ETSI EG 202 057-4 V1.1.1 (2005-10)

ETSI Guide

Speech Processing, Transmission and Quality Aspects (STQ);
User related QoS parameter definitions and measurements;
Part 4: Internet access

Reference
DEG/STQ-00042

Keywords
Internet, QoS

ETSI

650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
Association à but non lucratif enregistrée à la
Sous-Préfecture de Grasse (06) N° 7803/88

Contents

Intellectual Property Rights 5
Foreword 5
Introduction 6
1 Scope 7
2 References 7
3 Definitions and abbreviations 8
3.1 Definitions 8
3.2 Abbreviations 9
4 General considerations 10
4.1 Scope 10
4.1.1 Understanding of the term "Internet access" 10
4.1.2 Internet accesses covered 10
4.2 Use of the parameters 11
4.3 Parameters and measurement issues 11
4.3.1 Real traffic monitoring versus test calls 12
4.3.2 Measurement 13
4.4 Data collection issues 14
4.4.1 Reporting for different classes of customers 14
4.4.2 Non standard levels of QoS 14
4.4.3 Data processing 14
4.4.4 Data collection period 14
4.5 Comparability of measurements 14
4.6 Publication of QoS parameters 15
5 QoS parameters for Internet access 15
5.1 Login time 16
5.1.1 Definition 16
5.1.2 Application 16
5.1.3 Measurement and statistics 16
5.2 Data transmission speed achieved 16
5.2.1 Definition 16
5.2.2 Application 16
5.2.3 Measurement and statistics 16
5.3 Unsuccessful data transmission ratio 17
5.3.1 Definition 17
5.3.2 Application 17
5.3.3 Measurement and statistics 17
5.4 Successful log-in ratio 17
5.4.1 Definition 17
5.4.2 Application 17
5.4.3 Measurement and statistics 18
5.5 Delay (one way transmission time) 18
5.5.1 Definition 18
5.5.2 Application 18
5.5.3 Measurement and statistics 18
Annex A (normative): Reference connection 19
Annex B (normative): Measurement set-up 20
Annex C (normative): Guidance on the determination of representative test calls 23
Annex D (normative): Specification of test file 25

Annex E (normative): Combination of weekly or monthly results **26**

Annex F (normative): Guidance on technical performance aspects of Internet accesses: Delay, packet loss and jitter **27**

F.1 QoS versus network performance 27

F.2 Implications of delay, packet loss and jitter 27
F.2.1 Delay 27
F.2.2 Jitter 28
F.2.3 Packet loss 28
F.2.4 Performance considerations for different applications 28

F.3 Further information 31

Annex G (normative): Further explanation of "X % of" **32**

Annex H (informative): Bibliography **33**

History 34

Intellectual Property Rights

IPRs essential or potentially essential to the present document may have been declared to ETSI. The information pertaining to these essential IPRs, if any, is publicly available for **ETSI members and non-members**, and can be found in ETSI SR 000 314: *"Intellectual Property Rights (IPRs); Essential, or potentially Essential, IPRs notified to ETSI in respect of ETSI standards"*, which is available from the ETSI Secretariat. Latest updates are available on the ETSI Web server (http://webapp.etsi.org/IPR/home.asp).

Pursuant to the ETSI IPR Policy, no investigation, including IPR searches, has been carried out by ETSI. No guarantee can be given as to the existence of other IPRs not referenced in ETSI SR 000 314 (or the updates on the ETSI Web server) which are, or may be, or may become, essential to the present document.

Foreword

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

The present document is part 4 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 contains general user related QoS parameter definitions and measurement methods that can be applied to any service as well as user related QoS parameter definitions and measurement methods for voice, data and fax services accessed via the public telecommunication network.

EG 202 057-2 contains user related QoS parameter definitions and measurement methods for voice, modem data, fax services and SMS accessed via the public telecommunication network. The data parameters are specified for the case where a V.9x series modem is used since this kind of modem is in common use.

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

EG 202 057-4 (the present document) 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 must be carried out at the NTP. However, the NTP in PLMNs is not precisely defined. Other methods must be used to achieve an adequate representation of the quality that would be perceived at the NTP for the parameters defined in the present document.

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 the standard. The standard 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.

Introduction

The present document provides definitions and measurement methods for various QoS parameters for Internet access. The parameters were developed on the basis of the user's Quality of Service criteria identified in the TR 102 276 [1].

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. This Guide 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 outside the scope of this Guide. The QoS parameters listed in this Guide are also not intended to assess the complete QoS of a telecommunication service. This guide 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 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.
- The degree to which the parameters will provide a reliable comparison of performance.
- The cost of measuring and reporting each parameter.

2 References

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

- References are either specific (identified by date of publication and/or edition number or version number) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.

Referenced documents which are not found to be publicly available in the expected location might be found at http://docbox.etsi.org/Reference.

[1] ETSI TR 102 276: "User Group; Users' Quality of Service Criteria for Internet Access in Europe".

[2] ETSI TS 102 250-5: "Speech Processing, Transmission and Quality Aspects (STQ); QoS aspects for popular services in GSM and 3G networks; Part 5: Definition of typical measurement profiles".

[3] ETSI TS 102 250-6: "Speech Processing, Transmission and Quality Aspects (STQ); QoS aspects for popular services in GSM and 3G networks; Part 6: Post processing and statistical methods".

[4] ITU-T Recommendation G.1010: "End-user multimedia QoS categories".

[5] ITU-T Recommendation G.1020: "Performance parameter definitions for quality of speech and other voiceband applications utilizing IP networks".

[6] ITU-T Recommendation Y.1540: "Internet protocol data communication service - IP packet transfer and availability performance parameters".

[7] ITU-T Recommendation Y.1541: "Network performance objectives for IP-based services".

[8] IETF RFC 792: "Internet Control Message Protocol".

[9] ITU-T Recommendation I.350: "General aspects of quality of service and network performance in digital networks, including ISDNs".

3 Definitions and abbreviations

3.1 Definitions

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

NOTE: Since the purpose of the present document is to formulate definitions for QoS parameters, these definitions are given in the main body of the text and are not repeated here.

authentication: process of verifying a claimed identity to ensure that the stated identity of a user is correct

authorization: process of determining if the presenter of certain credentials is authorized to access a resource or make use of a service

email: messages automatically passed from one computer user to another, often through computer networks and/or via modems over telephone lines

File Transfer Protocol (FTP): protocol that allows users to copy files between their local system and any system they can reach on the network

host: computer that provides client stations with access to files and printers as shared resources to a computer network

Internet: computer network consisting of a worldwide network of computer networks that use the TCP/IP network protocols to facilitate data transmission and exchange

Internet access: making available of facilities and/or services for the purpose of providing an access to the public Internet in order to provide a user with access to services or resources of the Internet

NOTE 1: The Internet access can be separated in two parts, the physical and the logical access. The physical access provides a connection form the user's premises to, but not including, the POP (normally a dial-up circuit or broadband link or leased line) whereas the logical access consist of the setting up of an account that later on enables the user by a login process with the ability to access to the services and resources of the Internet (normally by assigning an IP address).

NOTE 2: The physical and logical access may be provided by different service providers.

NOTE 3: The function of the physical access may be provided by several interconnected networks.

Internet Access Provider (IAP): organization that provides users with an Internet access

Internet Protocol (IP): main internetworking protocol used in the Internet. Used in conjunction with the Transfer Control Protocol (TCP) to form TCP/IP

IP address: four-byte number uniquely defining each host on the Internet, usually written in dotted-decimal notation with periods separating the bytes

EXAMPLE: 217.111.27.1 for IP Version 4.

login process: multi-step process which includes both authentication and authorization as well as other system start-up tasks in order to provide a user with access to services or resources.

public Internet: part of the Internet that is available to the general public

NOTE: The access is normally provided by Internet access and Internet service providers.

physical access provider: organization that arranges the provision of physical access from the user's premises to the POP

NOTE 1: Excluding, the POP.

NOTE 2: Usually a dial-up circuit or an ADSL link or leased line are used.

NOTE 3: The function of the physical access provider may be provided by several interconnected networks.

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

router: device which forwards packets between networks

NOTE: The forwarding decision is based on network layer information and routing tables, often constructed by routing protocols. An IP router forwards data based on IP source and destination addresses.

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

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 e.g. 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

3.2 Abbreviations

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

DSL	Digital Subscriber Line
FQDN	Fully Qualified Domain Name
GPRS	General Packet Radio Service
GSM	Global System for Mobile communications
IAP	Internet Access Provider
ICMP	Internet Control Message Protocol
kbit/s	kilobit per second
NTP	Network Termination Point
PC	Personal Computer
PING	Packet InterNet Groper
POP	Point Of Presence
PTN	Public Telecommunication Network
QoS	Quality of Service
SMS	Short Message Service
UMTS	Universal Mobile Telecommunications System
WLAN	Wireless Local Area Network

4 General considerations

4.1 Scope

4.1.1 Understanding of the term "Internet access"

Unlike to other (traditional) telecommunication services the Internet access - as it is used in these days - consists of different connections and services that have to be available in combination in order to allow for a functional Internet access. All these different connections and services can be understood as separate telecommunication services with their own QoS aspects. However, the quality of services as perceived by the user accessed via the Internet like e.g. web browsing and email will be influenced by the quality of each single element of the end-to-end connection/service.

Furthermore the term "Internet access" is not only understood by the naive user as an access in the real meaning of the word, i.e. the provision of a physical connection and being able to establish connections to other parts or accesses of the network. The user understands Internet access as having access to (end-to-end) services that rely on the transport mechanisms of the Internet. This is because for most users the pure (physical) access to the internet is of no practical use; only if the user gains access via e.g. web browsing to information and applications available on servers he can use the "Internet".

However, from a technical point of view the (end-to-end) services/applications are offered independently to the (physical) Internet access. This is also reflected in the fact that the majority of end users need to have two contracts in order to have full Internet access: one for an access to the PTN in order to be able to connect to an IAP and another one with an IAP/ISP in order to access services provided via the Internet.

In order to use the Internet, the user needs first to have access to the Internet (in most cases via the public PTN). Technically spoken, he must be able to have access to the transport mechanisms of the Internet, i.e. having access to IP layer transmission. This provides him with the ability to connect to other entities of the Internet (IP based network). From there on the user may access advanced services that involve higher layers (above the IP layer) of communication. These services may be offered totally independent of the physical access. Thus Internet access can be understood as a (transportation) platform to access advanced services.

Therefore the term Internet access should primarily be understood as physical access to the core of the Internet, i.e. the access includes all functionalities that are needed to enable the user to establish connections to other entities within the Internet and engage advanced services. All issues beyond that basic understanding of an Internet access are highly dependent on the specific end-to-end service used and therefore should be subject to additional service specific considerations.

4.1.2 Internet accesses covered

The present document provides QoS and network performance parameters for assessing the quality of Internet access as perceived by the user. The purpose of the parameters is to inform the user on the transmission performance of the assessed Internet access.

The term Internet access includes the physical access between the user's terminal equipment and the access to the network of the IAP who is providing the end user with access to the Public Internet itself. The scope of the parameters is limited to the Internet access itself, i.e. the connection between end user and IAP and the availability and reliability of the access. The quality of end-to-end services accessed via this connection is outside the scope of the present document.

Internet access is normally not provided by a single service or network provider as it is possible in the case of other telecommunication services like telephony services. The Internet access consists of a combination of different connections and services.

The user gains access to the Public Internet via a suitable terminal equipment e.g. a PC that can be connected to the PTN. The Internet access itself is normally provided by an IAP. The connection between the terminal equipment and the IAP is established via a transit network. In most cases this will be the PTN, but it may also be a LAN or WLAN. The overall quality of the Internet access is a combination of the performance of each element of this connection.

Annex A provides an reference scenario for the Internet access and illustrates the application of the QoS parameters.

The parameters are in principle applicable for any kind of Internet access technologies. This includes the following access types:

- fixed narrowband access technologies like modem dial-up/ISDN connections;
- fixed broadband access (DSL, cable modem);
- wireless access technologies like WLAN, GSM, GPRS and UMTS.

NOTE: Although it is stated here that the measurement methods defined in this Guide can be applied in principle to any access technologies including wireless, care should be taken when applying the measurement methods. The proposed measurement methods are focused access technologies for the provision of Internet access at fixed locations. The measurement set-up does not take into account effects due to a moving user as it is the case with wireless accesses. Thus the QoS parameters can only be applied to Internet accesses with wireless technologies when they are provided at a fixed location.

The definitions and measurement methods of the parameters were elaborated primarily in order to assess QoS aspects of "standard", i.e. typical accesses. Therefore mainly common aspects and technologies were considered and are reflected in the present parameters. In principle the 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. Depending on the set of parameters used by the stakeholders the scope of the services covered may vary.

In many cases the provider of Internet access services may depend on other providers for part of the service. An example is an IAP who is offering access to Internet services but does not provide the access from and to the NTP. 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 by stakeholders.
- 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 Parameters and measurement issues

Quality of Service as perceived by the user is affected by operational and technical aspects. The QoS parameters listed in this Part of the Guide are covering aspects that are related to the network performance of connections set up via Internet accesses. In other words the parameters cover the technical connection based quality aspects when the Internet access is in use. Operational Aspects like e.g. supply time, billing and customer relations are dealt with in Part 1 of the Guide.

Users are often interested in specific network performance parameters, i.e. delay, jitter, packet loss. These parameters however cannot directly be related to the quality a user will perceive. One always has to take into account the specific service that is in use. Some services may react very sensitive to variations or degradation of this parameters whereas other services can compensate them very well. Therefore there are no specific QoS parameters in this Guide that provide separate statistics for theses parameters themselves. Instead of doing so in annex F basic information on the influence of these parameters on the user perceived quality is given.

The QoS parameters defined in the present document are designed in order to assess the quality of an Internet access that is understood as being the connection between an end user and an IAP. Furthermore it is assumed that an end user will use this access in order to get access to and/or make use of more advanced services that rely on the Internet Protocol.

Thus the basic functions an Internet access must provide is to allow for establishing IP connections. Besides that the end user is also interested in the reliability and availability of the access. Some basic transmission characteristics of the connection between the UNI and the IAP are also important because the performance of the services the end user may use are affected and limited by the performance of these parameters.

The QoS parameters to be measured include the performance of end user equipment as well as equipment at the IAP side of the connection. The influence of the terminal equipment on the resulting quality of the connection is significant.

4.3.1 Real traffic monitoring versus test calls

Network performance related parameters are measured either by using test calls or by monitoring real traffic (or by a combination of both). Both methods have their advantages and disadvantages.

Real traffic monitoring is on the one hand a low cost alternative but on the other hand the measuring party has no influence on the connection details (termination, services, end equipment etc.) and therefore reproducibility is affected and results may be less reliable. Whereas test call measurements are more complex and expensive but on the other hand they are reproducible, i.e. it is exactly known what is measured.

It is up to the measuring party to decide which approach should be taken in order to establish the required statistics. In the following list several aspects of the different measurement approaches that have to be considered are given.

The approach of test calls has the following advantages:

- It measures the network from an external point as would be seen by the user and so does not depend on any correct functioning of the network to enable a measurement to be made.
- The same test system can be used to compare results for different networks and so the comparability of results at the same point and time is high, although the results are not necessarily highly representative of the performance of the whole network.

This approach however has the following disadvantages:

- The test configurations (i.e. the terminal and its method of use) are not indicative of how users actually handle their terminals.
- In order to obtain adequate accuracy (representing a whole network) for comparison purposes, a large number of samples is needed.
- It is difficult to set up test plans that are representative of users' behaviour in both location and timing and thus not be necessarily be representative of the performance of the whole network. Normally the number of test connections is rather low compared to real traffic monitoring, therefore the representativity of test calls is very important.

The real traffic monitoring approach is based on the use of signalling and transmission characteristic related information that can be gathered in the gateways and routers of the network. This approach has the following advantages:

- It includes the effects of all calls, and so provides better comparability of congestion and network failures.
- It takes account of changes in terminal equipment and the actual performance achieved by real terminals used by real users.

This approach however has the following disadvantages:

- It does only measure the performance of actual transmissions and thus failures and faults concerning the physical access to the network may not be covered.
- It depends on software algorithms in the routers and gateways, and the algorithms of different manufacturers may differ and there may be differences between algorithms in different versions of the same software. Thus the measurement results may not be reliable and additional effort must be spent in the interpretation of the raw data.
- Terminal related effects are unknown, so it may be impossible to distinguish between network/access related faults and terminal/user related ones.

- The distribution of terminating NTPs/destinations is unknown prior to the measurement. Thus depending on the geographic coverage of the measurements long observation times may be needed in order to achieve a fair distribution and collection of enough calls for a secure statistical basis.

- Signalling information may not be used in accordance with the standards and it may be impossible to find out what each network section is using since routing tables may be altered frequently.

For both measurement methods additional information has to be known and preparation time to be spent prior to the actual measurements. For test calls a precise measurement plan has to be set-up in order to establish a reliable statistical basis. Real traffic monitoring has to rely on several assumptions (e.g. set-up of routers and gateways in other networks, distribution of terminating NTPs, kind of terminals involved, traffic distribution) that have to be checked first. In practice this will be done on a sample based examination.

If the measurements are performed by a party other than the access network provider, special care must be taken that all relevant information concerning the access is known (signalling system, set-up of switches, tones, etc.).

In principle both methods or a combination of them can be used in order to assess the quality of an Internet access based on the parameters defined in the present document.

4.3.2 Measurement

Even though the parameters can in principle be measured by using either real traffic monitoring or test calls, the measurement methods defined here are all based on a test call scenario. For the time being there is no known real traffic measurement scenario that could be used to assess the quality of an Internet access and provide comparable QoS statistics. This is for further study.

The basic measurement set-up consists of a Test-PC and a Test-Server with specified software and hardware. Test calls have to be established between the Test-PC and Test-Server and measurements must be made for the respective QoS parameters given in clause 5. Also the necessary number of test calls and their distribution must be defined.

The measurement set-up is given by the following annexes:

- Annex B: Measurement set-up.

- Annex C: Number and Distribution of test calls.

- Annex D: Specification of a test file.

Specific measurement conditions for the respective parameters are dealt with in clause 5.

The reference configuration given in annex B is based on test calls and the test-server is located as near as possible to the gateway providing the interconnection between access network and IAP network. Thus the measurements will not take into account the influence of the IAP network itself (defined between this gateway and the gateway interconnecting with the Internet) on the QoS offered to the user. Therefore the parameters will measure the quality of the connection between the end-user and its IAP access server (i.e. the test server). This quality is not the quality the user will perceive when he actually will access Internet services via this access since the quality of the IAP network and the Internet itself is not taken into account.

Despite of these deficiencies this approach has been chosen since it allows for comparable and reproducible results. The parameters in this Guide are intended to measure the quality of an Internet access while trying to avoid any influence on the measurements caused by the use of specific services in combination with network infrastructure. For the time being the only know practical solution is to trigger the measurements on the allocation of an IP address which happens at an access server behind the gateway of the IAP network to the access network. In this way comparable and reproducible QoS statistics can be produced.

Even if the measurements will not reflect the QoS as perceived by the user when accessing Internet services, they are a good estimation that allow for comparing different offers of Internet access.

4.4 Data collection issues

4.4.1 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.

4.4.2 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.

4.4.3 Data processing

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 E for combining the weekly or monthly results.

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.

4.4.4 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.

4.5 Comparability of measurements

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

- Measurement methods may be implemented differently.
- Measurements based on signalling information may be unreliable because the signalling systems and are not implemented in a fully standardized manner.
- 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.

4.6 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 internet supported 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 Internet access

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
Login time	number of successful log-ins	Test calls	all IAP services that are accessed via a login process
Data transmission speed achieved	a) The maximum data transmission rate in kbit/s achieved. b) The minimum data transmission rate in kbit/s achieved. c) The mean value and standard deviation of the data transmission rate in kbit/s.	Test calls	all IAP
Unsuccessful data transmissions ratio	% unsuccessful data transmission	Test calls	all IAP
Successful log-in ratio	% Successful log-ins	Test calls	all IAP services that are accessed via a login process
Delay (one way transmission time)	a) The mean values of the delay in milliseconds. b) The standard deviation of the delay.	Test calls	all IAP

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
Login time	Time of fastest 80 % and 95 % of logins	IAP
Data transmission speed achieved	Maximum, minimum, mean value and standard deviation of transmission rate	IAP
Unsuccessful data transmissions ratio	Percentage of unsuccessful data transmissions	IAP
Successful log-in ratio	Percentage of successful log-ins	IAP
Delay (one way transmission time)	Time in seconds	IAP

5.1 Login time

5.1.1 Definition

The login time is the period starting when the data connection between the Test-PC and the Test-Server has been established and finishing when the login process is successfully completed.

An attempt to login is unsuccessful if the login process fails for any reason. If more than 5 consecutive attempts to login fail, an ISP outage is assumed.

5.1.2 Application

This parameter is applicable to all IAP services that are accessed via a login process.

5.1.3 Measurement and statistics

The time in seconds within the fastest 80 % and 95 % of logins are achieved should be provided.

The statistics should be calculated from test calls made according to the measurement set-up given in annex B and taking into account the representativeness requirements given in annex C.

The number of successful log-ins is counted. An attempt to login is unsuccessful, if it fails for any reason independent whether the fault is caused by the access network or the IAP.

Attempts to login that are classified as unsuccessful should be excluded.

5.2 Data transmission speed achieved

5.2.1 Definition

The data transmission speed is defined as the data transmission rate that is achieved separately for downloading and uploading specified test files between a remote web site and a user's computer.

5.2.2 Application

This parameter is applicable to all IAP.

5.2.3 Measurement and statistics

The following statistic should be provided separately for download and upload direction:

a) The highest 95 % of the data transmission rate in kbit/s achieved.

b) The lowest 5 % of the data transmission rate in kbit/s achieved.

c) The mean value and standard deviation of the data transmission rate in kbit/s.

NOTE: An explanation of the highest 95 % and the lowest 5 % of the data transmission rate is given in annex G.

The statistics should be calculated from test calls made according to the measurement set-up given in annex B and taking into account the representativeness requirements given in annex C. The data transmission rate is measured by downloading/uploading a test file specified in annex D.

The data transmission rate is calculated by dividing the size of the test file by the transmission time required for a complete and error-free transmission.

The transmission time is the time period starting when the access network has received the necessary information to start the transmission and ending when the last bit of the test file has been received.

5.3 Unsuccessful data transmission ratio

5.3.1 Definition

The unsuccessful data transmission ratio is defined as the ratio of unsuccessful data transmissions to the total number of data transmission attempts in a specified time period.

A data transmission is successful if a test file is transmitted completely and with no errors.

5.3.2 Application

This parameter is applicable to all IAP services.

5.3.3 Measurement and statistics

The percentage that is the sum total of unsuccessful data transmissions, divided by the sum total of all attempts to transmit a test file should be provided.

The statistics should be calculated from test calls made according to the measurement set-up given in annex B and taking into account the representativeness requirements given in annex C.

The unsuccessful data transmission is measured by downloading/uploading a test file specified in annex D when the connection to the IAP is available. An attempt to transmit the test file should be considered unsuccessful if it takes longer than 60 seconds.

NOTE: The threshold of 60 seconds refers to the limit for acceptable performance for bulk data transmission/retrieval of ITU-T Recommendation G.1010 [4].

5.4 Successful log-in ratio

5.4.1 Definition

The successful log-in ratio is defined as the ratio of successful log-ins to access the Internet when both the access network and the IAP network are available in full working order.

NOTE: This parameter is a measure for the availability of the Internet access. The access network and the IAP network are normally available and failures/unavailability of the networks only occur in exceptional cases. The most likely reason for unavailability of the Internet access is caused by congestion or malfunction of the access server of the IAP which leads to unsuccessful log-ins.

5.4.2 Application

This parameter is applicable to all IAP services.

5.4.3 Measurement and statistics

The percentage that is the sum total of successful log-ins, divided by the sum total of all attempt to login should be provided.

The statistics should be calculated from test calls made according to the measurement set-up given in annex B and taking into account the representativeness requirements given in annex C.

The number of successful log-ins is counted. An attempt to login is unsuccessful, if it fails for any reason independent whether the fault is caused by the access network or the IAP.

NOTE: It should be noted that test measurements/samples have to be statistically independent. Therefore after detecting an unsuccessful log-in subsequent measurements cannot be done immediately.

If an attempt to login takes longer then 10 seconds it is classified as unsuccessful.

5.5 Delay (one way transmission time)

5.5.1 Definition

The delay is half the time in milliseconds, that is needed for an ICMP Echo Request/Reply (Ping) to a valid IP address.

5.5.2 Application

This parameter is applicable to all IAP services.

5.5.3 Measurement and statistics

The following statistic should be provided:

a) The mean values of the delay in milliseconds.

b) The standard deviation of the delay.

The statistics should be calculated from test calls made according to the measurement set-up given in annex B and taking into account the representativeness requirements given in annex C.

The delay is assessed by measuring half the time for a Echo Reply Message according to RFC 792 [8].

The standard deviation of the delay is a measure for the jitter.

Annex A (normative): Reference connection

The following figure provides a generic overview on the elements and network sections the Internet access consists of.

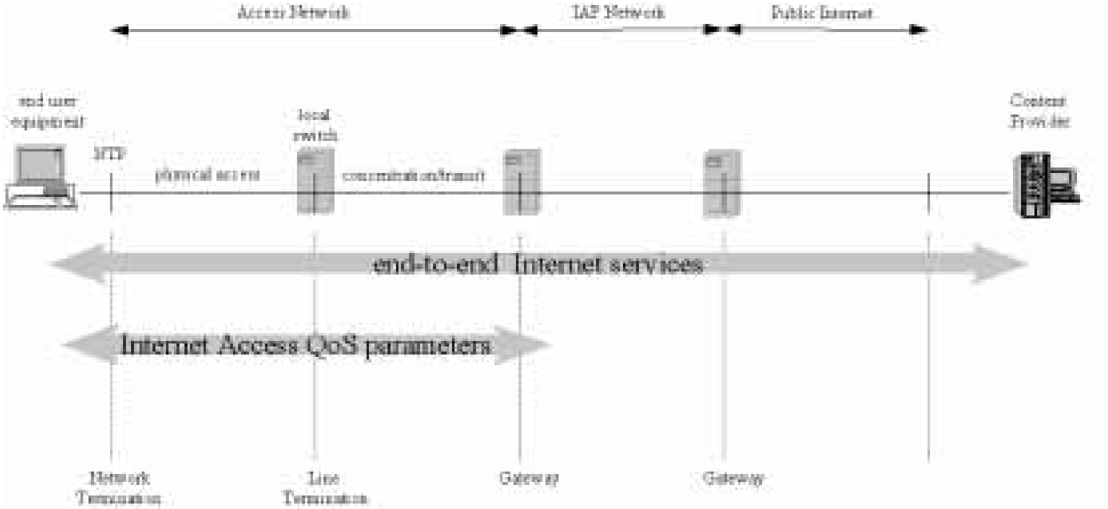

Figure A.1: Internet access elements and network sections

The end user's terminal equipment is connected to a NTP of the access network. In most cases the access network will be a PTN. In principle the access network can be divided in two parts: a section providing the physical access and a concentration/transit section. The physical access is stable with respect to its transmission characteristics, i.e. its characteristics are defined by the interface specification of the respective network and are not influenced by traffic variations. In other words, the maximum achievable transmission quality is determined/limited by the capabilities of the physical access. Further quality degradations are caused by the other sections of the connection to the IAP.

The physical access is connected to a local switch were several (hundred) accesses are combined. From there on the traffic of these end users is concentrated and routed to the respective IAPs. In this section of the connection it is most likely that the quality of the Internet access is significantly affected. The available bandwidth of this network section is limited and therefore each single connection will encounter variations of transmission quality due to traffic variations. Furthermore it is possible that the IAP may only be reached via several transit networks each adding additional impairments.

Last but not least the type of interconnection between IAP and access network, i.e. maximum possible bandwidth is influencing the Internet access quality. End-to-end Internet services will of course also be influenced by the quality of the IAP network, the public network and the content provider. But these effects are out of the scope of the QoS parameters. The QoS parameters defined in the present document are only assessing the quality of the connection between end user and IAP.

Annex B (normative): Measurement set-up

The measurement set-up consists of an Test-PC that is connected to the Access Network and a dedicated Test-Server that is situated within the IAP network, as shown in figure B.1.

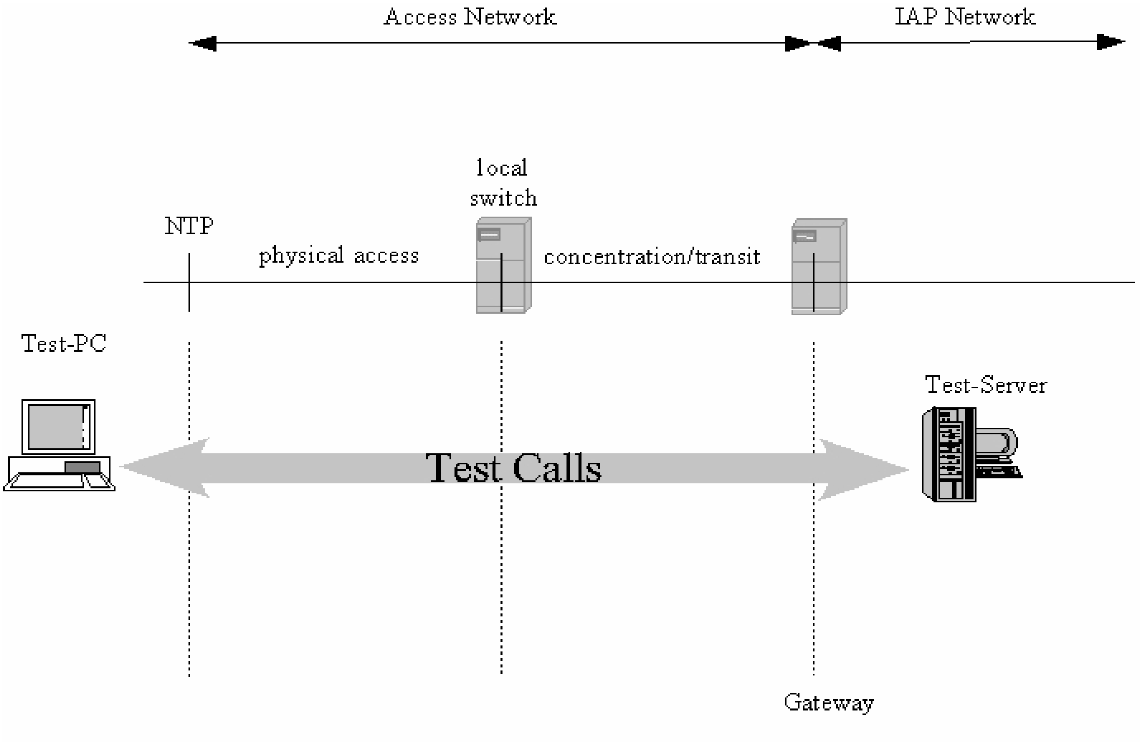

Figure B.1: Measurement set-up

Ideally the Test-Server should be placed as near as possible to the gateway providing the interconnection between access network and IAP network. This position has been chosen due to the understanding of the term "Internet access" as given in clause 4.1.1 in combination with the basic functionality of an Internet access which is to allow for establishing IP connections (see clause 4.3). However, the final choice of the positioning of the test server has to be made by the IAP taking into account network infrastructure and accessibility of facilities.

NOTE 1: The location of the test-server as near as possible to the gateway providing the interconnection between access network and IAP network implies that the measurements will no reflect the influence in the QoS of the IAP network, between that gateway and the gateway interconnecting with the Internet.
If one would intent to also take into account the influence of the IAP network on QoS, the test-server should then be located as near as possible to the gateway providing the interconnection between IAP network and the Internet. However this solution may imply scalability and cost issues, so it is left for further study.

Due to different implementations of the TCP/IP stack of various operating systems measurement result may vary depending on the configuration chosen for the measurement campaign. Therefore the Test-PC and Test-Server should always use the same operating system.

NOTE 2: The following specification of the Test-PC and Test-Server is based on TS 102 250-5 [2] and is provided as an example. When performing a measurement campaign specific values like e.g. window sizes and operating system may be changes. This might be necessary in order to take into account the network infrastructure and access types under consideration. In any case the same measurement set-up should be used for the whole measurement campaign.
The given settings also allow for using other operating systems than the one taken here as an example.

Requirements for the Test-PC:

For all data measurements TCP settings may be chosen at will.

If the measurements shall be used for comparison with other networks the following settings shall be used on the measurement client (based on the assumption, that the majority of the customers will use Microsoft WINDOWS XP™ Professional SP1 English):

- Maximum Segment Size between 1 380 Bytes and 1 460 Bytes.
- TCP RX Window Size = 16 384 Bytes.
- SACK enabled.
- ECN disabled.
- TCP Window Scaling disabled.
- TCP Timestamping disabled.
- PMTU Discovery disabled (but DF-bit set).
- TCP Fast Retransmit.
- TCP Fast Recovery enabled.
- Delayed ACK enabled (200 ms).

NOTE 3: The recommended TCP settings represent one of the possible (out-of-the-box) implementations of a common operating system of a client. The reason why this option was chosen was due to the fact that these are closer to the "default" user settings.

NOTE 4: Although the same TCP parameters may be used for benchmarking purposes, when different OSs are used in the tests, different results maybe obtained. This is due to the different implementations of the TCP/IP stack in the different OSs. This needs to be considered when comparing the results from a benchmarking exercise especially when the client and/or server OSs of the compared networks are not the same. This is a limitation which should not be overlooked, and a possible solution is to use the same version of the OSs in the client, as well as the server for all networks compared in a benchmark campaign (this does not imply that the OS of the client must be the same as the server).

NOTE 5: Proxy servers installed in the networks IP core network may act as the TCP peer instead of the application server the tests are performed against.

NOTE 6: Measurements with other settings may not be called conforming to the present document. There is no preference concerning the used operating system as long as these settings are used.

Requirements for the Test-Server:

For all tests a dedicated test server should be used as a well defined reference. Under no circumstances should a commercial server (e.g. www.yahoo.com) be used, since the content on such a server may change over time. This makes later reproduction of the results impossible.

The test server should be identified by an IP address and not by its FQDN in order to avoid issues with DNS lookup and including the DNS caching strategies of the used operating system into the measurement.

The TCP settings of the server tested against should also be recorded. Since the number of host operating systems for internet servers is larger than on the client side, no detailed recommendation concerning the TCP settings of the server is given. However, the TCP stack of the reference server should at least be capable of the following:

- Maximum Segment Size between 1 380 Bytes and 1 460 Bytes.
- TCP RX Window Size > 4 096 Bytes.
- SACK enabled.

- TCP Fast Retransmit.
- TCP Fast Recovery enabled.
- Delayed ACK enabled (200 ms).

Annex C (normative): Guidance on the determination of representative test calls

Spatial and temporal distribution of test calls:

The choice of adequate test calls, i.e. geographical locations of origin and destination of calls as well as traffic variations, is a crucial point with respect to the comparability and validation of the statistics to be calculated for the measured parameters.

Only general guidance on the correct choice of the temporal and spatial distribution of test calls can be given since this choice is highly dependent on the kind of networks under consideration. The test plan should be designed in order to ensure that the results adequately reflect the QoS as perceived by the user. The spatial distribution should take into account the actual network infrastructure, especially how homogeneous the accesses are whereas the temporal distribution of the test calls need to reflect the traffic variations of the real traffic.

NOTE: In Germany e.g. the approach is chosen that a spatial representativeness is achieved if the geographical locations at least cover the all eight one-digit numbering areas of the national numbering plan. A temporal representativeness is achieved if the test calls are scheduled according to the daily traffic distribution.

Number of test calls:

The necessary number of samples (test calls) with a given precision may be calculated as described in the following text. Generally it must be distinguished whether quantitative or qualitative characteristics are measured.

The QoS parameters clauses 5.2 and 5.5 are quantitative characteristics, whereas clauses 5.1, 5.3 and 5.4 are qualitative characteristics.

- Quantitative characteristics.

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

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

whereas:

n: Is the number of samples.

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

s: Is the expected standard deviation (calculated from former measurements or taken from a pilot study).

mean(x): Is the expected mean value (calculated from former measurements or taken from a pilot study).

a: Is the relative accuracy.

The number of observations must be chosen such that an absolute accuracy of X % or a relative accuracy of Y % with a confidence level of 95 % is achieved.

The following table gives the resulting values where:

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

- Qualitative characteristics

If k unsuccessful calls are observed out of N call attempts, then the true value of the unsuccessful call ratio lies between $k/N - \Delta$ and $k/N + \Delta$ with a confidence level $1-\alpha$, Δ being approximated (for large value of N) by:

$$\Delta \approx \sigma(\alpha)\sqrt{p\frac{(1-p)}{N}}$$

Where p is the expected unsuccessful call ratio and $\sigma(\alpha)$ is the $(1-(\alpha/2))*100$ percentile of the normal distribution with mean 0 and standard deviation 1 (N(0,1)). I.e. the number of call attempts to be observed should be:

$$N = \frac{\sigma(\alpha)^2\, p(1-p)}{\Delta^2}$$

The number of observations must be chosen such that an absolute accuracy of X % or a relative accuracy of Y % with a confidence level of 95 % is achieved.

If the confidence level is $1 - \alpha = 0{,}95$ then $\sigma(\alpha) = 1{,}96 \approx 2$.

If the required accuracy for $p \leq 0{,}01$ is $\Delta p = 0{,}001$, then the number of call attempts to be observed should be $N = 4 \times 10^6 \times p(1 - p)$ for a confidence level of 95 %.

If the required accuracy for $p > 0{,}01$ is $\Delta p/p = 0{,}1$, then the number of call attempts to be observed should be $N = 400 \times ((1 - p)/p)$ for a confidence level of 95 %.

For example, if the expected unsuccessful call ratio is 1 %, the number of call attempts to be observed should be $N = 4 \times 10^6 \times 0{,}01(1 - 0{,}01) = 39\,600$ for an accuracy of $\Delta p = 0{,}001$ with a confidence level of 95 %.

If the unsuccessful call ratio is expected to be 3 %, then the number of call attempts should be $= 400 \times ((1 - 0{,}03)/0{,}03) \approx 13\,000$ for a relative accuracy of $\Delta p/p = 0{,}1$ and with a confidence level of 95 %.

Further information:

Additional information on post processing and statistical methods can be found in TS 102 250-6 [3].

Annex D (normative): Specification of test file

The test file should consist of incompressible data. This is normally achieved by generating a sequence of random numbers. Another practical solution can be to use a data file that is already compressed, e.g. like a zip or jpg file, or to use the digits of the number Pi.

The test file should have at least twice the size (in kbit) of the theoretically maximum data transmission rate per second (in kbit/s) of the Internet access under consideration.

Annex E (normative): 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 F (normative): Guidance on technical performance aspects of Internet accesses: Delay, packet loss and jitter

F.1 QoS versus network performance

Technical performance aspects of telecommunication networks are assessed by Network Performance (NP) parameters rather than QoS parameters. While QoS and NP parameters are different in nature and serve different purposes, it is clear that there exist intrinsic relationships between QoS and NP parameters, one having a direct or indirect, and sometimes even inverse, influence on the other. It should be noted that the distinction between QoS and NP is not always clear-cut.

The following table that is taken out of ITU-T Recommendation I.350 [9] provides a conceptual categorization of Quality of Service (QoS) and Network Performance (NP) metrics.

Quality of Service parameter	Network Performance parameter
User oriented	Network provider oriented
Service related attributes	Network element and technology related attributes
Focus on user observable effects	Focus on planning development (design), operations and maintenance
Observed at service access points for the users, independent of network process and events	Observed at network connection element boundaries, e.g. relating to protocol specific interface signals

Network performance parameters are measured in order to assess the quality of a connection, e.g. call set-up time, signal-to-noise ratio, packet loss. The measurement results can than be used to estimate the quality or better a specific quality aspect of a service. Normally during the planning phase of a telecommunication network and hence service limits/performance targets are allocated for these parameters. The parameters are monitored regularly and if the parameters stay within their limits, an adequate QoS performance is assumed.

The QoS parameters defined in this Guide are end user related parameters that are intended to assess the QoS as perceived by the end user. The objective is to provide a comparable and objective basis for end users in order to allow for benchmarking of the different service offers available at the market.

Concerning the performance of Internet accesses the parameters delay, packet loss and jitter are the most important ones and have a significant influence on the resulting quality of services accessed via Internet accesses. Although users are normally not familiar with the technical implications of these parameters they are aware of them and are interested to receive information on the performance of these parameters.

When defining QoS parameters for Internet accesses it was felt that it is not appropriate to include QoS parameters for delay, packet loss and jitter as they are network performance related and no direct relationship between the performance of these parameters and the resulting user perceivable quality can be established by the naïve user.

Therefore some basic information on the impairments produced by delay, packet loss and jitter and their effect on the quality of different kind of applications that are commonly used in combination with an Internet access has been collected in this annex.

F.2 Implications of delay, packet loss and jitter

F.2.1 Delay

Delay manifests itself in a number of ways, including the time taken to establish a particular service from the initial user request and the time to receive specific information once the service is established. Delay has a very direct impact on user satisfaction depending on the application, and includes delays in the terminal, network, and any servers. Note that from a user point of view, delay also takes into account the effect of other network parameters such as throughput.

F.2.2 Jitter

Jitter (delay variation) is generally included as a performance parameter since it is very important at the transport layer in packetized data systems due to the inherent variability in arrival times of individual packets. However, services that are highly intolerant of delay variation will usually take steps to remove (or at least significantly reduce) the delay variation by means of buffering, effectively eliminating delay variation as perceived at the user level (although at the expense of adding additional fixed delay).

F.2.3 Packet loss

Packet loss has a very direct effect on the quality of the information finally presented to the user, whether it is voice, image, video or data. In this context, information loss is not limited to the effects of bit errors or packet loss during transmission, but also includes the effects of any degradation introduced by media coding for more efficient transmission (e.g. the use of low bit-rate speech codecs for voice).

F.2.4 Performance considerations for different applications

The following table provides an overview on performance consideration of delay, jitter and packet loss when using different applications.

Table F.1: Performance consideration of delay, jitter and packet loss using different applications

Application		Performance consideration
Audio	Conversational voice	Conversational voice is heavily influenced by one-way delay. In fact, there are two distinct effects of delay. The first is the creation of echo in conjunction with two-wire to 4-wire conversions or even acoustic coupling in a terminal. This begins to cause increasing degradation to voice quality for delays of the order of tens of milliseconds, and echo control measures must be taken at this point. The second effect occurs when the delay increases to a point where it begins to impact conversational dynamics, i.e. the delay in the other party responding becomes noticeable. This occurs for delays of the order of several hundred milliseconds. However, the human ear is highly intolerant of short-term delay variation (jitter). As a practical matter, for all voice services, delay variation due to variability in incoming packet arrival times must be removed with a de-jitterizing buffer. Effects of packet loss are influenced by the fact that the human ear is tolerant to a certain amount of distortion of a speech signal. In IP-based transmission systems a prime source of voice quality degradation is due to the use of low bit-rate speech compression codecs and their performance under conditions of packet loss.
	Voice messaging	Requirements for information loss are essentially the same as for conversational voice (i.e. dependent on the speech coder), but a key difference here is that there is more tolerance for delay since there is no direct conversation involved. The main issue, therefore becomes one of how much delay can be tolerated between the user issuing a command to replay a voice message and the actual start of the audio. There is no precise data on this, but based on studies related to the acceptability of stimulus-response delay for telecommunications services, a delay of the order of a few seconds seems reasonable for this application. In fact, a distinction is possible between recording and playback, in that user reaction to playback is likely to be the more stringent requirement.
	Streaming audio	Streaming audio is expected to provide better quality than conventional telephony, and requirements for information loss in terms of packet loss will be correspondingly tighter. However, as with voice messaging, there is no conversational element involved and delay requirements for the audio stream itself can be relaxed, even more so than for voice-messaging, although control commands must be dealt with appropriately.

Application		Performance consideration
Video	Videophone	Videophone as used here implies a full-duplex system, carrying both video and audio and intended for use in a conversational environment. As such, in principle the same delay requirements as for conversational voice will apply, i.e. no echo and minimal effect on conversational dynamics, with the added requirement that the audio and video must be synchronized within certain limits to provide "lip-synch". The human eye is tolerant to some loss of information, so that some degree of packet loss is acceptable depending on the specific video coder and amount of error protection used. It is expected that the latest MPEG-4 video codecs will provide acceptable video quality with frame erasure rates up to about 1 %.
	One-way video	The main distinguishing feature of one-way video is that there is no conversational element involved, meaning that the delay requirement will not be so stringent, and can follow that of streaming audio.
Data	Web-browsing	In this category we refer to retrieving and viewing the HTML component of a Web page, other components e.g. images, audio/video clips are dealt with under their separate categories. From the user point of view, the main performance factor is how quickly a page appears after it has been requested. Delays of several seconds are acceptable, but not more than about 10 seconds.
	Bulk data	This category includes file transfers, and is clearly influenced by the size of the file. As long as there is an indication that the file transfer is proceeding, it is reasonable to assume somewhat longer tolerance to delay than for a single Web-page.
	Highpriority transaction services	The main performance requirement here is to provide a sense of immediacy to the user that the transaction is proceeding smoothly, and a delay of no more than a few seconds is desirable.
	Command/control	Clearly, command/control implies very tight limits on allowable delay, much less than a second. Note that a key differentiator from conversational voice and video services with similar low delay requirements is the zero tolerance for information loss.
	Still image	This category includes a variety of encoding formats, some of which may be tolerant to information loss since they will be viewed by a human eye. However, given that even single bit errors can cause large disturbances in other still image formats, it is argued that this category should in general have zero information loss. However, delay requirements for still image transfer are not stringent and may be comparable to that for bulk data transfer, given that the image tends to be built up as it is being received, which provides an indication that data transfer is proceeding.
	Interactive games	Requirements for interactive games are obviously very dependent on the specific game, but it is clear that demanding applications will require very short delays of the order of a fraction of a second, consistent with demanding interactive applications.
	Telnet	Telnet is included here with a requirement for a short delay of a fraction of a second in order to provide essentially instantaneous character echo-back.
	E-mail (server access)	E-mail is generally thought to be a store and forward service which, in principle, can tolerate delays of several minutes or even hours. However, it is important to differentiate between communications between the user and the local email server and server, to server transfer. When the user communicates with the local mail server, there is an expectation that the mail will be transferred within a few seconds.
	Instant messaging	Instant messaging primarily relates to text, but can also include audio, video and image. In any case, despite the name, it is not a real-time communication in the sense of conversational voice, and delays of several seconds are acceptable.
	Background applications	In principle, the only requirement for applications in this category is that information should be delivered to the user essentially error free. However, there is still a delay constraint, since data is effectively useless if it is received too late for any practical purpose.
	Fax	Fax is included in this category since it is not normally intended to be an accompaniment to highly interactive real-time communication. Nevertheless, for so-called "real-time" fax there is an expectation in most business scenarios that a fax will be received within about 30 seconds. Delay for store and forward fax can be much higher. Note that fax does not require zero information loss.

Application		Performance consideration
	Low priority transaction services	An example in this category is Short Message Service (SMS). 10s of seconds are an acceptable delivery delay value.
	Email (server-to-server)	This category is included for completeness, since as mentioned earlier, the prime interest in email is in the access time.
	Usenet	Usenet is a world-wide distributed discussion system. It consists of a set of "newsgroups" with names that are classified hierarchically by subject. "Articles" or "messages" are "posted" to these newsgroups by people on computers with the appropriate software. These articles are then broadcast to other interconnected computer systems via a wide variety of networks. This is a very low priority service, with corresponding relaxed delay requirements. However, it is desirable that messages are received by the user in the order that they are posted, to avoid seeing a reply prior to the original message.

In ITU-T Recommendation G.1010 [4] the following performance targets are given for audio, voice and data applications.

Table F.2: Targets for audio and voice applications [4]

Medium	Application	Degree of symmetry	Typical data rates	Key performance parameters and target values			
				One-way delay	Delay variation	Information loss (Note 2)	Other
Audio	Conversational voice	Two-way	4 kbit/s to 64 kbit/s	< 150 ms preferred (Note 1) < 400 ms limit (Note 1)	< 1 ms	< 3 % packet loss ratio (PLR)	
Audio	Voice messaging	Primarily one-way	4 kbit/s to 32 kbit/s	< 1 s for playback < 2 s for record	< 1 ms	< 3 % PLR	
Audio	High quality streaming audio	Primarily one-way	16 kbit/s to 128 kbit/s (Note 3)	< 10 s	<< 1 ms	< 1 % PLR	
Video	Videophone	Two-way	16 kbit/s to 384 kbit/s	< 150 ms preferred (Note 4) < 400 ms limit		< 1 % PLR	Lip-synch: < 80 ms
Video	One-way	One-way	16 kbit/s to 384 kbit/s	< 10 s		< 1 % PLR	
NOTE 1: Assumes adequate echo control. NOTE 2: Exact values depend on specific codec, but assumes use of a packet loss concealment algorithm to minimize effect of packet loss. NOTE 3: Quality is very dependent on codec type and bit-rate. NOTE 4: These values are to be considered as long-term target values which may not be met by current technology.							

Table F.3: Targets for data applications [4]

Medium	Application	Degree of symmetry	Typical amount of data	Key performance parameters and target values		
				One-way delay (Note)	**Delay variation**	**Information loss**
Data	Web-browsing - HTML	Primarily one-way	~10 KB	Preferred < 2 s /page Acceptable < 4 s /page	N.A.	Zero
Data	Bulk data transfer/retrieval	Primarily one-way	10 KB-10 MB	Preferred < 15 s Acceptable < 60 s	N.A.	Zero
Data	Transaction services - high priority e.g. e-commerce, ATM	Two-way	< 10 KB	Preferred < 2 s Acceptable < 4 s	N.A.	Zero
Data	Command/control	Two-way	~ 1 KB	< 250 ms	N.A.	Zero
Data	Still image	One-way	< 100 KB	Preferred < 15 s Acceptable < 60 s	N.A.	Zero
Data	Interactive games	Two-way	< 1 KB	< 200 ms	N.A.	Zero
Data	Telnet	Two-way (asymmetric)	< 1 KB	< 200 ms	N.A.	Zero
Data	E-mail (server access)	Primarily one-way	< 10 KB	Preferred < 2 s Acceptable < 4 s	N.A.	Zero
Data	E-mail (server to server transfer)	Primarily one-way	< 10 KB	Can be several minutes	N.A.	Zero
Data	Fax ("real-time")	Primarily one-way	~ 10 KB	< 30 s/page	N.A.	< 10^{-6} BER
Data	Fax (store & forward)	Primarily one-way	~ 10 KB	Can be several minutes	N.A.	< 10^{-6} BER
Data	Low priority transactions	Primarily one-way	< 10 KB	< 30 s	N.A.	Zero
Data	Usenet	Primarily one-way	Can be 1 MB or more	Can be several minutes	N.A.	Zero
NOTE: In some cases, it may be more appropriate to consider these values as response times.						

F.3 Further information

Additional more detailed information on the impact of delay, packet loss and jitter can be found in the following documents:

- ITU-T Recommendation G.1010 [4]: "End-user multimedia QoS categories".
- ITU-T Recommendation G.1020 [5]: "Performance parameter definitions for quality of speech and other voiceband applications utilizing IP networks".
- ITU-T Recommendation Y.1540 [6]: "Internet protocol data communication service - IP packet transfer and availability performance parameters".
- ITU-T Recommendation Y.1541 [7]: "Network performance objectives for IP-based services".

Annex G (normative): 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 H (informative): Bibliography

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".

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

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

ETSI EG 202 057-2: "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".

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)".

ETSI EG 202 009-1: "User Group; Quality of Telecom Services; Part 1: Methodology for identification of parameters relevant to the Users".

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

History

Document history		
V1.1.1	August 2005	Membership Approval Procedure MV 20051014: 2005-08-16 to 2005-10-14
V1.1.1	October 2005	Publication