RECOMMENDATION ITU-R M.1079-2

Performance and quality of service requirements for International Mobile Telecommunications-2000 (IMT-2000) access networks

(Question ITU-R 229/8)

(1994-2000-2003)

Summary

This Recommendation defines the speech/data quality and performance requirements for IMT-2000 access networks taking into consideration the end-to-end requirements. It also defines the connection/session performance, concerning issues such as call set-up time, delay characteristics and handover probability, to be achieved in the IMT-2000 access network that the user will expect in a network of comparable performance to the fixed network.

1 Introduction

IMT-2000 are third generation mobile systems which provide access to a wide range of telecommunication services, supported by the fixed telecommunication networks (e.g. public switched telephone network (PSTN)/ISDN/Internet Protocol (IP)), and to other services that are specific to mobile users.

Key features of IMT-2000 are:

– high degree of commonality of design worldwide;

– compatibility of services within IMT-2000 and with the fixed networks;

– high quality;

– small terminal suitable for worldwide use;

– worldwide roaming capability;

– capability for multimedia applications, and a wide range of services and terminals.

The capabilities of IMT-2000 systems are being continuously enhanced in line with market and technology trends.

This Recommendation on performance and QoS requirements defines the requirements for speech quality, data quality, connection/session performance and the radio interface performance to be achieved in IMT-2000.

Annex 1 contains information on planning tools to address end-to-end voice transmission quality.

2 Scope

This Recommendation defines the speech/data quality and performance requirements for IMT-2000 access networks, taking into consideration the end-to-end requirements.

This Recommendation lists the basic Recommendations essential for:

– achieving speech quality comparable to the fixed network by specifying natural speech, free, for example from excessive delay and echoes, that will enable users to converse easily using the IMT-2000 network, taking account of the full range of impairments like transcoding and environmental noise that are to be expected; and

– acceptable data quality and performance requirements.

This Recommendation also defines the connection/session performance, concerning issues like call set-up time, delay characteristics and handover probability, to be achieved in the IMT-2000 access network that the user will expect in a network of comparable performance to the fixed network. The scope of the Recommendation includes both the terrestrial and the satellite radio interfaces, as defined in Recommendation ITU-R M.1457.

3 Structure of the Recommendation

This Recommendation contains recommendations dealing with speech/data quality, connection/session performance, data, the requirements for other services and the radio performance requirements. In particular, QoS requirements are given in this Recommendation to meet end-to-end quality for services in wireless mobile networks.

4 Related documents

The following are the applicable related documents:

Recommendation ITU-R M.816: Framework for services supported on International Mobile Telecommunications-2000 (IMT-2000)

Recommendation ITU-R M.818: Satellite operation within International Mobile Telecommunications-2000 (IMT-2000)

Recommendation ITU-R M.819: International Mobile Telecommunications-2000 (IMT-2000) for developing countries

Recommendation ITU-R M.1034: Requirements for the radio interface(s) for International Mobile Telecommunications-2000 (IMT-2000)

Recommendation ITU-R M.1224: Vocabulary of terms for International Mobile Telecommunications-2000 (IMT-2000)

Recommendation ITU-R M.1225: Guidelines for evaluation of radio transmission technologies for IMT-2000

Recommendation ITU-R M.1311: Framework for modularity and radio commonality within IMT-2000

Recommendation ITU-R M.1457:	Detailed specifications of the radio interfaces of International Mobile Telecommunications-2000 (IMT-2000)
ITU-T Recommendation E.770:	Land mobile and fixed network interconnection traffic grade of service concept
ITU-T Recommendation E.771:	Network grade of service parameters and target values for circuit-switched land mobile services
ITU-T Recommendation E.800:	Terms and definitions related to quality of service and network performance including dependability
ITU-T Recommendation F.116:	Service features and operational provisions in IMT-2000
ITU-T Recommendation G.107:	The E-model, a computational model for use in transmission planning
ITU-T Recommendation G.114:	One-way transmission time
ITU-T Recommendation G.131:	Control of talker echo
ITU-T Recommendation G.173:	Transmission planning aspects of the speech service in digital public land mobile networks
ITU-T Recommendation G.174:	Transmission performance objectives for terrestrial digital wireless systems using portable terminals to access the PSTN
ITU-T Recommendation G.726:	40, 32, 24, 16 kbit/s adaptive differential pulse code modulation (ADPCM)
ITU-T Recommendation G.728:	Coding of speech at 16 kbit/s using low-delay code excited linear prediction
ITU-T Recommendation G.729:	Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear-prediction (CS-ACELP)
ITU-T Recommendation G.1010:	End-user multimedia QoS categories
ITU-T Recommendation P.79:	Calculation of loudness ratings for telephone sets
ITU-T Recommendation P.313:	Transmission characteristics for cordless and mobile digital terminals
ITU-T Recommendation P.800:	Methods for subjective determination of transmission quality
ITU-T Recommendation P.830:	Subjective performance assessment of telephone-band and wideband digital codecs
ITU-T Recommendation P.831:	Subjective performance evaluation of network echo cancellers

ITU-T Recommendation P.862:	Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs
ITU-T Recommendation Q.1701:	Framework of IMT-2000 networks
ITU-T Recommendation Q.1711:	Network functional model for IMT-2000
ITU-T Recommendation Y.1540:	Internet protocol data communication service – IP packet transfer and availability performance parameters
ITU-T Recommendation Y.1541:	Network performance objectives for IP-based services

5 Abbreviations and acronyms (see Note 1)

ADPCM	adaptive differential pulse code modulation
BER	bit error ratio
CN	core network
CRC	cyclic redundancy code
DCME	digital circuit multiplex equipment
FTP	file transfer protocol
GoB	good or better
GoS	grade of service
IP	Internet protocol
ISP	Internet service provider
MOS	mean opinion score
MT	mobile terminal
PDP	packet data protocol
PDU	protocol data unit
PLR	packet loss ratio
PoW	poor or worst
QoS	quality of service
RAN	radio access network
RANI	radio access network interface

RLP	radio link protocol
RRM	radio resource management
SDU	service data unit
SMS	short message service
VoIP	voice over IP

NOTE 1 – Additional abbreviations are provided in Recommendation ITU-R M.1224.

6 Definitions

6.1 QoS

The collective effect of service performances which determine the degree of satisfaction of a user of a service. It is characterized by the combined aspects of performance factors applicable to all services, such as:

– service operability performance

– service accessibility performance

– service retainability performance

– service integrity performance

– other factors specific to each service.

6.2 Speech quality

The speech quality expresses the degree of customer satisfaction with conversational speech transmission. Speech quality depends on the quality of the whole speech path from the talker at one end of the connection to the listener at the other, and can be categorized into two types of quality: quality which is mainly dependent on handset acoustics and quality which is mainly dependent on the transmission medium. Telecommunications services where special attention needs to be paid to speech quality, such as audio teleconferencing and voice mail, should also be considered.

6.3 Connection performance

Connection performance is expressed in ITU-T Recommendation E.770 as Grade of Service (GoS). GoS parameters consist of the signalling delay for call set-up and call release, and the probability of end-to-end blocking, as well as the probability of unsuccessful handover, etc.

6.4 Service retainability performance

Service retainability performance is defined in ITU-T Recommendation E.800 as the probability that a service, once obtained, will continue to be provided for a communication under given conditions, for example conditions of fading, shadowing and co-channel interference.

6.5 Reliability performance

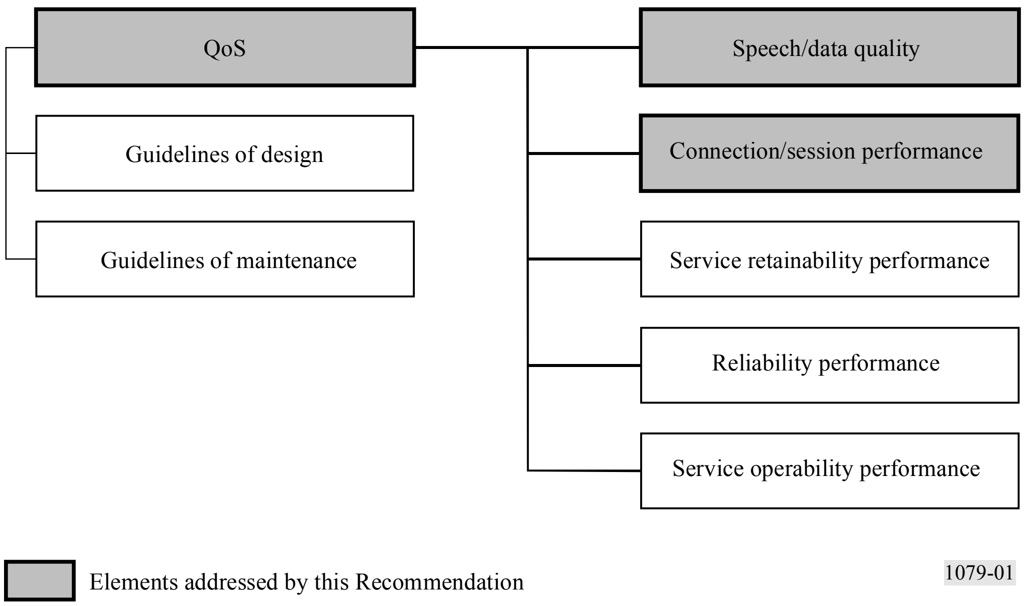
Reliability performance is defined in ITU-T Recommendation E.800 as the probability that an item can perform a required function under stated conditions for a given time interval. Faults in the telephone network can be classified as two types. One type is where the user encounters a small scale fault in the network segment other than the user's own segment, in which case service can be re-established if the user calls again at once. The other type is where the fault occurs in the user's segment or a large-scale fault occurs in the network segment, in which case, no service can be provided even if the user calls many times. A measure of reliability performance of the user's segment is the failure rate, and a measure of the network segment is unavailability.

6.6 Guidelines of design

To realize telecommunication services which achieve the criteria specified in quality of service, guidelines for the design of the network are needed. The quality of systems which are designed in accordance with these guidelines will be expected to meet the recommendations made below (see Fig. 1).

FIGURE 1

An example of the functional structure of quality for telecommunication service



6.7 Guidelines of management

Guidelines to maintain and operate the facilities are needed. These guidelines are the basis on which a service provider or a network operator maintains the service, judges the quality in order to improve the service, and takes remedial action.

7 Considerations

In developing this Recommendation it was considered:

a) that ITU-R has been studying IMT-2000 and has issued Recommendations ITU-R M.687, ITU-R M.816, ITU-R M.817, ITU-R M.818, ITU-R M.819, ITU-R M.1034, ITU-R M.1035, ITU-R M.1036, ITU-R M.1078, ITU-R M.1224, ITU-R M.1311, and ITU-R M.1457 which relate to these systems;

b) that the ITU-R studies are continuing;

c) that IMT-2000 encompasses a number of different systems;

d) that users will expect the speech/data quality, information transmission quality, reliability of connection, and degree of blocking to be comparable to those for the same services provided by the fixed networks, recognizing the limitations imposed by the radio environment;

e) that service availability will be dependent on a number of factors which could include: mobile terminal type, speed of motion, and geographic factors; for example hand portable sized/vehicle mounted terminals, indoor/outdoor, residential or business areas, urban/suburban/rural areas, etc.;

f) the relevant ITU-T Recommendations and on-going studies;

g) that there is a need for mobile terminals to roam between public land mobile telecommunication networks in different countries and between networks in the same country;

h) that IMT-2000 will offer voice and data services which interconnect with the PSTN/ISDN/B-ISDN/Internet and other public fixed and mobile networks;

j) that voiceband data applications will be an important early part of IMT-2000 and of the application of IMT-2000 to developing countries;

k) that the choice of speech codec and the speech quality achieved in the mobile network will have a major impact on the penetration of the telephone market place. If the quality is poor and the delay in the speech path is too great, the adoption of IMT-2000 by the general public may not reach the expected level; data quality achieved in the mobile network will have major impact also for introduction of high-speed multimedia and Internet services;

l) that this issue has not been exposed fully in first and second generation systems because these are used to serve people to whom mobility is imperative. In a mass market, with many users in a static or semi-mobile environment, mobility may not be sufficient to justify poor quality and excessive delay, in competition with a fixed network offering high quality;

m) that with a competitive mass market a significant number of calls will be mobile to mobile, or make use of cascaded connections, and that in such circumstances quality must be adequately maintained;

n) users will expect the speech quality to be maintained in connections through the PSTN/Internet involving transcoding to 64 kbit/s PCM, DCME, ADPCM and analogue circuits;

o) that Internet and IP-based services such as Web browsing are growing at rapid speed.

8 Recommendations

The ITU Radiocommunication Assembly recommends the following requirements to determine QoS performance for the various services:

8.1 Overview of different levels of QoS

Network services are considered end-to-end, this means from a TE to another TE. An end-to-end service may have a certain QoS which is provided for the user of a network service. It is the user that decides whether he is satisfied with the provided QoS or not.

To realize a certain network QoS a bearer service with clearly defined characteristics and functionality is to be set up from the source to the destination of a service.

A bearer service includes all aspects to enable the provision of a contracted QoS. These aspects are among others the control signalling, user plane transport and QoS management functionality. An IMT-2000 bearer service layered architecture is depicted in Fig. 2, each bearer service on a specific layer offers its individual services using services provided by the layers below. In this Figure additional functional groupings that support the external bearer service between the "CN gateway" and the "TE" on the right side have been omitted for simplicity.

8.1.1 The end-to-end service and IMT-2000 bearer service

On its way from the TE to another TE the traffic has to pass different bearer services of the network(s). A TE is connected to the IMT-2000 network by use of an MT. The end-to-end service on the application level uses the bearer services of the underlying network(s). As the end-to-end service is conveyed over several networks (not only IMT-2000) it is not subject to further elaboration in this Recommendation.

The end-to-end service used by the TE will be realized using a TE/MT local bearer service, an IMT-2000 bearer service, and an external bearer service.

TE/MT local bearer service is not further elaborated here as this bearer service is outside the scope of the IMT-2000 network.

Having said that the end-to-end bearer service is beyond the scope of this Recommendation it is however the various services offered by the IMT-2000 bearer service that the IMT-2000 operator offers. It is this bearer service that provides the IMT-2000 QoS.

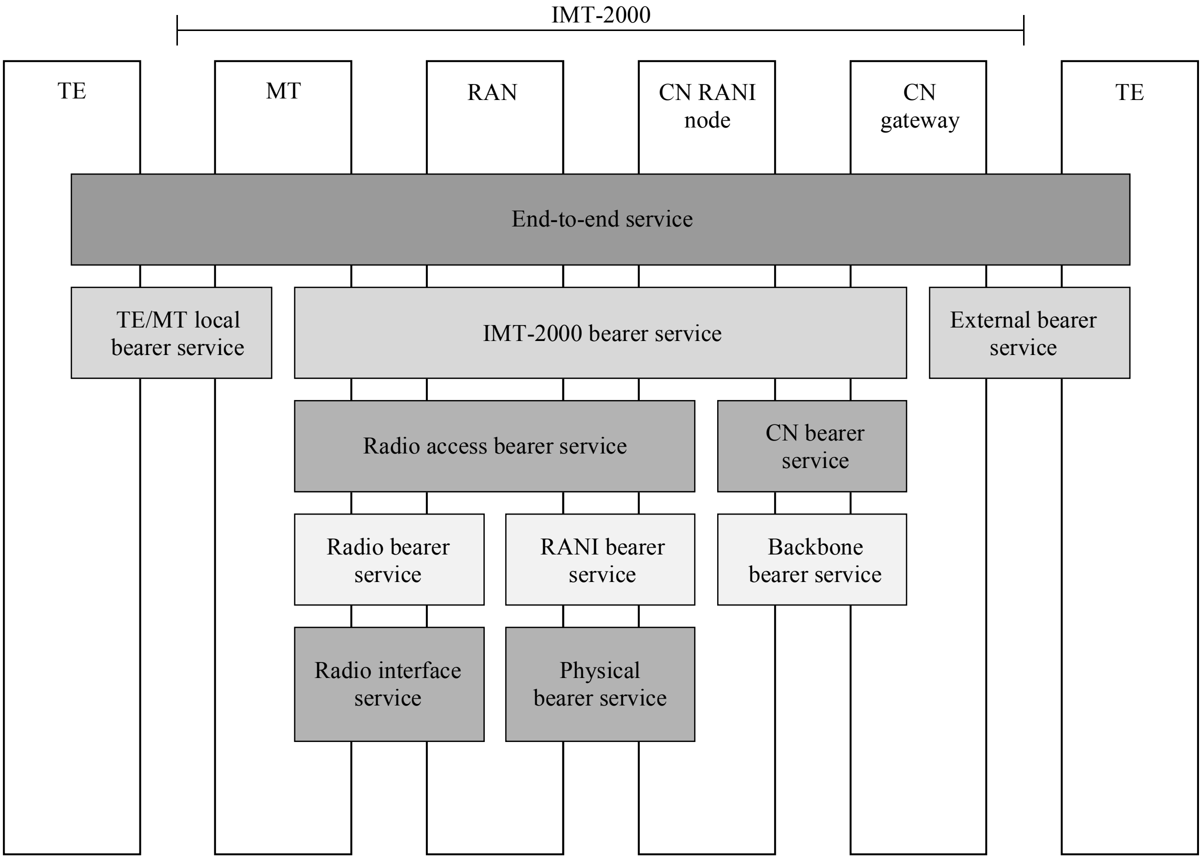
The external bearer service is not further elaborated here as this bearer may be using several network services, e.g. another IMT-2000 bearer service.

8.1.2 The radio access bearer service and the CN bearer service

As described above it is the IMT-2000 bearer service that provides the IMT-2000 QoS. The IMT-2000 bearer service consists of two parts, the radio access bearer service and the CN bearer service. Both services reflect the optimized way to realize the IMT-2000 bearer service over the respective IMT-2000 network topology taking into account such aspects as, e.g. mobility and mobile user profiles.

FIGURE 2

IMT-2000 bearer service layered architecture for the assessment of QoS*



* The functional blocks shown in this Figure are not intended to imply that interfaces between blocks need to be defined by ITU. They are just useful functional groupings used in the development of QoS ideas for IMT-2000.

1079-02

The radio access bearer service provides confidential transport of signalling and user data between MT and CN RANI node with the QoS adequate to the negotiated IMT-2000 bearer service or with the default QoS for signalling. This service is based on the characteristics of the radio interface and is maintained for a moving MT.

The CN bearer service of the IMT-2000 CN connects the IMT-2000 CN RANI node with the CN gateway to the external network. The role of this service is to efficiently control and utilize the backbone network in order to provide the contracted IMT-2000 bearer service.

8.1.3 The radio bearer service and the RANI bearer service

The radio access bearer service is realized by the radio bearer service and a RANI-bearer service.

The role of the radio bearer service is to cover all the aspects of the radio interface transport. This bearer service uses the radio interface(s), which is not elaborated further in this Recommendation.

The RANI bearer service together with the physical bearer service provides the transport between RAN and CN.

8.1.4 The backbone network service

The CN bearer service uses a generic backbone network service.

The backbone network service covers the layer 1/layer 2 functionality and is selected according to the operator's choice in order to fulfil the QoS requirements of the CN bearer service. The backbone network service is not specific to IMT-2000 but may reuse an existing standard.

8.2 IMT-2000 QoS classes

A typical user is not concerned with how a particular service is provided. However, the user is interested in comparing one service with another in terms of universal, user-oriented performance parameters which apply to any end-to-end service. From a user's perspective, performance needs to be expressed by parameters which:

– focus on user-perceivable effects, rather than their causes within the network;

– are independent of the networks internal design;

– take into account all aspects of the service from the user's point of view which can be objectively measured at the service access point;

– can be assured to a user by the service provider(s).

With these considerations in mind, Table 1 illustrates four QoS classes for IMT-2000 from the user perspective:

– conversational class of service

– interactive class of service

– streaming class of service

– background class of service.

TABLE 1

IMT-2000 QoS classes from a user perspective

QoS class of service	Conversational class of service Real-time conversation	Interactive class of service Interactive best effort	Streaming class of service Real-time streaming	Background class of service Background best effort
Fundamental characteristics from the user perspective	– Preserve time relation (variation) between information entities of the stream – Conversational pattern (stringent and low delay)	– Request response pattern – Preserve payload content	– Preserve time relation (variation) between information entities of the stream	– Destination is not expecting the data within a certain time – Preserve payload content
Example of the application	– Voice	– Web browsing	– Streaming video	– Background download of e-mails

The main distinguishing factor between these classes is how delay-sensitive the application is: conversational class refers to applications which are very delay-sensitive while background class is the most delay-insensitive QoS class.

It should be noted that for any particular application more than one QoS class of service may be required.

8.2.1 Conversational class

The most well-known use of this scheme is telephony speech. But with Internet and multimedia a number of new applications will require this scheme, for example VoIP and videoconferencing tools. Real-time conversation is always performed between peers (or groups) of live (human) end-users. This is the only scheme where the required characteristics are strictly given by human perception. Therefore this scheme has the strongest and most stringent QoS requirements.

The real-time conversation scheme is characterized by the transfer time that must be low because of:

– the conversational nature of the scheme;

– at the same time the time relation (variation) between information entities of the stream must be preserved in the same way as for real-time streams.

The maximum transfer delay is given by the human perception of video and audio conversation. Therefore the limit for acceptable transfer delay is very strict, as failure to provide low enough transfer delay will result in unacceptable lack of quality. The transfer delay requirement is therefore both significantly lower and more stringent than the round trip delay of interactive applications.

Real-time conversation – fundamental characteristics for QoS:

– preserve time relation (variation) between information entities of the stream;

– conversational pattern (stringent and low delay).

8.2.2 Interactive class

When the end-user, that is either a machine or a human, is online requesting data from remote equipment (e.g. a server) this scheme applies. Examples of human interaction with the remote equipment are: Web browsing, database retrieval, server access. Examples of machine interaction with remote equipment are: polling for measurement records and automatic database enquiries (tele-machines).

Interactive traffic is the other classical data communication scheme that on an overall level is characterized by the request response pattern of the end-user. At the message destination there is an entity expecting the message (response) within a certain time. Round-trip delay time is therefore one of the key attributes. Another characteristic is that the content of the packets must be transparently transferred (with low BER).

Interactive traffic – fundamental characteristics for QoS:

– request response pattern;

– preserve payload content.

8.2.3 Streaming class

When the user is looking at (listening to) real-time video (audio) the scheme of real-time streams applies. The real-time data flow is always aiming at a live (human) destination. It is a one-way transport.

This scheme is one of the newcomers in data communication, raising a number of new requirements in both telecommunication and data communication systems. It is characterized by the time relations (variation) between information entities (i.e. samples, packets) within a flow which must be preserved, although it does not have any requirements on low transfer delay.

The delay variation of the end-to-end flow must be limited, to preserve the time relation (variation) between information entities of the stream. But as the stream normally is time aligned at the receiving end (in the user equipment), the highest acceptable delay variation over the transmission media is given by the capability of the time alignment function of the application. Acceptable delay variation is thus much greater than the delay variation given by the limits of human perception.

Real-time streams – fundamental characteristics for QoS:

– unidirectional continuous stream;

– preserve time relation (variation) between information entities of the stream.

8.2.4 Background class

When the end-user, that typically is a computer, sends and receives data-files in the background, this scheme applies. Examples are background delivery of e-mails, SMS, download of databases and reception of measurement records.

Background traffic is one of the classical data communication schemes where an overall level is characterized by the absence of any parameter at the destination expecting to receive the data within a certain time limit. The scheme is thus more or less delivery time insensitive. Another characteristic is that the content of the packets must be transparently transferred (with low BER).

Background traffic – fundamental characteristics for QoS:

– the destination is not expecting the data within a certain time;

– preserve payload content.

A background application is one that does not carry delay information. 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.

8.3 Range of QoS requirements

It should be possible for one application to specify its QoS requirements to the network by requesting a bearer service with any of the specified traffic type, traffic characteristics, maximum transfer delay, delay variation and BER.

Table 2 indicates the range of values that should be supported by IMT-2000. These requirements are valid for both connection and connectionless traffic. It should be possible for the network to satisfy these requirements without wasting resources on the radio and network interfaces due to granularity limitations in QoS. The values in Table 2 show the range of values for the maximum transfer delay and BER at the bearer transport level, while the detailed end-user requirements for various applications are given in § 8.3.1 and 8.4.

TABLE 2

BER and delay requirements for IMT-2000 operating environments

<table>
<tr><th rowspan="2">Operating environment</th><th>Real time (constant delay)</th><th>Non-real time (variable delay)</th></tr>
<tr><th>BER/maximum transfer delay</th><th>BER/maximum transfer delay</th></tr>
<tr><td>Satellite (terminal relative speed to ground up to 1 000 km/h for plane)</td><td>Maximum transfer delay, less than 400 ms

BER: 1×10^{-3} to 1×10^{-7} (1)</td><td>Maximum transfer delay, 1 200 ms or more(2)

BER: 1×10^{-5} to 1×10^{-8}</td></tr>
<tr><td>Rural outdoor (terminal relative speed to ground up to 500 km/h) (3)</td><td rowspan="3">Maximum transfer delay, 20-300 ms

BER: 1×10^{-3} to 1×10^{-7} (1),(4)</td><td rowspan="3">Maximum transfer delay, 150 ms or more(2)
BER: 1×10^{-5} to 1×10^{-8}</td></tr>
<tr><td>Urban/suburban outdoor (terminal relative speed to ground up to 120 km/h)</td></tr>
<tr><td>Indoor/low range outdoor (terminal relative speed to ground up to 10 km/h)</td></tr>
</table>

(1) There is likely to be a compromise between BER and delay.

(2) The maximum transfer delay should be regarded here as the target value for 95% of the data.

(3) The value of 500 km/h as the maximum speed to be supported in the rural outdoor environment was selected in order to provide service on high-speed vehicles (e.g. trains). This is not meant to be the typical value for this environment (250 km/h is more typical).

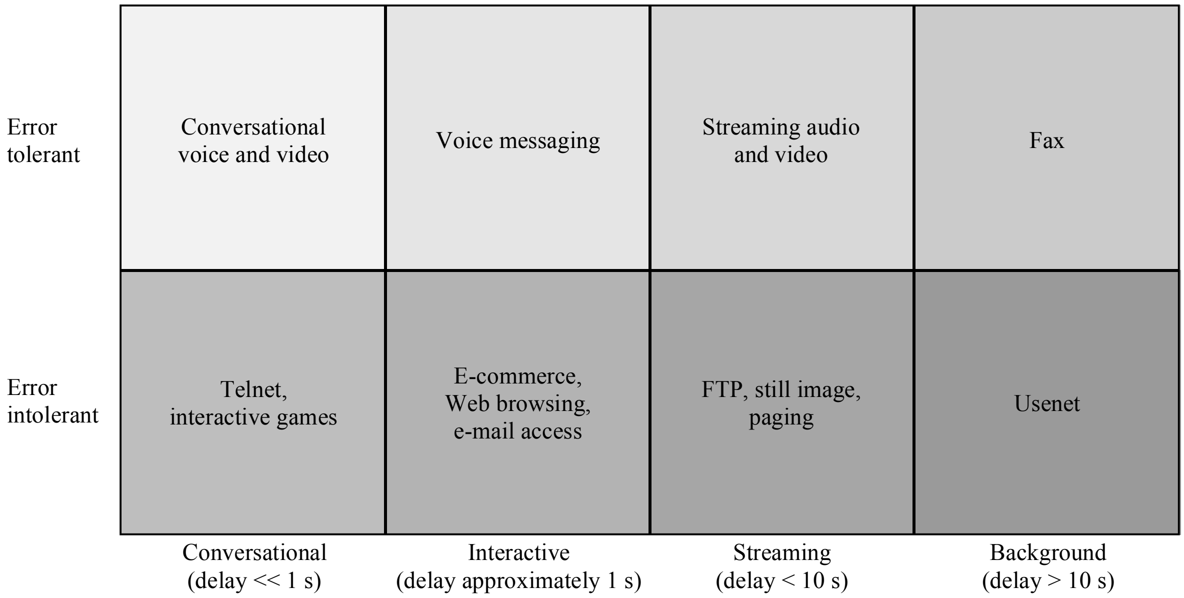
(4) See § 8.4.4 for further information about delay.

8.3.1 Supported end user QoS

This section outlines the QoS that should be provided to the end user/applications. Figure 3 summarizes the major groups of application in terms of QoS requirements. Applications and new applications may be applicable to one or more groups.

FIGURE 3

Groups of applications behaviour in terms of QoS requirements



1079-03

Tables 3 to 5 further elaborate IMT-2000 end user/application QoS requirements for the classes conversational (Table 3), interactive (Table 4) and streaming services (Table 5). These Tables specify end-to-end delay and are written from an applications service viewpoint and typically define both a preferred delay and a maximum delay for that service. The values in Tables 3 to 5 are aligned with those in ITU-T Recommendation G.1010. Since these QoS requirements are end-to-end from the user perspective, they need to be suitably apportioned for the IMT-2000 access network.

Tables 3 to 5 complement Table 2. Table 2 indicates the performance likely to be achieved by IMT-2000 systems in different operating environments.

8.4 Principal speech quality requirements

8.4.1 Subjective quality

The quality of the speech should be comparable to the fixed network, for users of different age, sex and language, according to the requirements described below (reference ITU-T Recommendation G.174).

8.4.2 Natural speech quality and speaker recognition

The speech shall sound like natural human speech. It is essential that the user should be able to recognize the voice of callers whose voice is familiar to the user.

8.4.3 Ease of conversation

Users should find the system easy to use for tasks which require the exchange of information in conversations over the connection, including the occurrence of double talk, where both parties talk at once.

TABLE 3

End-user performance expectations – Conversational/real-time services

Medium	Application	Degree of symmetry	Data rate (kbit/s)	Key performance parameters and target values		
				One-way delay (ms)	Delay variation (ms)	Information loss
Audio	Conversation at narrow-band speech	Two-way	4-13	< 150 preferred < 400 limit	< 1	< 3% PLR
Audio	Conversation at voice wideband speech	Two-way	4-13 10-64	< 150 preferred < 400 limit	< 1	< 3% PLR
Video	Videophone	Two-way	32-384	< 150 preferred < 400 limit Lip-synch: < 100		< 1% PLR
Data	Telemetry – two-way control	Two-way	< 28.8	< 250	Not applicable	Zero
Data	Interactive games	Two-way	< 1	< 250	Not applicable	Zero
Data	Telnet	Two-way (asymmetric)	< 1	< 250	Not applicable	Zero

TABLE 4

End-user performance expectations – Interactive services

Medium	Application	Degree of symmetry	Data rate (kbit/s)	Key performance parameters and target values		
				One-way delay	Delay variation (ms)	Information loss
Audio	Voice messaging	Primarily one-way	4-32	< 1 s for playback < 2 s for record	< 1	< 3% PLR
Data	Web-browsing – HTML	Primarily one-way		Preferred: < 2 s/page Acceptable: < 4 s/page	Not applicable	Zero
Data	Transaction services – high priority, e.g. ATM, e-commerce	Two-way		Preferred: < 2 s Acceptable: < 4 s	Not applicable	Zero
Data	E-mail (server access)	Primarily one-way		Preferred: < 2 s Acceptable: < 4 s	Not applicable	Zero

TABLE 5

End-user performance expectations – Streaming services

Medium	Application	Degree of symmetry	Data rate (kbit/s)	Key performance parameters and target values		
				One-way delay (s)	Delay variation (ms)	Information loss
Audio	High quality streaming audio	Primarily one-way	16-128	< 10	<< 1	< 1% PLR
Video	One-way	One-way	32-384	< 10		< 1% PLR
Data	Bulk data transfer/retrieval	Primarily one-way		Preferred: < 15 Acceptable: < 60	Not applicable	Zero
Data	Still image	One-way		Preferred: < 15 Acceptable: < 60	Not applicable	Zero
Data	Telemetry – monitoring	One-way	< 28.8	< 10	Not applicable	Zero

8.4.4 Loss of interactivity due to delay in the speech path

One-way delay is defined as the delay associated with processing, encoding, decoding, radio propagation between a mobile and the PSTN connection (public land mobile network, PLMN).

Conversations between users shall not suffer from a lack of proper interactivity due to excessive delay in the connection. Delay can interfere with user applications, such as the ease with which interactive conversations can be maintained. Therefore, it is critical to control the delay introduced by IMT-2000.

A mean one-way delay of less than 40 ms is a long-term objective for IMT-2000. However, it is recognized that in the short term attaining that value may be extremely difficult or impractical. Therefore, in calculating transmission delay budgets a one-way value of around 100 ms should be considered for the IMT-2000 access part.

8.4.5 Freedom from echo

For IMT-2000, the expected transmission delay will require the use of echo control in the system.

The issue of echo control in the IMT-2000 environment is complex. Reference should be made to ITU-T Recommendation G.174 for guidance on echo control requirements looking towards the PSTN, and to ITU-T Recommendation P.313 for requirements looking towards the terminal.

8.4.6 Uniformity in different environments

Where different radio interfaces are used for access in different environments (e.g. pico cell, large cells, etc.) the same speech quality requirements shall be used. The user should find a uniformity of speech quality throughout the system.

It is recognized that more complex codecs with greater power consumption may be needed to achieve the required IMT-2000 speech quality in large cells, where lower bit rates are needed to achieve spectral efficiency.

8.4.7 Effects of transcoding

End-to-end connections in IMT-2000 may typically start in one type of cell, pass through the fixed network and be terminated in another type of cell, possibly passing through a satellite component in either the IMT-2000 or the fixed network. If different speech codecs are selected in these different wireless access environments and in the fixed network, it will result in the concatenation of a variety of speech codecs, with consequent loss in speech quality as a result of the necessary transcoding.

The techniques that will minimize the need for and the impact of transcoding such as tandem-free operation or transcoder free operation should be used. The effects of transcoding should be fully considered in meeting the speech quality requirements given in this Recommendation.

8.4.8 Quality of end-to-end connections

The speech quality requirements shall be achieved in complete end-to-end connections, including impairments arising from the radio interfaces (with typical interference and propagation conditions), transcoding, delay and echoes in the connection, etc.

8.4.9 Terminal acoustics

Handset acoustics play an important role in determining overall audio quality in wireless systems. A prime consideration is to ensure that the send, receive and sidetone signal levels are compatible with conventional wireline telephony, and sufficient terminal coupling loss is provided to control echo. However, other considerations such as handset shape (positioning of the microphone relative to the user's mouth and sealing of the earcap against the user's ear) are also important, particularly under noisy operating conditions. More detailed information is given in ITU-T Recommendation P.313.

8.4.10 Call progress tones, announcements and music

No annoying effects should be imposed on call progress tones, network announcements or music on hold.

8.4.11 Handover

The user should be unaware of the effects of handover on speech quality or voiceband data performance.

8.4.12 Robustness

The ability to withstand random errors, burst errors and high BERs over the whole service area is important. The ranking of potential speech/channel codec combinations may be different under good and marginal conditions.

8.4.13 Background acoustic noise

IMT-2000 environments are expected to result in a higher level of background acoustic noise than with wireline, for example from road traffic, railway and bus station concourses, etc. The speech codec and associated transducers should therefore be robust to such background acoustic noise.

The speech codec shall also be robust to the presence of other talkers in the background.

8.4.14 Interconnection of IMT-2000 users in different networks

Any speech quality impairment that results from transcoding between two IMT-2000 users should be minimized.

8.4.15 Speech performance testing

The ability of IMT-2000 to meet the speech quality requirements given above should be judged with a realistic selection method which takes account of the impairments of the mobile radio channel.

Tests could include two-way speech conversations in which the speakers have realistic tasks that make demands on the use of the channel.

The range of connection scenarios should be represented, including mobile to fixed, mobile to mobile, inclusion of satellite links in the mobile interface, satellite links in the network, etc. System impairments such as handover and network echoes and delays should be included.

During testing, the speech connection should be stressed with an error pattern generated by an error model related to the radio interface.

General guidance on subjective testing methods applicable to IMT-2000 access systems is given in ITU-T Recommendation P.800. Subjective testing methods applicable to speech codecs are covered in ITU-T Recommendation P.830. Subjective testing methods applicable to network echo cancellers are covered in ITU-T Recommendation P.831. Objective testing methods applicable to speech codecs are covered in ITU-T Recommendation P.862.

Annex 1

Planning tool to assess end-to-end voice transmission quality

ITU-T Recommendation G.107 is the approach recommended by ITU-T for assessing end-to-end voice transmission quality, including wireless access. More details on how to apply the E-model for specific scenarios are available in ITU-T Recommendation G.108.

For example, in Fig. 4 there will be degradations including the following:

– Mobile network:
  – voice codec impairments
  – propagation errors
  – propagation and processing delay
  – handset echo

– IP network:
  – voice codec impairments
  – packet loss
  – propagation delay
  – packet jitter
– PSTN:
  – voice codec impairments (negligible for 64 kbit/s PCM)
  – propagation errors
  – propagation and processing delay
  – handset echo
– Gateways:
  – voice codec conversion impairments
  – propagation delay.

Thus, this planning tool provides relative comparisons of systems under various transmission conditions to help make engineering decisions related to performance/cost trade-offs.

FIGURE 4

End-to-end system