

**Speech and multimedia Transmission Quality (STQ);
Transmission requirements for
wideband VoIP loudspeaking and
handsfree terminals 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), and is now submitted for the ETSI standards Membership Approval Procedure.

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. 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 with specifications for traditional digital terminals. However, 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, advanced signal processing is used to address the IP specific issues.

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

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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

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>.

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2.1 Normative references

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 telephony terminals; Part 6: Wideband (7 kHz), loudspeaking and hands free telephony".
- [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.108: "Application of the E-model: A planning guide".
- [4] ITU-T Recommendation G.109: "Definition of categories of speech transmission quality".
- [5] ITU-T Recommendation G.122: "Influence of national systems on stability and talker echo in international connections".
- [6] ITU-T Recommendation G.131: "Talker echo and its control".
- [7] ITU-T Recommendation G.711: "Pulse code modulation (PCM) of voice frequencies".
- [8] ITU-T Recommendation G.722: "7 kHz audio-coding within 64 kbit/s".
- [9] 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".
- [10] 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".
- [11] ITU-T Recommendation G.1020: "Performance parameter definitions for quality of speech and other voiceband applications utilizing IP networks".
- [12] ITU-T Recommendation P.50: "Artificial voices".