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ETSI STANDARD

**Speech and multimedia Transmission Quality (STQ);
Transmission requirements for wideband
VoIP loudspeaking and handsfree terminals
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, wideband terminals providing higher audio-bandwidth and 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. Due to the unique characteristics of the IP networks including packet loss, delay, etc. new performance specification, as well as appropriate measuring methods, will have to be developed. Terminals are getting increasingly complex.

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 requirements for wideband VoIP loudspeaking and hands-free 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 8 kHz wideband VoIP loudspeaking and hands-free terminals; it addresses all types of IP based terminals, including wireless, softphones and group audio terminals.

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.

NOTE: The present document does not concern headset terminals.

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 I-ETS 300 245-6: "Integrated Services Digital Network (ISDN); Technical characteristics of telephone terminals; Part 6: Wideband (7 kHz), loudspeaking and hands free telephony".
- [2] Recommendation ITU-T G.108: "Application of the E-model: A planning guide".
- [3] Recommendation ITU-T G.109: "Definition of categories of speech transmission quality".
- [4] Recommendation ITU-T G.722: "7 kHz audio-coding within 64 kbit/s".
- [5] Recommendation ITU-T G.722.1: "Low-complexity coding at 24 and 32 kbit/s for hands-free operation in systems with low frame loss".
- [6] Recommendation ITU-T G.722.2: "Wideband coding of speech at around 16 kbit/s using Adaptive Multi-Rate Wideband (AMR-WB)".
- [7] 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".
- [8] Recommendation ITU-T P.56: "Objective measurement of active speech level".
- [9] Recommendation ITU-T P.58: "Head and torso simulator for telephonometry".
- [10] Recommendation ITU-T P.79: "Calculation of loudness ratings for telephone sets".
- [11] Recommendation ITU-T P.310: "Transmission characteristics for narrow-band digital handset and headset telephones".
- [12] Recommendation ITU-T P.340: "Transmission characteristics and speech quality parameters of hands-free terminals".