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ETSI Guide

Speech Processing, Transmission and Quality Aspects (STQ); Objectives and principles for the transmission performance of multiple interconnected networks that aim to provide "traditional quality" telephony services

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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

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Foreword

This ETSI Guide (EG) has been produced by ETSI Technical Committee Speech Processing, Transmission and Quality Aspects (STQ).

1 Scope

The present document specifies a simple set of objectives, principles and responsibilities for the transmission performance of multiple interconnected networks that provide "traditional quality (POTS)" circuit switched telephony services.

The objectives, principles and responsibilities take account of the liberalization of telephony services and the interconnection of several separate networks, each with different topologies, in the provision of telephony connections.

The present document applies to:

- national and international networks;
- Digital Networks and Integrated Digital Networks (i.e. networks where the only analogue component may be the local loop).

The present document applies in the cases where the telephony service is:

- contracted between the network operator and the customer/end user; and
- contracted through a service provider.

It applies where:

- the caller pays for the call;
- the recipient pays for the call (e.g. the 800 service); or
- the cost of the call is shared by the caller and the recipient.

The present document does not apply to any segment of calls where either the calling or called terminal is a mobile "terminating network".

2 References

References are either specific (identified by date of publication and/or edition number or version number) or non-specific.

- For a specific reference, subsequent revisions do not apply.
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  - if it is accepted that it will be possible to use all future changes of the referenced document for the purposes of the referring document;
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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 indispensable for the application of the present document. For dated references, only the edition cited applies. For non-specific references, the latest edition of the referenced document (including any amendments) applies.

[1] ITU-T Recommendation G.113 (1996): "Transmission impairments".

[2] ITU-T Recommendation G.114 (1996): "One -way transmission time".

[3] ITU-T Recommendation G.126 (1993): "Listener echo in telephone networks".

[4] ITU-T Recommendation G.131 (1996): "Control of talker echo".

[5] ITU-T Recommendation G.168 (1997): "Digital network echo cancellers".

[6] ITU-T Recommendation G.175 (1997): "Transmission planning for private/public network interconnection of voice traffic".

[7] ITU-T Recommendation G.711 (1988): "Pulse code modulation (PCM) of voice frequencies".

[8] ITU-T Recommendation G.821 (1996): "Error performance of an international digital connection operating at a bit rate below the primary rate and forming part of an integrated services digital network".

[9] ITU-T Recommendation G.822 (1988): "Controlled slip rate objectives on an international digital connection".

[10] ITU-T Recommendation G.826 (1996): "Error performance parameters and objectives for international, constant bit rate digital paths at or above the primary rate".

[11] ITU-T Recommendation Q.551 (1996): "Transmission characteristics of digital exchanges".

[12] ITU-T Recommendation Q.552 (1996): "Transmission characteristics at 2-wire analogue interfaces of digital exchanges".

[13] ETSI EG 201 050: "Speech Processing, Transmission and Quality Aspects (STQ); Overall Transmission Plan Aspects for Telephony in a Private Network".

[14] ETSI EN 300 462-1-1: "Transmission and Multiplexing (TM); Generic requirements for synchronization networks; Part 1-1: Definitions and terminology for synchronization networks".

[15] ETSI EN 300 462-6-1: "Transmission and Multiplexing (TM); Generic requirements for synchronization networks; Part 6-1: Timing characteristics of primary reference clocks".

[16] ISO/IEC 11573 (1994): "Information technology - Telecommunications and information exchange between systems - Synchronization methods and technical requirements for Private Integrated Services networks".

2.2 Informative references

The following referenced documents are not essential to the use of the present document but they assist the user with regard to a particular subject area**.** For non-specific references, the latest version of the referenced document (including any amendments) applies.

Not applicable.

3 Definitions and abbreviations

3.1 Definitions

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

network operator: organization that runs a network (switches and transmission) on which a service is provided

service provider: organization that offers and provides the telephony service to the customer

NOTE: This may be the same organization that is the network operator.

3.2 Abbreviations

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

A/D	Analogue to Digital
ADPCM	Adaptive Differential Pulse Code Modulation
ATM	Asynchronous Transfer Mode
D/A	Digital to Analogue
NTP	Network Termination Point
QDU	Quantizing Distortion Units
RLR	Receive Loudness Rating
SLR	Send Loudness Rating
VAD	Voice Activity Detection

4 Principles

4.1 Responsibility

With the development of competition so that a substantial proportion of calls pass across more than one network, it is no longer realistic to consider that any operator has overall responsibility for the quality of a call from one user-network interface to the other unless the call remains wholly on-net. Responsibility implies choice and control, but in many cases this choice and control do not exist.

EXAMPLE 1: Call termination, where normally there is only one operator who serves a given number and the operator who is paid to handle the call has no choice over its delivery.

EXAMPLE 2: Call origination for carrier selection or pre-selection, where the subscriber chooses their access operator and their call handling operator separately such that the call handling operator has no control over the choice of access operator.

In both these cases the operator who is paid to handle the call may have to pay the other operator but has no effective control since the other operator is in a monopoly position. Furthermore the calling user/subscriber has no control over the terminating operator selected by their correspondents.

In this situation, adequate end-to-end quality can be achieved only by all operators following standards and where there are choices that might involve incompatibilities by exchanging information and cooperating so that incompatibilities can be avoided.

4.2 Technical requirements

The following technical requirements apply to the segment of the call for which the operator is responsible, in accordance with clause 4.1.

4.2.1 Loudness

The transmission level point at the digital exchange should be 0 dBr.

Loss in analogue access sections (i.e. local exchange lines) should be such that the SLR at the A/D conversion point is in the range 7 dB to 10 dB and the RLR in the range 1 dB to 4 dB, when analogue telephones with the nominal SLR/RLR values defined in national or harmonized standards are attached to the NTPs.

Wherever possible, signal level adjustment should be made in the analogue domain. Digital loss or gain pads limit the available level range and increase signal distortion and should not be used if possible.

4.2.2 Encoding and Low Bit Rate Coding

The encoding method used at interconnection points should be agreed bilaterally and conform to an ITU-T or European standard.

NOTE 1: The most common method is the one specified in ITU-T Recommendation G.711 [7].

Consideration should be given to the ability of the network to carry in-band data, facsimile and modem access to Internet providers. Particular consideration should be given to the trend towards increasing data rates.

Low bit rate coding adversely affects speech quality. The number of these devices should be limited and the impairments introduced should be minimized.

NOTE 2: The effects of transcoding and compression are given as equipment impairment factors described in ITU-T Recommendation G.113 [1]. The use of quantizing distortion units (QDU) is restricted to the effects of encoding and decoding in accordance with ITU-T Recommendation G.711 [7]. QDUs should not be used for other coding methods. Guidance for planners can be found in ITU-T Recommendation G 175 [6] and EG 201 050 [13].

4.2.3 Voice activity detection and speech multiplexing

Some systems are especially designed to provide a flexible "bandwidth on demand" feature, utilizing a number of 64 kbit/s-channels in part of the connection in a more economical way, mainly for data transmission. For speech channels "Voice Activity Detection" (VAD) will reduce costs in a way similar to low bit-rate coding. The reduction of the transmitted bit-rate is performed by detecting speech pauses (VAD).

The additional delay introduced by VAD should be taken into account. Systems that introduce significant delay may already be equipped with integrated echo cancellers. The transmission parameters of VAD devices should be considered carefully during planning, mainly in conjunction with echo cancelling in other sections of the connection.

VAD devices generally introduce a measure of temporal clipping of speech signals. Care should be taken with tandeming of such devices in order to minimize degradation to the intelligibility of the speech.

NOTE: Some non-standard VAD systems that include integrated echo cancellers may also insert loss.

4.2.4 Bit integrity

Bit integrity is possible across a network only where the path is wholly digital. It may be required for services such 64 kbit/s unrestricted but is not required for speech. Signalling processing devices such as echo cancellers, low bit rate coders and digital loss and gain pads corrupt bit integrity. If there is to be an option for bit integrity then is should be possible to disable such devices.

4.2.5 Absolute delay

Irrespective of the effect of delay on echo, absolute delay should be minimized. Absolute delay does not impair the intelligibility of speech but if the total delay exceeds around 150 ms from mouth to ear, it begins to affect the interactivity of conversations. Therefore if practicable large delays of this magnitude should be avoided for speech. Detailed guidance is given in ITU-T Recommendation G.114 [2].

NOTE: Some low bit rate coders introduce high values of delay. Some terminal equipment also introduce high values of delay.

4.2.6 Echo control

The network operator should plan on the assumptions that:

- there will be up to 5 ms of one-way delay on the customer side of the NTP;
- terminals will not include echo reduction techniques.

Echo control should normally be provided using echo cancellation rather than echo suppression.

The provision by the network of additional echo cancellation facilities (or additional performance by these facilities) should be a matter for commercial negotiation. For example, extra cancellation may be provided at an appropriate extra charge.

NOTE 1: Terminals that introduce higher delays (e.g. some cordless telephones) normally provide their own echo reduction to compensate for the additional delay.

NOTE 2: Echo control devices in the network do not normally reduce acoustic echo in terminals because they are designed to reduce electrical echo originating in hybrids.

ITU-T Recommendations G.126 [3] and G.131 [4] provide guidance on echo. Echo cancellers should meet the requirements of ITU-T Recommendation G.168 [5].

4.2.7 Noise, crosstalk and group delay distortion

Analogue local access loops and A/D and D/A conversion systems should be designed to achieve performance in respect of noise, crosstalk, and group delay distortion that at least meets the levels recommended in ITU-T Recommendations Q.551 [11] and Q.552 [12].

4.2.8 Error performance

Digital transmission equipment should be used that is designed to ensure that the error performance specified in the G.820 series of Recommendations is exceeded by a substantial margin in normal operating conditions.

NOTE: When faults are not present, errors in transmitted information are normally caused by temporary local electromagnetic phenomena. The levels in these recommendations allow for the occurrence of these phenomena. Equipment should be designed to perform much better than these requirements in the absence of these phenomena.

4.2.9 Synchronization

Proper synchronization design is part of the network planning strategy, because synchronization impairments will affect the quality of the calls: the networks should be synchronized as defined in the series of documents EN 300 462-1-1 [14] to EN 300 462-6-1 [15] and ISO/IEC 11573 [16], in order to achieve the slip rate objectives defined in ITU-T Recommendation G.822 [9].

4.3 Provision of Information

To facilitate the provision of transmission quality in the manner described in the present document, network operators should either:

- publish; or
- provide on request from customers or operators of interconnected networks,information on the transmission impairments incurred within their own networks.

Information on the expected values of delay and the use of any compression techniques in fixed networks is especially important.

Resellers who provide a reduced level of transmission quality for lower price calls should inform their customers of the nature and extent of the quality reduction in terms understandable to the customer.