Technical Specification Group, Radio Access Network **TSGR#4(99)372**
Meeting #4, Miami, 17-19 June 1999 (Tdoc TSGR-AH1-99059)

Source: ITU Ad Hoc Contact Person

Title: Proposed liaison to TSG SA on the ITU-R TG 8/1 revision of Recommendation M.1079

Document for: Approval

Agenda Item: 5.6

The attached document is the revised version of the ITU-R TG8/1 Recommendation M.1079 entitled “Performance and Quality of Service Requirements for International Mobile Telecommunications-2000 (IMT-2000)”. This version was developed at the recent TG8/1 meeting held in Bejing, May 31 – June 11, 1999. Much of the new material in the attached document has been taken from the 3GPP documents. The schedule within ITU-R TG8/1 is to finalise the revision on the Recommendation M.1079 at the next TG8/1 meeting to be held in Helsinki, October 25 – November 5, 1999.

TSG RAN asks TSG SA to develop detailed contribution(s) to comment and propose changes to the present revision of Recommendation M.1079. These needs to be submitted well in time for the next RAN#5 meeting, October 6-8, 1999, so that these in turn can then be submitted to TG8/1 according to the 3GPP working procedures (coordinated by the TSG RAN AH ITU).

Out from the present version of Recommendation M.1079, the TSG RAN understanding is that the document should be relevant for TSG SA WG1 (Services), WG2 (Architecture), and WG4 (Codec), but ask the TSG SA Miami meeting for further handling of the document within TSG SA.

Attachment: Revised version of ITU-R Recommendation M.1079, ITU-R TG8/1 Document 8-1/TEMP/198Rev1, “Performance and Quality of Service Requirements for International Mobile Telecommunications-2000 (IMT-2000)”

Task Group 8/1

DOCUMENT CATEGORY	:	Draft of updated Recommendation
SOURCE	:	WG 4
APPROVAL STATUS	:	Adopted for further study
DATE	:	June 1999
TO	:	TG 8/1 Plenary
ACTION	:	To be adopted by TG 8/1 for further study and to be included in the Beijing TG 8/1 report
SCHEDULE	:	11 June 1999

RECOMMENDATION ITU-R M.1079*-NEW

~~SPEECH AND VOICEBAND DATA~~ PERFORMANCE and Quality of Service (QoS) REQUIREMENTS FOR INTERNATIONAL MOBILE TELECOMMUNICATIONS-2000 (IMT-2000)

(Question ITU-R 39/8)

1. Introduction

International Mobile Telecommunications-2000 (IMT-2000) are third generation mobile systems (TGMS) which are scheduled to start service around the year 2000 subject to market consideration. 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,

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

– high quality,

– ~~use of a small pocket terminal worldwide~~small terminals for worldwide use

- capability for multimedia applications, wide range of services and terminals

– worldwide roaming capablility~~.~~

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 by World Administrative Radio Conference for Dealing with Frequency Allocations in Certain Parts of the Spectrum ~~(Malaga-Torremolinos, 1992) (WARC-92) (1 885-2 025 and 2 110-2 200 MHz, with the satellite component limited to 1 980-2 010 and 2 170-2 200 MHz).~~

. The subject matter of IMT-2000 is complex and its representation in the form of Recommendations is evolving. To maintain the pace of progress on the subject it is necessary to produce a sequence of Recommendations on a variety of aspects. The Recommendations strive to avoid apparent conflicts between themselves. Future Recommendations, or revisions, will be used to resolve any discrepancies.

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

In addition there are annexes concerning objectives for planning and assessement of end-to-end network performance, ~~a~~ speech codec(s) and error models of the radio interface to be used in testing codecs.

~~Further transmission performance objectives are contained in ITU-T draft Recommendation G.174.~~

2. Scope

This Recommendation defines the speech/data quality and performance requirements for the International Mobile Telecommunications-2000 (IMT-2000), including the satellite aspects.

This Recommendation lists the basic Recommendations essential for a) 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 b) 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.

~~In addition to speech this Recommendation is also concerned with voiceband data.~~

~~The Recommendation contains Annex 1, which sets out the parameters required in speech codecs for the IMT-2000 applications, and Annexes 2 and 3 concerned with defining an error model to be used in testing and selecting proposed speech codecs for use in the IMT-2000 network.~~ This Recommedation contains Annexes related to end-to-end network performance and error models.

3. Structure of the Recommendation

The document contains Recommendations dealing with speech/data quality, connection/session performance, ~~voiceband~~ data, the requirements for other services and the radio performance requirements. ~~There are also important annexes dealing with objectives for speech codecs for IMT-2000 and error models for testing and selecting speech codecs.~~ In particular, QoS (Quality of Service) 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:

– ITU-T Recommendation G.114:	Mean one-way transmission time
– ITU-T Recommendation G.131:	Stability and echo
– ITU-T Recommendation G.173:	Transmission planning aspects of the speech service in digital public land mobile networks
– ITU-T Recommendation G.726:	40, 32, 24, 16 kbit/s adaptive differential pulse code modulation (ADPCM)
- ITU-T Recommendation G.728	
- ITU-T Recommendation G.729	
– Recommendation ITU-R M.1034:	Requirements for the radio interface(s) for International Mobile Telecommunications-2000 (IMT-2000)
- Recommendation ITU-R M.1225	
– ITU-T Recommendation E.800:	Quality of service and dependability vocabulary
– 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
– 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.1311	Framework for modularity and radio commonality within IMT-2000
– ITU-T ~~draft~~ Recommendation G.174:	Transmission performance of terrestrial wireless personal communication systems
– ITU-T Recommendation P.79:	Calculation of loudness ratings
- ITU-T Recommendation G.107	E-model, a computational model for use in transmission planning, 1998
- ITU-T Recommendation G.109	Definition of categories of speech transmission quality
- ITU-T Recommendation P.313	Transmission Characteristics for cordless and mobile digital terminals
- ITU-T Recommendation F.116	Service features and operational provisions in IMT-2000
- ITU-T Recommendation Q.1701	
- ITU-T Recommendation Q.1711	

ADD appropriate X.series references for data quality

5. Abbreviations

RAN- Radio Access Network

RANI- Radio Access Network Interface

CN- Core Network

MT- Mobile terminal

RT- Real Time

NRT – Non-real Time

QoS- Quality of Service

BER- Bit Error Rate

FER- Frame Erasure Rate

~~5.~~6. ~~5.~~ Definitions

6.1 Quality of Service : the collective effect of service performances which determine the degree of satisfaction of a user of a service. It is characterised 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; and
- other factors specific to each service.

~~5.1~~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.

~~5.2~~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.

~~5.3~~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.

~~5.4~~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.

~~5.5~~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

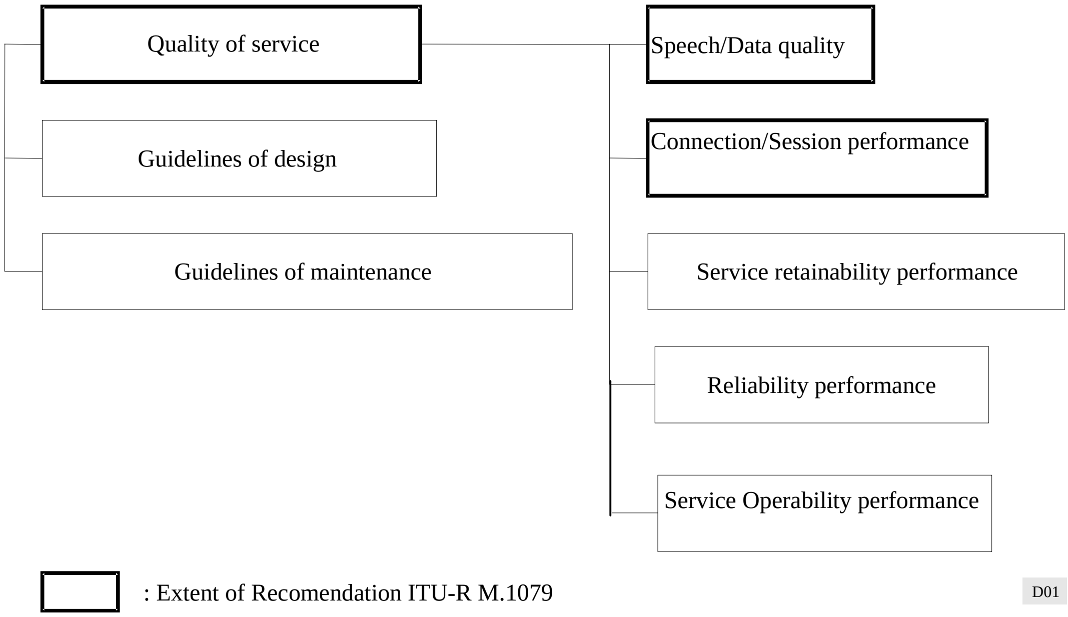
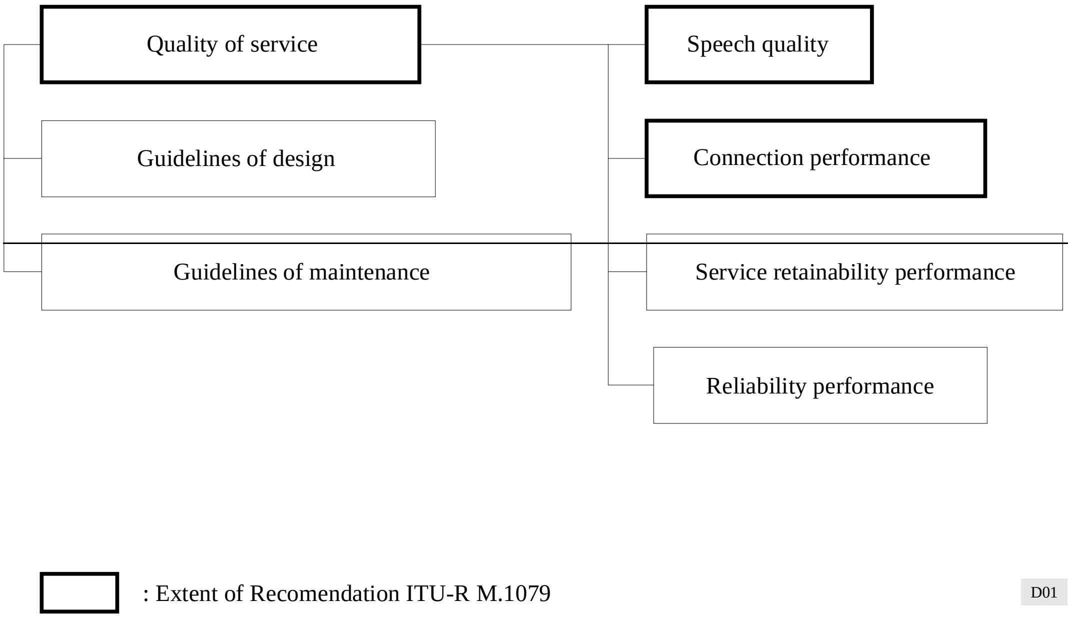


FIGURE 1

An example of the functional structure of quality for telecommunication service



~~FIGURE 1/M~~

~~5.6~~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.

~~5.7~~6.8 Gross speech bit rate

The gross speech bit rate is defined as the bit rate required for the speech codec to meet the speech quality requirements, including the redundant bits for the error control of the coded speech bits and the internal synchronization bits if they are required, but excluding the synchronization word for the radio transmission and the associated control channel for call control and housekeeping of the radio channel.

6.9 Session (Packet drop rate?)

6.10 Data quality

~~6.~~7. Considerations

In developing this Recommendation the following factors were considered:

a) that ITU-R has been studying International Mobile Telecommunications-2000 (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, M.1311 and ITU-R M.1078 which relate to these systems;

b) that the ITU-R studies are continuing ~~and are being conducted by Task Group 8/1 under Decision ITU-R 116;~~

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 ~~non-voice~~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~~j) that internet services such as web browsing are growing at rapid speed~~;~~

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 studies of speech quality and the impact of audio path delay are continuing in the ITU-T and other bodies.~~

~~7.~~8. Recommendations

The ITU Radiocommunication Assembly *recommends* that the following ~~requirements be met~~approach to determine QoS performance requirements for various services.

8.1 Overview of Different Levels of QoS

Network Services are considered end-to-end, this means from a Terminal Equipment (TE) to another TE. A End-to-End Service may have a certain Quality of Service (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 realise 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. A IMT-2000 bearer service layered architecture is depicted in Figure 2, each bearer service on a specific layer offers it’s individual services using services provided by the layers below.

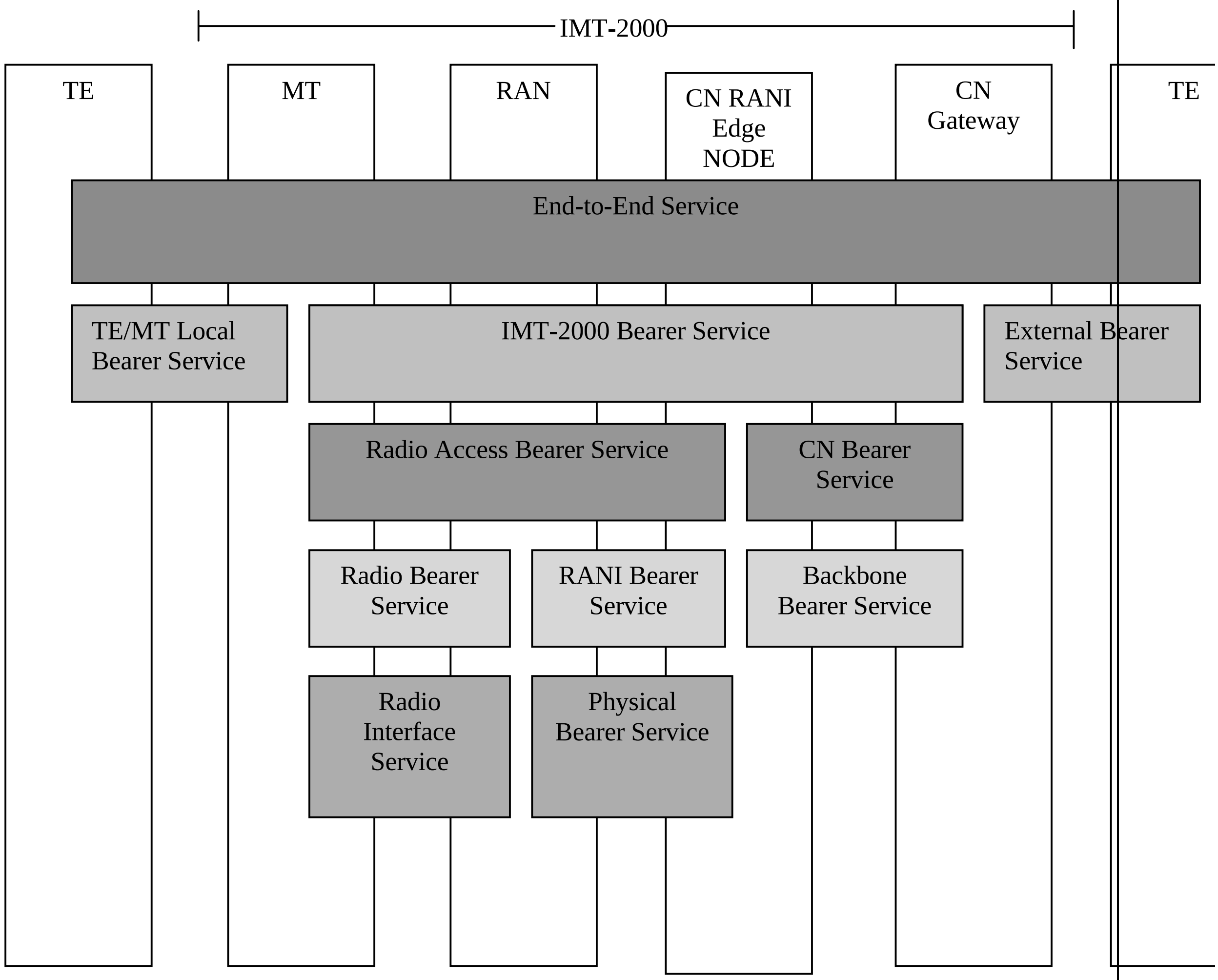


Figure 1: IMT-2000 QoS Functional Architecture

Note: The functional blocks shown in Figure 2 above are not intended to imply that interfaces between blocks need to be defined by the ITU. They are just useful functional groupings used in the development of QoS ideas for IMT-2000.

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 a Mobile Termination (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 for further elaboration in this document.

The *End-to-End-Service* used by the TE will be realised using a *TE/MT Local Bearer Service*, a *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 document 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 Core Network Bearer Service

As described in the previous chapter 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 Core Network Bearer Service. Both services reflects the optimised way to realise the IMT-2000 Bearer Service over the respective cellular network topology taking into account such aspects as e.g. mobility and mobile subscriber profiles.

The Radio Access Bearer Service provides confidential transport of signalling and user data between MT and CN Iu Edge 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 Core Network Bearer Service of the IMT-2000 core network connects the IMT-2000 CN RANI Edge Node with the CN Gateway to the external network. The role of this service is to efficiently control and utilise 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 realised by Radio Bearer Service and an 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 document.

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 Core Network Bearer Service uses a generic Backbone Network Service.

The Backbone Network Service covers the Layer 1/Layer2 functionality and is selected according to operator’s choice in order to fulfil the QoS requirements of the Core Network 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 air interface have to be taken into account. It is not reasonable to define complex mechanisms as have been in fixed networks due to different error characteristics of the air interface. The QoS mechanisms provided in the cellular 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 and
- 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 video telephony, 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, Email, Telnet, FTP and News. Due to looser delay requirements , compare 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 Email or interactive Web browsing, while Background class is meant for background traffic, e.g. background download of Emails 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 wireless environment where the bandwidth is low compared to fixed networks.

8.2.1 Conversational class

The most well known use of this scheme is telephony speech (e.g. GSM). But with Internet and multimedia a number of new applications will require this scheme, for example voice over IP and video conferencing 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.

Real time conversation scheme is characterised by that the transfer time must be low because of the conversational nature of the scheme and at the same time that 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 characterised by that the time relations (variation) between information entities (i.e. samples, packets) within a flow 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 on line requesting data from remote equipment (e.g. a server), this scheme applies. Examples of human interaction with the remote equipment are: web browsing, data base retrieval, server access. Examples of machines interaction with remote equipment are: polling for measurement records and automatic data base enquiries (tele-machines).

Interactive traffic is the other classical data communication scheme that on an overall level is characterised 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 bit error rate).

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 that on an overall level is characterised by that the destination is not expecting the data within a certain time. 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 bit error rate).

Background traffic - fundamental characteristics for QoS:

- the destination is not expecting the data within a certain time
- preserve payload content

Table 1: IMT-2000 QoS classes

Traffic class	**Conversational class** conversational RT	**Streaming class** streaming RT	**Interactive class** Interactive best effort NRT	**Background** Background best effort NRT
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 emails

8.3 QoS Parameters

8.3.1 General

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

8.3.2 IMT-2000 Bearer Service Attributes

8.3.2.1 List of attributes

[Note: The text within square brackets explaining the purpose of each attribute can be excluded later if that information is given elsewhere in the technical report.]

Potential purpose and definition of a possible additional attribute 'average bitrate' is FFS.

Traffic class ['conversational', 'streaming', 'interactive', 'background']

Definition: type of application for which the IMT-2000 bearer service is optimised

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

Maximum bitrate [kbps]

Definition: maximum number of bits delivered by IMT-2000 at a SAP within a measurement period, divided by the duration of the measurement period. The definition of the period is FFS.

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

Guaranteed bitrate [kbps]

Definition: guaranteed number of bits delivered by IMT-2000 at a SAP within a measurement period (provided that there is data to deliver), divided by the duration of the measurement period. The definition of the measurement period is FFS.

[Purpose: Guaranteed bitrate 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 bitrate.]

Delivery order [y/n]

Definition: indicates whether the *IMT-2000* bearer shall 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]

SDU size information [bits]

Definition: list of possible exact sizes of SDUs

[Purpose: 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 re-transmission is not used. Thus, if the application can specify SDU sizes, the bearer is less expensive.]

Reliability

Definition: information on the reliability in terms of error ratio etc that the *IMT-2000* bearer provides. The error ratio is the relation between erroneous SDUs/bits and SDUs/bits requested to be transferred. Whether one or both of SDU and bit error ratio apply to a certain *IMT-2000* bearer is FFS. Additionally, reliability specifies whether the *IMT-2000* bearer shall deliver SDUs with detected errors or discard these SDUs. Whether this is implicitly given by the two above mentioned error ratios is FFS.

[Purpose: used to specify the reliability needed by the application. Especially RAN may set internal parameters based on this value; the reliability impact on the CN is assumed to be negligible.]

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 FFS.

[Purpose: used to specify the delay tolerated by the application. It allows RAN to set transport formats and ARQ parameters.]

[Note: 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.

8.3.2.2 Attributes discussed per class

Conversational class

Although the bitrate of a conversational source codec may vary, conversational traffic is assumed to be relatively non-bursty. **Maximum bitrate** specifies the upper limit of the bitrate with which the IMT-2000 bearer delivers SDUs at the SAPs. The IMT-2000 bearer is not required to transfer traffic exceeding the **Guaranteed bitrate**. Maximum and guaranteed bitrate attributes are used for resource allocation within IMT-2000. Minimum resource requirement is determined by guaranteed bitrate (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 realised in RAN without RLC re-transmissions. Hence, 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 size information**.

By using the **reliability** attribute(s), 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 forward error correction means can be selected.

Streaming class

As for conversational class, streaming traffic is assumed to be rather non-bursty. **Maximum bitrate** specifies the upper limit of the bitrate the IMT-2000 bearer delivers SDUs at the SAPs. The IMT-2000 bearer is not required to transfer traffic exceeding the Guaranteed bitrate. Maximum and guaranteed bitrate attributes are used for resource allocation within IMT-2000. Minimum resource requirement is determined by guaranteed bitrate. (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 re-transmission 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.

By using the **reliability** attribute(s), 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 optimised 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 bitrate** 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, bitrate etc, which however at present seems complex to implement within 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, bitrate, 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 a part of the **reliability**.

Background class

The background class is optimised 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 bitrate** is included.

No other guarantee than bit integrity in the delivered data, according to the **reliability**, is needed.

8.3.2.3 IMT-2000 bearer attributes: summary

In Table 2, the defined IMT-2000 bearer attributes and their relevancy for each bearer class are summarised. 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 bitrate	X	X	X	X
Delivery order	X	X	X	X
SDU size information	X			
Reliability	X	X	X	X
Transfer delay	X	X		
Guaranteed bit rate	X	X		
Traffic handling priority			X	
Allocation/Retention priority	X	X	X	X

8.3.3 Radio Access Bearer Service Attributes

8.3.3.1 List of attributes

[Note: The text within square brackets explaining the purpose of each attribute can be excluded later if that information is given elsewhere in the technical report.]

Traffic class [‘conversational’, ’streaming’, ’interactive’, ’background’]
Definition: type of application for which the Radio Access Bearer service is optimized

[Purpose: By including the traffic class itself as an attribute, RAN can make assumptions about the traffic source and optimize the transport for that traffic type. In particular, buffer allocation may be based on traffic class.]

Maximum bitrate [kbps]
Definition: maximum number of bits delivered by RAN at a SAP within a measurement period, divided by the duration of the measurement period. The definition of the period is FFS.

[Purpose: to limit the delivered bitrate to applications or external networks with such limitations]

Guaranteed bitrate [kbps]
Definition: guaranteed number of bits delivered at a SAP within a measurement period (provided that there is data to deliver), divided by the duration of the measurement period. The definition of the measurement period is FFS.

[Purpose: Guaranteed bitrate may be used to facilitate admission control based on available resources, and for resource allocation within RAN. Quality requirements expressed by e.g. delay and reliability attributes only apply to incoming traffic up to the guaranteed bitrate. The guaranteed bitrate at the RAB level may be different from that on IMT-2000 bearer level, for example due to header compression.]

Delivery order [y/n]
Definition: indicates whether the IMT-2000 bearer shall provide in-sequence SDU delivery or not.

[Purpose: 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]

SDU size information [bits]
Definition: list of possible exact sizes of SDUs.

[Purpose: 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 re-transmission is not used. Thus, if the application can specify SDU sizes, the bearer is less expensive.

Reliability

Definition: information on the reliability in terms of error ratio etc that the Radio Access Bearer provides. The error ratio is the relation between erroneous SDUs/bits and SDUs/bits requested to be transferred. Whether one or both of SDU and bit error ratio apply to a certain Radio Access bearer is FFS. Additionally, reliability specifies whether the Radio Access bearer shall deliver SDUs with detected errors or discard these SDUs. Whether this is implicitly given by the two above mentioned error ratios is FFS.

[Purpose: RAN may set internal parameters based on specified reliability.]

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 FFS.

[Purpose: specifies the RAN part of the total transfer delay for the IMT-2000 bearer. It allows RAN to set transport formats and ARQ parameters.]

Traffic handling priority
Definition: specifies the relative importance for handling of all SDUs belonging to the radio access 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 RAN 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 Radio access bearers for allocation and retention of the Radio access 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.

8.3.3.2 Attributes discussed per class

[Note: the use of new attributes introduced on Radio Access Bearer service level will be described here.]

Conversational class

Streaming class

Interactive class

Background class

8.3.3.3 Radio Access Bearer attributes: summary

In Table 3, the defined Radio Access Bearer attributes and their relevancy for each bearer class are summarized. Observe that traffic class is an attribute itself.

Table 3. Radio Access Bearer attributes defined for each bearer class.

Traffic class	Conversational class	Streaming class	Interactive class	Background class
Maximum bitrate	X	X	X	X
Delivery order	X	X	X	X
SDU size information	X			
Reliability	X	X	X	X
Transfer delay	X	X		
Guaranteed bit rate	X	X		
Traffic handling priority			X	
Allocation/ Retention priority	X	X	X	X

~~7.1 Principal speech quality requirements~~

~~7.1.1 Subjective quality~~

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

~~7.1.2 Natural speech quality and speaker recognition~~

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

~~7.1.3 Ease of conversation~~

~~Subscribers shall 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.~~

~~7.1.4 Loss of interactivity due to delay in the speech path~~

~~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 Public Land Mobile Network 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 the one way delay may exceed 40 ms, due to the propagation 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.~~

~~7.1.5 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 draft Recommendation G.174.~~

~~By keeping the access delay sufficiently small the need for echo control can be avoided and significant cost saving achieved.~~

~~7.1.6 Uniformity in different environments~~

~~Where different air interfaces are used for access in different environments (e.g. pico cell, large cells, etc.) the same speech quality requirements shall be used. The subscriber shall 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.~~

~~7.1.7 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.~~

~~Consideration should be given to techniques which will minimize the need for and the impact of transcoding.~~

~~The effects of transcoding should be fully considered in meeting the speech quality requirements given in this document.~~

~~7.1.8 Quality of end-to-end connections~~

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

~~7.1.9 Handset 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 draft 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.~~

~~7.1.10 Call progress tones, announcements and music~~

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

~~7.1.11 Voice recognition~~

~~Due regard shall 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.~~

~~7.1.12 Handover~~

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

~~7.1.13 Gross speech bit rate~~

~~The gross speech bit rate (rather than the codec bit rate) required in the radio interface to support both the digital speech and the necessary error control coding, shall be considered in selecting the speech codec (see § 5.7).~~

~~Alternatively, the figure of merit for selecting a speech codec could be the resulting capacity of the system.~~

~~7.1.14 Robustness~~

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

~~7.1.15 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.~~

~~7.1.16 Cost and power consumption~~

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

~~7.1.17 Interconnection of IMT-2000 users in different networks~~

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

~~7.1.18 Speech codec objectives~~

~~Annex 1 contains a table of parameters for speech codecs for the IMT-2000 application. Required values of the parameters are given together with values to be used as objectives.~~

~~7.1.19 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 should 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 shall 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 shall be included.~~

~~During testing, the speech connection shall be stressed with an error pattern generated by an error model related to the air interface. At the present time the air interface technology has not yet been selected and consequently an interim error model must be used.~~

~~The interim error model is the Bellcore model described in Annex 3, which is representative of the burst errors found by slow-moving or stationary users of mobile systems. Task Group 8/1 expects to generate further error models appropriate to the air interface technologies developed for IMT-2000 and to the range of environments and vehicle speeds to be expected in the system. An explanation of generation of error models for IMT-2000 interfaces is provided in Annex 2.~~

~~7.2 Principal voiceband data requirements~~

~~7.2.1 DTMF signalling requirements~~

~~The transmission of DTMF information shall be supported by IMT-2000 with a performance comparable to the fixed network (see ITU-T draft 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.~~

~~7.2.2 Voiceband data requirements~~

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

~~7.3 Radio performance requirements~~

~~7.3.1 Speech quality requirements~~

~~The speech quality in a connection in IMT-2000 involving two radio interfaces, under the error conditions defined by the current IMT-2000 error model, together with any necessary transcoding shall not be degraded more than 0.5 MOS compared with error free G.726 at 32 kbit/s.~~

~~7.3.2 Connection performance requirements~~

~~International Mobile Telecommunications-2000 will need to meet the performance requirements listed in Table 1, for which values have yet to be specified (also see ITU-T draft Recommendation G.174).~~

~~TABLE 1~~

~~Connection performance requirements~~

~~Probability of blocking on the radio link~~	~~For further study~~
~~Probability of blocking between IMT-2000 and the fixed network~~	~~For further study~~
~~Post selection delay~~	~~For further study~~
~~Answer signal delay~~	~~For further study~~
~~Call release delay~~	~~For further study~~
~~Probability of unexpected disconnect~~	~~For further study~~
~~Probability of unsuccessful handover~~	~~For further study~~

8.4 Range of QoS requirements

It shall 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 & bit error ratios.

The following table indicates the range of values that shall be supported by IMT-2000.. These requirements are valid for both connection and connectionless traffic. It shall 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 4 BER and Delay requirements for IMT-2000 operating environments

	Real Time (Constant Delay)	Non Real Time (Variable Delay)
Operating environment	**BER/Max Transfer Delay**	**BER/Max Transfer Delay**
Satellite (Terminal relative speed to ground up to 1000 km/h for plane)	Max Transfer Delay less than 400 ms BER 10-3 - 10-7 (Note 1)	Max Transfer Delay 1200 ms or more (Note 2) BER = 10-5 to 10-8
Rural outdoor (Terminal relative speed to ground up to 500 km/h) (Note 3)	Max Transfer Delay 20 - 300 ms BER 10-3 - 10-7 (Note 1)	Max Transfer Delay 150 ms or more (Note 2) BER = 10-5 to 10-8
Urban/ Suburban outdoor (Terminal relative speed to ground up to 120 km/h)	Max Transfer Delay 20 - 300 ms BER 10-3 - 10-7 (Note 1)	Max Transfer Delay 150 ms or more (Note 2) BER = 10-5 to 10-8
Indoor/ Low range outdoor (Terminal relative speed to ground up to 10 km/h)	Max Transfer Delay 20 - 300 ms BER 10-3 - 10-7 (Note 1)	Max Transfer Delay 150 ms or more (Note 2) BER = 10-5 to 10-8
NOTE 1; There is likely to be a compromise between BER and delay. NOTE 2; The Max Transfer Delay should be here regarded as the target value for 95% of the data. NOTE 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).		

~~5.5~~8.4.1 Supported End User QoS

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

Error tolerant	Conversational voice and video	Voice messaging	Streaming audio and video	Fax
Error intolerant	Telnet, interactive games	E-commerce, WWW browsing, Email access,	FTP, still image, paging	Usenet
	Conversational (delay <<1 sec)	Interactive (delay approx1 sec)	Streaming (delay <10 sec)	Background (delay >10 sec)

Figure 3 Groups of applications behavior in terms of QoS requirements

The following tables further elaborate IMT-2000 end user / application QoS requirements.

Table 5: End-user Performance Expectations - Conversational / Real-time Services

Medium	Application	Degree of symmetry	Data rate	Key performance parameters and target values		
				One-way Delay	Delay Variation	Information loss
Audio	Conversational voice	Two-way	4-13 kb/s	<150 msec preferred <400 msec limit	< 1 msec	< 3% FER
Video	Videophone	Two-way	32-384 kb/s	< 150 msec preferred <400 msec limit Lip-synch : < 100 msec		< 1% FER
Data	Telemetry - two-way control	Two-way	<28.8 kb/s	< 250 msec	N.A	Zero
Data	Interactive games	Two-way	< 1 KB	< 250 msec	N.A	Zero
Data	Telnet	Two-way (asymmetric)	< 1 KB	< 250 msec	N.A	Zero

Table 6: End-user Performance Expectations - Interactive Services

Medium	**Application**	**Degree of symmetry**	**Data rate**	**Key performance parameters and target values**		
				One-way Delay	Delay Variation	Information loss
Audio	Voice messaging	Primarily one-way	4-13 kb/s	< 1 sec for playback < 2 sec for record	< 1 msec	< 3% FER
Data	Web-browsing - HTML	Primarily one-way		< 4 sec /page	N.A	Zero
Data	Transaction services – high priority e.g. e-commerce, ATM	Two-way		< 4 sec	N.A	Zero
Data	E-mail (server access)	Primarily One-way		< 4 sec	N.A	Zero

Table 7: End-user Performance Expectations - Streaming Services

Medium	**Application**	**Degree of symmetry**	**Data rate**	**Key performance parameters and target values**		
				One-way Delay	Delay Variation	Information loss
Audio	High quality streaming audio	Primarily one-way	32-128 kb/s	< 10 sec	< 1 msec	< 1% FER
Video	One-way	One-way	32-384 kb/s	< 10 sec		< 1% FER
Data	Bulk data transfer/retrieval	Primarily one-way		< 10 sec	N.A	Zero

Data	Still image	One-way		< 10 sec	N.A	Zero
Data	Telemetry - monitoring	One-way	<28.8 kb/s	< 10 sec	N.A	Zero

8.5 Principal speech quality requirements

8.5.1 Subjective quality

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

8.5.2 Natural speech quality and speaker recognition

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

8.5.3 Ease of conversation

Subscribers shall 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.

8.5.4 Loss of interactivity due to delay in the speech path

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.

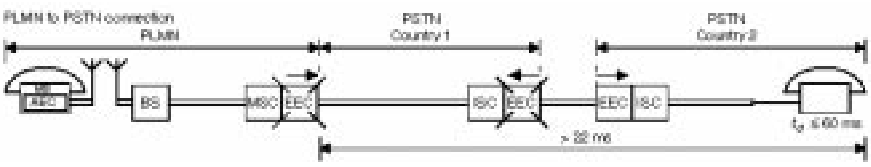
In a digital Public Land Mobile Network with sufficient echo control, ITU-T Recommendation G.173 recommends a mean one way delay objective of 20 ms and that the one way delay should not exceed 40 ms. It is recognized that in the satellite component the one way delay may exceed 40 ms, due to the propagation 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, air propagation between a mobile and the PSTN connection (PLMN):

FIGURE
A.4/G.173



The results of subjective tests are reported in ITU-T Recommendation G.114, based on the Mean Opinion Score (MOS) degradation over a range of one way delay transmission times from 0 to 1 500 msec. The results are plotted in terms of Percent Poor or Worst (POW):

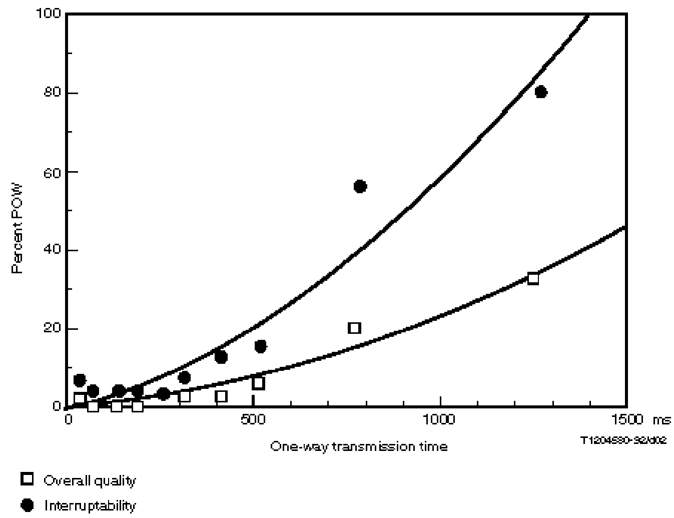


FIGURE B.2/G.114

Comparison of POW for overall quality and interruptability

The above results clearly indicate that there is no significant difference in the overall quality or interruptablity when the one-way-delay transmission time is maintained below 300 msec. 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.

Based on the above results and the ITU-T Study Group 12 Liaison Statement to Task Group 8/1, WG 4 recommends that a mean one way delay of less than 40 ms is an important objective for IMT-2000. However, it recognizes 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.

8.5.5 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 draft Recommendation G.174.

By keeping the access delay sufficiently small the need for echo control can be avoided and significant cost saving achieved.

8.5.6 Uniformity in different environments

Where different air interfaces are used for access in different environments (e.g. pico cell, large cells, etc.) the same speech quality requirements shall be used. The subscriber shall 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.

8.5.7 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.

Consideration should be given to techniques which will minimize the need for and the impact of transcoding.

The effects of transcoding should be fully considered in meeting the speech quality requirements given in this document.

8.5.8 Quality of end-to-end connections

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

8.5.9 Handset 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 draft 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.

8.5.10 Call progress tones, announcements and music

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

8.5.11 Voice recognition

Due regard shall 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.

8.5.12 Handover

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

8.5.13 Gross speech bit rate

The gross speech bit rate (rather than the codec bit rate) required in the radio interface to support both the digital speech and the necessary error control coding, shall be considered in selecting the speech codec (see § 5.7).

Alternatively, the figure of merit for selecting a speech codec could be the resulting capacity of the system.

8.5.14 Robustness

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

8.5.15 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.

8.5.16 Cost and power consumption

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

8.5.17 Interconnection of IMT-2000 users in different networks

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

8.5.18 Speech codec objectives

Annex 2 contains a table of parameters for speech codecs for the IMT-2000 application. Required values of the parameters are given together with values to be used as objectives.

8.5.19 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 should 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 shall 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 shall be included.

During testing, the speech connection shall be stressed with an error pattern generated by an error model related to the air interface. At the present time the air interface technology has not yet been selected and consequently an interim error model must be used.

The interim error model is the Bellcore model described in Annex 3, which is representative of the burst errors found by slow-moving or stationary users of mobile systems. Task Group 8/1 expects to generate further error models appropriate to the air interface technologies developed for IMT-2000 and to the range of environments and vehicle speeds to be expected in the system. An explanation of generation of error models for IMT-2000 interfaces is provided in Annex 2.

8.6 Principal voiceband data requirements

8.6.1 DTMF signalling requirements

The transmission of DTMF information shall be supported by IMT-2000 with a performance comparable to the fixed network (see ITU-T draft 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.

8.6.2 Voiceband data requirements

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

8.7 Radio performance requirements

8.7.1 Speech quality requirements

The speech quality in a connection in IMT-2000 involving two radio interfaces, under the error conditions defined by the current IMT-2000 error model, together with any necessary transcoding shall not be degraded more than 0.5 MOS compared with error free G.726 at 32 kbit/s.

8.7.2 Connection performance requirements

International Mobile Telecommunications-2000 will need to meet the performance requirements listed in Table 1, for which values have yet to be specified (also see ITU-T draft Recommendation G.174).

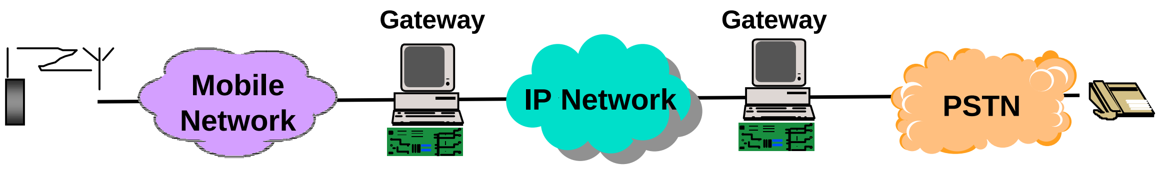
TABLE 8

Connection performance requirements

Probability of blocking on the radio link	For further study
Probability of blocking between IMT-2000 and the fixed network	For further study
Post selection delay	For further study
Answer signal delay	For further study
Call release delay	For further study
Probability of unexpected disconnect	For further study
Probability of unsuccessful handover	For further study

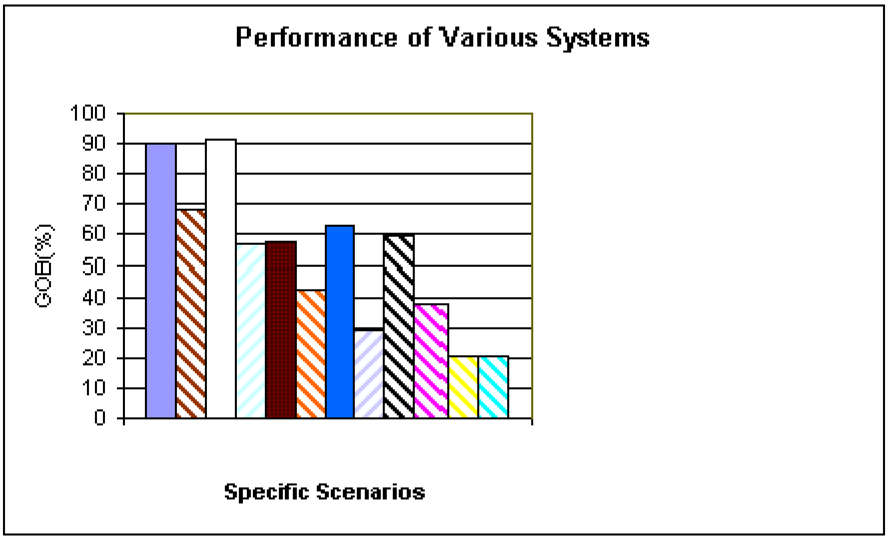
Annex 1: Planning tool to assess end-to-end voice transmission quality

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



For example in the above figure, 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



Using the E-model, output results (for example expressed as "Good or Better" (GoB)) can be obtained for a variety of possible scenarios – some may be very good (90% GoB), others may be very bad (20% GoB).

[ANNEX 1

IMT-2000 speech codec objectives

1. Introduction

Recommendation ITU-R M.1034 identifies a number of terrestrial and satellite environments that will place different requirements on the speech codecs used in these environments. For example at one extreme there will be users accessing the system in city pico cells, using small handsets, where any shortfall in quality will contrast strongly with conventional wired access, which will often be immediately at hand. Speech codecs will have to meet the IMT-2000 quality requirement at a low power consumption and also suffer the minimum quality degradation as a result of transcoding in the network. In this situation it may be desirable to use a speech codec with a higher bit rate, and a lower power consumption and therefore complexity. This could be considered to be a class A type speech codec (see below).

In cells where spectrum considerations require a lower bit rate, it will be desirable to use a speech codec with a lower bit rate, resulting in greater complexity and power consumption in the codec to meet the IMT-2000 quality requirement. Such a speech codec could be considered a class B codec.

Again, when considering satellite connections, the spectrum consideration may result in an even lower bit rate being adopted with consequent increase in complexity and power consumption to meet the IMT-2000 quality requirement. This progression from high bit rate, low complexity waveform codecs to low bit rate higher complexity algorithm codecs to meet the need for increasing spectrum efficiency is followed in the speech codec objectives identified later.

Should more than one speech codec be adopted for IMT-2000, consideration should be given to ways of minimising the impact of the transcoding that will occur, by for example adopting common frame structures or methods of synchronization.

It should be emphasized that all the speech codecs must meet the single IMT-2000 quality requirement, however they are likely to have different parameters, particularly in their bit rate and their power consumption. In order to meet the IMT-2000 requirement of the minimum number of speech codecs, and also to achieve spectrum efficiency and wide acceptance of the system in the market place, it is necessary to relate the required parameters of the IMT-2000 speech codecs to the anticipated environments. Table 2 lists the environments given in Recommendation ITU-R M.1034 and relates them to categories of speech codec. These categories are then defined by listing their parameters in order to provide the objectives for speech codec development which are required by Telecommunication Standardization Study Group 15.

2. Radio environments

The following radio environments are taken from Recommendation ITU-R M.1034:

TABLE 2

Environment	Class
Business indoor	A
Neighbourhood indoor/outdoor	A and B
Home	A
Urban vehicular outdoor	B
Urban pedestrian outdoor	A and B
Rural outdoor	B
Fixed outdoor	B
Terrestrial aeronautical	B
Local high bit rate	A
Urban satellite	B and C
Rural satellite	C
Satellite fixed-mounted	C
Indoor satellite	C

The classes A, B and C identify the commonality of requirement in the different radio environments in the sense that a speech codec that satisfies for example the business environment is likely to satisfy the home environment as well. It should be recognized that IMT-2000 requires the minimum number of codecs so that a codec that meets the requirements of more than one class may be a desirable choice, but would raise the question of the trade-off between having fewer codecs, or having codecs that more closely match the need.

3. Speech codec parameters

TABLE 3

Parameter	Class					
	A		B		C	
	Required value	Objective	Required value	Objective	Required value	Objective
Speech quality without errors	G.726		G.726		G.726	
Codec delay (one way) (ms)	10	2	20	10	40	20
Power consumption (mW)	2	1	20	5	300	200
Speech codec bit rate (kbit/s)	32	16	16	4	4	2-3
Adaptive bit rate capability (kbit/s)	No	Yes	Yes	Yes	Yes	Yes
Voice activity detection capability	No	No	Yes	Yes	Yes	Yes
Transparent to DTMF	No	Yes	No	Yes	No	Yes

Note 1 – Power consumption figures are based on the technology to be expected in the year 2000.

4. Test methods

Testing should take account of both the errors in the mobile radio channel and also the para-conversational use of the telephone connection.

The wireless transmission technologies are not yet defined in IMT-2000 and may be narrow-band or broadband TDMA or CDMA or some combination of these, which would alter the error pattern suffered by the digital speech. The existing Bellcore model is a good first indication of burst error performance, but may need to be refined to give a satisfactory result for transmission technologies which are not directly block oriented. Task Group 8/1 will be responsible for producing an improved model related to the chosen wireless access channel.

Whilst the conversational use of the connection would indicate the need for conversational testing, it is recognized that subjective testing using conversations would be difficult. However, it is felt that in IMT-2000 where there is a cascade of segments in the connection, the build-up of impairments may make it misleading to simply add their degradations. At the least, consideration should be given to calibrating the test method with some conversational tests of key cases to ensure the validity of the results.

ANNEX 2

Generation of error models for IMT-2000 air interfaces

1. Introduction

Error masks are needed to evaluate and select speech codecs for the IMT-2000 application which realistically represent the errors to be expected in the IMT-2000 air interfaces when used by subscribers moving at typical speeds in the mobile environment. The IMT-2000 air interface technology has yet to be selected, and the choice of say CDMA or TDMA or a combination of the two will significantly alter the pattern of errors that occurs.

No single error mask can properly represent the wide range of candidate access techniques or the range of users' speeds. However, the development of speech codecs continues and some interim error mask is needed for tentative assessment and selection. Such an interim position is given in Annex 3 where the Bellcore Error Model is described, which generates error bursts typical of slow moving users in a mobile environment.

Such models are based on simulation of the air interface to generate the statistics of the errors resulting from channel impairments, like fading and noise, and interference from other users. These statistics can then be fed into the model, which is then used to generate error masks. Alternatively, an error mask can be taken directly from the simulation.

The following sections describe the simulation of example TDMA and CDMA air interfaces and give the resulting error statistics, which are shown to have a wide range which would require several error masks to properly represent them. In addition it is shown that Frame Error Bursts (FEBs) occur where a number of consecutive frames are in error over periods as long as 150 ms, which may be too long to be tolerated by speech codecs aimed at toll quality speech.

The problem of long FEBs leads to an alternative view of both testing speech codecs and specifying the air interface, in which the maximum duration on FEBs is limited to say 50 ms, with a ratio of occurrence less than say 1%, which is perhaps the worst that speech codecs can cope with and therefore the minimum performance the air interface technology must realize.

A further problem is that in some systems, different classes of bits may be specified by the speech codec for different degrees of channel protection. For instance, the North American TDMA Digital Cellular standard (IS-54) specifies three classes of bits, each protected differently. In other words, for the same channel conditions and transceiver structure, the different classes of bits may undergo different error statistics. The Bellcore model does not take this situation into account. Consideration should be given to the use of the actual bit-error and frame-error masks obtained from the transceiver simulations when the air interface access technology is selected.

The following material indicates the kind of simulations that can be made of IMT-2000 air interfaces in order to produce statistics for error models. Simulations of this kind will be needed when the IMT-2000 air interfaces are chosen so that a definitive error model or models for IMT-2000 can be defined.

2. Simulation of TDMA and CDMA wireless access

This section provides a brief description of the transceivers and the channel models simulated to arrive at typical FEB statistics. The radio channel is modelled by a single Rayleigh fading path narrow-band model. The interference is assumed to be AWGN. The Rayleigh fading channel was simulated using Jakes' model.

An example TDMA/FDMA system, with carrier spacing of 350 kHz and TDMA frames of 8 ms long is simulated. Each frame is divided into 40 time slots, each carrying 8 kbit/s of speech data. $\pi/4$ QPSK modulation with coherent reception is considered. A 2-antenna selection diversity is employed.

In an example CDMA system, the uplink (mobile-to-base) and downlink (base-to-mobile) transceivers are simulated. The user data rate is 9.6 kbit/s and the frame length is 20 ms. The uplink of the CDMA system first spreads the user data using a rate 1/3 convolutional code. The resulting coded symbols are interleaved and then each 6 symbols are mapped onto 64-ary orthogonal signals. The receiver employs 2-antenna diversity with a square-law non-coherent combiner on the uplink. A closed-loop power control strategy is used. The base estimates the received signal power. A power control command bit is generated every 1.25 ms by comparing the estimated power against a threshold. Based on power control command bits received from the base, the portable terminal adjusts its transmitted power every 1.25 ms by a fixed step size. The uplink channel per antenna is modelled as a single Rayleigh fading path.

In the CDMA system downlink, a rate 1/2 convolutional code is used to spread the user data, followed by QPSK spreading with PN codes. BPSK modulation with coherent reception and multipath diversity are considered. The downlink channel is modelled by two independently Rayleigh fading paths with equal average powers. The two paths are assumed to be generated artificially using two transmit antennas at the base.

3. TDMA and CDMA Frame-Error-Burst (FEB) statistics

This section presents simulation results on the statistics of FEB for TDMA and CDMA systems. Two types of statistics are presented:

– the number of FEBs per second with duration greater or equal to a given value, and

– the CDF (cumulative probability distribution function) of the FEB length.

Terminal speeds of 1, 10 and 100 km/h at a carrier frequency of 2 GHz are considered. The above statistics on the FEB are obtained under the constraint that the bit-error-ratio (BER) be less than 0.001.

Figure 2 shows the rate of occurrence of FEBs of different durations for the TDMA system, at portable speeds of 1, 10 and 100 km/h. The results of Fig. 2 were obtained for SNR of 18.5 dB corresponding to a bit-error ratio of less than 0.001. The corresponding frame-error-ratio (FER) was 0.1. Figure 3 shows the CDF of FEB duration for TDMA at speeds of 1 and 10 km/h.

FIGURE 2

Number of FEBs per second whose duration is greater or equal to abscissa (TDMA system)

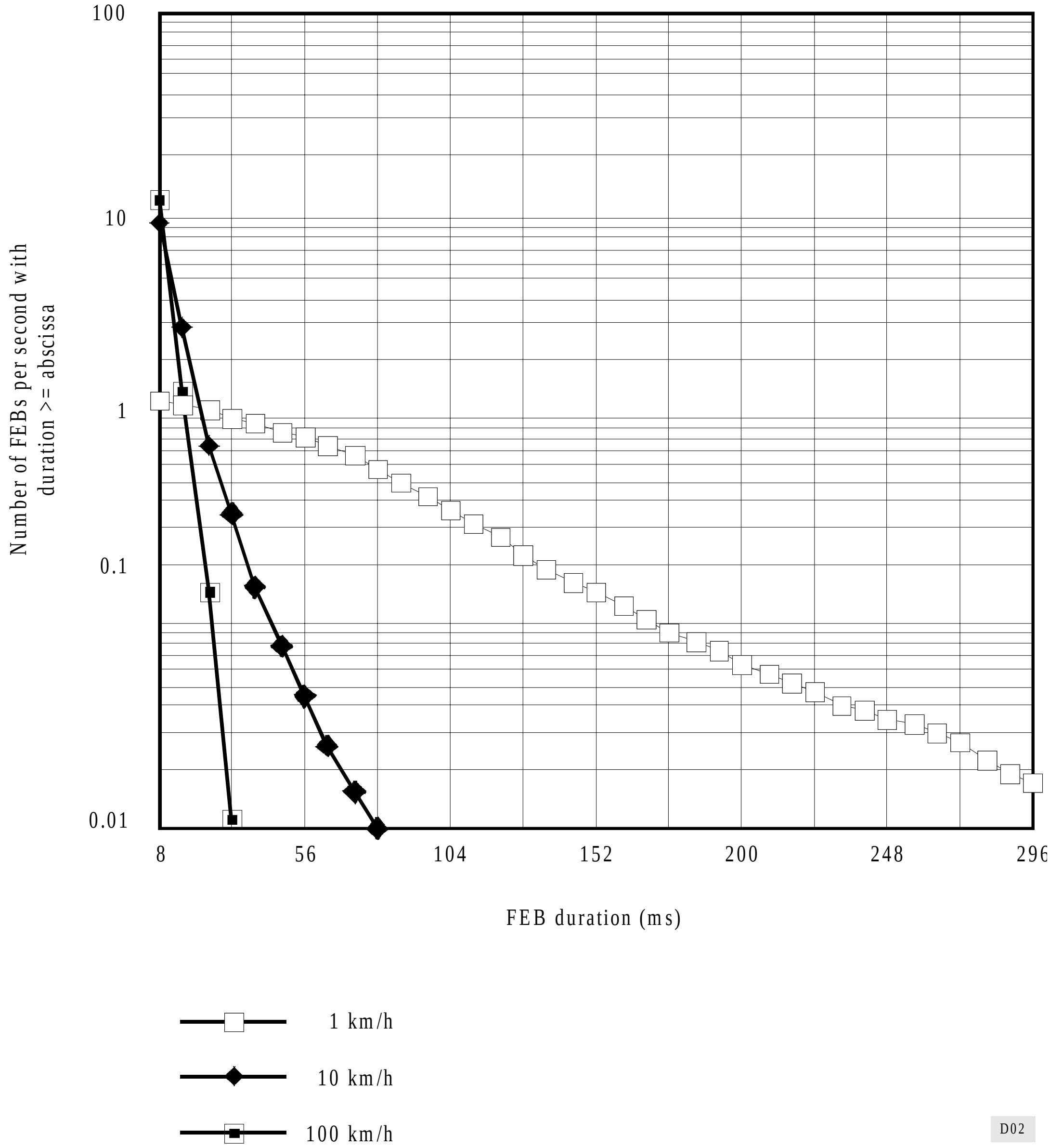
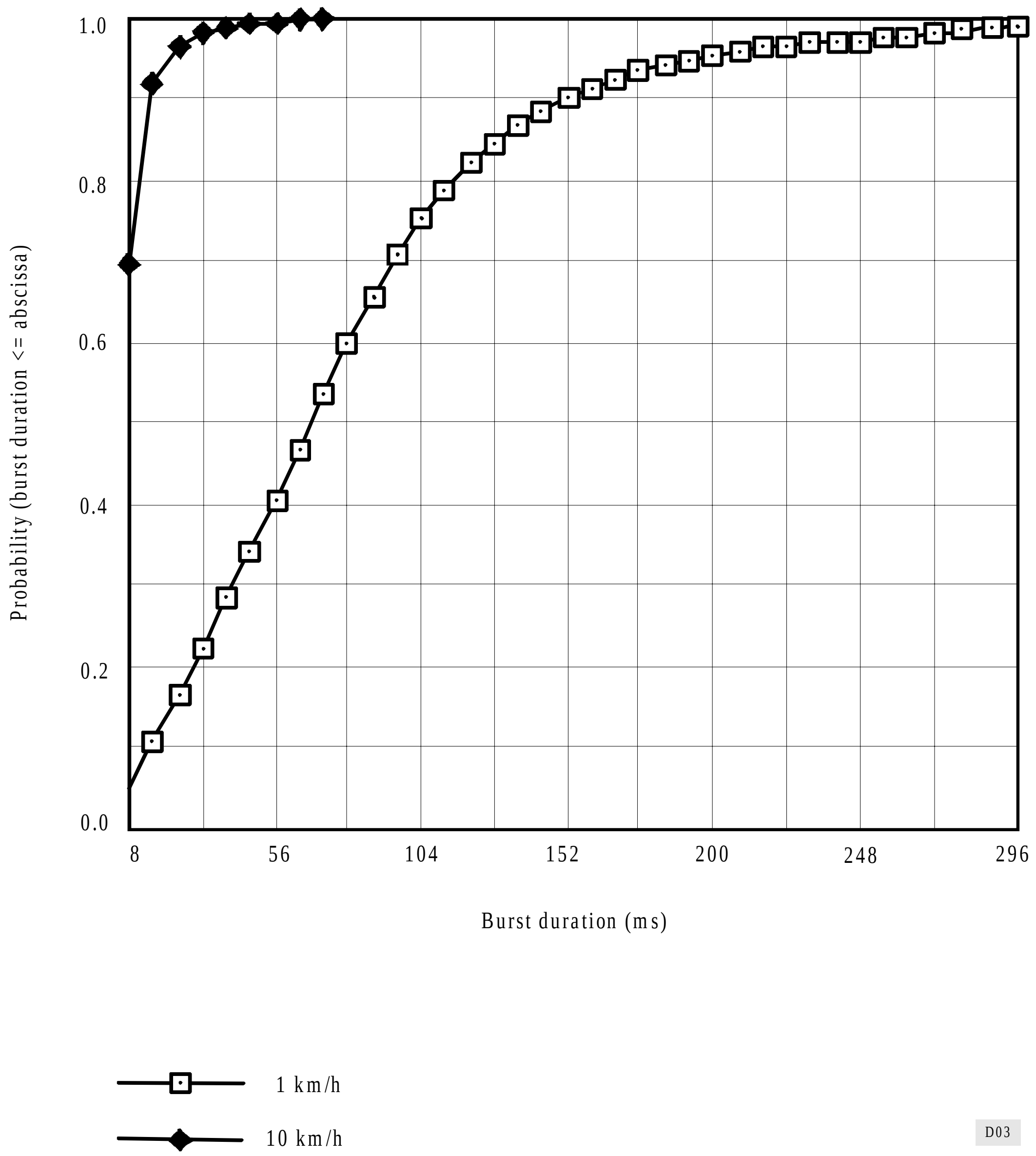


Figure 4 gives the rate of occurrence of FEBs of various durations for the CDMA downlink at a terminal speed of 1 km/h. The SNR of 16 dB corresponds to a BER of less than 0.001. The values for 10 km/h are not shown on the graph because the number of FEBs per second with duration greater than 1 frame was less than 0.01. FEB values for the 1 and 10 km/h cases are 0.0057 and 0.0043. The CDF of FEB duration is given in Fig. 5.

For the uplink, it was observed that the number of FEBs per second with a duration greater than 2 frames was less than 0.01 for a speed of 1 km/h. At 10 km/h, the number of FEBs per second with a duration greater than 1 frame was less than 0.01. The FEBs for speeds of 1 and 10 km/h are 0.024 and 0.012. The closed-loop power control considered on the uplink is fast enough to partially mitigate the effects of Rayleigh fading at low Doppler frequency. This has significantly reduced the FEB duration at low speeds. At higher speeds (high Doppler frequencies), the combination of channel coding and interleaving reduce the duration frequency of FEBs. Although fast power control is not simulated on the downlink of the CDMA system, the duration and frequency of FEBs on the downlink of CDMA is smaller than that of the TDMA system. This is due to the channel coding in the CDMA system.

FIGURE 3

CDF of FEB length for the TDMA system



4. Conclusions from the simulations

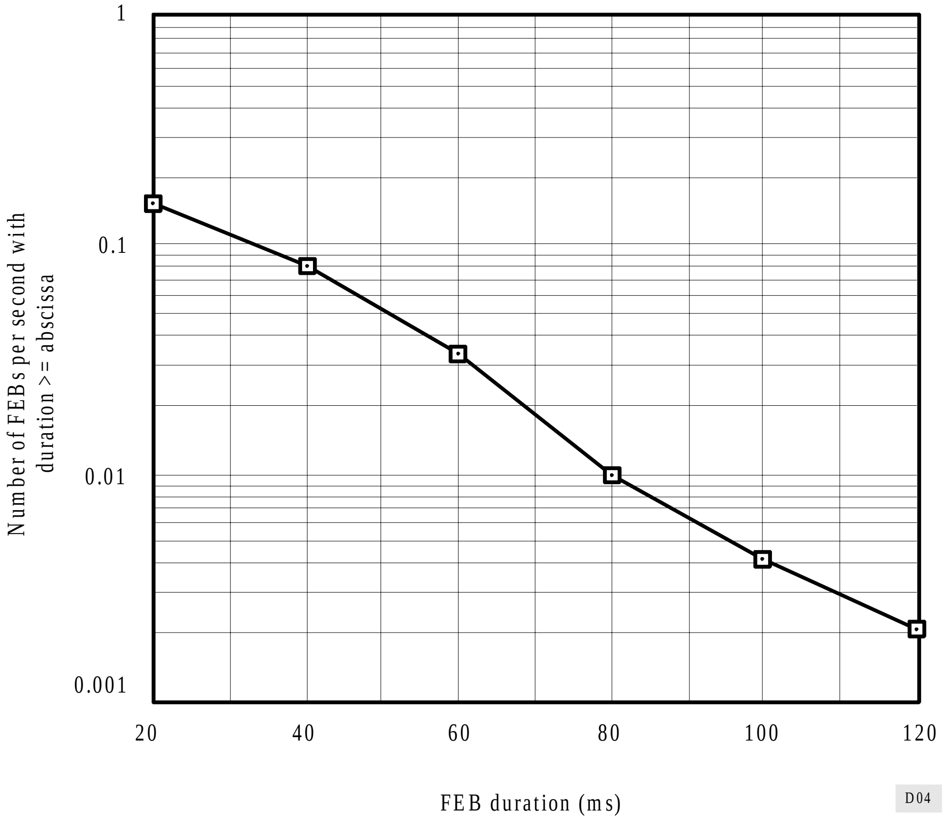
4.1 The simulation results show a wide range of statistics depending on the particular wireless access technique and the terminal speed. These different statistics impose different robustness requirements on the speech codec.

4.2 A speech codec that is optimized for one wireless access technique may not be optimal with another access scheme. Thus, one overall error model may not suffice for the purpose of speech codec evaluation and selection.

4.3 Figures 3 and 4 illustrate that for the TDMA system, frame-error-bursts longer than 152 ms (19 frames of duration 8 ms each) occur with significant probability. Therefore, the finite state Markov chain error model proposed by Bellcore needs more than 8 states to represent a wide range of access techniques.

FIGURE 4

Number of FEBs per second with duration greater or equal to abscissa, speed of 1 km/h (downlink of CDMA system)



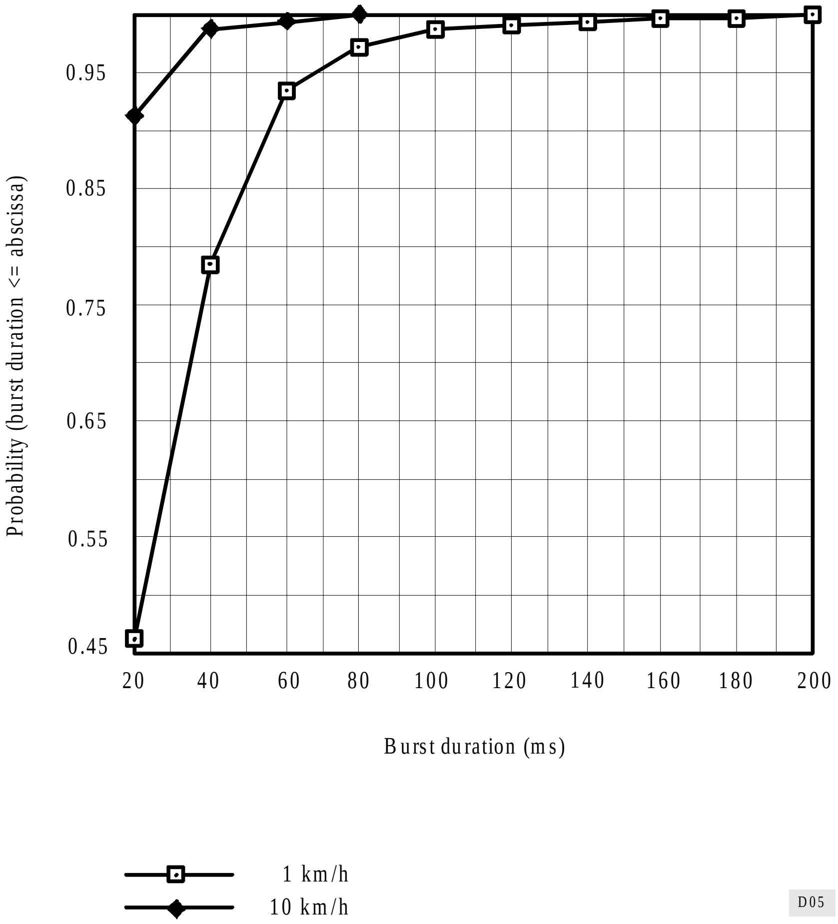
5. Alternative criterion for speech codec evaluation

Consideration should be given to an alternative criterion in which a limit is imposed on the number of FEBs per second with duration greater than a certain threshold value. The speech codecs would be designed to achieve high quality under the specified constraints on the frequency and duration of FEBs.

The candidate wireless access techniques would then be required to ensure that the constraints on the statistics of FEBs are met. The threshold value on the duration of FEB and the constraint on the number of such bursts per second must be obtained as a result of subjective testing. Based on preliminary results, a threshold value of 50 ms for the duration of FEBs, and a threshold value on the number of occurrences of such FEBs in the range of 1 to 5% per second provide reasonable starting values. It is recommended that further subjective testing of fade masking techniques be made to establish a precise value.

FIGURE 5

CDF of FEB length for the downlink of the CDMA system



ANNEX 3

Interim IMT-2000 error model – Bellcore

1. Introduction

Analysis of radio wave propagation in urban and suburban environments for proposed PCS/IMT-2000 services indicates that due to multipath signal cancellations (i.e. Rayleigh fading), slow moving or stationary users may experience sporadic bursts of errors. The frequency and duration of these error bursts are dependent on factors such as user motion, frequency, type of diversity used, detection scheme, and mean SNR, and durations may range from a few to several hundred milliseconds. Speech codecs that are to be used over radio channels in such applications should therefore be evaluated while subjected to burst error patterns typical of this environment.

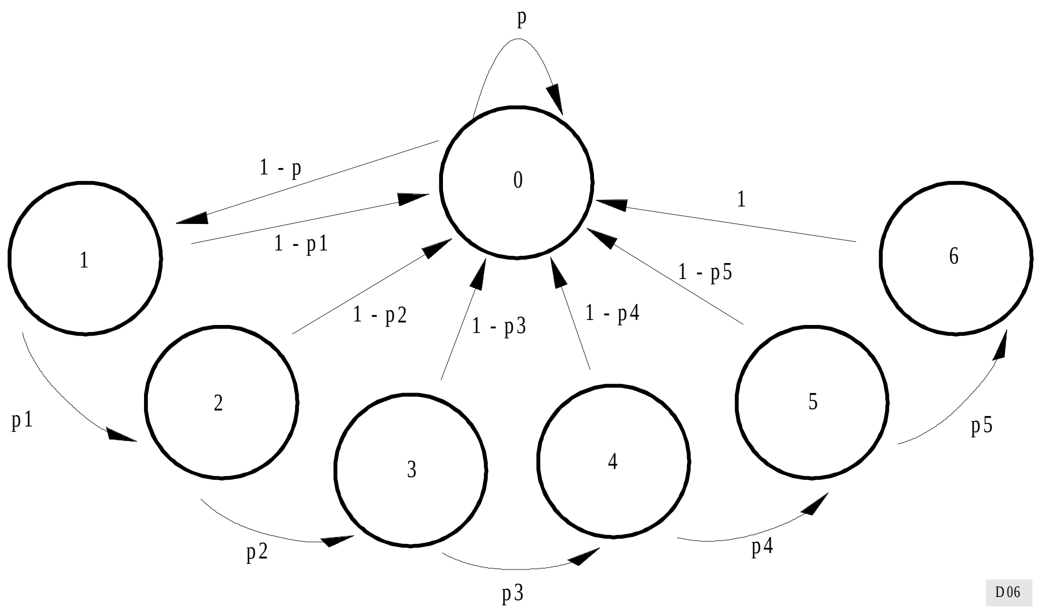
The extreme variability of the parameters means that the possible range of fade durations and fades per second that a user could encounter would create a test matrix too large to be suitable for use in subjective testing. Since no one set of parameter values can adequately represent the error environment “seen” by the typical user, an alternative method is to define a test that would characterize a speech codec's performance by the way it reacts to a fixed pattern of short, medium and long error bursts. The definitions of short, medium and long are somewhat arbitrary and would likely be based from the frame length used by the codec. For example, for a codec using a 16 ms frame length, the definitions could be: "short" is one to two frames, "medium" is three to four frames, and "long" is five to six frames.

2. Overview of the burst error model

A 7-state Markov model has been proposed to generate correlated bursts of errors for use in subjective testing. This Annex gives additional information about the model, and explains how it can be used to generate the desired burst error patterns. A diagram of the model is shown in Fig. 6.

FIGURE 6

Burst error model (7-State Markov chain)



The model has seven states (0 to 6) with the name of each state corresponding to the number of frames lost during an error burst. For example, in state 0 there are no missed frames, state 1 represents a single missed frame, and state 6 represents an error burst that causes six frames to be lost.

The model is controlled by the conditional probabilities (p, p1, p2, ... p6) that define the probability of going from one state to the other. For example, starting in state 0, we see that there is a probability of p that we will remain in state 0. Conversely, we could say that the probability of going from state 0 to state 1 would be $1 - p$. Once in an error state (states 1 through 6), the options for each successive trial are to continue to the next error state or return to state 0.

One feature of this model is that once in state 6, the probability of returning to state 0 is 1, thereby truncating the maximum number of lost frames in a single error burst to six. For a frame length of 16 ms, this means that the longest error burst that the model can generate is 96 ms. The assumption is that error bursts in excess of about 100 ms introduce impairments independent of a codec's ability to recover, reducing its value for comparative testing.

3. Desired burst error statistics

The theoretical average error ratio and the probabilities of each error state can be mathematically derived from the conditional probabilities. Table 4 shows an example using an arbitrarily selected set of conditional probabilities.

TABLE 4

Conditional probabilities p = 0.99 p1 = 0.7 p2 = 0.6 p3 = 0.5 p4 = 0.4 p5 = 0.3		
State	Probability of state *n*	Formula (K_n = probability of being in state *n*)
0	0.975608	$K_0 = 1 - K_1 - 2 \times K_2 - 3 \times K_3 - 4 \times K_4 - 5 \times K_5 - 6 \times K_6$
1	0.003000	$K_1 = (1 - p) \times (1 - p1)$
2	0.002800	$K_2 = (1 - p) \times p1 \times (1 - p2)$
3	0.002100	$K_3 = (1 - p) \times p1 \times p2 \times (1 - p3)$
4	0.001260	$K_4 = (1 - p) \times p1 \times p2 \times p3 \times (1 - p4)$
5	0.000588	$K_5 = (1 - p) \times p1 \times p2 \times p3 \times p4 \times (1 - p5)$
6	0.000252	$K_6 = (1 - p) \times p1 \times p2 \times p3 \times p4 \times p5$
Theoretical average error ratio = $1 - K_0$ = 2.44%		

The conditional probabilities should be chosen such that two requirements are satisfied:

– that the burst error states are uniformly distributed, i.e. each error state should occur about as often as any of the other error states, and

– that the average burst error ratio be limited to about 3%.

It may be desirable to slightly limit the occurrence of longer error bursts, so as to allow room under the 3% error ratio constraint for an adequate number of short bursts. Based on these design considerations, a reasonable distribution of burst errors might be:

TABLE 5

Number of missed frames in error burst	Desired ratio of occurrence (%)
1	0.2
2	0.2
3	0.2
4	0.2
5	0.1
6	0.1

The set of conditional probabilities that would produce such a distribution (given a large enough sample) are shown in Table 6. As can be seen from Table 6, these values result in a very close match between the theoretical and desired ratio of occurrence.

TABLE 6

Conditional probabilities	State	Theoretical probability of state *n*
p = 0.99	0	0.968965
p1 = 0.8	1	0.002000
p2 = 0.75	2	0.002000
p3 = 0.67	3	0.001980
p4 = 0.5	4	0.002010
p5 = 0.5	5	0.001005
	6	0.001005
Theoretical average error ratio = $1 - K_0$ = 3.10%		

4. Model performance: theoretical vs. actual

For actual use in subjective testing, a computer program was written to implement the 7-state Markov model and generate the desired distribution of burst errors. Running this program with the conditional probabilities specified in Table 6, we obtain the following results:

TABLE 7

Number of trials: 1 000					
Conditional probability	Frames in burst	Theoretical probability	Actual ratio of occurrence	Actual number of occurrences	Total missed frames
p = 0.99	0	0.968965	0.958	958	0
p1 = 0.80	1	0.002000	0.003	3	3
p2 = 0.75	2	0.002000	0.002	2	4
p3 = 0.67	3	0.001980	0.004	4	12
p4 = 0.50	4	0.002010	0.002	2	8
p5 = 0.50	5	0.001005	0.003	3	15
p6 = 1.0	6	0.001005	0.000	0	0
Total missed frames = 42 Actual missed frame ratio = 4.2% Theoretical missed frame ratio = 3.1%					

Note that the actual ratios of occurrence and missed frame ratio differ from the theoretical ratios. This is because the computer simulation of the Markov model makes use of a random number generator (the *rand()* function of ANSI C) so the actual outcomes depend on the start seed, the spectral properties of the random number generator, and (most importantly) the number of trials. As the number of trials increases, the actual ratios of occurrence and actual missed frame ratios converge towards the theoretical ratios. Table 8 shows the results for 10 000 trials. Note that the actual results converge closer to the theoretical results.

TABLE 8

Number of trials: 10 000					
Conditional probability	Frames in burst	Theoretical probability	Actual ratio of occurrence	Actual number of occurrences	Total missed frames
p = 0.99	0	0.968965	0.9709	9 709	0
p1 = 0.80	1	0.002000	0.0021	21	21
p2 = 0.75	2	0.002000	0.0018	18	36
p3 = 0.67	3	0.001980	0.0016	16	48
p4 = 0.50	4	0.002010	0.0022	22	88
p5 = 0.50	5	0.001005	0.0010	10	50
p6 = 1.0	6	0.001005	0.0008	8	48
Total missed frames = 291 Actual missed frame ratio = 2.91% Theoretical missed frame ratio = 3.1%					

5. Generating error patterns for use in subjective testing

The above results show that although the computer simulation of the model can generate a distribution of burst errors whose total average missed frame ratio can be specified, the bursty nature of the errors requires that care must be taken when extracting a partial segment from the entire generated error pattern. Otherwise, the total error ratio and error distributions in the segment may not match the distributions that were sought.

Since the computer simulation will be used to generate error patterns for use in subjective testing, the solution is to generate an error stream whose length matches that of the speech sample used in the subjective tests; and then adjust the conditional probabilities (through trial and error) to obtain the desired distributions. For example, if the speech samples used in the testing are 20 s long, then the computer simulation would be programmed to generate a pattern of 1 250 16 ms frames. For 12 ms frames, a pattern of 1 667 frames would be generated. Table 9 shows the results for 1 500 trials.

TABLE 9

Number of trials: 1 500					
Conditional probability	Frames in burst	Theoretical probability	Actual ratio of occurrence	Actual number of occurrences	Total missed frames
p = 0.99	0	0.968965	0.9687	1 453	0
p1 = 0.80	1	0.002000	0.0020	3	3
p2 = 0.75	2	0.002000	0.0020	3	6
p3 = 0.67	3	0.001980	0.0033	5	15
p4 = 0.50	4	0.002010	0.0013	2	8
p5 = 0.50	5	0.001005	0.0020	3	15
p6 = 1.0	6	0.001005	0.0000	0	0
Total missed frames = 47 Actual missed frame ratio = 3.1% Theoretical missed frame ratio = 3.1%					

Since the actual error distribution (see Table 9) is not very even (state 3 occurs too often, state 6 does not occur), the conditional probabilities can be adjusted to obtain a better distribution as shown in Table 10.

TABLE 10

Number of trials: 1 500					
Conditional probability	Frames in burst	Theoretical probability	Actual ratio of occurrence	Actual number of occurrences	Total missed frames
p = 0.99	0	0.970084	0.9624	1 457	0
p1 = 0.77	1	0.002070	0.0024	3	3
p2 = 0.75	2	0.001733	0.0024	3	6
p3 = 0.76	3	0.001247	0.0040	3	9
p4 = 0.70	4	0.001185	0.0016	2	8
p5 = 0.75	5	0.000691	0.0024	2	10
p6 = 1.0	6	0.002074	0.0000	1	6
Total missed frames = 42 Actual missed frame ratio = 2.8% Theoretical missed frame ratio = 3.0%					

6. Scaling the model to accommodate different frame lengths

As mentioned previously, the model is designed to limit the longest single error burst to a maximum of about 100 ms. The current model has 7-states based on the assumption that a 16 ms frame will be used (i.e. 6 x 16 ms = 96 ms). If the model is to be used to generate error patterns for use with a different frame length, then the model will have to be scaled appropriately. For example, if a 12 ms frame is used, then the model should be expanded by two additional states so that the maximum number of frames that can be lost in a single error burst will be 8, corresponding to 96 ms (8 × 12 ms).

Since the model is realized with software, expanding the states of the model can be done quickly. Therefore, once the frame length of a candidate codec becomes known, a burst error model can be quickly designed and brought on line to generate the burst error pattern required for subjective testing.

7. Summary

The proposed burst error model is based on a multi-state Markov chain and can generate burst errors of various durations for use in subjective testing. The distribution of error durations is controlled by selecting a set of conditional probabilities. As mentioned previously, the conditional probabilities should be chosen such that two requirements are satisfied:

– that the burst error states are uniformly distributed, i.e. each error state should occur about as often as any of the other error states, and

– that the average burst error ratio be limited to about 3%.

It may be desirable to slightly limit the occurrence of longer error bursts, so as to allow room under the 3% error ratio constraint for an adequate number of short bursts.

The model should be used to generate an error pattern approximately equal in length to the speech sample used in the subjective tests. The conditional probabilities can then be adjusted on a trial-and-error basis until the exact distribution of burst error durations is obtained.]