RECOMMENDATION ITU-R M.1079-1*

PERFORMANCE AND QUALITY OF SERVICE REQUIREMENTS FOR INTERNATIONAL MOBILE TELECOMMUNICATIONS-2000 (IMT-2000)

(1994-2000)

1 Introduction

IMT-2000 are third generation mobile systems which are scheduled to start service around the year 2000 subject to market considerations. They will provide access, by means of one or more radio links, to a wide range of telecommunication services supported by the fixed telecommunication networks (e.g. PSTN/ISDN/IP networks), and to other services which are specific to mobile users.

A range of mobile terminal types is encompassed, linking to terrestrial and/or satellite-based networks, and the terminals may be designed for mobile or fixed use.

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 pocket terminal for worldwide use;

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

– worldwide roaming capability.

IMT-2000 are defined by a set of interdependent ITU Recommendations of which this one is a member.

This Recommendation forms part of the process of specifying the radio interfaces of IMT-2000. IMT-2000 will operate in the worldwide bands identified in the RR.

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, including the satellite aspects.

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 network that the user will expect in a network of comparable performance to the fixed network.

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* This Recommendation should be brought to the attention of the Telecommunication Standardization Sector.

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.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
ITU-T Recommendation P.79:	Calculation of loudness ratings for telephone sets
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 F.116:	Service features and operational provisions in IMT-2000
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:	C source code and test vectors for implementation verification of the G.729 8 kbit/s CS-ACELP speech coder
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:	Quality of service and dependability vocabulary
ITU-T Recommendation Q.1701:	Framework of IMT-2000 networks
ITU-T Recommendation Q.1711:	Network functional model for IMT-2000

5 **Abbreviations** (see Note 1)

ADPCM	adaptive differential pulse code modulation
CN	core network

CRC	cyclic redundancy code
DCME	digital circuit multiplex equipment
FER	frame erasure rate
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
PoW	poor or worst
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

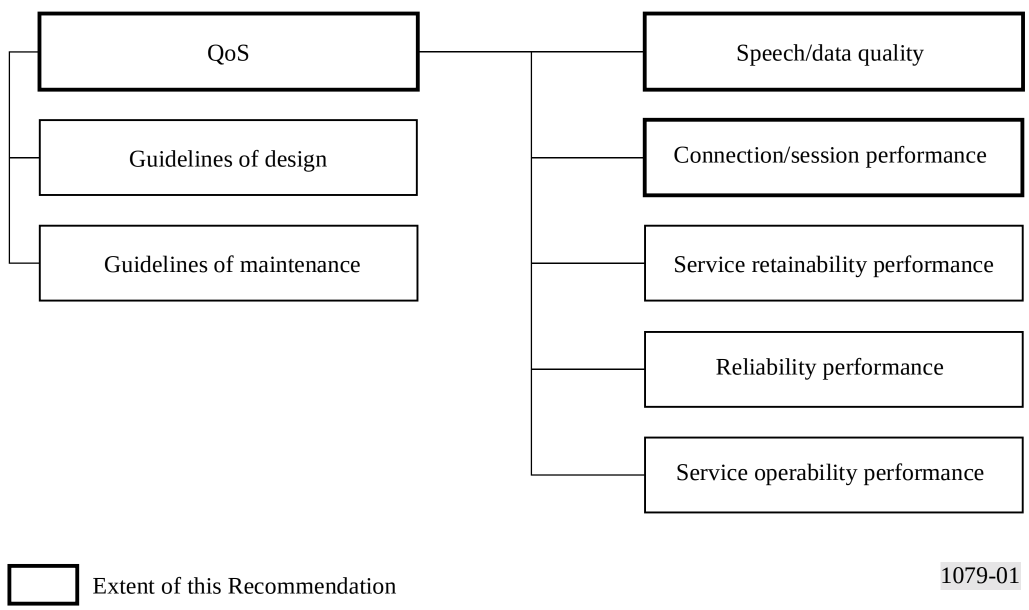
Reliability performance is defined in ITU-T Recommendation E.800 as the probability that an item can perform a required function under stated conditions for a given time interval. Faults in the telephone network can be classified as two types. One type is where the user encounters a small scale fault 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 and ITU-R M.1311 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

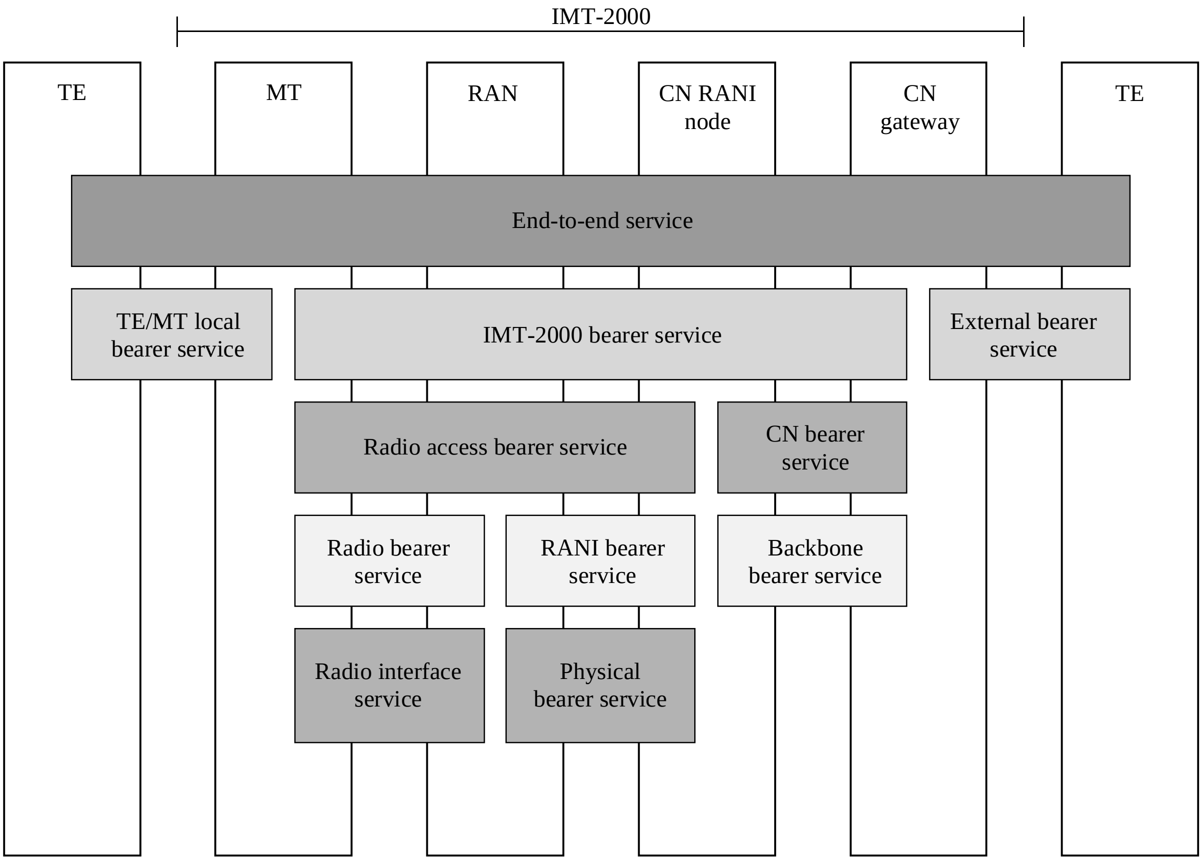
Network services are considered end-to-end, this means from a TE to another TE. An end-to-end service may have a certain QoS which is provided for the user of a network service. It is the user that decides whether he is satisfied with the provided QoS or not.

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.

FIGURE 2

IMT-2000 QoS functional architecture*



* 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.

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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 reflects 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.

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

When defining the IMT-2000 QoS classes the restrictions and limitations of the radio interface have to be taken into account. The QoS mechanisms provided in the IMT-2000 network have to be robust and capable of providing reasonable QoS resolution. Table 1 illustrates proposed QoS classes for IMT-2000.

In the proposal there are four different QoS classes (or traffic classes):

– conversational class

– streaming class

– interactive class

– background class.

The main distinguishing factor between these classes is how delay sensitive the traffic is: conversational class is meant for traffic which is very delay sensitive while background class is the most delay insensitive traffic class.

Conversational and streaming classes are mainly intended to be used to carry real-time traffic flows. The main divider between them is how delay sensitive the traffic is. Conversational real-time services, like videotelephony, are the most delay sensitive applications and those data streams should be carried in conversational class.

Interactive class and background are mainly meant to be used by traditional Internet applications like WWW, e-mail, Telnet, FTP and news. Due to looser delay requirements, compared to conversational and streaming classes, both provide better error rate by means of channel coding and retransmission. The main difference between interactive and background class is that interactive class is mainly used by interactive applications, e.g. interactive e-mail or interactive

Web browsing, while background class is meant for background traffic, e.g. background download of e-mails or background file downloading. Responsiveness of the interactive applications is ensured by separating interactive and background applications. Traffic in the interactive class has higher priority in scheduling than background class traffic, so background applications use transmission resources only when interactive applications do not need them. This is very important in a wireless environment where the bandwidth is low compared to fixed networks.

TABLE 1

IMT-2000 QoS classes

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

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.

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 the interactive traffic case.

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 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:

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

8.2.3 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 machines 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.

8.2.4 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.

8.3 IMT-2000 QoS manager functions

The following key functions are identified for the IMT-2000 QoS manager:

– *Mapping:* Mapping to/from the IMT-2000 service classes to external network service classes. This mapping will be based on agreements between the two networks.

– *Resource allocation and routing:* The mapping information will be used to allocate appropriate radio resources for providing the needed QoS using RRM. RRM will decide the RLP/RLC and physical layer mechanisms to be used, e.g. ARQ techniques at link layer, turbo codes at physical layer, etc. for each service class using the performance parameters.

– *Scheduling:* When a physical layer resource is shared by multiple information flows, there is a necessity of a link layer scheduling algorithm in the IMT-2000 QoS manager. This manager will schedule the RLP/RLC frames from multiple information flows to the same radio resource depending on service priority.

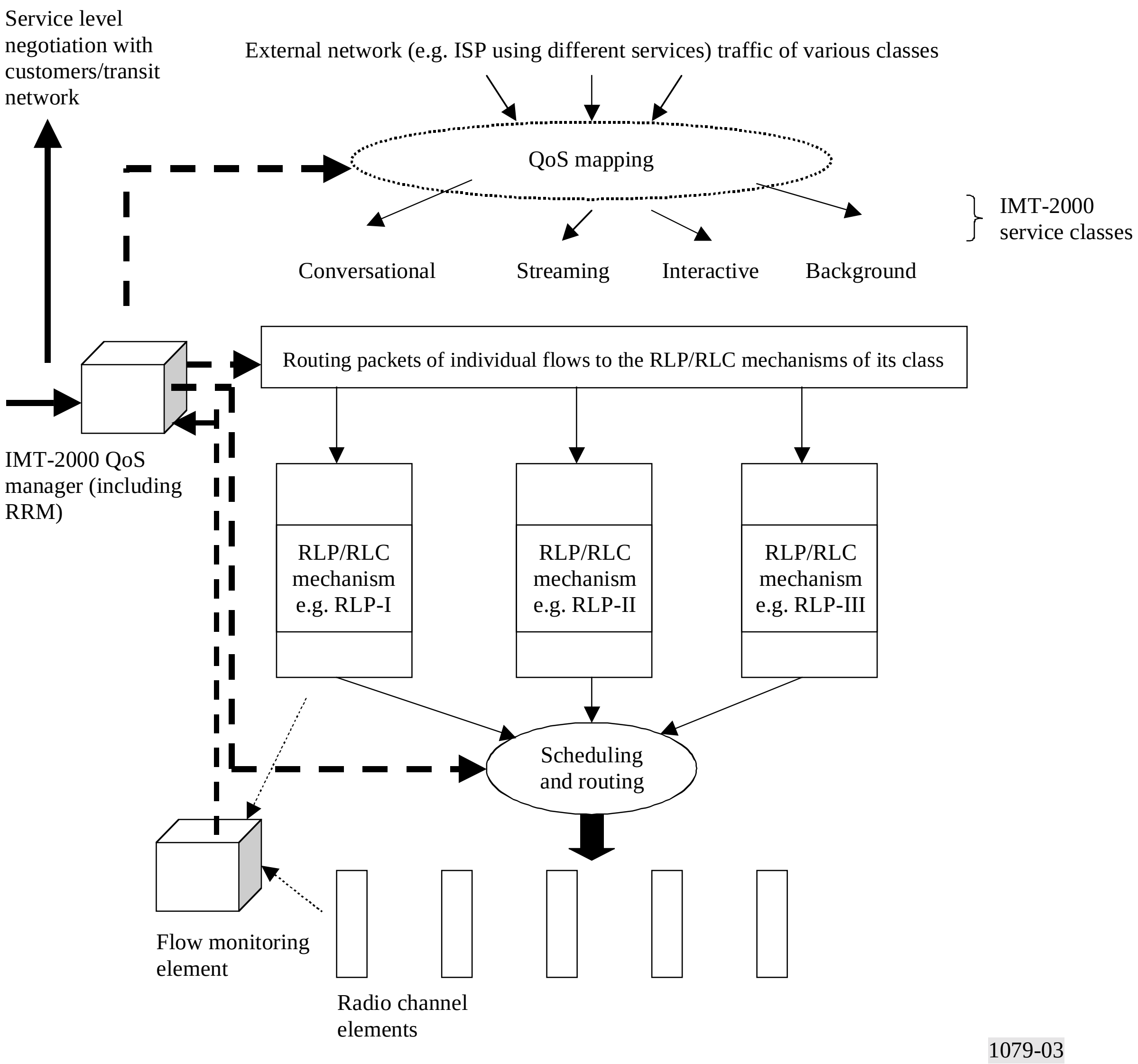
– *Service negotiation:* The IMT-2000 QoS manager will be able to do service negotiation with the end user and will also be capable of exchanging QoS parameters with the QoS manager in the external network. The service negotiation will determine the QoS mapping and traffic conditioning requirements (metering, marking, shaping and packet dropping) as the traffic transverses from one network to another.

– *Dynamic in-session service adaptation (optional):* The QoS manager monitors the service performance of a flow via the flow monitoring element. If the service performance is not meeting the required level, the QoS manager may initiate service adaptation in order to meet the required level of performance. This function is optional depending on the implementation complexity and network operator needs.

Figure 3 illustrates the IMT-2000 framework that is used for deriving the functions of the IMT-2000 QoS manager.

FIGURE 3

IMT-2000 QoS framework



8.4 QoS parameters

8.4.1 Asymmetric bearers

The parameters related to throughput/bit rate should be separated for uplink/downlink in order to support asymmetric bearers.

8.4.2 Sources of IMT-2000 bearer service parameters

IMT-2000 bearer service parameters describe the service provided by the IMT-2000 network to the user of the IMT-2000 bearer service. A set of QoS parameters (QoS profile) specifies this service. At IMT-2000 bearer service establishment or modification different QoS profiles have to be taken into account:

– The TE and MT capabilities form a QoS profile which may limit the IMT-2000 bearer service which can be provided.

– The TE and MT in the IMT-2000 network or the TE within the terminating network may request a QoS profile at IMT-2000 bearer establishment or modification. The application using the user equipment may request the user equipment to provide a IMT-2000 bearer service with a specific QoS profile. If the application requests no specific QoS the MT/TE may use a QoS profile configured within the MT/TE.

– A QoS profile in the IMT-2000 subscription describes the upper limits for the provided service if the service user requests specific values. Otherwise, this QoS profile may describe a default QoS service profile requested by the user.

– Default QoS profile(s) may be configured by the operator for the IMT-2000 bearer services provided by the network.

– A network specific QoS profile characterizing for example the current resource availability or other network capabilities or limitations may limit the provided IMT-2000 bearer service or initiate a modification of an established IMT-2000 bearer service.

8.4.3 IMT-2000 bearer service attributes

The attributes listed in this section also apply to the radio access bearer service (see Fig. 2).

8.4.3.1 List of attributes

Traffic class (conversational, streaming, interactive, background)

Definition: Type of application for which the IMT-2000 bearer service is optimized.

Purpose: By including the traffic class itself as an attribute, IMT-2000 networks can make assumptions about the traffic source and optimize the transport for that traffic type.

Maximum bit rate (kbit/s)

Definition: Maximum number of bits delivered by IMT-2000 at a SAP within a period of time, divided by the duration of the period. The traffic is conformant with maximum bit rate as long as it follows a token bucket algorithm where token rate = maximum bit rate and bucket size = maximum SDU size. The conformance definition should not be interpreted as a required implementation algorithm.

Purpose: Maximum bit rate can be used to make reservations in the downlink of the radio interface. Its purpose is to limit the delivered bit rate to applications or external networks with such limitations.

Guaranteed bit rate (kbit/s)

Definition: Guaranteed number of bits delivered by IMT-2000 at a SAP within a period of time (provided that there is data to deliver), divided by the duration of the period. The traffic is conformant with the guaranteed bit rate as long as it follows a token bucket algorithm where token rate = guaranteed bit rate and bucket size = $k \times$ maximum SDU size. The conformance definition should not be interpreted as a required implementation algorithm.

Purpose: Guaranteed bit rate may be used to facilitate admission control based on available resources, and for resource allocation within IMT-2000. Quality requirements expressed by, e.g. delay and reliability attributes only apply to incoming traffic up to the guaranteed bit rate.

Delivery order (yes/no)

Definition: Indicates whether the IMT-2000 bearer should provide in-sequence SDU delivery or not.

Purpose: The attribute is derived from the user protocol (PDP type) and specifies if out-of-sequence SDUs are acceptable or not. This information cannot be extracted from the traffic class. Whether out-of-sequence SDUs are dropped or re-ordered depends on the specified reliability.

Maximum SDU size (bits)

Definition: The maximum allowed SDU size.

Purpose: The maximum SDU size is used for admission control and policing.

SDU format information (bits)

Definition: List of possible exact sizes of SDUs.

Purpose: IMT-2000 RAN needs SDU size information to be able to operate in transparent RLC protocol mode, which is beneficial to spectral efficiency and delay when RLC retransmission is not used. Thus, if the application can specify SDU sizes, the bearer is less expensive.

SDU error ratio

Definition: Indicates the fraction of SDUs lost or detected as erroneous. SDU error ratio is defined only for conforming traffic.

Note that by reserving resources, SDU error ratio performance is independent of the loading conditions, whereas without reserved resources, such as in interactive and background classes, SDU error ratio is used as target value.

Purpose: Used to configure the retransmission protocol on layer 2 and the error detection coding on layer 1.

Residual BER

Definition: Indicates the undetected BER in the delivered SDUs. If no error detection is requested, residual BER indicates the BER in the delivered SDUs.

Purpose: Used to configure channel coding and error detection coding on layer 1.

Delivery of erroneous SDUs (yes/no/-)

Definition: Indicates whether SDUs detected as erroneous should be delivered or discarded.

NOTE 1 – “yes” implies that error detection is employed and that erroneous SDUs are delivered together with an error indication, “no” implies that error detection is employed and that erroneous SDUs are discarded, and “–” implies that SDUs are delivered without considering error detection.

Purpose: Used to decide whether frames with failed CRC on layer 1 should be forwarded or not.

Transfer delay (s)

Definition: Time between request to transfer an SDU at one SAP to its delivery at the other SAP. Transfer delay is specified for one or more fixed SDU sizes. Exact statistical transfer delay definition and fixed SDU sizes are to be determined.

Purpose: Used to specify the delay tolerated by the application. It allows IMT-2000 RAN to set transport formats and ARQ parameters.

NOTE 2 – Transfer delay of an arbitrary SDU is not meaningful for a bursty source, since the last SDUs of a burst may have long delay due to queuing, whereas the meaningful response delay perceived by the user is the delay of the first SDU of the burst.

Traffic handling priority

Definition: Specifies the relative importance for handling of all SDUs belonging to the IMT-2000 bearer compared to the SDUs of other bearers.

Purpose: Within the interactive class, there is a definite need to differentiate between bearer qualities. This is handled by using the traffic handling priority attribute, to allow IMT-2000 to schedule traffic accordingly. By definition, priority is an alternative to absolute guarantees, and thus these two attribute types cannot be used together for a single bearer.

Allocation/retention priority

Definition: Specifies the relative importance compared to other IMT-2000 bearers for allocation and retention of the IMT-2000 bearer.

Purpose: Priority is used for differentiating between bearers when performing allocation and retention of a bearer, and the value is typically related to the subscription.

8.4.3.2 Attributes discussed per class

Conversational class

Although the bit rate of a conversational source codec may vary, conversational traffic is assumed to be relatively non-bursty. Maximum bit rate specifies the upper limit of the bit rate with which the IMT-2000 bearer delivers SDUs at the SAPs. The IMT-2000 bearer is not required to transfer traffic exceeding the guaranteed bit rate. Maximum and guaranteed bit rate attributes are used for resource allocation within IMT-2000. Minimum resource requirement is determined by guaranteed bit rate. (When a conversational source generates less traffic than allocated for the bearer, the unused resources can of course be used by other bearers.)

Since the traffic is non-bursty, it is meaningful to guarantee a transfer delay of an arbitrary SDU.

Conversational bearers are likely to be realized in IMT-2000 RAN without RLC retransmissions. Hence, IMT-2000 RAN transport is more efficient and thereby cheaper if RLC PDU size is adapted to IMT-2000 bearer SDU size (RLC transparent mode). This motivates the use of SDU format information. The SDU periodicity knowledge needed to operate in RLC transparent mode is obtained through dividing the largest defined SDU format by maximum bit rate. This must be considered when setting the attribute values in a service request.

The maximum SDU size is only applicable if SDU format information is not specified and is used for admission control and policing. If maximum SDU size is specified the SDU size is variable. If SDU format information is specified, with one or several possible sizes, each SDU must exactly conform to one of the specified sizes. By using the SDU error ratio, residual BER and delivery of erroneous SDUs attribute, the application requirement on error rate can be specified, as well as whether the application wants IMT-2000 to detect and discard SDUs containing errors and an adequate FEC means can be selected.

Streaming class

As for conversational class, streaming traffic is assumed to be rather non-bursty. Maximum bit rate specifies the upper limit of the bit rate the IMT-2000 bearer delivers SDUs at the SAPs. The IMT-2000 bearer is not required to transfer traffic exceeding the guaranteed bit rate. Maximum and guaranteed bit rate attributes are used for resource allocation within IMT-2000. Minimum resource requirement is determined by guaranteed bit rate. (When a streaming source generates less traffic than allocated for the bearer, the unused resources can of course be used by other bearers.)

Since the traffic is non-bursty, it is meaningful to guarantee a transfer delay of an arbitrary SDU.

The transfer delay requirements for streaming are typically in a range where at least in a part of this range RLC retransmission may be used. It is assumed that the application's requirement on delay variation is expressed through the transfer delay attribute, which implies that there is no need for an explicit delay variation attribute.

It should be possible for streaming bearers to be realized in IMT-2000 RAN without RLC retransmissions. Hence, IMT-2000 RAN transport is more efficient and thereby cheaper if RLC PDU size is adapted to IMT-2000 bearer SDU size (RLC transparent mode). This motivates the use of SDU format information. The SDU periodicity knowledge needed to operate in RLC transparent mode is obtained through dividing the largest defined SDU format by maximum bit rate. This must be considered when setting the attribute values in a service request.

The maximum SDU size is only applicable if SDU format information is not specified and is used for admission control and policing. If maximum SDU size is specified the SDU size is variable. If SDU format information is specified, with one or several possible sizes, each SDU must exactly conform to one of the specified sizes.

By using the SDU error ratio, residual BER and delivery of erroneous SDUs attribute, the application requirement on error rate can be specified, as well as whether the application wants IMT-2000 to detect and discard SDUs containing errors.

Interactive class

This bearer class is optimized for transport of human or machine interaction with remote equipment, such as Web browsing. The source characteristics are unknown but may be bursty.

To be able to limit the delivered data rate for applications and external networks by traffic conditioning, maximum bit rate is included.

There is a definite need to differentiate between quality for bearers within the interactive class. One alternative would be to set absolute guarantees on delay, bit rate, etc. which however at present seems complex to implement within IMT-2000 RAN/CN. Instead, traffic handling priority is used. SDUs of a IMT-2000 bearer with higher traffic handling priority is given priority over SDUs of other bearers within the interactive class, through IMT-2000-internal scheduling.

It is principally impossible to combine this relative approach with attributes specifying delay, bit rate, packet loss, etc. so an interactive bearer gives no quality guarantees, and the actual bearer quality will depend on the load of the system and the admission control policy of the network operator.

The only additional attribute that is reasonable to specify is the bit integrity of the delivered data, which is given by SDU error ratio, residual BER and delivery of erroneous SDUs. Because there are no reserved resources for interactive class, SDU error ratio should be used as a target value. SDU error ratio cannot be guaranteed under abnormal load conditions.

Background class

The background class is optimized for machine-to-machine communication that is not delay sensitive, such as messaging services. Background applications tolerate a higher delay than applications using the interactive class, which is the main difference between the background and interactive classes.

IMT-2000 only transfers background class SDUs when there is definite spare capacity in the network. To be able to limit the delivered data rate for applications and external networks by traffic conditioning, maximum bit rate is included.

No other guarantee than bit integrity in the delivered data, given by SDU error ratio, residual BER and delivery of erroneous SDUs, is needed. Because there are no reserved resources for background class, SDU error ratio should be used as a target value. SDU error ratio cannot be guaranteed under abnormal load conditions.

8.4.3.3 IMT-2000 bearer attributes: summary

In Table 2, the defined IMT-2000 bearer attributes and their relevancy for each bearer class are summarized. Observe that traffic class is an attribute itself.

TABLE 2

IMT-2000 bearer attributes defined for each bearer class

Traffic class	Conversational class	Streaming class	Interactive class	Background class
Maximum bit rate	X	X	X	X
Delivery order	X	X	X	X
Maximum SDU size	X	X	X	X
SDU format information	X	X		
SDU error ratio	X	X	X	X
Residual BER	X	X	X	X
Delivery of erroneous SDUs	X	X	X	X
Transfer delay	X	X		
Guaranteed bit rate	X	X		
Traffic handling priority			X	
Allocation/retention priority	X	X	X	X

NOTE 1 – Source statistics descriptor can be defined and added to the Table above to determine the specific radio access bearer characteristics of SDUs to support in the IMT-2000 RAN.

8.5 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 3 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.

TABLE 3

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 § 8.6.4 for further information about delay.

8.5.1 Supported end user QoS

This section outlines the QoS that should be provided to the end user/applications. Figure 4 summarizes the major groups of application in terms of QoS requirements. Applications and new applications may be applicable to one or more groups.

Tables 4 to 6 further elaborate IMT-2000 end user/application QoS requirements for the classes conversational (Table 4), interactive (Table 5) and streaming services (Table 6). 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.

FIGURE 4

Groups of applications behaviour in terms of QoS requirements

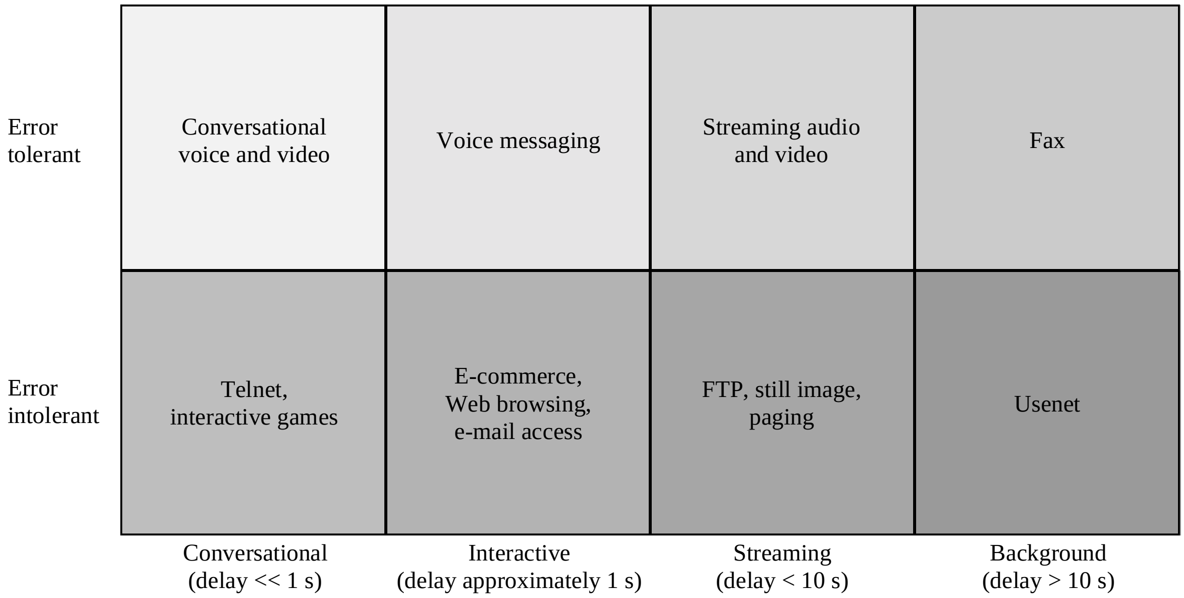


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TABLE 4

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% FER
Audio	Conversation at voice wideband speech	Two-way	4-13 10-64	< 150 preferred < 400 limit	< 1	< 3% FER
Video	Videophone	Two-way	32-384	< 150 preferred < 400 limit Lip-synch: < 100		< 1% FER
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 5

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	4-13	< 1 s for playback < 2 s for record	< 1	< 3% FER
Data	Web-browsing – HTML	Primarily one-way		< 4 s/page	Not applicable	Zero
Data	Transaction services – high priority, e.g. ATM, e-commerce	Two-way		< 4 s	Not applicable	Zero
Data	E-mail (server access)	Primarily one-way		< 4 s	Not applicable	Zero

TABLE 6

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	32-128	< 10	< 1	< 1% FER
Video	One-way	One-way	32-384	< 10		< 1% FER
Data	Bulk data transfer/retrieval	Primarily one-way		< 10	Not applicable	Zero
Data	Still image	One-way		< 10	Not applicable	Zero
Data	Telemetry – monitoring	One-way	< 28.8	< 10	Not applicable	Zero

Tables 4 to 6 complement Table 3. Table 3 indicates the performance likely to be achieved by IMT-2000 systems in different operating environments.

NOTE 1 – The term, transfer delay, is defined in § 8.4.3.1.

8.6 Principal speech quality requirements

8.6.1 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).

8.6.2 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.

8.6.3 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.

8.6.4 Loss of interactivity due to delay in the speech path

The recommended mean one-way delay of less than 40 ms is an important 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 value of around 100 ms should be considered for the IMT-2000 access part.

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.

In a digital PLMN with sufficient echo control, ITU-T Recommendation G.173 recommends a mean one-way delay objective of 20 ms and that the one-way delay should not exceed 40 ms. It is recognized that in the satellite component and in the PLMN, the one-way delay may exceed 40 ms, due to the propagation and processing delay (see ITU-T Recommendation G.114).

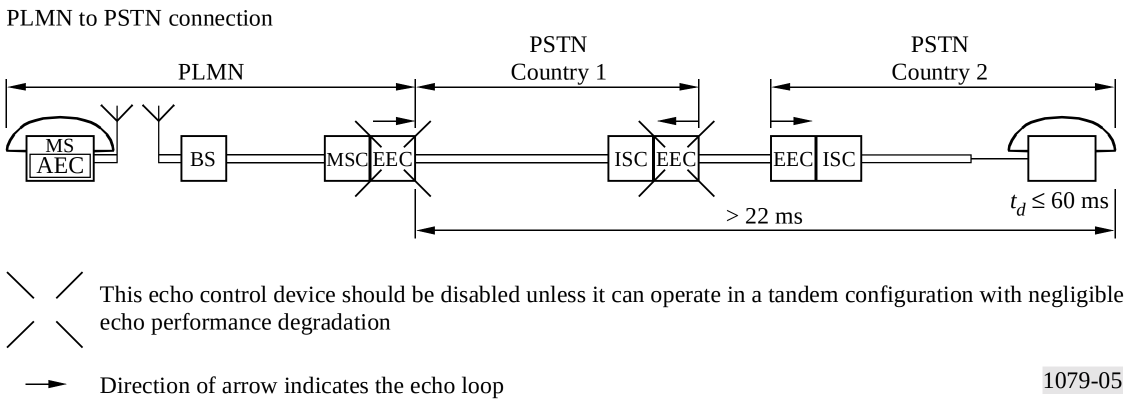
Even though a greater delay may occur in a satellite connection, delay shall be minimized in the wireless access to the network for the majority of calls, which use terrestrial connections.

Further study is needed on how to apportion the allowed delay between the speech codec and the radio physical layer.

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

FIGURE 5

Part of Fig. A.4/G.173 of ITU-T Recommendation G.173



The results of subjective tests using Fig. 5 configuration have been reported in ITU-T Recommendation G.114, based on the MOS degradation over a range of one-way delay transmission times from 0 to 1 500 ms. The results are plotted in terms of per cent PoW:

FIGURE 6

Comparison of PoW for overall quality and interruptability
(taken from ITU-T Recommendation G.114)

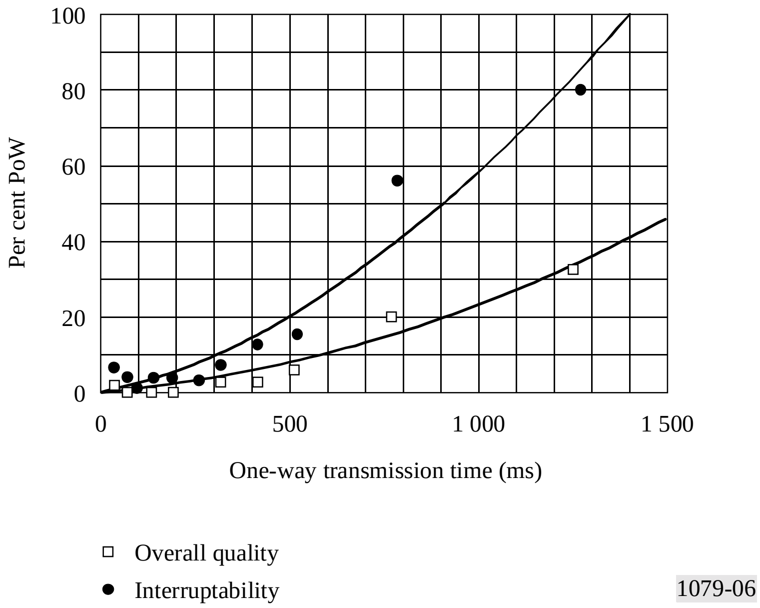


Figure 6 results clearly indicate that there is no significant difference in the overall quality or interruptability when the one-way delay transmission time is maintained below 300 ms. Thus, even considering a mobile-to-mobile call scenario, a one-way delay transmission time of 100 ms for a terrestrial wireless access system seems acceptable.

8.6.5 Freedom from echo

The issue of echo control in the IMT-2000 environment is complex. Experience from other systems should be treated with caution. Delays which may be considered tolerable in stand-alone systems may not be acceptable for IMT-2000. Reference should be made to ITU-T Recommendation G.174.

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

8.6.6 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.

8.6.7 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.

8.6.8 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.

8.6.9 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. These signal levels are usually specified in terms of loudness ratings (see ITU-T Recommendation P.79) and suitable values are given in ITU-T Recommendation G.174. 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.

8.6.10 Call progress tones, announcements and music

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

8.6.11 Voice recognition

Due regard should be given to the need to retain those aspects of speech that are used in speech recognition systems that already work well with wireline originated speech and in expected future speech recognition systems.

8.6.12 Handover

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

8.6.13 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.

8.6.14 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.

8.6.15 Cost and power consumption

Speech and channel coding proposals should be assessed for their expected cost, power consumption and complexity.

8.6.16 Interconnection of IMT-2000 users in different networks

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

8.6.17 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. At the present time the radio interface technology has not yet been selected and consequently an interim error model must be used.

8.7 Voiceband data requirements

The transmission of DTMF information should be supported by IMT-2000 with a performance comparable to the fixed network (see ITU-T Recommendation G.174).

DTMF tones can originate from either the handset keypad or from an acoustically coupled separate device.

It would be desirable to transmit DTMF tones by passing them transparently through the speech codec in order to minimize the cost of the handset and the network infrastructure. However, there is a danger that adequate error performance will not be realized due to impairments on the radio interface. There may also be an unnecessary burden on the speech codec. Consequently DTMF tones that enter the handset by acoustic coupling will be recognized as such, and carried in IMT-2000 as data signals, unless adequate error performance can be ensured with transparent transmission.

Voiceband data signals supported by IMT-2000 should be transmitted with a quality comparable with the fixed network (see ITU-T Recommendation G.174). An important example of voiceband data is Group 3 facsimile.

The performance reference for clean background conditions is 32 kbit/s (see ITU-T Recommendation G.726).

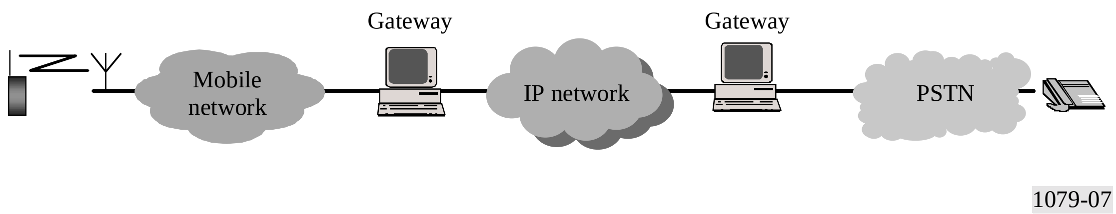
ANNEX 1

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

Modern communications frequently traverse multiple networks and it can be difficult to assess what the final quality perceived by the user will be. The E-model of ITU-T Recommendation G.107 can be employed to estimate the combination of degradations from each of the subnetworks.

FIGURE 7

End-to-end system



For example, in Fig. 7 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

The relative performance of different configurations based on the E-model as described in ITU-T Recommendation G.107 can be plotted on a bar chart. The E-model provides a statistical estimation of quality measures.

Using the ITU-T Recommendation G.107 E-model, output results (for example expressed as GoB) can be obtained for a variety of possible scenarios and can be plotted for comparison - some may be very good (90% GoB), others may be very bad (20% GoB). Thus, this planning tool provides relative comparisons of systems under various transmission conditions to help make engineering decisions related to performance/cost trade-offs.