3GPP TSG-SA ***TSGS4#6(99)253R***
TSG-S4#6: September 8-10, 1999, Sophia-Antipolis, France

Liaison To: **TSG-S2**

From: **TSG-S4**

cc: **TSG-RAN**

Subject: **Comments on the revised version of ITU-R TG8/1 Recommendation M.1079**

TSG-S4 has reviewed the revised version of the ITU-R recommendation M.1079 entitled "Performance and Quality of Service Requirements for International Mobile Telecommunications-2000 (IMT-2000)" contained in the document SP-99312/RP-99372, forwarded by TSG-SA to S1, S2 and S4.

Understanding that S2 is responsible for gathering comments to be submitted to ITU on this recommendation, the following inputs are provided to S2 for further consideration:

Note that S4 comments are essentially addressing technical aspects of the recommendation of relevance to S4. It was also mentioned that the document as a whole would benefit from a general editing and structuring work.

[S4 Note: No comments are provided on the content of the Annexes since they appear unchanged from the previous version of M.1079, however, S4 believes that the parameters specified in the second Annex 1 should be updated]

S4 comments or proposed modifications to ITU-R M.1079:

4. Related documents

Add ITU-T Recommendation G.116: Transmission performance objectives applicable to end to end international connections.

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 three~~two~~ types of quality: quality ~~which is mainly~~ dependent on terminal~~handset~~ acoustics, quality dependent on the speech coding performance 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.

Further definitions of categories of speech transmission quality can be found in ITU-T Recommendation G.109.

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6.7 Guidelines of maintenance~~nagement~~

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.

~~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.~~

[S4 Note: Definition does not appear to be correct or needed].

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8.4 Range of QoS requirements

[S4 Note: Table 4 specifies the performance level for the IMT-2000 environments in terms of BER while Tables 5-7 specify the acceptable information loss in terms of FER. It could be useful to use the same criteria, for example a Residual Bit Error Rate to link both tables]

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 Narrowband Speech~~voice~~	Two-way	4-13 kb/s	<150 msec preferred <400 msec limit	< 1 msec	< 3% FER
Audio	Conversational Wideband Speech	Two-way	10-64 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

[S4 Note: The data rate for wideband speech is based on existing wideband codecs (MPEG-4 for lowest rate, G.722 for highest rate)]

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8.5.4 Loss of interactivity due to delay in the speech path

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.

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 and in PLMN the one way delay may exceed 40 ms, due to the propagation and processing delay (see ITU-T Recommendation G.114).

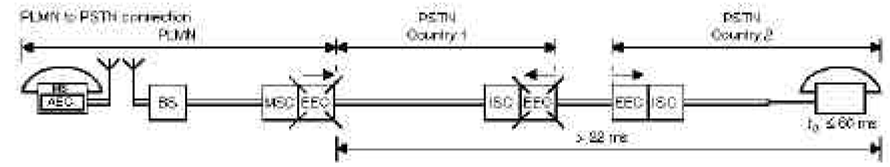
Even though a greater delay may occur in a satellite connection, delay shall be minimized in the wireless access to the network for the majority of calls, which use terrestrial connections.

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

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

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):

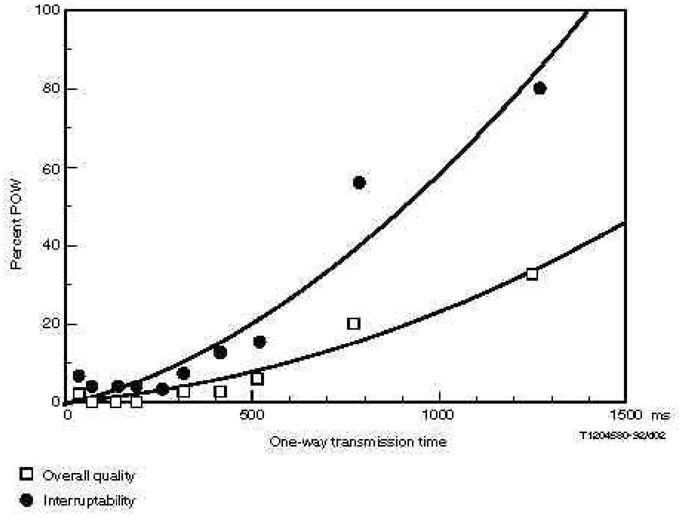


FIGURE B.2/G.114

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

…/…

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 (ex: Tandem Free Operation or Transcoder Free Operation).

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 terminal acoustics, impairments arising from the air interfaces (with typical interference and propagation conditions), transcoding, delay and echoes in the connection, etc.

8.5.9 ~~Handset~~Terminal acoustics

~~Handset~~Terminal 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.

~~8.5.13 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.~~

[Note S4: The combination of the codec bit rate and the robustness are the key design factor, not the Gross bit rate, especially for wideband systems. The reference to the improvement in system capacity is not relevant for the Gross speech bit rate]

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

The range of connection scenarios 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.

[S4 Note: It is unsure whether two way conversational tests can actually be handled/funded in practice]

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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.

[S4 Note: It is more usual to specify codec performances in reference to other well-known speech codecs without using MOS differences. It appears that this comment was already made by ITU-T SG16 to ITU-R TG 8/1].