INTERNATIONAL TELECOMMUNICATION UNION

# ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

# G.1010

(11/2001)

SERIES G: TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

Quality of service and performance

## End-user multimedia QoS categories

ITU-T Recommendation G.1010

ITU-T G-SERIES RECOMMENDATIONS

**TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS**

| | |
|---|---|
| INTERNATIONAL TELEPHONE CONNECTIONS AND CIRCUITS | G.100–G.199 |
| GENERAL CHARACTERISTICS COMMON TO ALL ANALOGUE CARRIER-TRANSMISSION SYSTEMS | G.200–G.299 |
| INDIVIDUAL CHARACTERISTICS OF INTERNATIONAL CARRIER TELEPHONE SYSTEMS ON METALLIC LINES | G.300–G.399 |
| GENERAL CHARACTERISTICS OF INTERNATIONAL CARRIER TELEPHONE SYSTEMS ON RADIO-RELAY OR SATELLITE LINKS AND INTERCONNECTION WITH METALLIC LINES | G.400–G.449 |
| COORDINATION OF RADIOTELEPHONY AND LINE TELEPHONY | G.450–G.499 |
| TESTING EQUIPMENTS | G.500–G.599 |
| TRANSMISSION MEDIA CHARACTERISTICS | G.600–G.699 |
| DIGITAL TERMINAL EQUIPMENTS | G.700–G.799 |
| DIGITAL NETWORKS | G.800–G.899 |
| DIGITAL SECTIONS AND DIGITAL LINE SYSTEM | G.900–G.999 |
| **QUALITY OF SERVICE AND PERFORMANCE** | **G.1000–G.1999** |
| TRANSMISSION MEDIA CHARACTERISCTICS | G.6000–G.6999 |
| DIGITAL TERMINAL EQUIPMENTS | G.7000–G.7999 |

*For further details, please refer to the list of ITU-T Recommendations.*

**ITU-T Recommendation G.1010**

# End-user multimedia QoS categories


## Summary

This Recommendation defines a model for multimedia Quality of Service (QoS) categories from an end-user viewpoint. By considering user expectations for a range of multimedia applications, eight distinct categories are identified, based on tolerance to information loss and delay. It is intended that these categories form the basis for defining realistic QoS classes for underlying transport networks, and associated QoS control mechanisms.



## Source

ITU-T Recommendation G.1010 was prepared by ITU-T Study Group 12 (2001-2004) and approved under the WTSA Resolution 1 procedure on 29 November 2001.

## FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications. The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

## NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

## INTELLECTUAL PROPERTY RIGHTS

ITU draws attention to the possibility that the practice or implementation of this Recommendation may involve the use of a claimed Intellectual Property Right. ITU takes no position concerning the evidence, validity or applicability of claimed Intellectual Property Rights, whether asserted by ITU members or others outside of the Recommendation development process.

As of the date of approval of this Recommendation, ITU had not received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementors are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database.

CONTENTS

**ITU-T Recommendation G.1010**

## End-user multimedia QoS categories

## 1 Scope

The intent of this Recommendation is to provide guidance on the key factors that influence Quality of Service (QoS) from the perspective of the end-user. By considering a range of applications involving the media of voice, video, image and text, and the parameters that govern end-user satisfaction for these applications, a broad classification of end-user QoS categories is determined. It is intended that these categories be used as the basis for deriving realistic QoS classes and associated QoS control mechanisms for the underlying transport networks.

## 2 References

The following ITU-T Recommendations and other references contain provisions, which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published.

NOTE – The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

[1] ITU-T Recommendation F.700 (2000), *Framework Recommendation for multimedia services*.

[2] ITU-T Recommendation G.131 (1996), *Control of talker echo*.

[3] ITU-T Recommendation G.114 (2000), *One-way transmission time*.

[4] ETSI TS 101329-2 (2002), *Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; End-to-end Quality of Service in TIPHON Systems; Part 2: Definition of speech Quality of Service (QoS) classes*.

## 3 User-driven performance requirements

A major challenge for emerging wireline and wireless IP-based networks is to provide adequate Quality of Service (QoS) for different services. To do this requires a detailed knowledge of the performance requirements for particular services and applications. The starting point for deriving these performance requirements must be the user.

A typical user is not concerned with how a particular service is implemented. However, the user is interested in comparing the same service offered by different providers in terms of universal, user-oriented performance parameters. This implies that performance should be expressed by parameters that:

- Take into account all aspects of the service from the user's point of view;
- Focus on user-perceivable effects, rather than their causes within the network;
- Are independent of the specific network architecture or technology;
- Can be objectively or subjectively measured at the service access point;
- Can be easily related to network performance parameters;

- Can be assured to a user by the service providers(s).

# 4 Key parameters impacting the user

## 4.1 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.

## 4.2 Delay variation

Delay variation is generally included as a performance parameter since it is very important at the transport layer in packetised 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).

## 4.3 Information loss

Information loss has a very direct effect on the quality of the information finally presented to the user, whether it be 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).

# 5 Performance considerations for different applications

## 5.1 Audio

A general classification of audio into five levels of quality, and a mapping to various services, is given in [1]. More specific details are given below.

### 5.1.1 Conversational voice

Requirements for conversational voice are 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 (provision of echo cancellers etc [2]). 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 [3].

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.

Requirements for information 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.

### 5.1.2 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.

### 5.1.3 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 (see 5.3.4).

## 5.2 Video

A general classification of video into six levels of quality, and a mapping to various services, is given in [1]. More specific details are given below.

### 5.2.1 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 synchronised within certain limits to provide "lip-synch".

Once again, 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%.

### 5.2.2 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.

Taking into account the above considerations, performance targets for audio and video applications are shown in Table I.1.

## 5.3 Data

From a user point of view, a prime requirement for any data transfer application is to guarantee essentially zero loss of information. At the same time, delay variation is not generally noticeable to the user, although there needs to be a limit on synchronisation between media streams in a multimedia session (e.g. audio in conjunction with a white-board presentation). The different applications therefore tend to distinguish themselves on the basis of the delay which can be tolerated by the end-user from the time the source content is requested until it is presented to the user.

#### 5.3.1 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.

#### 5.3.2 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.

#### 5.3.3 High-priority transaction services (E-commerce)

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.

#### 5.3.4 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.

#### 5.3.5 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.

#### 5.3.6 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.

#### 5.3.7 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.

#### 5.3.8 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.

#### 5.3.9 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.

### 5.4 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.

#### 5.4.1 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.

#### 5.4.2 Low priority transaction services

An example in this category is Short Message Service (SMS). 10s of seconds are an acceptable delivery delay value.

#### 5.4.3 Email (server-to-server)

This category is included for completeness, since as mentioned earlier, the prime interest in email is in the access time.

#### 5.4.4 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.

Taking into account the above considerations, performance targets for data applications are summarised in Table I.2.

## 6 Classification of performance requirements into end-user Quality of Service categories

Based on the target performance requirements identified in Appendix I, the various applications can be mapped onto axes of packet loss and one-way delay as shown in Figure 1. The size and shape of the boxes provide a general indication of the limit of delay and information loss tolerable for each application class.

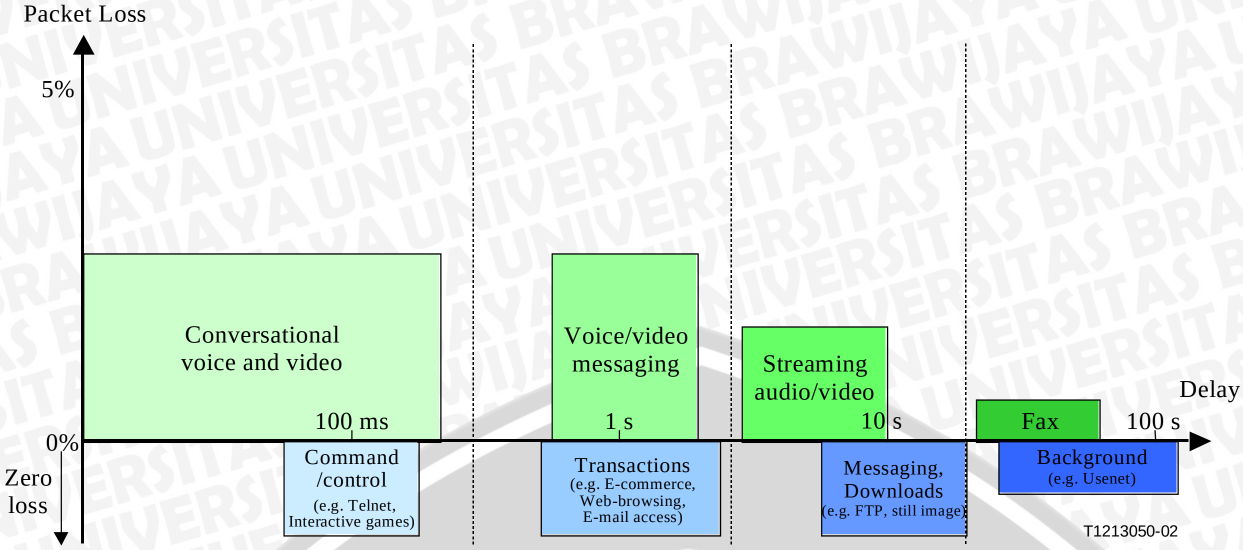


**Figure 1/G.1010 – Mapping of user-centric QoS requirements**

It can be seen that there are eight distinct groupings which encompass the range of applications identified. Within these eight groupings there is a primary segregation between applications that can tolerate some information loss and those that can not tolerate any information loss at all, and four general areas of delay tolerance.

This mapping can be formalised in Figure 2, to provide a recommended model for end-user QoS categories, where the four areas of delay are given names chosen to illustrate the type of user interaction involved. Of course, it is possible that each category could be subdivided into further categories to provide a range of quality levels for a specific service, as has been done for conversational voice in ETSI TS 101329-2 [4].

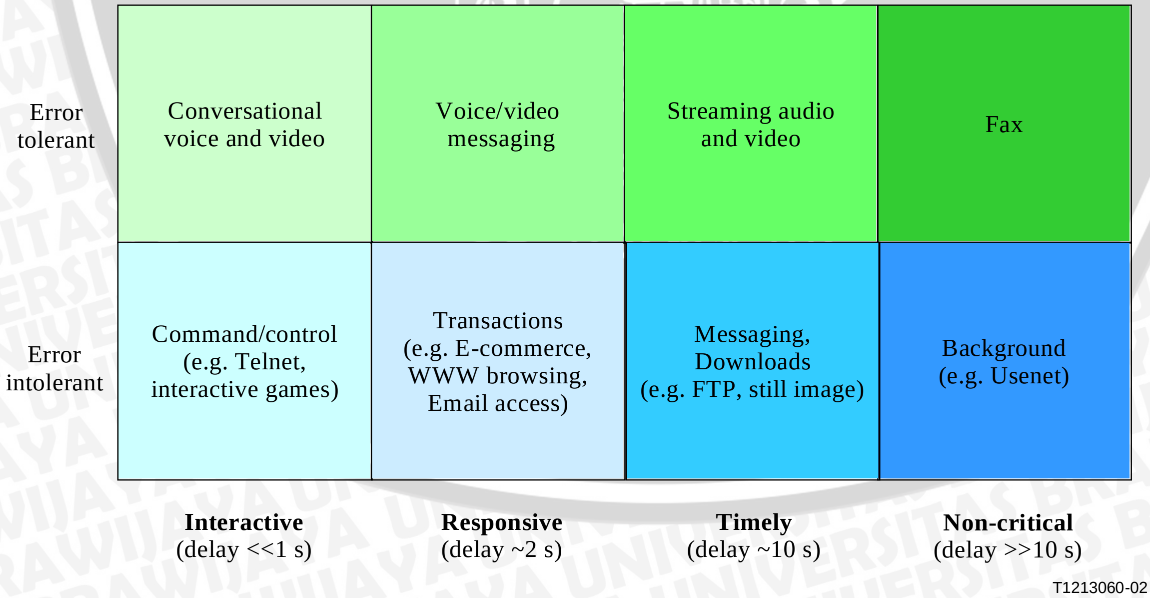


**Figure 2/G.1010 – Model for user-centric QoS categories**

Several useful features of this model are:

1) The model is based on end-to-end user perception of impairments and is therefore not dependent on any specific technology for its validity. This feature means that the model can be applied regardless of the underlying transport technology (IP, ATM, wireline, wireless etc).

2) The model provides an indication of the upper and lower boundaries for applications to be perceived as essentially acceptable to the user. Exceed an upper boundary (in loss or delay) and the service will be considered unsatisfactory; exceed a lower boundary, and the service will still be considered acceptable but may be wasteful from a network resource point of view because the service is using unnecessary resources.

3) The model provides a simple means of determining whether a bearer channel qualifies to carry a given application's data. For example, a channel with one-way delay of 1 second cannot support effective, natural real-time communication such as voice and Telnet. Furthermore, even if the one-way delay were reduced to 100 ms, Telnet would still be compromised if there were any information loss: loss of a single character would be conspicuous.

4) The model shows how the underlying impairments of information loss and delay can be grouped appropriately, without implying that one class is "better" than another (as in categorisations that use Gold, Silver, etc.). This can be used as the basis for deriving realistic and meaningful network QoS classes for differentiating service performance.

5) Note that the particular applications cited are exemplars rather than exhaustive. Other applications can be located in the schema by their similarity to these exemplars.

APPENDIX I

## Performance targets

Based on information in the Bibliography (Appendix II), Table I.1 provides an indication of suitable performance targets for audio and video applications.

**Table I.1/G.1010 – Performance targets for audio and video applications**

| **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 | 4-64 kbit/s | <150 ms preferred (Note 1)<br><400 ms limit (Note 1) | < 1 ms | < 3% packet loss ratio (PLR) | |
| Audio | Voice messaging | Primarily one-way | 4-32 kbit/s | < 1 s for playback<br>< 2 s for record | < 1 ms | < 3% PLR | |
| Audio | High quality streaming audio | Primarily one-way | 16-128 kbit/s (Note 3) | < 10 s | << 1 ms | < 1% PLR | |
| Video | Videophone | Two-way | 16-384 kbit/s | < 150 ms preferred (Note 4)<br><400 ms limit | | < 1% PLR | Lip-synch: < 80 ms |
| Video | One-way | One-way | 16-384 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 minimise 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.

Based on information in the Bibliography (Appendix II), Table I.2 provides an indication of suitable performance targets for data applications.

**Table I.2/G.1010 – Performance targets for data applications**

| 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<br>Acceptable < 4 s /page | N.A. | Zero |
| Data | Bulk data transfer/retrieval | Primarily one-way | 10 KB-10 MB | Preferred < 15 s<br>Acceptable < 60 s | N.A. | Zero |
| Data | Transaction services – high priority e.g. e-commerce, ATM | Two-way | < 10 KB | Preferred < 2 s<br>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<br>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<br>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.

APPENDIX II

## Bibliography


– KWOK (T.): Residential broadband Internet services and application requirements, *IEEE Communications Magazine*, June 1997.

– STEINMETZ (R.): Human Perception of Jitter and Media Synchronisation, *IEEE J-SAC*, Vol. 14, Issue 1, January 1996.

– SHNEIDERMAN (B.): Response Time and Display Rate in Human Performance with Computers, *ACM-Computing Surveys*, Vol. 16, No. 3, September 1984.

– MACDONALD (D.M.), ARCHAMBAULT (S.): Using customer perception in setting objectives for Intelligent Network services, *ITC-14*, 1994 .

– GALLAWAY (Glen R.): Response time to user activities in Interactive Man/Machine computer systems, *Proceedings of the Human Factors Society*, 25th Annual meeting, 1981.

– ETSI ETR 160, 1995, *Human factors aspects of multimedia telecommunications*.

– SHUR and MORTON, Performance of character-echo oriented applications on X.25 packet networks, *Globecom'92*.

– DANESHMAND (M.F.), ROY (R.R.) and SAVOLAINE (C.G.): Framework and requirements of Quality of Service for multimedia applications, *IIS'97*.

– GOLICK (Jerry): Catching the Multimedia Wave, *Data Communications Tech Tutorials*, 21st March 1999.

– WANG (S.Y.), MEYER (S.E.) and MY LAI: Quality of Service for Residential Broadband Video Service, *Globecom'96*.

– ANSI T1.522-2000, *Quality of Service for Business Multimedia Conferencing*.

– ZANETTIN (Alberto): Applications Prioritization: Giving the Green Light to Business Apps., *Data Communications Tech Tutorials*, March 1999.

– ISO-IEC/JTC 1/SC 29/WG 11 Report N2604, *Report of the formal verification tests on MPEG-4 video error resilience*, December 1998.

# SERIES OF ITU-T RECOMMENDATIONS

| | |
|---|---|
| Series A | Organization of the work of ITU-T |
| Series B | Means of expression: definitions, symbols, classification |
| Series C | General telecommunication statistics |
| Series D | General tariff principles |
| Series E | Overall network operation, telephone service, service operation and human factors |
| Series F | Non-telephone telecommunication services |
| **Series G** | **Transmission systems and media, digital systems and networks** |
| Series H | Audiovisual and multimedia systems |
| Series I | Integrated services digital network |
| Series J | Cable networks and transmission of television, sound programme and other multimedia signals |
| Series K | Protection against interference |
| Series L | Construction, installation and protection of cables and other elements of outside plant |
| Series M | TMN and network maintenance: international transmission systems, telephone circuits, telegraphy, facsimile and leased circuits |
| Series N | Maintenance: international sound programme and television transmission circuits |
| Series O | Specifications of measuring equipment |
| Series P | Telephone transmission quality, telephone installations, local line networks |
| Series Q | Switching and signalling |
| Series R | Telegraph transmission |
| Series S | Telegraph services terminal equipment |
| Series T | Terminals for telematic services |
| Series U | Telegraph switching |
| Series V | Data communication over the telephone network |
| Series X | Data networks and open system communications |
| Series Y | Global information infrastructure and Internet protocol aspects |
| Series Z | Languages and general software aspects for telecommunication systems |

*21883*

Printed in Switzerland
Geneva, 2002