clock accuracy send	M	M	MM	M
6.2.19.2 Clock accuracy receive	M	M	MM	M
6.2.19.3 Send delay variation	M	M	MM	M
6.2.20 Immunity to DTMF false detection in send direction	M	M	MM	M
6.3 Codec Specific Requirements				
6.3.1 Send Delay	M	M	NA	M
6.3.2 Receive Delay	M	M	NA	M
6.3.3 Delay for IP-to-IP MGW	NA	NA	MM	NA
6.3.4 Objective Listening Speech Quality MOS-LQO in Send direction	M	M	M	M
6.3.5 Objective Listening Speech Quality MOS-LQO in Receive direction	M	M	M	M
6.3.5.1 Efficiency of Packet Loss Concealment (PLC)	FFS	FFS	FFS	FFS
6.3.5.2 Efficiency of Delay Variation Removal	FFS	FFS	FFS	FFS

M: Mandatory
MM: Mandatory for both interfaces of the MGW
NA: Not applicable
FFS: For further study

5 Test equipment

5.1 IP half channel measurement adaptor

The IP half channel measurement adaptor is described in EG 202 425 [i.2]. Such an apparatus is required to code and insert audio signals into IP packets send to the IP receive interface of the gateway under test, as well as to capture and decode audio signals constituting the payload of IP packets received from the IP sending interface of the gateway under test.

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 2: Measurement Accuracy

Item	Accuracy
Electrical signal level	±0,2 dB for levels ≥ -50 dBV ±0,4 dB for levels < -50 dBV
Frequency	±0,2 %
Time	±0,2 %

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

Table 3: Accuracy of test signal generation

Quantity	Accuracy
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.	

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 Netem™.

The key points of Netem™ can be summarized as follows:

- Netem™ is 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™) 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™.

6 Requirements and associated Measurement Methodologies

Differences between different media gateway types are dealt with in the respective requirements.

In the case of IP-IP MGW packet based interfaces are provided at both sides of the gateway. Therefore the receive requirements apply, for both interfaces.

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 standard.

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.

6.1 Test setup

The preferred way of testing a gateway is to connect its interfaces to network simulators with exact defined settings and access points. The test sequences are fed in electrically, using a reference codec or using the direct signal processing approach.

When VoIP runs on the gateway under test only in conjunction with a registration by an application server (e.g. SIP proxy), the network simulator may need to provide also the registration functionality.

Alternatively, if for the IP-interfaces another technology than Ethernet is used (for instance DSL access, it may be necessary to add additional equipment in the test setup for connecting the measurement equipment (e.g. a DSLAM, if the IP-interface works over DSL). There should be no speech signal processing in this additional equipment (the media payload has to be passed transparent through this equipment, while e.g. header manipulation is allowed). The influence of this additional equipment (delay and eventually delay variation) has to be taken in account for the measurements.

NOTE 1: It is up to the testlab to identify potential time invariances or non linearities in the network used for interconnection and to take those effects into account properly.

With this setup it is possible to measure the parameters listed in the present document over a whole network, if the behaviour of the network is known.

In the present document, the terms "send" and "receive" can be found in the pictures of the relevant test setup.

When a coder with variable bit rate is used for testing the MGW parameters, the bit rate recognized giving the best characteristics and/or the ones commonly used should be selected, e.g.:

- AMR-NB (TS 126 171 [2]): 12,2 kbit/s.
- AMR-WB (G. 722.2 [28]): 12,65 kbit/s.
- ITU-T Recommendation G.729.1 [13]: 32 kbit/s.

NOTE 2: Although packet capturing and network simulation in figures 1 to 4 are shown in one box they may be separate devices.