



**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

RES/STQ-256

Keywords

narrowband, quality, speech, telephony, terminal,
VoIP

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 - NAF 742 C
Association à but non lucratif enregistrée à la
Sous-Préfecture de Grasse (06) N° 7803/88

Important notice

The present document can be downloaded from:

<http://www.etsi.org/standards-search>

The present document may be made available in electronic versions and/or in print. The content of any electronic and/or print versions of the present document shall not be modified without the prior written authorization of ETSI. In case of any existing or perceived difference in contents between such versions and/or in print, the only prevailing document is the print of the Portable Document Format (PDF) version kept on a specific network drive within ETSI Secretariat.

Users of the present document should be aware that the document may be subject to revision or change of status.

Information on the current status of this and other ETSI documents is available at

<https://portal.etsi.org/TB/ETSIDeliverableStatus.aspx>

If you find errors in the present document, please send your comment to one of the following services:

<https://portal.etsi.org/People/CommiteeSupportStaff.aspx>

Copyright Notification

No part may be reproduced or utilized in any form or by any means, electronic or mechanical, including photocopying and microfilm except as authorized by written permission of ETSI.

The content of the PDF version shall not be modified without the written authorization of ETSI.

The copyright and the foregoing restriction extend to reproduction in all media.

© ETSI 2017.
All rights reserved.

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 protected for the benefit of its Members.

GSM® and the GSM logo are trademarks registered and owned by the GSM Association.

Contents

Intellectual Property Rights	5
Foreword.....	5
Modal verbs terminology.....	5
Introduction	5
1 Scope	6
2 References	6
2.1 Normative references	6
2.2 Informative references.....	7
3 Definitions and abbreviations.....	8
3.1 Definitions.....	8
3.2 Abbreviations	8
4 General considerations	9
4.1 Default coding algorithm.....	9
4.2 End-to-end considerations	10
5 Test equipment	10
5.1 IP half channel measurement adaptor.....	10
5.2 Environmental conditions for tests.....	10
5.3 Accuracy of measurements and test signal generation	11
5.4 Network impairment simulation.....	11
5.5 Acoustic environment.....	12
5.6 Influence of terminal delay on measurements	12
6 Requirements and associated measurement methodologies.....	13
6.1 Notes	13
6.2 Test setup.....	13
6.2.1 General.....	13
6.2.2 Setup for handsets and headsets.....	14
6.2.3 Position and calibration of HATS.....	14
6.2.4 Test signal levels.....	15
6.2.5 Setup of background noise simulation	15
6.2.6 Setup of variable echo path.....	15
6.3 Coding independent parameters	16
6.3.1 Send frequency response	16
6.3.2 Send Loudness Rating (SLR).....	17
6.3.3 Mic mute.....	17
6.3.4 Linearity range for SLR	18
6.3.5 Send distortion	19
6.3.6 Out-of-band signals in send direction	19
6.3.7 Send noise.....	20
6.3.8 Sidetone Masking Rating STMR (mouth to ear)	20
6.3.9 Sidetone delay.....	21
6.3.10 Terminal Coupling Loss weighted (TCLw).....	22
6.3.11 Stability loss.....	22
6.3.12 Receive frequency response.....	23
6.3.13 Receive Loudness Rating (RLR)	26
6.3.14 Receive distortion	26
6.3.15 Out-of-band signals in receive direction	27
6.3.16 Minimum activation level and sensitivity in receive direction	27
6.3.17 Receive noise	27
6.3.18 Automatic level control in receive	28
6.3.19 Double talk performance	28
6.3.19.1 General	28
6.3.19.2 Attenuation range in send direction during double talk $A_{H,S,dt}$	28

6.3.19.3	Attenuation range in receive direction during double talk $A_{H,R,dt}$	29
6.3.19.4	Detection of echo components during double talk	30
6.3.19.5	Minimum activation level and sensitivity of double talk detection.....	31
6.3.20	Switching characteristics	31
6.3.20.1	Note.....	31
6.3.20.2	Activation in send direction	31
6.3.20.3	Silence suppression and comfort noise generation.....	32
6.3.21	Background noise performance	32
6.3.21.1	Performance in send direction in the presence of background noise.....	32
6.3.21.2	Speech quality in the presence of background noise.....	33
6.3.21.3	Quality of background noise transmission (with far end speech).....	34
6.3.22	Quality of echo cancellation	34
6.3.22.1	Temporal echo effects	34
6.3.22.2	Spectral echo attenuation	35
6.3.22.3	Occurrence of artefacts	36
6.3.22.4	Variable echo path.....	36
6.3.23	Variant impairments; network dependant	36
6.3.23.1	Clock accuracy send.....	36
6.3.23.2	Clock accuracy receive	36
6.3.23.3	Send packet delay variation.....	37
6.3.24	Send and receive delay - round trip delay	37
6.4	Codec specific requirements.....	39
6.4.1	Objective listening speech quality MOS-LQO in send direction.....	39
6.4.2	Objective listening quality MOS-LQO in receive direction	40
6.4.3	Quality of jitter buffer adjustment	42
Annex A (informative):	Processing delays in VoIP terminals	44
Annex B (informative):	Example IP delay variation.....	47
Annex C (informative):	Bibliography	48
History		49

Intellectual Property Rights

Essential patents

IPRs essential or potentially essential to the present document may have been declared to ETSI. The information pertaining to these essential IPRs, if any, is 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 IPR Policy, no investigation, 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.

Foreword

This final draft ETSI Standard (ES) has been produced by ETSI Technical Committee Speech and multimedia Transmission Quality (STQ), and is now submitted for the ETSI standards Membership Approval Procedure.

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

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".