EG 202 086 V1.1.1 (1999-02)

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

Reference

DEG/STQ-00003 (jbc00icq.PDF)

Keywords

interworking, quality, speech, telephony, transmission

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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

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.
- A non-specific reference to an ETS shall also be taken to refer to later versions published as an EN with the same number.

[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] Void.

[9] 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".

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

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

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

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

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

[15] EN 300 462-1-1: "Transmission and Multiplexing (TM); Generic requirements for synchronization networks".

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

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

3 Definitions and abbreviations

3.1 Definitions

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

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

service provider: the organization that offers and provides the telephony service to the customer. 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
ATM	Asynchronous Transfer Mode
D/A	Digital to Analogue
NTP	Network Termination Point

POI	Point of Interconnection
QDU	Quantizing Distortion Units
RLR	Receive Loudness Rating
SLR	Send Loudness Rating
VAD	Voice Activity Detection

4 Principles

4.1 Responsibility

The network operator who charges the customer (calling or called party or their agent) should be responsible for the transmission quality of the segment of the connection for which the charge is made.

NOTE 1: In most cases, one operator will thus be responsible for transmission quality from NTP to NTP. In the case where the customer calling party exercises a carrier selection option, the network operator that charges for the call and takes responsibility for quality may not be the operator of the network to which the customer's terminal is connected directly.

In some cases calls may be carried across interconnected networks where the customer makes separate payments to different operators, for example, separate payments for local access and trunk segments. In these cases, there is separate responsibility for different segments of the call.

Where the call is paid by the recipient, or shared between the caller and recipient, the operator that provides the service to the recipient is responsible (because the tariff level is set through this operator).

Where the service is sold through a service provider, the network operator should be responsible to the service provider, and the service provider should be responsible to the customer.

Where a network operator does not provide the whole of the connection for which it is responsible, the operator should fulfil its responsibility through the conditions of its interconnection contract with the operator to which it interconnects physically. There may be a chain of interconnections to which this principle applies.

NOTE 2: This is not intended to be a regulatory or legal document. The responsibility referred to is that which arises naturally from the provision of a service within a commercial relationship. The precise nature of the responsibility will depend on the details of the service and transmission performance offered, and the regulatory environment of the country concerned. Where network operators are not able to obtain adequate commitments from other interconnected operators, they may not be able to specify a particular level of transmission quality for connections involving those operators.

4.2 Technical requirements

The following technical requirements apply to the segment of the call for which the operator is responsible, in accordance with subclause 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 to 10 dB and the RLR in the range 1 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 [14].

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 [12] and Q.552 [13].

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 planing strategy, because synchronization impairments will affect the quality of the calls: the networks should be synchronized as defined in the series of documents EN 300 462-1-1/6-1 [15], [16] and ISO/IEC 11573 [17], in order to achieve the slip rate objectives defined in ITU-T Recommendation G.822 [10].

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.

4.4 Future developments

Due to the following factors:

- the replacement of analogue technology with digital;
- the introduction of ATM based transmission and switching;
- greater use of dynamic management of network configurations;
- increasing use of digital radio systems in the local loop and in cordless terminal equipment.

The following changes have occurred, or are occurring, in transmission quality:

- the dominant impairment is echo rather than loss;
- the transmission delays in connections are increasing and becoming less easy to predict.

These changes make it more difficult for network operators to follow an efficient strategy for the deployment of echo control devices. Strategies for echo control need further study. One option is to devise a system for the on-line exchange of delay information through the signalling system and this option is under study.

Bibliography

The following material, though not specifically referenced in the body of the present document (or not publicly available), gives supporting information.

ITU-T Recommendation G.763 (1998): "Digital circuit multiplication equipment using ADPCM (Recommendation G.726) and digital speech interpolation".

History

Document history		
V1.1.1	December 1998	Membership Approval Procedure MV 9905: 1998-12-01 to 1999-01-29
V1.1.1	February 1999	Publication

ISBN 2-7437-2879-5
Dépôt légal : Février 1999