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ETSI Guide

**Speech Processing, Transmission and Quality Aspects (STQ);
User related QoS parameter definitions and measurements;
Part 2: Voice telephony, Group 3 fax,
modem data services and SMS**

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Contents

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Foreword

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

The present document is part 2 of a multi-part deliverable covering Speech Processing, Transmission and Quality Aspects (STQ); User related QoS parameter definitions and measurements, as identified below:

Part 1: "General";

Part 2: "Voice telephony, Group 3 fax, modem data services and SMS";

Part 3: "QoS parameters specific to Public Land Mobile Networks (PLMN)";

Part 4: "Internet access".

EG 202 057-1 [i.21] contains general user related QoS parameter definitions and measurement methods that can be applied to any service.

EG 202 057-2 (the present document) contains user related QoS parameter definitions and measurement methods for voice, Group 3 fax, modem data services and SMS accessed via the public telecommunication network. The data parameters are specified for the case where an ITU-T Recommendations V.90 [i.18] and V.92 [i.19], compliant modem is used since this kind of modem is in common use.

EG 202 057-3 [i.22] contains user related QoS parameter definitions and measurement methods specific to public land mobile networks (PLMN).

EG 202 057-4 [i.23] contains user related QoS parameter definitions and measurement methods specific to Internet access.

The present document takes into account as far as practicable the following eight principles:

1) QoS parameters should be easily understood by the public, and be useful and important to them.

2) All parameters are applicable at the network termination point (where appropriate).

3) Where measurements are possible they should be made on the customer's premises, using in-service lines.

NOTE: Literally principles 2 and 3 imply that all measurements should be carried out at the NTP, which would require co-operation by users and be excessively intrusive, as it would require many visits to the premises of users. Measurements at the subscriber side of the local exchange (e.g. at the MDF or other possible connection point/distribution frame in the access network) generally give an adequate representation of the quality that would be perceived at the NTP for the parameters defined in the present document, and so this approach is used because it is more practicable and meets the underlying objectives of these principles.

4) To be as realistic as possible, real traffic rather than test calls should be used as a basis of the measurements, wherever possible.

5) Parameters should be capable of verification by independent organizations. This verification might be made by direct measurements or by audit of service provider's measurements.

6) The accuracy of QoS values should be set to a level consistent with measurement methods being as simple as possible with costs as low as possible.

7) The parameters are designed for both statistical and individual application. The statistical values should be derived by the application of a simple statistical function to the individual values. The statistical function should be specified in this multi-part deliverable. This multi-part deliverable should also contain guidelines on how statistically significant samples should be selected.

8) The statistical functions should be designed so QoS figures from different service providers can be compared easily by users and in particular consumers.

1 Scope

The present document contains definitions and measurement methods for a range of user perceivable Quality of Service (QoS) parameters. The purpose of these parameters is to define objective and comparable measures of the QoS delivered to users/customers for use by users/customers. The present document applies to any telecommunication service, however, some parameters may have a limited application.

The present document is intended to provide a menu from which individual items can be selected. There is no obligation to use any or all of the parameters.

The QoS parameters are related primarily to services and service features and not to the technology used to provide the services. Therefore the parameters should be capable of use when the services are provided on new technologies such as IP and ATM or other packet switched technologies as well as on circuit switched technologies.

The establishment of target values for QoS is beyond the scope of the present document. The QoS parameters listed in the present document are also not intended to assess the complete QoS of a telecommunication service. The present document provides a set of QoS parameters that covers specific user related QoS aspects rather than a complete list of QoS parameters. This set has been chosen to address areas where monitoring of QoS is likely to be most worthwhile, i.e. the areas that are most likely to be affected by any QoS problems.

If stakeholders wish to examine other QoS aspects they are recommended to follow the general approach of the present document - as far as practicable - as a basis for the development of definitions and measurement methods for new specific QoS parameters.

The set of QoS parameters is designed to be understood by the users of various telecommunications services. Sub-sets of these parameters can be selected for use in different circumstances. For example a specific parameter might be relevant for many users in some countries or markets but the same parameter might not be of relevance in others. Therefore stakeholders - users, customers, regulators, service providers, network operators and other parties interested in the use of QoS parameters - should decide in co-operation, which parameters and which measures should be used in their particular situation. This decision should take account of:

- The precise purpose for which they will be used.
- The general level of quality achieved by most operators/providers.
- The degree to which the parameters will provide a reliable comparison of performance.
- The cost of measuring and reporting each parameter.

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.
- Non-specific reference may be made only to a complete document or a part thereof and only in the following cases:
  - 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;
  - for informative references.

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.

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.

Not applicable.

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.

[i.1] ETSI EG 201 769: "Speech Processing, Transmission and Quality Aspects (STQ); QoS parameter definitions and measurements; Parameters for voice telephony service required under the ONP Voice Telephony Directive 98/10/EC".

[i.2] ETSI ETS 300 905: "Digital cellular telecommunications system (Phase 2+) (GSM); Teleservices supported by a GSM Public Land Mobile Network (PLMN) (GSM 02.03)".

[i.3] ETSI EN 300 659 (all parts) : "Access and Terminals (AT); Analogue access to the Public Switched Telephone Network (PSTN); Subscriber line protocol over the local loop for display (and related) services".

[i.4] Directive 98/10/EC of the European Parliament and of the Council of 26 February 1998 on the application of open network provision (ONP) to voice telephony and on universal service for telecommunications in a competitive environment.

[i.5] ITU-T Recommendation E.180: "Technical characteristics of tones for the telephone service".

[i.6] ITU-T Recommendation E.425: "Internal automatic observations".

[i.7] ITU-T Recommendation E.451: "Facsimile call cut-off performance".

[i.8] ITU-T Recommendation E.452: "Facsimile modem speed reductions and transaction time".

[i.9] ITU-T Recommendation E.453: "Facsimile image quality as corrupted by transmission-induced scan line errors".

[i.10] ITU-T Recommendation E.800: "Terms and definitions related to quality of service and network performance including dependability".

[i.11] ITU-T Recommendation G.107: "The E-model, a computational model for use in transmission planning".

[i.12] ITU-T Recommendation P.862.1: "Mapping function for transforming P.862 raw result scores to MOS-LQO".

[i.13] ITU-T Recommendation P.862.2: "Wideband extension to Recommendation P.862 for the assessment of wideband telephone networks and speech codecs".

[i.14] ITU-T Recommendation G.109: "Definition of categories of speech transmission quality".

[i.15] ITU-T Recommendation T.4: "Standardization of Group 3 facsimile terminals for document transmission".

[i.16] ITU-T Recommendation I.210: "Principles of telecommunication services supported by an ISDN and the means to describe them".

[i.17] ITU-T Recommendation T.22: "Standardized test charts for document facsimile transmissions".

[i.18] ITU-T Recommendation V.90: "A digital modem and analogue modem pair for use on the Public Switched Telephone Network (PSTN) at data signalling rates of up to 56 000 bit/s downstream and up to 33 600 bit/s upstream".

[i.19] ITU-T Recommendation V.92: "Enhancements to Recommendation V.90".

[i.20] ITU-T Recommendation Q.850: "Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part".

[i.21] ETSI EG 202 057-1: "Speech Processing, Transmission and Quality Aspects (STQ); User related QoS parameter definitions and measurements; Part 1: General".

[i.22] ETSI EG 202 057-3: "Speech Processing, Transmission and Quality Aspects (STQ); User related QoS parameter definitions and measurements; Part 3: QoS parameters specific to Public Land Mobile Networks (PLMN)".

[i.23] ETSI EG 202 057-4: "Speech Processing, Transmission and Quality Aspects (STQ); User related QoS parameter definitions and measurements; Part 4: Internet access".

[i.24] ITU-T Recommendation P.862: "Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs ".

[i.25] ITU-T Recommendation P.561: "In-service non-intrusive measurement device - Voice service measurements".

[i.26] ITU-T Recommendation P.562: "Analysis and interpretation of INMD voice-service measurements".

[i.27] ITU-T Recommendation P.563: "Single-ended method for objective speech quality assessment in narrow-band telephony applications".

[i.28] ETSI TR 101 949: "Speech Processing, Transmission and Quality Aspects (STQ); QoS parameter definitions and measurements for use in network-to-network narrowband interconnection".

[i.29] ITU-T Recommendation G.114: "One-way transmission time".

[i.30] ITU-T Recommendation G.113: "Transmission impairments due to speech processing".

3 Definitions and abbreviations

3.1 Definitions

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

access line: connection from the Network Termination Point (NTP) to the entry point to the local switch or remote concentrator, whichever is the nearer

NOTE: In many cases this is the main distribution frame.

access network operator: organization that provides the access line

NOTE: In many cases the access network operator will be the direct service provider, but if the line is unbundled, the direct service provider would be a separate organization.

call by call carrier selection: form of carrier selection where the user dials a carrier access code to indicate which carrier is to route the call

carrier access code: code that the user may or needs to dial before the national (significant) number when dialling an access line in another telecommunications network, so that the call is routed by the carrier of his choice

customer: party that pays for the telecommunication service(s) provided

NOTE: Customers can generally be categorized as business or residential; the definition of business and residential customers is left to individual service providers. Service providers who receive interconnect services from other service providers are not considered to be customers for the purpose of the present document. The term "customer" is equivalent to "subscriber", which is used in Directive 98/10/EC [i.4].

data service: telecommunications service involving the transport of data via the PTN such that any user can use equipment connected to a network termination point to exchange data with another user of equipment connected to another termination point

direct service: service where the service provider that provides the telecommunication service(s) also provides the access network or rents an unswitched local loop (unbundled local loop) to use for the provision of the service to the customer

fax service: telecommunications service of transport of facsimile via the PTN such that any user can use equipment connected to a network termination point to exchange facsimiles with another user of equipment connected to another termination point

indirect service: service where the service provider that provides the telecommunication service(s) does not provide the access network but is selected by the customer or user using a form of call by call carrier selection or carrier preselection

network operator: organization that provides a network for the provision of a public telecommunication service

NOTE: If the same organization also offers services it also becomes a service provider.

Network Termination Point (NTP): physical point at which a user is provided with access to a public telecommunications network

ported number: subscriber number (directory number) where the location of the NTP and/or the identity of the service provider has changed after the number was originally allocated

preselection: form of carrier selection where the customer informs his access network operator which carrier is to route all or a particular subset of his calls, unless call by call carrier selection is used

Public Telecommunications Network (PTN): telecommunications network used wholly or partly for the provision of publicly available telecommunications services

Quality of Service (QoS): collective effect of service performance which determines the degree of satisfaction of a user of the service

NOTE: See ITU-T Recommendation E.800 [i.10].

service provider: organization that offers a telecommunication service to the customer and/or user

NOTE: A service provider need not be a network operator.

Short Message Service (SMS): telecommunications service involving the transport of a short alphanumeric message (160 alphanumeric characters) via the PTN such that any user can use equipment connected to a network termination point to exchange these messages with another user of equipment connected to another termination point

NOTE: See also ETS 300 905 [i.2] (GSM networks) and EN 300 659 [i.3] (fixed networks).

stakeholder: party having an interest in the level of quality of a service

supplementary service: additional service that modifies or supplements a basic telecommunication service

NOTE: Consequently, it cannot be offered to a customer as a stand-alone service; it has to be offered in association with a basic telecommunication service. The same supplementary service may be common to a number of basic telecommunication services. (See ITU-T Recommendation I.210 [i.16]).

telecommunications: technical process of sending, transmitting and receiving any kind of message in the form of signs, voice, images or sounds by means of telecommunications systems

telecommunication services: provision of telecommunications and the provision of other additional services that are closely related to the provision of telecommunications like

EXAMPLE: Billing, directory services.

telecommunications systems: technical equipment or systems capable of sending, transmitting, switching, receiving, steering or controlling as messages identifiable electromagnetic signals

user: individuals, including consumers, or organizations using or requesting publicly available telecommunications services

NOTE: See Directive 98/10/EC [i.4].

voice service: telecommunications service of direct transport of real-time speech via the PTN such that any user can use equipment connected to a network termination point to communicate with another user of equipment connected to another termination point

3.2 Abbreviations

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

AMR	Adaptive Multi-Rate
ATM	Asynchronous Transfer Mode
BAS	Broadband Access
DECT	Digital European Cordless Telephone
DSLAM	Digital Subscriber Line Access Multiplexor
EFR	Enhanced Full Rate
FR	Full Rate
GMSC	Gateway Mobile Switching Centre
GSM	Global System for Mobile communications
HLR	Home Location Register
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
MDF	Main Distribution Frame
MOS	Mean Opinion Score
MSC	Mobile Switching Centre
MVNO	Mobile Virtual Network Operator
NER	Network Effectiveness Ratio
NRA	National Regulatory Authority
NTP	Network Termination Point
PLMN	Public Land Mobile Network
PSTN	Public Switched Telephone Network
PTN	Public Telecommunications Network
QDU	Quantization Distortion Unit
QoS	Quality of Service
RLR	Receive Loudness Rating
SAP	Service Access Point
SLR	Send Loudness Rating
SMS	Short Message Service
STQ	Speech Transmission and Quality (Technical Committee)
TELR	Talker Echo Loudness Rating
TrFO	Transcoder Free Operation

4 General considerations

4.1 Services covered

The QoS parameters of the present document cover aspects of telecommunications services which are typically provided via the public telecommunications network such as voice, fax or data services. These services may be accessed via terminals connected to fixed network termination points or via mobile accesses e.g. GSM.

The definitions and measurement methods of the QoS parameters were elaborated primarily in order to assess QoS aspects of "standard" telecommunication services. Therefore mainly common aspects and applications of telecommunication services were considered and are reflected in the present parameters. In principle the QoS parameters may also be used for the investigation of special or non-standard telecommunication services but further enhancements/additions to the definitions and measurements methods may be necessary.

Most parameters are in principle applicable to any service provided via the public telecommunication network. Some parameters are however only applicable to specific services depending on technical aspects of the provision of those services, e.g. mobile, data, fixed NTP. Depending on the set of QoS parameters used by the stakeholders the scope of the services covered may vary.

The parameters are end-user/customer and end-to-end orientated and are not intended to address the quality of interconnect services explicitly. Any dependence on interconnect services is included implicitly in the measures of QoS provided to the end user. Separate Guides in this series deal with the QoS of interconnect arrangements.

In many cases the provider of telecommunications services to the customer may depend on other providers for part of the service. An example is an international call where several service providers are normally involved. In such cases the provider of the service to the customer is responsible for all elements for which it receives payment from the customer. In order to provide satisfactory QoS, this service provider will need to ensure that adequate QoS is provided by the other interconnected service providers. QoS figures for the responsible service provider will reflect both its own capability and that of the interconnected service providers.

4.2 Use of the parameters

The parameters may be used for various purposes including:

- Specifying the level of quality of service in customer telecommunication service contracts or in the description or terms and conditions of the service.
- Comparing the quality of service of different service providers.
- Comparing the quality of service aspects of different service offers.
- Preparing long term studies on the quality of service aspects of a specific service.

4.3 Reporting for different classes of customers

For each parameter, statistics may be produced or requested that are aggregated over all classes of customer or, where a distinction between different classes is desired, e.g. residential and business, separate statistics may be used, or both. This recognizes the voluntary nature of these measures and the fact that some stakeholders may only wish to target specific sections or to provide a rough overview of the market.

NOTE: Due to the fact that a variety of different service offers is available at the market, it is not always possible to clearly distinguish between classes of customers like residential or business. Furthermore it may not be fair to compare different service offers on the basis of different classes of customers because the results may be misleading. Also statistics may be falsified when aggregating over all classes of customers. See also clause 4.9.

4.4 Non standard levels of QoS

Statistics produced should normally be based on the standard level of QoS for each telecommunication service. The standard level is defined in the terms and conditions of the services as published by the service providers. Stakeholders may choose to produce or request specific statistics for cases where customers are able to pay more for enhanced or less for lower QoS. It is recommended to provide additional information on the kind and scope of services the QoS statistics are referring to when covering non-standard levels of QoS.

4.5 Reporting for directly- and indirectly-serviced customers

The principle used is that the service provider who charges the customer should be responsible for the quality of the service and for providing QoS statistics relevant to the service provided. Thus, in the case of carrier selection, the indirect service provider has the responsibility for QoS and provision of QoS statistics when it is selected to carry a call.

For each parameter in clause 5 a statement is made on whether it is applicable to indirect services.

Some service providers provide both direct and indirect services. Where there are likely to be significantly different levels of performance for these two service types or where the services are understood as being two different not comparable service offers (even though the same telecommunication service is offered), the production of separate statistics for each service type is recommended.

The treatment of direct and indirect services is summarized in the last column of table 1.

NOTE: Where only a combined statistic for both types of service is specified, separate statistics for each service type may be provided in addition if the stakeholders to do so.

4.6 Data processing issues

Where the measures are based on all actual occurrences rather than samples, the measuring party may prefer to process data on a weekly or monthly basis, discard the detailed data and use a statistical method such as that specified in annex A for combining the weekly or monthly results.

For one parameter the statistic required is "X % of ...". This statistic is explained in annex B.

In some cases disasters, freak weather, etc. may distort measured QoS figures. Such occurrences may not necessarily damage a network, but could degrade QoS by inducing exceptional traffic levels etc. In these cases, service providers should provide the measured QoS and may additionally provide a second figure which excludes the effects of the exceptional circumstances. A note clearly explaining the difference should also be provided. Service providers covering large geographical areas are likely to be more prone to these effects than service providers serving smaller areas. The effect on the reported QoS of a service provider covering a small area is likely to be more severe, however, should such an event occur.

4.7 Data collection period

Where the measurements are to be used for long term comparisons, it is recommended that QoS data should be collected and calculated on a quarterly basis starting on 1 January, 1 April, 1 July and 1 October.

Stakeholders may also decide to use longer or shorter data collection periods. For most QoS parameters a data collection period on a quarterly basis is suitable, and will provide adequately up-to-date information. But there may also be cases were a longer period is more practicable, e.g. extensive customer surveys. Shorter periods are advisable for QoS aspects where frequent and fast changes in quality are likely to occur.

4.8 Sampling and test calls

Where sampling and test calls are used the approach should ensure that the results adequately reflect the QoS perceived by customers for the period under review.

Guidance on the choice of adequate test calls with respect to choice of origin, destination, traffic variations etc. may be found in annex G.

4.9 Comparability of measurements

The following issues may affect the comparability of the measurement results:

- Measurement methods may be implemented differently (e.g. use of real traffic vs. test calls, choice of representative connections).
- Measurements based on signalling information or tones may be unreliable because the signalling systems and tones are not implemented in a fully standardized manner (e.g. different uses of cause values).
- Service offers that claim to be similar may differ in terms of significant service features/aspects.

NOTE: The parameters were elaborated with respect to "standard" service offers and so special care should be taken for non-standard services.

4.10 Publication of QoS parameters

Where measurements are made and published in accordance with the present document, it is recommended that an explicit reference to the present document should be given so that readers can be made aware of the background of the definitions and measurement methods. The reader should be enabled to understand the meaning, purpose and areas of application of the QoS parameters.

It is important that the reader is aware of the scope of the parameters and with that of the correct application of the QoS statistics, otherwise there is a high risk that the measurement results are misinterpreted. A fair and justified comparison of the published data of different service offers, i.e. quality aspects of different telecommunication services, is only possible if the data is strictly used according to the scope of the defined QoS parameters.

Stakeholders who publish QoS statistics in accordance with the present document should provide additional and explanatory text in order to facilitate the understanding of the statistics. It may be assumed that a reader who is interested in comparable QoS statistics and QoS parameters of different nature is willing and capable to understand technical and operational background information on telecommunication services. A balanced approach should be used taking into account on the one hand the need for easy understandable information and on the other hand the requirement of correctly edited data derived from the measurements.

5 QoS parameters for voice, data and fax services accessed via the PTN and SMS

Table 1 summarizes the QoS parameters defined in the present document.

NOTE: Many of the parameters have several subtleties associated with their definition, applicability and measurement. The parameters are fully explained in clause 5.

Table 1: Summary of QoS parameters

Parameter	Measure	Measurement Method	Application
Unsuccessful call ratio	a) the percentage of unsuccessful calls for national calls b) the percentage of unsuccessful calls for international calls c) the number of observations used for national and international calls together with absolute accuracy	Measurements on: - real traffic (all or sample) - test calls	directly and indirectly accessed fixed and/or mobile voice services

Parameter	Measure	Measurement Method	Application
Call set up time	a) the mean value in seconds for national calls b) the time in seconds within which the fastest 95 % of national calls are set-up c) the mean value in seconds for international calls d) the time in seconds within which the fastest 95 % of international calls are set-up e) the number of observations performed for national and international calls	Measurements on: - real traffic (all or sample) - test calls	directly and indirectly accessed fixed and/or mobile voice services
Speech connection quality	a) quality category according to ITU-T Recommendation G.109 [i.14] b) characteristics of terminals c) reference connections	Use of E-Model with input parameters either derived of measurements or planning values	Directly and indirectly accessed fixed and/or mobile voice services
Fax connection quality	% successful fax transactions	Test calls	directly and indirectly accessed fixed and/or mobile fax services
Data rate of Dial-up access to the Internet	transmission rate of modem data of 80 % of connections in bit/s	Test calls	Directly and indirectly accessed data services
Successful SMS Ratio	a) percentage of successfully sent short messages b) number of observations together absolute accuracy limits for 95 % confidence	Measurements on: - real traffic (all or sample) - test calls	short message service providers
Completion Rate for SMS	a) ratio of successfully sent and received short messages b) number of observations together absolute accuracy limits for 95 % confidence	Measurements on: - real traffic (all or sample) - test calls	short message service providers
End-to-End delivery time for SMS	a) the mean value in seconds for sending and receiving short messages b) the time in seconds within which the fastest 95 % of short messages are sent and received c) The number of observations performed	Measurements on: - real traffic (all or sample) - test calls	short message service providers

Table 2 summarizes the information to be provided from the perspective of the user, who may have both a direct service provider (whose service includes the access line) and one or more indirect service providers that may be selected for different calls using call-by-call carrier selection or pre-selection. For each parameter, the table shows what will be measured and which service provider will report an event covered by the parameter.

Table 2: QoS parameters from the perspective of the user

Parameter	Measure	Information provided by
Unsuccessful call ratio	Percentage of unsuccessful calls	direct, indirect and mobile service provider
Call set up time	Time in seconds	direct, indirect and mobile service provider
Speech connection quality	Quality category	voice service provider
Fax connection quality	Percentage of successful fax transactions	fax service provider
Data rate of Dial-up access to the Internet	Data transmission rate of 80 % of connections	data service provider
Successful SMS Ratio	percentage of successfully sent short messages	short message service providers
Completion Rate for SMS	ratio of successfully sent and received short messages	short message service providers
End-to-End delivery time for SMS	Time in seconds	short message service providers

5.1 Unsuccessful call ratio

5.1.1 Definition

Unsuccessful call ratio is defined as the ratio of unsuccessful calls to the total number of call attempts in a specified time period.

An unsuccessful call is a call attempt to a valid number, properly dialled following dial tone, where neither called party busy tone, nor ringing tone, nor answer signal, is recognized at the access of the calling user within 30 seconds from the instant when the last digit of the destination subscriber number is received by the network.

NOTE: The unsuccessful call ratio is comparable to the Network Effectiveness Ratio (NER) as defined in ITU-T Recommendation E.425 [i.6].

5.1.2 Application

The QoS parameter is applicable to directly and indirectly accessed fixed and/or mobile services.

5.1.3 Measurement and statistics

The following statistics should be provided separately:

a) The percentage of unsuccessful calls for national calls, together with the number of observations used and the absolute accuracy limits for 95 % confidence calculated from this number.

b) The percentage of unsuccessful calls for international calls, together with the number of observations used and the absolute accuracy limits for 95 % confidence calculated from this number.

The statistics should be calculated from:

a) measurements on all real traffic; or

b) measurements on real traffic for outgoing calls in a representative population of local exchanges to a representative set of destinations; or

c) test calls in a representative population of local exchanges or NTPs to a representative set of destinations; or

d) a combination of the above.

Guidance on the choice of adequate origin and destination NTPs and/or local exchanges may be found in annex G.

NOTE 1: These alternative methods each have different advantages and disadvantages. The use of test calls is often expensive. Observations based on signalling information can be a low cost alternative but may be unreliable because in real equipments cause values may not be assigned exactly in accordance with the standards and therefore extra care should be taken.

NOTE 2: Measurements may be based on the analysis of tones or on signalling information or on a combination of them. Extra care should be taken to set up the measuring equipment adequately in order to receive comparable results.

Methods for deciding if a call is unsuccessful are given in annex C.

Measurements should be scheduled so as to reflect accurately traffic variations over the hours of a day, the days of the week and the months of the year. When measuring values for different destination categories (national or international) this applies to each destination category separately. In the case of test calls the choice of destination exchanges (or NTPs) should be traffic weighted.

The number of observations may be chosen by the reporting operator and will determine the absolute accuracy to be given with the results but:

a) should fulfil the Laplace criterion for the applicability of calculations based on the normal distribution (see annex D); but

b) is not required to exceed a test call rate of 1 in 1 000.

Annex D gives information on how to calculate the absolute accuracy based on the measured result, the confidence level and the number of observations.

NOTE 3: This approach has been chosen because of the cost implications of trying to specify the accuracy level. Operators will improve the accuracy by using more observations but may decide themselves how many observations are worth taking.

For directly connected customers service providers should exclude from the statistics calls that they deliver to an indirect service provider who then completes the call and charges the customer.

For indirectly connected customers, either:

a) measurements should be based on call data from the processor of the originating local exchange for real calls; or

b) measurements should be made from the subscriber line side of the local exchange in the access network; or

c) measurements should be made from the NTP.

5.1.4 Further considerations

This parameter has been formulated in a general manner. In practice measurements and reporting should be focused on particular services, e.g. normal geographic services or free-phone numbers, and the scope of the measurements identified explicitly in any reporting.

NOTE 1: The indirect service provider may have to pay the access network operator to make measurements or special test calls from the local exchange. Special confidentiality requirements may apply to this information.

NOTE 2: Care should be taken not to degrade the customer's service by making an excessive number of test calls in periods of high traffic levels.

NOTE 3: No intrusive measurements should be made by the access network operator without the agreement of the indirect service provider.

NOTE 4: In view of the costs and administrative overhead involved with recording data or taking measurements from the originating local exchange, NRAs may accept simplifications or approximations for indirect and mobile services such as making measurements from the point of interconnection and using an appropriate adjustment to make allowance for the current performance of the access network to calculate the resulting end to end quality.

The first two arrangements for indirectly connected customers will require co-operation by the access network operator in making the measurements or providing the call data. For the third arrangement there is no co-operation with the access network operator necessary. Both the second and third arrangement may involve disturbance to the customer.

5.2 Call set up time

5.2.1 Definition

The call set up time is the period starting when the address information required for setting up a call is received by the network and finishing when the called party busy tone or ringing tone or answer signal is received by the calling party. Where overlap signalling is used the measurement starts when sufficient address information has been received to allow the network to begin routeing the call.

NOTE: Information on characteristics of tones may be found in ITU-T Recommendation E.180 [i.5].

5.2.2 Application

The QoS parameter is applicable to directly and indirectly accessed fixed and/or mobile services.

5.2.3 Measurement and statistics

The following statistics should be provided separately:

a) the mean value in seconds for national calls;

b) the time in seconds within which the fastest 95 % of national calls are set-up;

c) the mean value in seconds for international calls;

d) the time in seconds within which the fastest 95 % of international calls are set-up;

e) the number of observations performed for national and international calls.

Calls that are classified as unsuccessful calls should be excluded.

Calls to ported numbers should be included.

The statistics should be calculated from:

a) measurements on real traffic for outgoing calls; or

b) measurements on real traffic for outgoing calls in a representative population of local exchanges to a representative set of destinations; or

c) test calls in a representative population of local exchanges or NTPs to a representative set of destinations; or

d) a combination of the above.

Guidance on the choice of adequate origin and destination NTPs and/or local exchanges may be found in annex G.

Measurements should be scheduled so as to reflect accurately traffic variations over the hours of a day, the days of the week and the months of the year. Call monitoring can be done by monitoring every K^{th} call where K is to be calculated from the total expected number of calls in the relevant time intervals and from the needed number of observations. When measuring values for different destination categories (national or international) this applies to each destination category separately. In the case of test calls the choice of destination exchanges (or NTPs) should be traffic weighted.

NOTE 1: These alternative methods each have different advantages and disadvantages. The use of test calls is expensive and provides only an estimate of the actual performance but involves measurement at the access line side of the local exchange. Observations performed at the exchange processor are cheaper and more data can be obtained giving more accurate estimates, but the data does not come from so close to the NTP.

NOTE 2: Measurements may be based on the analysis of tones or on signalling information or on a combination of them. Extra care should be taken to set up the measuring equipment adequately in order to receive comparable results.

Annex E gives a formula for calculating the number of observations needed.

For mobile services, the addition of a correction factor to the measurements on real traffic (based on core network signalling information) is needed to take into account the set-up time through the radio access network.

For directly connected customers service providers should exclude from the statistics calls that they hand over to an indirect service provider who then completes the call and charges the customer.

For indirectly connected customers, either:

a) measurement should be based on call data from the processor of the originating local exchange for real calls; or

b) measurement should be made from the subscriber line side of the local exchange in the access network; or

c) measurement should be made from the NTP.

In addition the statistics should state if en bloc, overlap dialling or a mixture was used and if connections between fixed NTPs, mobile NTPs or a combination of fixed/mobile NTPs were measured.

Separate statistics may be produced for the above listed dialling procedures and connection scenarios. Where there are likely to be significantly different levels of performance, the production of separate statistics is recommended.

5.2.4 Further considerations

This parameter has been formulated in a general manner. In practice measurements and reporting should be focused on particular services, e.g. normal geographic services or freephone numbers, and the scope of the measurements identified explicitly in any reporting.

NOTE 1: The indirect service provider may have to pay the access network operator to make measurements or special test calls from the local exchange. Special confidentiality requirements may apply to this information.

NOTE 2: Care should be taken not to degrade the customer's service by making an excessive number of test calls in periods of high traffic levels.

NOTE 3: No intrusive measurements should be made by the access network operator without the agreement of the indirect service provider.

NOTE 4: In view of the costs and administrative overhead involved with recording data or taking measurements from the originating local exchange, NRAs may accept simplifications or approximations for indirect and mobile services such as making measurements from the point of interconnection and using an appropriate adjustment to make allowance for the current performance of the access network to calculate the resulting end to end quality.

The first two arrangements for indirectly connected customers will require co-operation by the access network operator in making the measurements or providing the call data. For the third arrangement there is no co-operation with the access network operator necessary. Both the second and third arrangement may involve disturbance to the customer.

When overlap signalling is used the measuring party has to know when the network starts routeing the call, i.e. the minimum number of digits of the subscriber number that needs to be transmitted to the network. This number depends on the setting of the switches and therefore this information is normally only available at the access network provider.

For the performance levels of call set up times in fixed circuit switched networks offering traditional telephone service it may be assumed that quality problems are not likely to occur. There are however scenarios that could decrease the performance levels.

EXAMPLE 1: Routing of calls via several networks, e.g. national calls may not be routed directly to the destinations (especially in combination with number portability).

EXAMPLE 2: Connections from and to mobile networks.

EXAMPLE 3: Use of packet based/switched networks.

5.3 Speech connection quality

5.3.1 General considerations

The migration of traditional telephone services based on circuit-switched networks towards Voice over IP services based on packet switched IP has created a need for quality supervision tools. These needs concern all actors of the telecommunication domain, but more especially, network operators, regulators and benchmarkers.

Traditional telephone connections in circuit-switched networks are established via 64 kbit/s PCM channels with well-known and stable transmission characteristics. This was achieved by complying with national and international transmission plans. In emerging IP networks VoIP service can be impacted by numerous types of degradations that are less well investigated and covered in transmission plans as those in legacy networks. Also in IP networks each operator and service provider can follow different strategies and implement different technologies. Having this in mind and seeing that due to deregulation and liberalization of telecommunication markets there are no longer regulatory imposed transmission plans, it seems to be necessary to supervise and compare the quality of voice telephony.

Several parameters influence the end-user perceived speech quality, like e.g.:

- Transmission delay (duration of routing, processing delay by routers)
- Transmission delay variation (or jitter)
- Available bandwidth to amount of traffic ratio (network dimensioning)
- Packet loss
- Processing capacity of equipment (PC, routers and gateways)
- Voice processing delay (coding, packetization)
- Type of codec (processing delay, distortion, frequency band)
- Number of frames per IP packet
- Lost packet processing (PLC mechanism)
- VAD (vocal activity detection) and comfort noise generation during non-active periods
- Efficiency of echo cancellation mechanism (acoustic and electric)
- Jitter buffer length and agility
- Acoustic proprieties of terminals (coupling between microphone and speakers, hands-free)
- Circuit noise

Several voice quality evaluation methods have already been developed. Each method has different areas of application, proprieties, advantages and drawbacks. An overview is given in the next clause.

5.3.2 Evaluation methods

Three types of vocal quality evaluation methods are identified and used:

- Intrusive methods
- Non-intrusive methods
- Parametrical methods (especially ITU-T Recommendation G.107 [i.11])

5.3.2.1 Methods based on intrusive measurements

This type of measurements is performed on artificially generated traffic (established test calls) and can provide detailed information since the traffic can be tailored to check almost everything. This method allows the QoS to be evaluated only between one point to another at time "t". To obtain a global view of the service, a large number of connections are needed, at different time periods and between several end-to-end points. Several network technical parameters can be evaluated by this means.

For the evaluation of end-to-end speech connection quality the intrusive methods make use of reference test signals that are transmitted through the channel under test. By comparing the sent test reference signal with the signal received at the end terminal a measure for the speech quality is obtained. The comparison of the two signals (sent and received) is performed by psycho-acoustic models, such as ITU-T Recommendations P.862 [i.24], P.862.1 [i.12] and P.862.2 [i.13], which allows the evaluation of quality as perceived by the end users.

These methods are based on test calls between two probes. They take into consideration different access types (PSTN/GSM and IP).

Probes with analogue interface (for PSTN, GSM or behind home gateway):

- MOS-LQO with reference signal (recommendations ITU-T Recommendation P.862 [i.24]).
- Possibility of several other test methods (signal and noise level at the reception, etc.).

Probes with IP interface:

- IP parameters (packet loss ratio, jitter, codec type) + transmission quality (E model).
- Evolution towards recommendations ITU-T Recommendation P.862 [i.24] of new generation probes.

Advantages:

- end to end vision of the telephone QoS (end to end measurement);
- fine vision of transmission quality of the tested network configuration;
- good correlation with subjective perception;
- provide access to several indicators calculated on the speech signal (signal and noise level at the reception, attenuation, echo delay, MOS note, transmission delay, etc.);
- possibility of test performance by third party;
- allow quality survey over a period of time (for the tested configurations);
- assessment of service availability.

Drawbacks:

- generation of additional traffic due to tests;
- additional expense associated with test traffic;
- significant cost of probes;
- limited vision of QoS which is related to restricted number of probes deployed in the network (because of their high price);
- end to end vision which is not easily extrapolated to other non-tested configurations;
- no possibility of obtaining distribution of call types per operator.

5.3.2.2 Methods based on non-intrusive measurements

This measurement method is performed on real customer "live-traffic". The tool is implemented directly in the service provider network, in general at a strategic interface (a demarcation point between two sub-networks or a point of traffic concentration like an international gateway) using high-impendence probes or mirroring ports on active equipment. The live traffic is not disturbed by this kind of tool. Non-intrusive methods analyse signals without a known reference, i.e. the measurement is based on the absolute estimation of the quality. They determine transmission signal indicators without disturbing the signal.

The most known systems for non-intrusive measurement are the in-service, non-intrusive measurement devices (or INMDs).

In PSTN the INMD probes (ITU-T Recommendation P.561 [i.25] recommendation) measure the following indicators:

- Signal level (speech, noise).
- Echo (delay, fading).
- MOS-LQO without reference signal (ITU-T Recommendations P.562 [i.26] and P.563 [i.27] recommendation).

In IP network the dedicated analysers of VoIP signalling protocols (H.323, SIP, MGCP, etc.) are used to determine:

- the IP parameters (packet loss ratio, jitter, codec type);
- transmission quality (E model or P.VTQ) which requires access to RTP flows.

In IP network, it is necessary to decode and reconstruct the signal in order to perform the measurements using INMD.

Advantages:

- absence of additional traffic due to tests;
- no additional expense associated with test traffic;
- the possibility of analysing of a large number of calls (in the user terminal or in the network node);
- fine macroscopic vision of transmission quality;
- possibility of obtaining distribution of call types per operator.

Drawbacks:

- the access to speech signal requires much CPU (therefore rarely implemented at the IP level);
- vocal quality often evaluated by parametrical model which is less efficient than psycho-acoustic models with reference;
- significant cost of probes;
- little (or no) possibility of measurement performance by third party without explicit operator's agreement;
- not adapted to evaluation of service availability.

5.3.2.3 Methods based on the usage of the E model

Models attempt to map objective measures of network performance to subjective opinions. The objective measurements needed as input values for the mapping function are normally taken from measurements. The mapping is done by the use of a computational model, known as the E-model and in ITU-T recommendation G.107 [i.11]. The primary output of the model is a scalar rating of transmission quality, the "Rating Factor" R or R factor. The R-value is an evaluation of a quality perception to be expected by the average user when communicating via the connection under consideration which characterizes the vocal quality during a telephone conversation.

The calculation of the R factor takes the following information into consideration:

- Connection type (fixed-fixed, fixed-mobile, mobile-mobile).
- Call type (local, national, international).
- Transmission type (copper, optical, radio or satellite link).
- Negotiated codecs.
- Performance of transmission equipment (especially regarding delay).
- Performance of terminals.
- Performance of echo cancellation.

The R factor can be transformed to give estimates of customer opinion, i.e. MOS scores.

Advantages:

- The main advantage of this method is its modest cost, because it is based on the information about network architecture and equipment proprieties.
- It allows a macroscopic and averaged vision of quality of service. It provides the telecommunication operator a means of comparison between different telephone communication types. The evaluation can be done during conception phase as well as after the deployment of service.

Drawbacks:

- No end to end vision because the signal from the terminal user is not analysed.
- Limited QoS supervision because the model is supplied by information which is related to nominal functioning and not to incidents (failure, wrong configuration, traffic overload, etc.).
- Not adapted to utilization by third party as it requires information about network architecture and equipment configuration which the operators are not willing to share.
- Difficult comparison of telephone services of different operators; not only because of the fact that operators are not willing to disclose certain information but also because the information cannot be verified by third party.
- Not adapted to evaluation of service availability.

5.3.3 Practical Implementation of parametrical model

Currently a practical implementation of the parametrical model, i.e. an implementation of the E model has been elaborated and is given below and is referred to as "design figure of merit". Descriptions of practical implementations of other evaluation methods are for further study.

The design figure of merit for speech quality is a measure of the extent to which the operator in question has used the best design methods and equipment available to support speech quality.

The figure is the percentage of the R-value for a hypothetical network based on the design of the reporting operator divided by the maximum R-value achievable by using those options in the relevant standards that provide for the best quality and an optimum network topology.

The figure of merit is calculated using the design choices of the reporting operator, which can be obtained using a questionnaire. The hypothetical network should be specified to be appropriate to the country/area concerned and include a range of different terminals, and traffic levels representative of the country/area concerned. The originating side of the hypothetical network is the same as that of the network that is reporting, i.e. if a fixed network is reporting then only fixed call origination is considered. The terminating side of the hypothetical network includes both fixed and mobile call terminations. All details of the hypothetical network and its terminals and traffic patterns, but excluding the design decisions, are the same for all networks that are reporting.

NOTE: The figure of merit applies only to an operator's own network. It indicates the quality of on-net calls and the contribution that the reporting network makes to the quality of off-net calls, but not the full end-to-end quality of off-net calls since this is also affected by interconnected networks.

5.3.3.1 Results

The following results should be provided separately:

1) Fixed to fixed calls.

2) Fixed to mobile calls.

3) Mobile to fixed calls.

4) Mobile to mobile calls.

An example for the calculation of the Design figure of merit for speech quality is layed out in annex I.

5.3.3.2 Further considerations

This parameter has been formulated to provide a simple estimator of the design quality of a network that can be calculated reasonable easily by all operators and provides a figure that is comparable between operators. Earlier attempts to quantify speech quality were based on reference connections but were found not be practicable because of the increasing proportion of calls that pass across interconnection points between different networks.

The operator who is reporting does not necessarily have access to the performance parameters for the whole of a reference connection and in any case does not have control over the design of the whole connection. Thus any figures calculated for a reference connection that involves more than one network could not be a measure of the commitment to quality of the operator who is reporting. In the case of call termination and call origination for carrier selection, the operator who is being paid for the call will normally have no control over the performance of the other networks used and no possibility of using an alternative network.

Annex I explains in more detail how the figure of merit can be calculated. All calculations treat each impairment separately and then calculate the combined effect of the impairments using the E-Model.

5.3.4 Application

The QoS parameter is applicable to all networks that provide 3,1 kHz handset telephony irrespective of whether they provide fixed and/or mobile access and whether they are accessed directly or indirectly. It does not apply to wideband speech.

5.4 Fax connection quality

5.4.1 Definition

Fax transaction success ratio is defined as the ratio of successful fax transactions between Group 3 facsimile terminals (see ITU-T Recommendation T.4 [i.15]) to the total number of attempted fax transactions.

A successful fax transaction is defined as a transaction of the standardized test chart as defined in annex F that:

- completes (i.e. all pages are sent);
- takes place at the highest mutual transmission speed of the send and receive fax machines; and
- has no severely errored pages.

NOTE: In case any further guidance is needed to decide on the completeness, transmission speed and errored pages the following ITU-T Recommendations should be taken into account:

- ITU-T Recommendation E.451 (call cut off performance) [i.7];
- ITU-T Recommendation E.452 (facsimile modem speed reduction and transmission times) [i.8];
- ITU-T Recommendation E.453 (Facsimile Image Quality) [i.9].

5.4.2 Application

The QoS parameter is applicable to fax services irrespective whether they are accessed directly or indirectly and via fixed or mobile accesses.

5.4.3 Measurement and statistics

The following statistics should be provided separately:

- the percentage of successful fax transactions;
- the number of test calls.

The statistics should be calculated from test calls in a representative population of local exchanges or NTPs to a representative set of destinations.

NOTE: Guidance on the choice of adequate origin and destination NTPs may be found in annex G.

The result, expressed as a % successful fax transactions (to one decimal place), can be derived from:

(Total number effective transactions / Total number of observations) × 100

5.5 Data rate of dial-up access to the Internet

5.5.1 Definition

The data signalling rate over connections between an analogue network termination point and an ISP using an analogue-digital modem pair that is determined during Phase 4 of modem start-up according to the procedures as described in ITU-T Recommendations V.90 [i.18] and V.92 [i.19].

The definition recognizes that modems are used in most cases to connect to ISPs in order to gain access to online services. The standard modem used is the V.90/V.92 modem; a residential user is typically connected to the public telephone network via an analogue access line whereas an ISP is connected via a digital access.

Any decrease of transmission rate related to any equipment on the user or ISP sides of the network termination points, or related to the ISP's backbone connection or related to the performance of the connection part of the Internet is excluded.

NOTE: The most likely limiting factor of dial-up connections to the internet lies within the capacity of the public telephone infrastructure, in particular within the customer's access line. Traditional analogue accesses are optimized for the support of voice telephony services which does not automatically include the provision of data communications. In most cases service providers offering analogue network access are not guaranteeing any data transmission rate at all. When using this access for data communication the resulting data transfer rate is determined by the characteristics of the existing access line. Circuit multiplication equipment and the use of non-linear codecs (optimized for speech signals) may further decrease the possible data transmission rate significantly, even down to zero kbit/s.

This means that users are experiencing a range of data transmission rates from zero up to the theoretical maximum rate of 56 kbit/s (V.90 standard) which is not the net bit rate but the actual bit rate the modem is transmitting at (including error control, protocol etc.). Therefore the parameter Internet connection quality is focused on the resulting quality (mostly) attributable to the customer's access line. As an indicator of this quality the data signalling rates the modem sets up to during the final training phase is chosen. Other factors influencing the quality of a user-internet connection like packet loss, delay, jitter, reliability of the ISP's backbone connection and the internet itself are out of scope.

5.5.2 Application

The QoS parameter is applicable to data connections established between an analogue fixed NTP and a digital NTP using an analogue-digital modem pair according to ITU-T Recommendations V.90 [i.18] and V.92 [i.19]. The parameter reflects the quality of the connection via the public telecommunication network between the NTPs, i.e. the data signalling rate the modem sets up to during the training phase. Any influences on the transmission quality caused by the terminals or the ISP's connection to the internet are out of scope. Figure 5.1 illustrates the scope of the parameter.

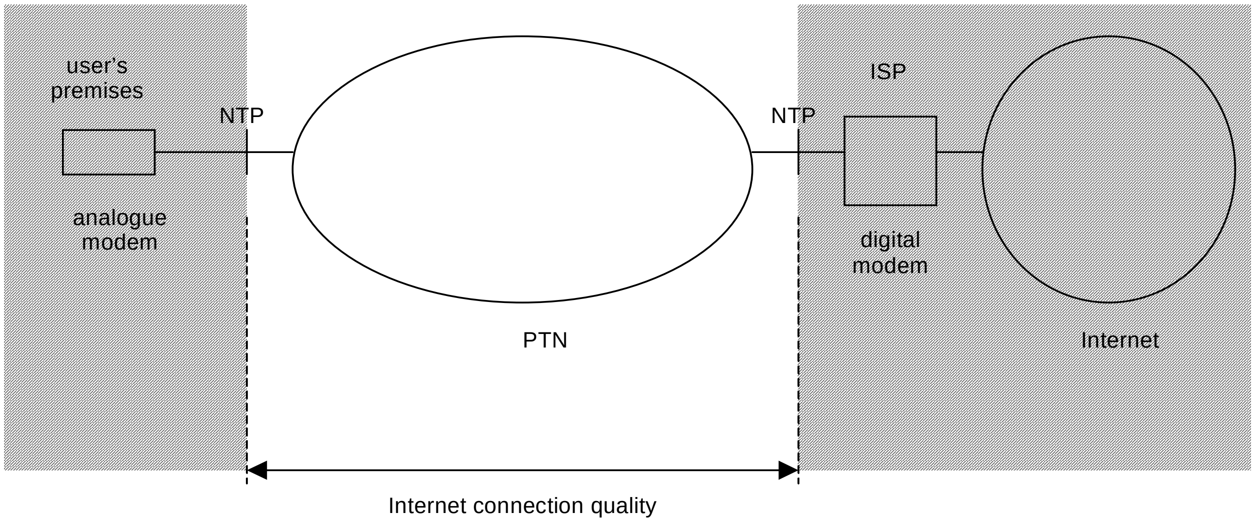


Figure 5.1: Application of QoS parameter Internet connection quality

5.5.3 Measurement and statistics

The following statistic should be provided separately for the upstream (analogue modem) or downstream (digital modem) direction.

The resulting data signalling rate of 80 % of connections in bit/s.

NOTE 1: The 80 % figures are calculated according to the principles explained in annex B.

NOTE 2: Due to the V.90 modem standard the data signalling rate will adjust in discrete steps in upstream direction from 4 800 bit/s to 28 800 bit/s in increments of 2 400 bit/s and for 31 200 bit/s and 33 600 bit/s and in downstream direction from 28 000 bit/s to 56 000 bit/s in increments of 8 000/6 bit/s.

The data signalling rate is the rate the analogue modem (upstream direction) and the digital modem (downstream direction) set to during Phase 4 "Final Training" of modem start-up according to the procedures described in ITU-T Recommendations V.90 [i.18] and V.92 [i.19].

The statistics should be calculated from test calls from a representative population of analogue NTPs to a representative population of ISPs. The test calls are set up and measured from an analogue-digital modem pair according to ITU-T Recommendations V. 90 [i.18] and V.92 [i.19].

NOTE 3: Guidance on the choice of adequate origin and destination NTPs and ISPs may be found in annex G.

Where service providers quote a standard data signalling rate for data connections this data signalling rate should also be provided.

5.5.4 Further considerations

The QoS parameter "Data rate of Dial-up access to the Internet" gives an indication on the quality of a connection between a user and an ISP with respect to the possible data signalling rate of the PTN part of the connection. This quality is not the quality the user subjectively perceives when being connected to the internet.

From the user's perspective the net transmission rate, i.e. the data throughput, is of importance. This transmission rate is influenced not only by the public network but also by the terminal (software settings, DSP performance, etc.), the ISP's connection to the Internet and the overall performance of the Internet (available bandwidth). It is not feasible to define a parameter that reflects all these interactions. The resulting parameter would be too complex to provide any useful and understandable information to the naïve user.

Therefore the approach was chosen to define a parameter that provides information on the ability of the public network (especially the analogue access) to support Dial-up access to the Internet. The scenario chosen represents a typical situation a normal user is facing when trying to get access to the Internet.

Thus the parameter enables the user to obtain information on the connection quality his access network provider is supporting. The user still does not know what the resulting overall performance of his Internet access will be, but he is in the position to decide whether his "traditional" telephone access is suitable for dial-up access to the Internet.

5.6 Short Message Service (SMS) QoS parameters

5.6.1 Successful SMS Ratio

5.6.1.1 Definition

Probability that a user can send a Short Message successfully from a terminal equipment to a Short Message Center.

5.6.1.2 Application

The QoS parameter is applicable to all service providers offering a short message service.

5.6.1.3 Measurement and statistics

The following statistics should be provided.

The percentage of successfully sent short messages, together with the number of observations used and the absolute accuracy limits for 95 % confidence calculated from this number.

The statistics should be calculated from:

a) measurements on real traffic for short messages; or

b) measurements on real traffic for short messages in a representative population of local exchanges or MSCs; or

c) test calls in a representative population of local exchanges or MSCs; or

d) a combination of the above.

Guidance on the choice of adequate origin local exchanges or MSCs may be found in annex G.

Measurements should be scheduled so as to reflect accurately traffic variations over the hours of a day, the days of the week and the months of the year. Call monitoring can be done by monitoring every K^{th} call where K is to be calculated from the total expected number of calls in the relevant time intervals and from the needed number of observations.

NOTE 1: These alternative methods each have different advantages and disadvantages. The use of test calls is expensive and provides only an estimate of the actual performance but involves measurement at the access line side of the local exchange. Observations performed at the exchange processor are cheaper and more data can be obtained giving more accurate estimates, but the data does not come from so close to the NTP.

NOTE 2: Measurements may be based on the analysis of signalling information or on a combination of them. Extra care should be taken to set up the measuring equipment adequately in order to receive comparable results.

Annex E gives a formula for calculating the number of observations needed.

5.6.1.4 Further considerations

NOTE: Concerning the mobile environment the parameter is meant to measure the combination of the network accessibility in the claimed area of coverage and congestion in the signalling channels and SMS system, i.e. the ability of a user to send an SMS when in a claimed area of coverage. Whilst operators may wish to distinguish the effects of coverage and access congestion, it is not necessary to distinguish them from the perspective of the user.

5.6.2 Completion Rate for SMS

5.6.2.1 Definition

The Ratio of correctly sent and received SMS between two terminal equipments.

5.6.2.2 Application

The QoS parameter is applicable to all service providers offering a short message service.

5.6.2.3 Measurement and statistics

The following statistics should be provided:

- The ratio of successfully sent and received short messages, together with the number of observations used and the absolute accuracy limits for 95 % confidence calculated from this number.

The statistics should be calculated from:

a) measurements on real traffic for short messages; or

b) measurements on real traffic for short messages in a representative population of NTPs/SAPs; or

c) test calls in a representative population of NTPs/SAPs; or

d) a combination of the above.

Guidance on the choice of adequate origin and destination NTPs/SAPs may be found in annex G.

Measurements should be scheduled so as to reflect accurately traffic variations over the hours of a day, the days of the week and the months of the year. SMS monitoring can be done by monitoring every K^{th} SMS where K is to be calculated from the total expected number of calls in the relevant time intervals and from the needed number of observations.

NOTE 1: These alternative methods each have different advantages and disadvantages. The use of test calls is expensive and provides only an estimate of the actual performance but involves measurement at the access line side of the local exchange. Observations performed at the exchange processor are cheaper and more data can be obtained giving more accurate estimates, but the data does not come from so close to the NTP.

NOTE 2: Measurements may be based on the analysis of signalling information or on a combination of them. Extra care should be taken to set up the measuring equipment adequately in order to receive comparable results.

Annex E gives a formula for calculating the number of observations needed.

5.6.3 End-to-End delivery time for SMS

5.6.3.1 Definition

The end-to-end delivery time for SMS is the period starting when sending a SMS from a terminal equipment to a Short Message Center and finishing when receiving the very same SMS on another terminal equipment.

5.6.3.2 Application

The QoS parameter is applicable to all service providers offering a short message service.

5.6.3.3 Measurement and statistics

The following statistics should be provided separately:

a) the mean value in seconds for sending and receiving short messages;

b) the time in seconds within which the fastest 95 % of short messages are sent and received;

c) the number of observations performed.

The statistics should be calculated from:

a) measurements on real traffic for short messages; or

b) measurements on real traffic for short messages in a representative population of NTPs/SAPs; or

c) test calls in a representative population of NTPs/SAPs; or

d) a combination of the above.

Guidance on the choice of adequate origin and destination NTPs/SAPs may be found in annex G.

Measurements should be scheduled so as to reflect accurately traffic variations over the hours of a day, the days of the week and the months of the year. SMS monitoring can be done by monitoring every K^{th} SMS where K is to be calculated from the total expected number of calls in the relevant time intervals and from the needed number of observations.

NOTE 1: These alternative methods each have different advantages and disadvantages. The use of test calls is expensive and provides only an estimate of the actual performance but involves measurement at the access line side of the local exchange. Observations performed at the exchange processor are cheaper and more data can be obtained giving more accurate estimates, but the data does not come from so close to the NTP.

NOTE 2: Measurements may be based on the analysis of signalling information or on a combination of them. Extra care should be taken to set up the measuring equipment adequately in order to receive comparable results.

Annex E gives a formula for calculating the number of observations needed.

Annex A: Combination of weekly or monthly results

Mean values and percentages produced weekly or monthly may be aggregated into quarterly statistics using one of the following formulae:

a) For weekly statistics:

$S_{quarterly} = (\Sigma\ N_i.S_i)/(\Sigma N_i)$ where i = 1, 2...13; and

N_i = the number of events in each week;

S_i = the statistic for each week.

b) For monthly statistics:

$S_{quarterly} = (\Sigma\ N_i.S_i)/(\Sigma N_i)$ where i = 1, 2, 3; and

N_i = the number of events in each month;

S_i = the statistic for each week.

For aggregating the median or the 95 % - quantile into quarterly statistics, one has to apply the same procedure as explained in annex B.

Annex B: Further explanation of "X % of"

One parameter requires a statistic of the form:

"X % of <relevant event>".

This annex explains what is meant.

The measurements give a list of <relevant event> recorded for the events. This list of events should be counted and sorted into ascending order.

X % of the total number of measurements counted should be calculated giving a number, say "n" which would be rounded down to the nearest integer.

The "n"th time in the sorted ascending list will then be "X % of <relevant event>" occurred and is the statistic to be reported.

Annex C: Decision about the success of a call attempt

Deciding whether a call attempt is successful or not is relatively easy for test calls made from a user's premises, because the equipment simulates a customer and therefore it can decide in a similar way (indicators are: answer from far end, busy or ringing tone).

In practise, the measurements are normally made by machines. For real traffic measured at exchanges, user tones are not available and another source of information is needed. This should be the Signalling System No. 7 between the switches. This annex defines a simple, but appropriate, form of an algorithm based on the information element Cause Value (see ITU-T Recommendation Q.850 [i.20]).

In principle, the Cause Values are not very reliable, because their setting (in the switches) in a living network may not be always correct. Normally they should be used as described in ITU-T Recommendation Q.850 [i.20] but it is in each operator's own responsibility. For these reasons, the proposed algorithm contains only a minimum set of Causes that are very frequently used. To make the algorithm more reliable the setting of the Cause Value can be part of a bilateral agreement.

The algorithm reads:

- A call that ends with the Cause:

  16: Normal call clearing; or

  17: User busy; or

  18: No user responding; or

  19: No answer from user (user alerted);

  should be add to the total number of call attempts.

- A call, which ends with the Cause:

  34: No circuit/channel available; or

  38: Network out of order; or

  41: Temporary failure; or

  42: Switching equipment congestion; or

  44: Requested circuit/channel not available; or

  46: Precedence call blocked; or

  47: Resource unavailable unspecified;

  should be add to the total number of call attempts and should be add to the total number of unsuccessful calls.

- A call, which ends with the Cause:

  31: Normal, unspecified, and its duration is 1 second or longer;

  should be added to the total number of call attempts.

- A call, which ends with the Cause:

  31: Normal, unspecified, and its duration is less than 1 second;

  should be added to the total number of call attempts and should be added to the total number of unsuccessful calls.

- A call that ends with any other Cause should be ignored.

If any other Cause arises in a remarkable amount (e.g. > 1 %) the network operators should negotiate how to handle it.

This algorithm is a recommendation. The interconnected network operators may use an alternative algorithm such as is described in ITU-T Recommendation E.425 [i.6].

Annex D: Relationship between the accuracy of the estimator of the unsuccessful call ratio and the number of calls to be observed

This annex explains that there is a four sided relationship between:

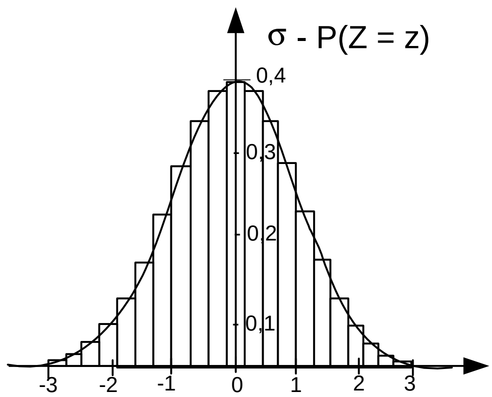
- the percentage of unsuccessful calls;
- the number of observations used in the measurements;
- the statistic interval (accuracy) required of measurements;
- the confidence level of that interval;

and gives guidance on how operators should determine the number of observations that they need to make.

D.1 Theory

In general, any measurement can provide only an estimate of the quantity being measured. Therefore a measurement is performed several times to build up an average out of all the individual measurements. The measured values make up an interval and it is assumed that the real value μ - which is most probably neither the average nor any one of the measured values - lies inside this interval. In the following some mathematics is provided to show the relationship between:

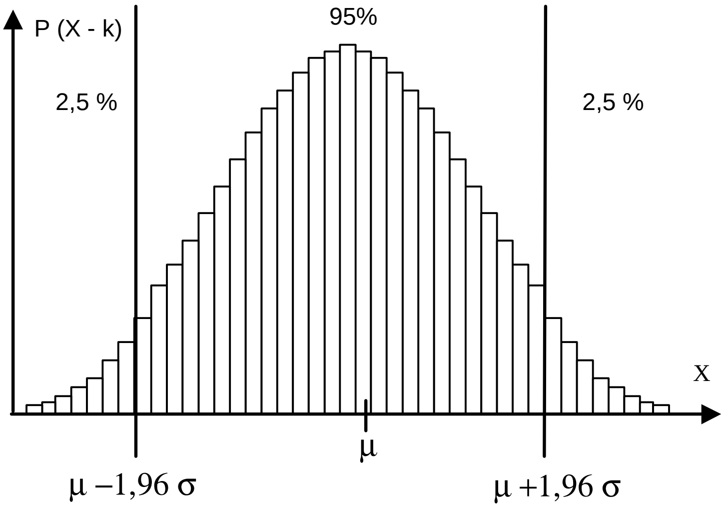
- the average of the measured quantity (in the case of the present document the quantity is the percentage of unsuccessful call ratio p;
- the number of measures;
- the statistic interval (later in this annex called accuracy); and
- the confidence (probability) that the real value lies inside the interval (95 % is assumed throughout this annex).



The starting point of the considerations is the assumption that the single measured values are distributed according a normal distribution around μ. This can be seen in a histogram (the pillars in the left picture). This assumption is correct for most natural processes, which are the result of a combination of individual more detailed processes.

If the number of values is sufficient (see Laplace criteria below) the pillars can be approximated by the Gaussian density distribution φ(z) - also called "Normal distribution". At the maximum of the function (at z = 0) lies μ. The values of the abscissa are the multiple of the standard deviation σ. The area beneath the graph can be interpreted as the totality of all possible (obtained by experiment) values. i.e.:

$$\int_{-\infty}^{\infty} \varphi(z)dz = 100\ \% \qquad \text{(D.1)}$$



For a measurement according to the normal distribution it is known that 95 % of the measured values are lying in the interval:

[μ - 1,96σ;μ + 1,96σ] around μ. This results, with formula (1), from:

$$\int_{\mu-k\sigma}^{\mu+k\sigma} \varphi(z)dz = 95\ \% \Rightarrow k = 1{,}96 \qquad \text{(D.2)}$$

Now, one can easily say that the relative accuracy is $\Delta p/p$, and – looking at the diagram – it is 1,96 σ/μ

NOTE: The unit of σ and μ is "number of unsuccessful calls".

This leads to:

$$\frac{1{,}96\sigma}{\mu} = \frac{\Delta p}{p} \qquad \text{(D.3)}$$

By combining this equation (D.3) with the formulas $\mu = np$ and $\sigma^2 = np(1-p)$, which are valid since we assume that the binomial process can be approximated to a normal distribution, one obtains the absolute accuracy:

$$\Delta p = 1{,}96\sqrt{\frac{p(1-p)}{n}} \qquad \text{(D.4)}$$

which is the same formula as in annex C of EG 201 769 [i.1].

By dividing formula (4) by p one obtains the relative accuracy:

$$\Delta p \big/ p = 1{,}96\sqrt{\frac{(1-p)}{pn}} \qquad \text{(D.5)}$$

Because of the approximation of the binomial by the normal distribution, the number of observations should be "large" which is defined by the Laplace criterion, $\sigma^2 > 9$. Therefore the number of observations, n, should always exceed $9/(p(1 - p))$. These limits are given in table D.1.

Table D.1: Minimum number of observations (n) for proportions of unsuccessful calls (p)

p	n >
0,5 %	1 809
1 %	909
2 %	459
4 %	234

D.2 Guidance

There is a trade-off between the accuracy (statistic interval) to be achieved and the number of observations needed and higher accuracy involves additional costs. The difficulty is that this trade-off itself depends on the percentage of unsuccessful calls that is being measured.

- For a given **relative** accuracy, **more** observations are needed when the percentage of unsuccessful calls is lower.
- For a given **absolute** accuracy, **fewer** observations are needed when the percentage of unsuccessful calls is lower.

When contracts for interconnection are prepared, the parties need to decide whether to specify:

- absolute accuracy;
- relative accuracy; or
- number of observations;

and they also need to state that they are using 95 % confidence (or a different specified level).

NOTE: This text has been copied from annex C of TR 101 949 [i.28]. Although the present document is intended to be applicable to end user QoS parameters and not interconnection contracts, the information was found to be useful and therefore was not deleted.

Practices vary and relative accuracy is quite commonly used. However operators who are unfamiliar with statistics could be unaware of the implications in terms of numbers of observations and hence costs if they specify high accuracy and have good performance. Therefore it is recommended that either the number of observations should be specified in the contracts, or an upper limit to the number of measurements should be specified.

When making measurements to achieve a specific accuracy, operators should therefore proceed by first obtaining a rough estimate of the proportion of unsuccessful calls so that they can use this value to calculate the number of observations required for a given accuracy. This rough estimate can be obtained either by making some initial observations or by using past data.

For relative accuracy, a value of 10 % (= $\Delta p/p$ = 0,1). is commonly used. This value leads, with equation (D.5), to the equation:

$$n = 384\left(\frac{1}{p} - 1\right) \tag{D.6}$$

which can be used to calculate the number of observations needed. Some values calculated with this formula are given in table D.2.

Table D.2: Number of observations (*n*) for proportions of unsuccessful calls (*p*) for 10 % relative accuracy

p	n
0,5 %	76 416
1 %	38 016
2 %	18 816
4 %	9 216

Annex E: Method of calculating the number of observations required for measures of time

The number of observations for quantitative variables depends on the variability of the measurements. It can be calculated by the formula:

$$n = \frac{z_{1-\alpha/2}^2}{a^2} \cdot \left(\frac{s}{mean(x)} \right)^2$$

Where:

$z_{1-\alpha/2}$: Is the 1-α/2-percentile of the standard normal distribution.

s: Is the expected standard deviation of the call set up time (calculated from former measurements).

mean(x): Is the expected mean value of the call set up time (calculated from former measurements).

a: Is the relative accuracy.

Even though there is no requirement to provide the standard deviation, an estimate should be available for use in this formula.

The following table gives the resulting values where:

$z_{1-\alpha/2}$ = 1,96 for a confidence level of 95 %;

a = 2 %.

s/mean(x)	observations
< 0,1	100
0,1 to 0,3	1 000
> 0,3 to 0,5	2 500
> 0,5 to 0,7	5 000
> 0,7 to 0,9	7 500
> 0,9	10 000

Annex F: Standardized test chart for fax connection quality

The following test charts are copyrighted to ITU and are taken from ITU-T Recommendation T.22 [i.17]. For additional information on the test charts please refer to the Recommendation.

There are two test charts:

- one for black-white patterns: a high contrast "facsimile test chart" for evaluating the technical quality of the page and the legibility of the text;
- the other for continuous tone: a "continuous tone addendum" for the evaluation of the technical quality of continuous tone information.

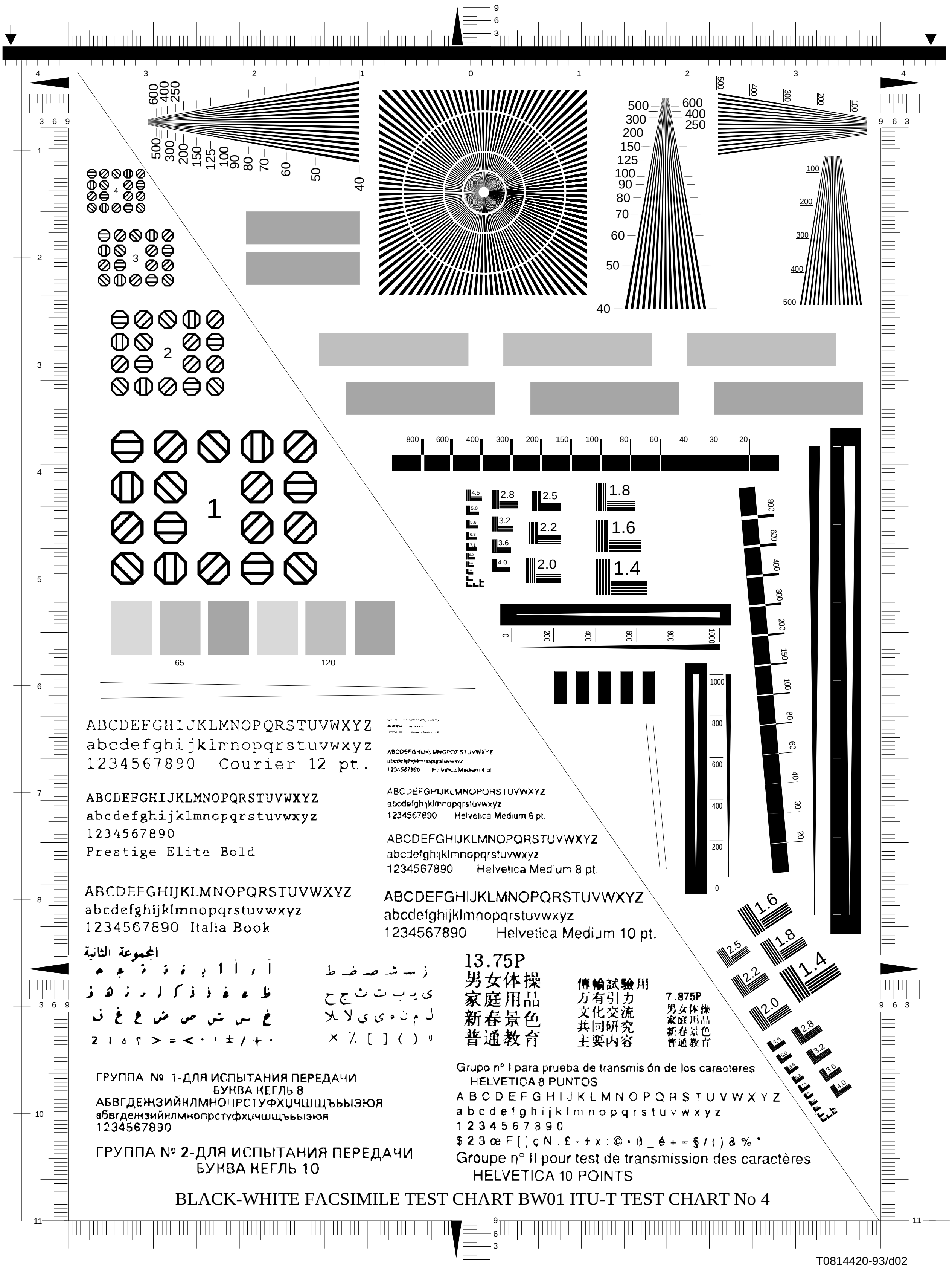
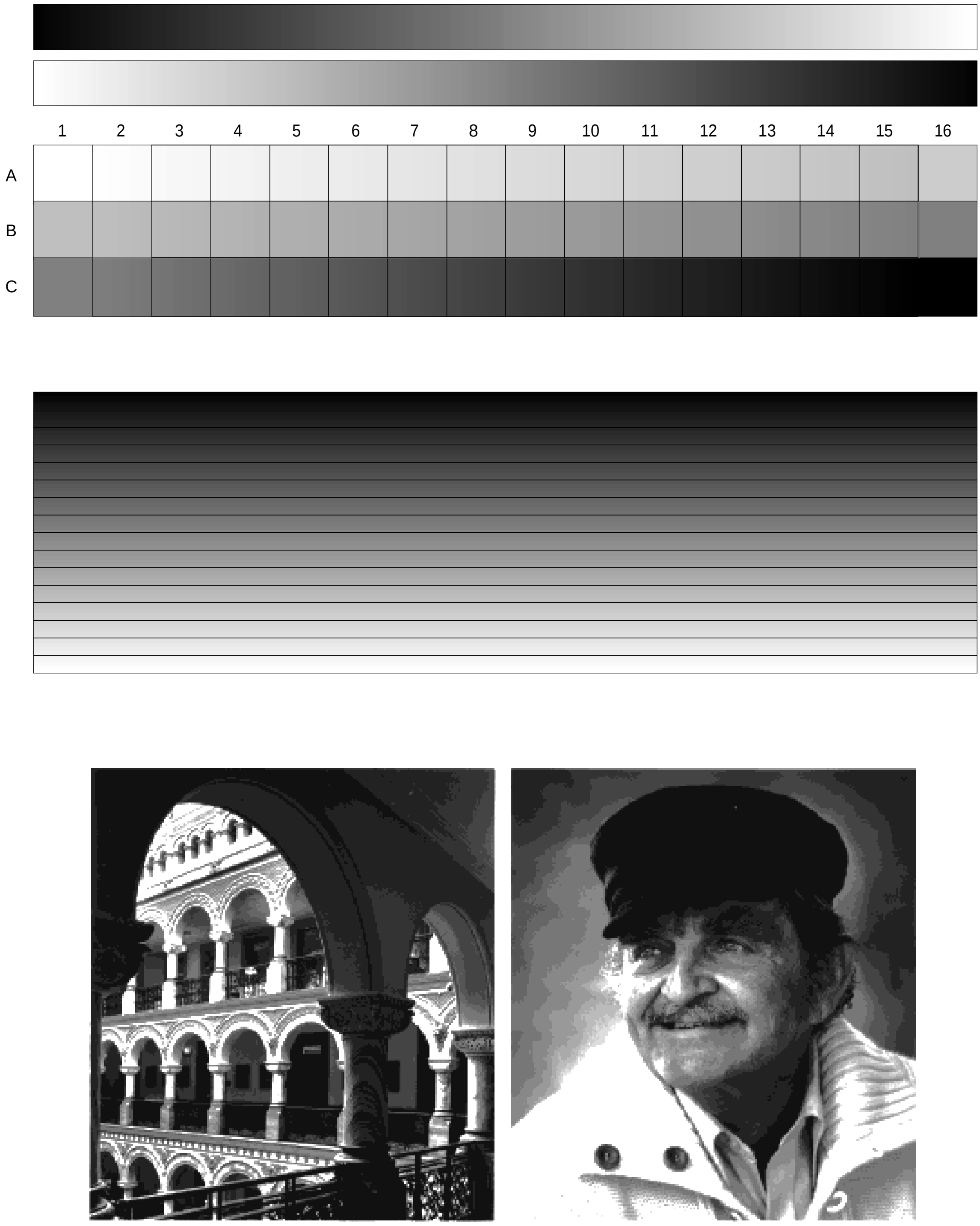


Figure F.1: Test chart No 1



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CONTINUOUS TONE FACSIMILE TEST CHART CTO1 ITU-T TEST CHART No 5

Figure F.2: Test chart No 2

Annex G: Guidance on the selection of representative samples and test calls

Where sampling and test calls are used the approach should ensure that the results adequately reflect the QoS perceived by customers for the period under review. The following aspects need to be taken into account; the list only contains general information and should be understood as a rough overview rather than a detailed description. The selection of representative samples and test calls is a process that is heavily influenced by specific technical and operational conditions of the measurement task. Therefore a detailed guidance cannot be given:

- The test program should be dedicated to the parameter that is to be measured (e.g. voice, fax, etc).
- In cases where the measurements are performed by parties other than the network provider (third parties) it should be ensured, that all relevant information that may influence the results is at hand. Normally only the network operator is aware of specific technical characteristics of the network access, software implementations, routing, etc. Depending on the parameters measured often additional information is needed in order to obtain comparable results. This is also valid for measurements of connections over more than one network (e.g. indirect services).
- Samples and test calls should ensure that traffic variations during the measurement period are taken adequately into account.
- The choice of adequate origin and destination NTPs for setting up test calls may be based on the national/international numbering plan or on traffic patterns/distribution or on geographic coverage.
- Depending on the kind of network(s) under study, i.e. fixed, mobile or combinations of them, network specific characteristics and user behaviour need to be taken into account.
- Network performance measurements are often based on the analysis of signalling information or on tones. When using such kind of information the measuring party should know in detail what kind of signalling system and/or tones are used in the network(s) under consideration. Especially any deviations to existing standards should be known e.g. the use of delta specifications to ITU-T Recommendation Q.850 [i.20].
- Measurements of parameters such as call set up time should take account of whether the calls are terminated on a user terminal or a function such as a mail box within the network. Such parameters will also be affected by some supplementary services (e.g. call forwarding). Also the performance for different number ranges may be different e.g. number translation services such as free phone and shared cost services may have increased call set up times.

G.1 Specific considerations for determining measurement locations and their distribution for voice quality measurements

Quality assessment of telephony service using an intrusive method (based on test calls=has the advantage to precisely analyze voice quality over defined configurations. This advantage is due to takes benefit of an end to end analysis using test signals. The whole transmission path is taken into account and the result is well correlated with user perception

The main drawback of this kind of analysis is to provide a limited view of the quality, because restricted to the analysed test configuration. It is known that for some kinds of transmission (e.g. packet transmission such as IP Protocol) the measurements done on one configuration are not easily applicable to the overall services.

Concerning VoIP several factors may influence telephony service quality and may introduce quality differences from a configuration to other, from a location to another one. Among these factors the enginery rules applied by the operator or service provider and the offer subscribed by the users. Other factors may be the network architecture including several equipments (home gateway, DSLAM, BAS, Call Server, Media Gateway, etc.) in the transmission path and several manufacturers for each type of equipment.

So a combination of configurations which may dramatically increase and with it may occur quality differences for the same telephony offer. It should also be indicated that the capabilities of traffic flow may change from a geographical location to another one and depending on the number of users and on network dimensions.

Only one measurement termination is not enough to obtain a macroscopic image of the service quality offered to the users. Ideally it should be necessary to deploy an analysis point per configuration but economic costs associated to probes (to buy and to install) are not realistic.

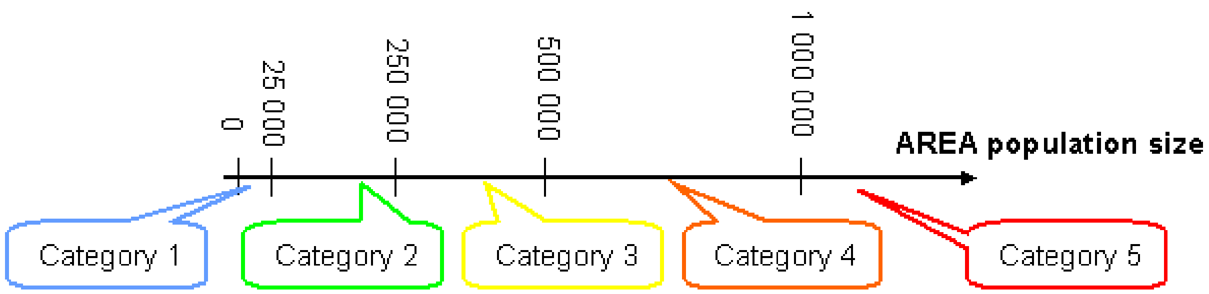
If only one analysis point and one analysis point per configuration are not satisfactory solutions for the supervision of the quality of IP telephony services, it is clear that more important the point number is better the service supervision will be. It is also important that the measurement points be located over the whole areas covered by the telephony offer but also that these measurement points be spread according to the size of AREAs.

In these conditions we can define 5 types of geographical areas to be covered by the analysis points:

- Areas with more than 1 000 000 inhabitants.
- Areas from 500 000 to 1 000 000 inhabitants.
- Areas from 250 000 to 500 000 inhabitants.
- Areas from 25 000 to 250 000 inhabitants.
- Areas with less than 25 000 inhabitants.

The whole process of identifying the minimum number of measurement points can be summarized in the following steps:

1) Identify the country population size to know the minimum number of measurement points:
   - population greater or equal to 10 000 000 → at least 10 analysis points;
   - population lower than 10 000 000 → at least 5 analysis points.
2) For this country define the term AREA:
   - strictly, the perimeter of cities;
   - perimeter of a set of cities making an urban continuity;
   - large perimeter of several cities not necessary in continuity.
3) Identify and classify the different AREAs in the 5 categories of size:



4) Determine the effective number of AREA categories for this country:
   - Depend of country size, population size, AREA perimeter, et.
5) Choose a measurement point for each effective AREA category:
   - Define a first series of measurement points by respecting a homogeneous geographical distribution.
6) Complete the location distribution.

Define the other measurement points by respecting the AREA typology of this country.

Two examples of measurement location determination are presented in annex H.

Annex H: Application examples of measurement location determination

H.1 Example for France

1) Identify and classify the different AREAs in the 5 categories of size:

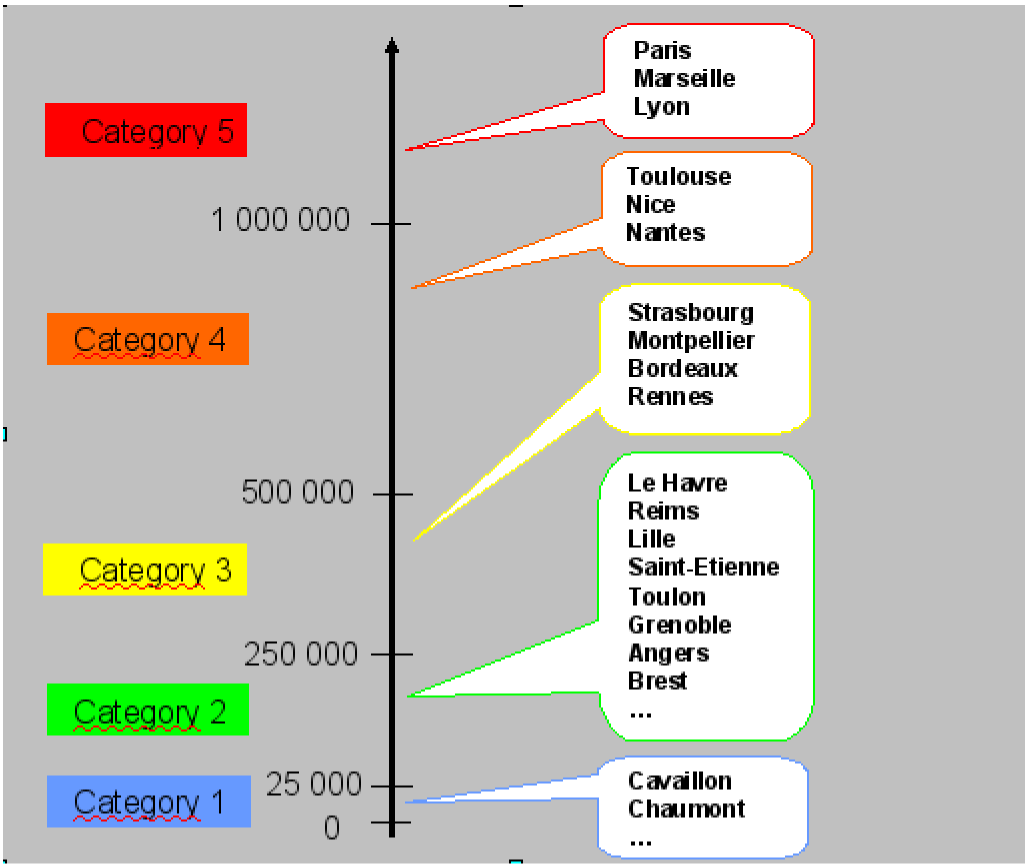


Figure H.1

2) Determine the effective number of AREA categories:

- → 5 effective AREA categories.

3) Choose a measurement point for each effective AREA category:

- Category 5: **Paris.**
- Category 4: **Toulouse.**
- Category 3: **Bordeaux.**
- Category 2: **Dijon.**
- Category 1: **Lannion.**

4) Complete the location distribution:

Several cities in every AREA category and an urban population ratio (75 %) showing that population is mainly situated in urban area.

As a consequence: it is conform to complete the measurement point distribution by choosing them principally in categories 2, 3, 4 and 5.

- Category 5: Marseille, Lyon.
- Category 4: Nantes.
- Category 3: Strasbourg.
- Category 2: Lille.

5) Overview.

H.2 Example for Switzerland

1) Identify and classify the different AREAs in the 5 categories of size:

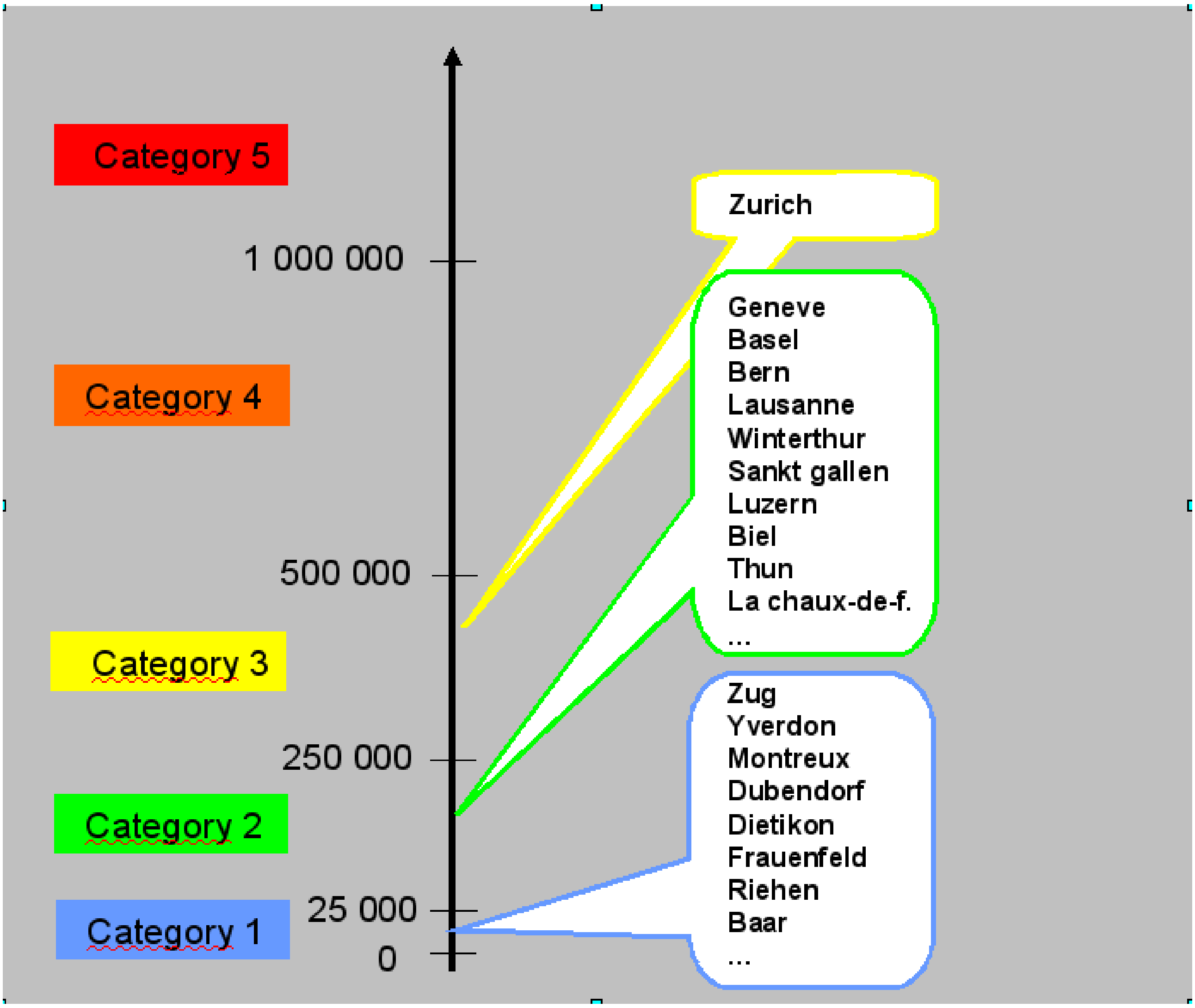


Figure H.2

2) Determine the effective number of AREA categories:

- → 3 effective AREA categories (no AREA in category 4 and 5).

3) Choose a measurement point for each effective AREA category:

- Category 3: **Zurich.**
- Category 2: **Geneva.**
- Category 1: **Locarno.**

4) Complete the location distribution:

Except Zurich, population is located in AREA of category 1 and 2.

Complete the measurement point distribution by choosing them in categories 1 and 2.

- Category 2: **Bern.**
- Category 1: **Altdorf UR.**

5) Overview:

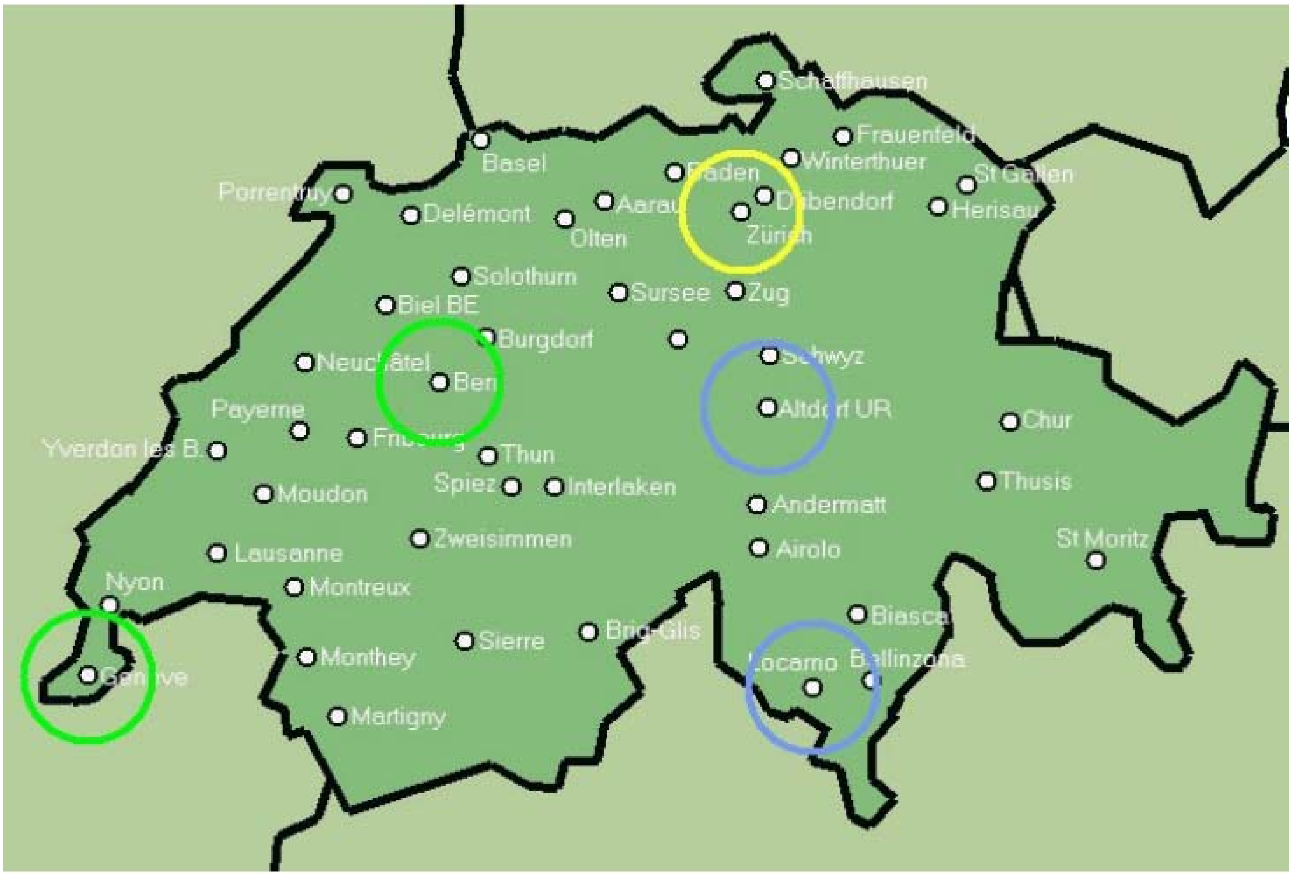


Figure H.3

Annex I: Calculation of the Design figure of merit for speech quality

There are three steps to calculate the design figure of merit for speech quality of a network:

1) Specifying the characteristics of the reference terminals to be used and their traffic weighting.

2) Obtaining the critical design parameters of the network and evaluating the main parameters of delay, qdu, Ie and echo.

3) Calculating the end-to-end performance for each combination of terminal and network and then calculating the traffic weighted mean value for all the combinations.

I.1 The reference terminals

Any number of different reference terminals may be used and their performance parameters should be specified in a way that includes the effect of the access network. For example in GSM the whole codec used across the radio access should be included, and for analogue fixed terminals the quantizing distortion for the analogue to digital conversations should be included with the terminals as should the loss effects of the analogue copper loop.

Figure I.1 shows three fixed terminals and three mobile terminals and their performance characteristics. The bottom row shows the traffic weighting percentage. This percentage will vary from country to country but in order to provide a real comparison the same percentages should be used by all operators whose results are to be compared with each other.

		Ph-1	Ph-2	Ph-3	Ph-4	Ph-5	Ph-6
		Analogue Set including analogue line and hybrid	DECT Set analogue connected	ISDN Set	GSM Set "large" including airpath and MSC	GSM Set "small" including airpath and MSC	GSM Set "Blackberry-style" including airpath and MSC
P	db(A)	35.0	35.0	35.0	50.0	50.0	50.0
SLR	dB	8.0	8.0	8.0	8.0	8.0	8.0
RLR	dB	2.0	2.0	2.0	2.0	2.0	2.0
STMR	dB	10.0	10.0	15.0	15.0	15.0	15.0
D		3.0	1.0	3.0	1.0	-1.0	-3.0
T	ms	0.0	0.0	3.0	95.0	95.0	95.0
Ta	ms	0.0	14.0	3.0	95.0	95.0	95.0
Tr	ms	0.0	28.0	6.0	190.0	190.0	190.0
qdu		0.5	0.5	0.5	0.5	0.5	0.5
Ie		0.0	7.0	0.0	5.0	5.0	5.0
A		0.0	0.0	0.0	0.0	0.0	0.0
TCLw	dB			45.0	55.0	50.0	45.0
EL	dB	25.0	25.0				
%		75	10	15	23	75	2

Figure I.1: Examples of terminal characteristics

I.2 The network performance characteristics

The characteristics to be obtained are:

1) Delay.

2) QDU.

3) Distortion (Ie).

4) Talker Echo Loudness Rating (TELR).

These characteristics can be obtained by asking a small number of relatively simple questions. The following is an example, but other methods can be used and the method can be adapted according to the characteristics of the country where the networks are located.

Different questions are asked of fixed and mobile operators. This clause explains each of the questions and the effect of their answers. The adjustments to the delay values are based on figures which can be found in ITU-T Recommendation G.114 [i.29]:

- Average delay per digital switch = 0,5 ms.
- Average delay per 100 km transmission = 0,5 ms.

Questions of the type asked of the fixed network operators do not normally need to be not asked of the mobile operators. The reasons are that mobile operators do not normally use low bit rate coding for internal transmission, and that the questions about network architecture and call path distance would not affect the results for mobile operators because the delay adjustments are very small compared to the delays involved in the mobile codecs.

Initial the delay, qdu and Ie values for the network should be set to zero and increased in response to the answers to the questions as described below.

I.2.1 Questions for fixed operators

F1: Does your network have two layers of digital switches/processors (i.e. with a separate transit layer) or one (do not count the layer for remote concentrators)?

If a network has two layers of switches rather than one layer then calls will on average pass through more switches and the length of the call path will increase. Therefore increase the network delay by 1 ms if a two-layer architecture is used. This increase covers one additional switch (0,5 ms) and 100 km of additional call path distance.

F2: What is the average area covered by each switch location that serves subscribers (i.e. local switch) in your network?

Where a network serves a wide area from a single switch location, the call paths from switch to subscriber will be longer than when the network provides more switches. The average area may be calculated by dividing the total area served by the number of switch locations. Note that switch locations rather than switches should be used because there may be more than one switch at each location.

In the case of new entrant operators, they may offer services only in the more populated parts of an area and not in the rural parts. For the purpose of assessing the area covered, the areas without coverage that are between areas with coverage should be included because the distances of the call paths will not be affected by the absence of coverage in some areas. For example, if the coverage of a switch looks like a chessboard with coverage only in the black areas, then the area should be taken as the whole area of the chessboard.

Calculate the typical additional path length as follows. Assume that the switch is at the centre of the area and that therefore the maximum path length is 0,5* Square root of the average area per switch location. The average path length is calculated as half the maximum because the switch will be located so as to minimize the path lengths and so the length will be less that the mathematical average distance.

This distance should be converted to additional delay using the formula that 100 km equates to 0,5 ms delay.

F3 What percentage of your calls is handled by low bit rate codecs between switches?

F4 What is the codec on these links?

This question applies only to the use of codecs within the operator's own network. Operators may use low bit rate codecs on links between their switches, especially if the links are leased lines rented from other operators. These codecs introduce additional delay and distortion.

The operator has to estimate the percentage of calls that pass through these codecs and to declare which type of codec is used. Where an operator runs a classical circuit switched digital network with no low bit rate coding (i.e. they use PCM A-Law - G.711 throughout their network), they should declare the percentage of calls as zero.

Table I.1 gives values of delay and distortion to be used.

Table I.1: Values for delay and distortion for different codecs

Codec	Additional delay (ms)	Distortion (Ie)
G.711	0	0
G.723.1-5.3	97	19
G.723.1-6.3	97	15
G.726-32	1	7
G.726-24	1	25
G.728	2	7
G.729	35	10
G.729A	35	11
Other	50	10

Multiply these values by the proportion of traffic that uses the given codec type and add the results to the network delay and distortion figures.

The values of delay and distortion for each codec are taken from ITU-T Recommendation G.113 [i.30] (Appendix I) and ITU-T Recommendation G.114 [i.29] (Appendix I).

F5: What percentage of your calls from directly connected subscribers is served by analogue copper loops?

F6: What is the average length of the analogue copper loops?

The characteristics of the fixed analogue terminals assume a zero length local loop. Account has to be taken of the effect of the analogue local loop on the loudness ratings and echo loss of the terminals. Therefore the operator is asked for the average length of its analogue loops and the percentage of calls that originate on such loops. This may not be the same as the fixed percentage assumed for analogue terminals.

If an operator is unable to calculate the percentage of calls then it should use the percentage of subscribers that are served by copper loops. This will normally be a higher figure than the percentage of calls and so there is an incentive for them to calculate the percentage of calls.

Where an operator uses carrier selection and pre-selection for call origination then the operator should calculate results separately for its own loops and for the loops of the access operator and weight the values according to the traffic proportions.

Calculate the loss of the local loop using a loss of 1,3 dB/km and adjust the SLR, RLR and echo (TELR) accordingly. This applies for analogue terminals as wells as for analogue connected DECT terminals.

F7: What percentage of your calls originates from carrier selection and pre-selection?

F8: What is the average area per interconnection points through which carrier selection traffic is received?

Carrier selection and pre-selection result in calls being carried over additional call paths and this adds delay, which needs to be weighted according to the percentage of calls originating from carrier selection or pre-selection. If an operator is unable to calculate the percentage of calls then they should enter the percentage of subscribers that use carrier selection. This will normally be a higher figure than the percentage of calls and so there is an incentive for them to calculate the percentage of calls.

It is also necessary to estimate the additional length of the call path that results from carrier selection or pre-selection. This is estimated form the average area per interconnection point through which carrier selection traffic is received.

Increase the network delay to account for the additional distance using the following formula:

Additional delay =
(% calls served by carrier selection)*0,25*(Square root of area per interconnection point) * 0,5 ms/100

I.2.2 Questions for mobile operators

The following applies to mobile networks based on the GSM/3GPP series of standards but where the calls are still carried in circuit switched form.

M1: Are you an MVNO with your own GMSC?

Mobile Virtual Network Operators (MVNOs) with their own GMSCs (= Gateway Mobile Switching Centre = provides an edge function within a PLMN (Public Land Mobile Network). It terminates the PSTN (Public Switched Telephone Network) signalling and traffic formats and converts this to protocols employed in mobile networks. For mobile terminated calls, it interacts with the HLR (Home Location Register) to obtain routing information. introduce extra delay into the call path compared to normal operators. Add 1,5 ms for such operators. This corresponds to one switching stage and 200 km transmission. Operators who are MVNOs but own only an HLR but not a GMSC should answer No (N).

Some mobile operators may use both their own network infrastructure and also national roaming on another network to provide service to their subscribers. National roaming may be used especially in rural areas. This situation really needs to be covered case by case by the operator who is reporting and they should adjust the figures for the support of EFR and TFO to represent the design performance of the network combination in accordance with the principles of this annex.

Treatment of codecs and Tandem Free Coding or Transcoder Free Operation (TrFO):

M2: What percentage of your network capacity supports both EFR and TFO?

M3: What percentage of your network capacity supports EFR but not TFO?

M4: What percentage of your network capacity does not support EFR but does support TFO?

M5: What percentage of your network capacity does not support either EFR or TFO?

Codecs

Mobile networks normally use any of three codecs:

- Adaptive Multi-Rate (AMR) for 3G access systems.
- Enhanced Full Rate (EFR).
- Full Rate (FR).

The choice of codec used in practice is the best that is supported by both the terminal and the network. Most networks support EFR and, if they use 3G, AMR.

The AMR codec uses the EFR algorithm when operating with a good signal to noise ratio and only uses different algorithms when the signal to noise is poor, i.e. at the edge of coverage. The AMR codec can therefore be treated as if it is the EFR codec as the calculations are concerned with the design for speech quality and not the coverage.

The parameters of the AMR, EFR and FR codecs together with the delays generated by the framing system for the mobile access are:

Table I.2: Mobile codec parameters without TFO

Codec	Delay (ms)	Distortion (Ie)
AMR	95	5
EFR	95	5
FR	95	20

These figures are taken from Appendix I to ITU-T Recommendation G.113 (Ie) [i.30] and Appendix I to ITU-T Recommendation G.114 [i.29].

Tandem free operation (TFO) or Transcoder Free Operation (TrFO)

Tandem free operation (TFO) (Transcoder Free Operation - TrFO- is the equivalent for 3GPP but for convenience we will just use the term TFO here) is a technique for improving quality where the calls between mobile terminals are not converted in the network from the AMR/EFR/FR codec to G.711 (A-Law) but are transmitted through the network in the coded form sent by the terminal. This reduces the impairments of a mobile call by one set of codec impairments. Failure to use TFO results in the following additional impairments:

Table I.3: Mobile codec parameters with TFO

Codec	Additional Delay (ms)	Additional Distortion (Ie)
AMR	50	5
EFR	50	5
FR	50	20

NOTE: TFO does not halve the delays because the framing in the radio access causes the majority of delay.

The effects on speech quality of codec choice and of TFO are large and therefore they need to be calculated in some detail.

EFR and TFO may be implemented only on certain switches in a network and for historical reasons the relationship between where each is implemented may vary. Therefore it may be necessary to ask separately for the capacity of the network that supports each combination.

The operator should calculate his answer by considering each mobile switching centre and its access network and the maximum number of calls that it can handle at the same time. This figure is the capacity for the functionality (EFR, TFO, etc.) of the switch.

Operators should not be asked separately about ERF and TFO because one could have 20 % supporting EFR and a different 20 % supporting TFO so both are never supported!

Note that the questions are about the proportion of network capacity not the proportion of actual calls.

For the following explanation, a variable is used for the proportion that corresponds to the percentage in each answer:

- M2: pEFRpTFO.
- M3: pEFRnTFO.
- M4: nEFRpTFO.
- M5: nEFRnTFO.

In addition account has to be taken of the fact that some calls will be on-net and benefit from EFR or TFO if provided whereas other calls will be off-net and terminated on other mobile networks whose capabilities may be different. Consequently a variable "pself" should be set for the typical proportion of calls that are terminated on-net, then (1-pself) are terminated off-net.

If the purpose is to produce a comparative figure for all networks in a specific country then the same value of pself should be used by all networks. If the objective is to produce a value that is a representative of the reporting network as possible, then the value that applies to real traffic on that network can be used.

An assumption also needs to be made for the terminating network for off-net mobile to mobile traffic. The following applies where it can be assumed that other networks support EFR and TFO. If this is unreasonable, then a different assumption should be agreed and the calculations changed accordingly.

Figure I.2 summarizes the situation for mobile to mobile calls where blue denotes EFR and red denotes TFO:

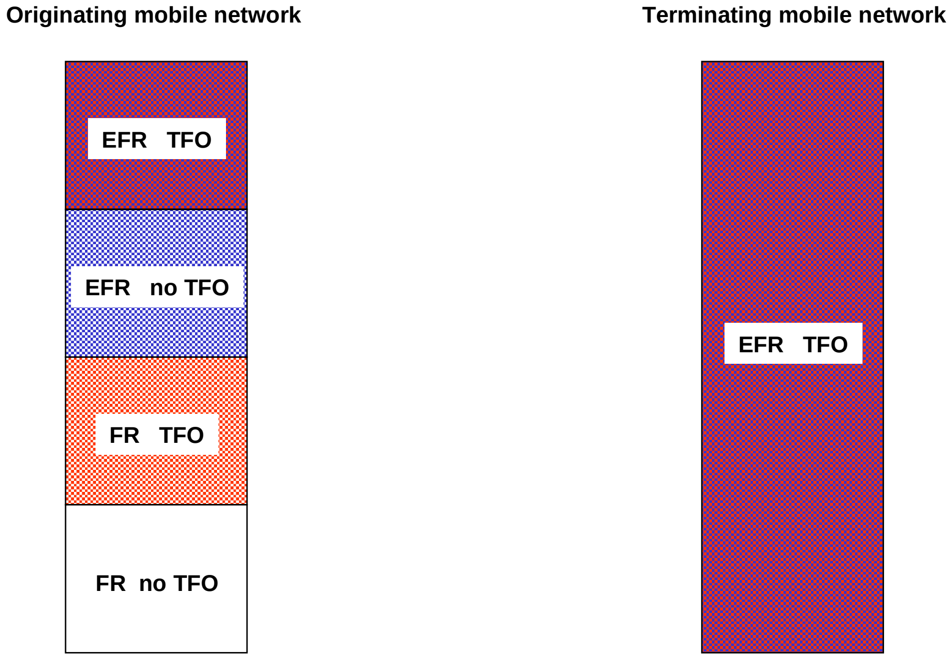


Figure I.2: EFR and TFO combinations assuming the terminating network supports EFR and TFO

There are five calls types to consider:

- EFR-EFR with TFO.
- EFR-EFR without TFO.
- FR-FR with TFO.
- FR-FR without TFO.
- FR-EFR and EFR- FR without TFO.

It is not necessary to consider the following types for the following reasons:

- EFR-FR with TFO because TFO requires support at both ends and so this reduces to FR-FR.
- FR-EFR with TFO because TFO requires support at both ends and so this reduces to FR-FR.

The proportions of mobile-mobile calls are then as follows, where pself is the proportion terminated on the originating operator's own mobile network.

Table I.4: Call proportions assuming that the terminating network supports EFR and TFO

	Call type	Proportion
A	EFR-EFR with TFO	PcallA = pEFRpTFO * ((1 - pself) + pself * pEFRpTFO)
B	EFR-EFR without TFO	PcallB = pEFRnTFO * ((1 - pself) + pself * pEFRnTFO) + 2 * pEFRpTFO * pself * pEFRnTFO
C	FR-FR with TFO	PcallC = nEFRpTFO * ((1 - pself) + pself * (2 * pEFRpTFO + nEFRpTFO))
D	FR-FR without TFO	PcallD = nEFRnTFO * pself * (nEFRnTFO + 2 * nEFRpTFO)
E	FR-EFR and EFR- FR without TFO	PcallE = nEFRnTFO * ((1 - pself) + 2 * pself * (pEFRpTFO + pEFRnTFO)) + 2 * pself * nEFRpTFO * pEFRnTFO

Since the terminal parameters given earlier have fixed values of delay and Ie corresponding to EFR, the network values should be calculated to compensate for this correctly.

Adjustments to delay

M-F calls

As the delay of EFR and FR is the same, there is no adjustment for M-F calls. TFO is not relevant.

M-M calls

For M-M, treat the different call types listed above with network values as follows.

Table I.5: Delay adjustments

	Call type	Originating terminal	Network	Receiving terminal	Total
A	EFR-EFR with TFO	95	-50	95	140
B	EFR-EFR without TFO	95	0	95	190
C	FR-FR with TFO	95	-50	95	140
D	FR-FR without TFO	95	0	95	190
E	FR-EFR and FR-EFR without TFO	95	0	95	190

Adjustments to Ie

M-F calls

Where the originating operator does not support EFR (call types A, D, E), Ie needs to be increased by 15.

M-M calls

Table I.6 shows the reality of the distortion values for the call types.

Table I.6: Ie adjustments

	Call type	Originating terminal	Network	Receiving terminal	Total
A	EFR-EFR with TFO	5	-5	5	5
B	EFR-EFR without TFO	5	0	5	10
C	FR-FR with TFO	20	-20	20	20
D	FR-FR without TFO	20	0	20	40
E (a)	EFR-FR without TFO	5	0	20	25
E (b)	FR-EFR without TFO	20	0	5	25

However the nominal terminals have an Ie value of 5 for each terminal and therefore the network adjustment factor has to compensate. Table I.7 shows the network compensation factors to achieve the correct end to end values.

Table I.7: Network compensation factors

	Call type	Network compensation factor
A	EFR-EFR with TFO	-5
B	EFR-EFR without TFO	0
C	FR-FR with TFO	10
D	FR-FR without TFO	30
E	FR-EFR and FR-EFR without TFO	15

The corrections in table H.7 should be added using the weightings given earlier.

M6: What is the average coverage area for each mobile switching centre?

Additional delay should be added using the same formula as in F2 but for the average coverage of each mobile switching centre.

I.2.3 Treatment of echo

Talker echo is the echo that speakers hears when they hear their own voice delayed and attenuated. The effect of talker echo on the speaker depends on both its delay and loudness.

Talker echo is generated by the sending path being connected to the receiving path. This connection occurs by either or both:

1) Imperfections in matching at a 4-wire to 2-wire hybrid in an analogue termination.
2) An acoustic path from ear piece to mouthpiece in the far-end handset.

Networks can use echo cancellers to reduce the level of echo by comparison of the sending and receiving paths and the application of cancellation techniques to the receiving path.

An assessment should be made of whether additional echo calculations are needed. The following gives guidance:

- Fixed terminals are unlikely to contain echo cancellation for the echo path through the terminal. This is the path that affects the caller to the fixed telephone. Therefore echo in calls to fixed terminals may be a problem and may need calculation.
- All mobile terminals contain echo cancellation for the echo path through the terminal. This is the path that affects the caller to the mobile. Therefore echo in calls to mobile terminals can be ignored.
- Fixed operators commonly do not use echo cancellation in their networks for national calls and therefore echo may occur with calls to fixed terminals. Echo does not occur on calls to mobile terminals because of the echo cancellation in the terminal.
- Mobile operators most commonly use echo cancellation in their networks for to fixed terminals, in which case echo for national calls need not be calculated.
- Mobile operators do not use echo cancellation in their networks for national mobile to mobile calls because the terminals provide the necessary cancellation.

In consequence echo calculations may be needed only on fixed to fixed calls.

For F-F calls the echo path attenuation is derived by the SLR plus RLR of the sending terminal plus the Echo Loss of the receiving terminal; this together with the end-to-end delay is entered into the E-model for the consideration of the impact of the talker echo.

I.3 Calculating the end-to-end performance

Having collected together all the data, E-Model calculations should be made for all the different terminal and network calculations. This clause explains how to make these calculations.

Fixed and mobile originating network should report separately, so if a network supports both fixed and mobile it should be considered as two separate networks.

An operator should make separate calculations for the following combinations:

1) Each type of originating terminal relevant to the type of network.
2) The network.
3) Each type of **fixed** terminal.

These resulting R-values should be weighted together using the weightings for each originating terminal and for each fixed terminal

This will give the weighted R-value for the fixed to fixed or mobile to fixed calls depending on the type of the originating network.

An operator should then make separate calculations for the following combinations:

1) Each type of originating terminal relevant to the type of network.

2) The network.

3) Each type of **mobile** terminal.

These resulting R-values should be weighted together using the weightings for each originating terminal and for each mobile terminal

This will give the weighted R-value for the fixed to mobile or mobile to mobile calls depending on the type of the originating network.

These are the results for the operator who is reporting.

To convert these results to the percentage of the best achievable, the exercise should be repeated using the following values for the network parameters but with the same values for the terminals.

Table I.8: Fixed network parameters for a perfect network

	Question	Answer
F1	Does your network have two layers of digital switches/processors (i.e. with a separate transit layer) or one (do not count the layer for remote concentrators)?	One
F2	What is the average area covered by each switch location that serves subscribers (i.e. local switch) in your network?	1 km^2
F3	What percentage of your calls is handled by low bit rate codecs between switches?	0
F4	What is the codec on these links?	Irrelevant
F5	What percentage of your calls from directly connected subscribers is served by analogue copper loops?	0
F6	What is the average length of the analogue copper loops?	Irrelevant
F7	What percentage of your calls originates from carrier selection and pre-selection?	0
F8	What is the average area per interconnection points through which carrier selection traffic is received?	1

Table I.9: Mobile network parameters for a perfect network

	Question	Answer
M1	Are you an MVNO with your own GMSC?	No
M2	What percentage of your network capacity supports both EFR and TFO?	100
M3	What percentage of your network capacity supports EFR but not TFO?	0
M4	What percentage of your network capacity does not support EFR but does support TFO?	0
M5	What percentage of your network capacity does not support either EFR or TFO?	0
M6	What is the average coverage area for each mobile switching centre?	1

For example, these inputs with the terminals specified in annex A give the maximum R-values as follows:

Table I.10: Example R-values for a perfect network

Call type	Maximum achievable R-value
Fixed to Fixed	88,47
Fixed to Mobile	76,66
Mobile to Fixed	85,54
Mobile to Mobile	76,79

Operators should then report separately the percentages for:

1) Calls to fixed.
2) Calls to mobile.

the values for their network over the values for a "perfect" network.

I.4 Terminals

Table I.11 show the parameters of example terminals that can be used, and at the bottom the example percentage of calls associated with each terminal type.

Table I.11: Example terminal parameters

		Ph-1	Ph-2	Ph-3	Ph-4	Ph-5	Ph-6
		Analogue Set including analogue line and hybrid	DECT Set analogue connected	ISDN Set	GSM Set "large" including airpath and MSC	GSM Set "small" including airpath and MSC	GSM Set "Blackberry-style" including airpath and MSC
P	db(A)	35.0	35.0	35.0	50.0	50.0	50.0
SLR	dB	8.0	8.0	8.0	8.0	8.0	8.0
RLR	dB	2.0	2.0	2.0	2.0	2.0	2.0
STMR	dB	10.0	10.0	15.0	15.0	15.0	15.0
D		3.0	1.0	3.0	1.0	-1.0	-3.0
T	ms	0.0	0.0	3.0	95.0	95.0	95.0
Ta	ms	0.0	14.0	3.0	95.0	95.0	95.0
Tr	ms	0.0	28.0	6.0	190.0	190.0	190.0
qdu		0.5	0.5	0.5	0.5	0.5	0.5
Ie		0.0	7.0	0.0	5.0	5.0	5.0
A		0.0	0.0	0.0	0.0	0.0	0.0
TCLw	dB			45.0	55.0	50.0	45.0
EL	dB	25.0	25.0				
%		75	10	15	23	75	2

The parameter values are taken from the default values of the E-Model and ETSI standards with the following exceptions:

- The STMR is reduced from 15 to 10 db to take account of older equipment.
- The D-factor is estimated based on experience of measurements.
- The delay parameters T of DECT is set to zero because it is assumed that there is echo cancellation otherwise the delay values are set according to the standards.

Annex J: Bibliography

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

ETSI TR 102 126: "Speech Processing, Transmission and Quality Aspects (STQ); Implementation of QoS parameter measurements according to ETSI EG 201 769".

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ETSI EG 202 009-2: "User Group; Quality of Telecom Services; Part 2: User related parameters on a service specific basis".

ITU-T Recommendation E.458: "Figure of merit for facsimile transmission performance".

ITU-T Recommendation G.Imp114/G.114 (02/01): "Implementors' Guides No. 1 and No. 2 for Recommendation G.114".

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ITU-T Recommendation T.30: "Procedures for document facsimile transmission in the general switched telephone network".

ITU-T Recommendation G.108: "Application of the E-model: A planning guide".

ITU-T Recommendation G.108.1: "Guidance for assessing conversational speech transmission quality effects not covered by the E-model".

History

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