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TRANSMISSION SYSTEMS AND MEDIA

GENERAL CHARACTERISTICS OF INTERNATIONAL TELEPHONE CONNECTIONS AND INTERNATIONAL TELEPHONE CIRCUITS

TRANSMISSION IMPAIRMENTS

ITU-T Recommendation G.113

(Previously "CCITT Recommendation")

FOREWORD

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

The World Telecommunication Standardization Conference (WTSC), which meets every four years, establishes the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

The approval of Recommendations by the Members of the ITU-T is covered by the procedure laid down in WTSC Resolution No. 1 (Helsinki, March 1-12, 1993).

ITU-T Recommendation G.113 was revised by ITU-T Study Group 12 (1993-1996) and was approved under the WTSC Resolution No. 1 procedure on the 6th of February 1996.

NOTE

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

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SUMMARY

This Recommendation was generated to provide planning guidance for designers of networks that would form part of an international telephone connection. For example, guidance on the maximum number of A/D conversions and the impairment impact of waveform type codecs, in the form of planning rules, was provided. This method has been referred to as the the Quantization Distortion Method. However, these planning rules did not adequately address the impairments of non-waveform encoders as well as a number of other impairments. This Recommendation has been reissued to make the document more useful by the retention of the Quantization Distortion Method and the incorporation of a new planning method, called the Equipment Impairment Factor Method. The Equipment Impairment Factor Method allows the impacts of: non-waveform codecs, non-optimum overall loudness rating, talker echo, and speech difficulty associated with long one-way delay to be evaluated.

TRANSMISSION IMPAIRMENTS

(revised in 1996)

1 Preamble

This Recommendation is intended to provide guidance to network and service planners who are concerned with overall transmission performance. Information is presented or referenced relating to transmission impairments found in analogue, analogue/digital, unintegrated network, integrated digital network and integrated services digital network connections.

Use of this information by national Administrations is dependent upon their network types and their national requirements.

The current regulatory operating environment in certain countries has made allowance for other networks to interconnect with the PSTN, e.g. Private Networks and Digital Cellular Networks, and for customers to provision their own terminal equipment. The information in this Recommendation will provide guidance for all parties that wish to operate in this changing environment.

2 Introduction

This Recommendation deals with impairments that affect a modern telephone connection with regards to the quality of speech communications, and to some extent, data transmission. In particular, the following parameters are considered in this Recommendation: attenuation distortion and loss, circuit noise, codec coding distortion, group delay distortion, one-way transmission time, talker echo, and some additional parameters of special importance for voiceband data. (Note that other transmission parameters also may influence the connection quality, however to a much lesser degree.)

The approach taken here is to list the individual transmission impairments, and provide guidance and/or references regarding the impact of these impairments. (These impairments should be considered individually or in combination, as needed by the planner.) In the case of digital processes, it may be useful to consider the accummulation of impairments. Accordingly, the methods in clauses 5 and 6 can be used as appropriate.

There are two provisionally recommended approaches which may be used by network planners and service planners. One approach, “The Quantization Distortion Method” is intended to assist in planning the placement/use of waveform-type codecs while the other approach, the “Equipment Impairment Factor Method” is intended to assist in planning the placement/use of non-waveform coders, but can also be used for waveform-type coders. (Additional information about a computational model for guidance in transmission planning can be found in draft Appendix I/G.101, COM 12-63). As these are two distinctly different methods of determining anticipated network connection quality, the user of this Recommendation is cautioned that the parameter values are not necessarily interchangeable between the two methods.

Networks are in various stages of evolution to digital. Thus, there will be instances where: connections will be routed utilizing all-digital components (end-to-end including the terminals); other connections will use all-digital network components and analogue access facilities; and still other connections will use portions of the network which are analogue while other network components may be digital. This Recommendation is intended to address each of these scenarios.

3 Definitions

For the purposes of this Recommendation, the following definitions apply.

3.1 analogue network: A network in which the access interface and all network elements are considered analogue.

3.2 **analogue/digital network**: A network in which the access interface is analogue and some network components are analogue while the remainder are digital.

3.3 **communication quality:** Quality as found in many mobile systems, characterized by good intelligibility, speaker identity maintained, but some loss in quality when directly compared to PSTN quality.

3.4 **equipment impairment factor**: A number allocated to a network element, in units of “eif”, that indicates the anticipated incremental level of impairment that would result when this element is inserted into a connection.

3.5 **equipment impairment factor unit**: The unit “eif” (Equipment Impairment Factor) is used to specify the impairment associated with a particular network element, e.g. transmission circuit or digital signal processing unit.

3.6 **expectation factor**: A normally positive quantity that represents the advantage of access that certain systems have over wirebound handset telephony.

3.7 **integrated digital network**: A network in which the access interface is analogue while the remainder of the network elements are digital.

3.8 **integrated services digital network**: A network in which the access interface and the network elements are all digital.

3.9 **PSTN quality**: Average quality of long distance Public Switched Telephone Network connections, i.e. good intelligibility, good speaker identification, naturalness, only minor disturbing impairments.

3.10 **quantization distortion unit**: A quantization distortion unit (qdu) was defined in 1982 as equivalent to the distortion that results from a single encoding and decoding by an average G.711 codec. Such a device has a Signal/Distortion ratio of 35 dB when measured according to Recommendation O.132.

3.11 **talker echo**: Echo produced by reflection near the listener’s end of a connection, and affecting the talker.

3.12 **total impairment value**: A numerical value, obtained by summing all of the equipment impairment factors of the end-to-end connection, which provides an indication of the expected quality of speech communication of the particular telephone connection. The Total Impairment Value consists of the sum of several Equipment Impairment Factors and is expressed in units of “eif”.

4 Guidelines regarding individual transmission parameter limits

4.1 Network performance objective for attenuation distortion

The resultant attenuation distortion on an international network connection is a function of the number of translations to and from voiceband that occur within the network. The objectives for the attenuation distortion of a maximum-length 4-wire chain are given in Recommendation G.132 for all-analogue and mixed analogue-digital equipment. All-digital connections, with analogue access interfaces, should meet the attenuation distortion requirements as given in the G.550-Series Recommendations (Transmission Characteristics of Digital Exchanges). On all-digital connections which utilize digital telephone sets and all-digital facilities, the attenuation response should meet the attenuation distortion requirements of Recommendation P.310 for narrow-band phones; or Recommendation P.311 for wideband handset telephones, or Recommendation P.314 wideband hands-free telephones. The evolution towards an all-digital network is resulting in a greater number of less complicated connections which will reduce overall attenuation and result in average and minimum loudness ratings that are advocated in Recommendation G.121 and will thereby ensure adequate transmission performance on international connections.

Should values of attenuation distortion or noise greatly different from those recommended by the ITU-T for systems and equipment be contemplated, then guidance concerning possible changes in transmission performance can be found in Recommendation P.11 and its annexes [1], with some indication of possible trade-offs between them.

4.2 Network performance objective for circuit noise

The ITU-T recommends that the network performance objective for the mean value, expressed in decibels and taken over a large number of worldwide connections (each including four international circuits), of the distribution of one-minute mean values of signal-independent noise power of the connections, should not exceed –43 dBm0p referred to the input of the first circuit in the chain of international circuits (see Recommendation G.143).

All-digital connections, with analogue access interfaces, will have an acceptable noise performance which is substantially independent of connection length [1] between the digital local exchange’s analogue interfaces. All-digital connections which utilize digital telephone sets will have an acceptable noise performance which is substantially independent of connection length [1] and which is controlled by the telephone sets [2].

4.3 Network performance objective for group delay distortion

The resultant group delay distortion on an international network connection is a function of the number of translations to voiceband that occur within the network. It is expected that a worldwide 4-wire chain of 12 analogue circuits (international plus national extensions) (see Recommendation G.132) and a worldwide chain of 4-wire circuits with 14 analogue to digital conversions will keep the group delay distortion within acceptable limits. The evolution towards an all-digital network is resulting in a greater number of less complicated connections which reduce overall group delay distortion and will thereby ensure adequate transmission performance on international connections.

4.4 Network performance objective for talker echo

With the increased use of digital technology in transmission and switching systems there is a trend towards lower loss and higher delays for connections. This makes talker echo effects more noticeable. Recommendation G.131 gives guidance in this matter.

4.5 One-way transmission time

Recommendation G.114 provides guidance in this matter.

4.6 Effect of random bit errors

As a general guideline, if the BER is $< 10^{-6}$ then voiceband services are not significantly impacted.

4.7 Effect of burst errors

Burst errors in a digital channel will affect voiceband services to varying degrees based on the length of the burst and the coding system used. At the present time the only meaningful guidance for speech quality in the presence of burst errors can be derived from subjective evaluations.

4.8 Effect of speech clipping

Speech clipping in DCME, PCME, or wireless accesses will affect speech quality to varying degrees based on the length of the clipped speech segments and the total per cent of time that clipping occurs. At the present time the only meaningful guidance for speech quality in the presence of speech clipping can be derived from subjective evaluations.

4.9 Transmission impairments due to digital processes

The incorporation of unintegrated digital processes in international telephone connections, particularly during the mixed analogue/digital and all-digital period, can result in an appreciable accumulation of transmission impairments. It is therefore necessary to ensure that this accumulation of impairments, due to digital processes, does not reach a point where it seriously degrades overall transmission quality. There are two provisionally recommended approaches which may be used by network planners and service planners. The following two clauses provide detail on these two methods which may be used to assess the impact of individual impairment processes.

5 The quantization distortion method

This method was the traditional method used for evaluating digital transmission impairments and is still useful for connections that do not include low-bit rate non-waveform coders. This method can be used in hybrid/mixed, and all digital networks.

5.1 General

The quantization distortion method is an approach which allocates a value of distortion to each digital process and which then allows the simple addition of these impairments to determine the overall anticipated connection quality.

This method is recommended for use where the digital processing is performed using coders that are waveform oriented. For example, the following could come under this category: A- and μ-Law encoders; 32 kbit/s coders designed according to Recommendation G.726; digital loss pads; PCM to ADPCM to PCM conversions; and μ-/A-Law converters. Examples of digital coders that do not lend themselves to this type of approach are as follows: LD-CELP, VSELP, RPE-LTP, and CELP+. For these latter coders and for the evaluation of connections using these coders, it is recommended that the Equipment Impairment Factor Method be used.

5.2 Discussion

The quantization distortion method is based on the principle that a distortion allocation for digital processes can be made by comparing the distortion of a process to that of a standard distortion unit. The standard distortion unit selected is called a quantization distortion unit (qdu) and is defined as being the level of distortion impairment that results from one ideal 8-bit G.711 PCM coding and decoding process. Specifically, the distortion allocation given to a digital process is given in units of qdus. Thus a unit assigned a value of 4 qdus is presumed to provide a level of impairment equivalent to four unintegrated 8-bit PCM processes in tandem.

Several methods of comparison are possible; these include objective measurements (or equivalent analysis), subjective tests, and data tests in which the effect on the bit error ratio at the output of a voiceband data modem receiver is used as a criterion.

At the present time no objective measurement capability exists which can produce results (e.g. SNR) that correlate closely with results obtained from subjective measurement of the effect of many of the digital processes now being studied on speech performance. Therefore, the number of units of quantization distortion for digital processes should, in general, be determined by subjective measurement methods, such as those found in Recommendation P.83. In some instances, the number of units of quantization distortion for a digital process can be determined without subjective measurement by decomposing a digital process into two or more parts and allocating to the parts suitable fractions of the total number of units assigned to the digital process. However, while this method may be considered an objective method for determining the qdu assignments for the parts, it uses as a starting point a subjectively determined value. Furthermore, except for relatively simple digital processes where the decomposition is uncomplicated, this method may not be reliable and should be used with care.

Planning rules should be applicable to all signals transmitted in the voice-frequency band. Therefore, in general, both speech quality and data performance must be considered. Speech quality should be evaluated by subjective tests and data performance should be evaluated by objective measurements which provide estimates of the expected bit error ratio and signaling performance. At present, however, because of the lack of an objective method for evaluating the effect of digital processes on voiceband data performance, the planning rule in this Recommendation is limited to voice connection planning purposes only. Clause 10 discusses some of the problems associated with developing a planning rule for connections carrying voiceband data and other non-speech signals. Such a rule would be based on a unit reflecting the contribution digital processes make to the impairment or impairments that affect voiceband data modems and/or signalling systems. Such a unit does not exist yet.

NOTE – qdu is defined in terms of quantizing distortion as present in PCM and other waveform coders and assumes that the quantizing distortion adds on a 15 $\log_{10}(n)$ law for n codec pairs in tandem. There is some evidence to suggest that while the 32 kbit/s ADPCM codec which complies with Recommendation G.726 exhibits the same distortion and additivity as PCM, the 16 kbit/s LD-CELP codec as tested and studied in 1991 exhibits additivity closer to 20 $\log_{10}(n)$. However, subjective tests carried out under the guidance of SQEG (Experts Group on Speech Quality) indicate that the 16 kbit/s codec pair closely tracks the subjective quality of the G.726 codec for up to four codecs in tandem. Beyond four in tandem, the 16 kbit/s codec performance decreases more rapidly than that of G.726 codec. Thus, it is proposed that the 16 kbit/s codec be treated the same as the G.726 codec determining network performance with the stipulation that no more than three 16 kbit/s codecs be allowed in the worldwide connection and noting that the LD-CELP distortion is not additive with the qdus of other codecs.

5.3 Threshold of acceptability

A threshold of acceptability of 14 qdus was selected for speech as this represents the subjective limit for overall signal to distortion ratio (*Red Book*, Fascicle III.1, Supplement 21). Similarly, a threshold of 14 qdus was selected as the threshold of acceptability for data. An average 8-bit PCM codec pair produces about 2 dB less quantizing distortion than the limits indicated in Recommendation G.712. This would correspond to a signal-to-distortion ratio of 35 dB for the sine-wave test method and approximately 36 dB for the noise test method. (Thus a total of fourteen 8-bit PCM processes in tandem, each of which just comply with the limits for signal-to-distortion ratio in Recommendation G.712, would provide unacceptable performance.) The same principle should be applied when proposing planning values of quantizing distortion units for other digital processes.

5.4 Limitations

The quantization distortion method assumes that it is permissible to simply add the units of quantizing distortion that have been assigned to the individual digital processes to determine the total or overall quantizing distortion, for telephone connections which incorporate unintegrated digital processes.

Conceptually the number of qdus assigned to a particular digital process should reflect the effect of only the quantization noise produced by the process on speech. In practice the qdus must be determined from subjective measurements of real or simulated processes, where subjects will be exposed to not only the quantization noise but other impairments produced by the digital process tested, including the departures from ideal frequency response in the anti-aliaising and reconstruction filters.

Therefore, the subjective test results will be biased by these other impairments if the levels of these other impairments differ to a greater or lesser extent from the levels produced by PCM coding/decoding (the reference). Such biases will cause the derived qdus not to be a true measure of the effect of quantization distortion. The qdu assignment will instead reflect the effect of all the impairments on speech quality. Thus, to reduce the chance for such a bias to occur when determining the qdu assignments for digital processes, it is important to design the subjective test so as to:

1) minimize the contributions of impairments other than quantization distortion to the subjective test results; or

2) equalize the levels of these other impairments in the test and reference conditions.

5.5 Quantization distortion allocations to digital processes

The units of quantizing distortion (qdu) tentatively assigned to a number of digital processes are given in Table 1. Background information on these assignments is given in Supplement Nos. 21 and 22, *Red Book*, Fascicles III.1 and III.2 (1985), respectively, and in the notes associated with Table 1.

5.6 Planning rule

From the point of view of quantizing distortion, for which qdus apply, it is recommended that no more than 14 units of quantizing distortion (qdu) should be introduced in an international telephone connection.

As a consequence of the relationships identified in 4.1, 4.2 and 4.3 above concerning quantizing distortion, attenuation distortion and group-delay distortion, it is possible to recommend a planning rule governing the incorporation of unintegrated digital processes in international telephone connections. This planning rule is in terms of units of quantization distortion (qdu) and the units of quantizing distortion allocated to specific digital processes are indicated in Table 1. The planning rule is as follows:

> The number of quantization distortion units in an international telephone connection should not exceed: 5 + 4 + 5 = 14 qdus.

Under the above rule, each of the two national portions of an international telephone connection are permitted to introduce up to a maximum of 5 qdus of transmission impairment and the international portion up to a maximum of 4 qdus.

NOTES

1 It is recognized that in the mixed analogue/digital period, it might for a time not be practical for some countries to limit their national contributions to a maximum of 5 qdus of transmission impairment. To accommodate such countries, a temporary relaxation of the provisional planning rule is being permitted. Through this relaxation, the national portion of an international telephone connection would be permitted to introduce up to 7 qdus of transmission impairment. Theoretically, this could result in international telephone connections with a total of 18 qdus of transmission impairment. Such connections would introduce an additional transmission penalty insofar as the voice telephone service is concerned. Administrations which find it indispensable to have a national allowance of more than 5 qdus (but no more than 7 qdus) should ensure that not more than a small percentage of traffic on national extensions exceeds 5 qdus.

2 It should be recognized that for economic reasons long international circuits employ DCME (Digital Circuit Multiplication Equipment), or other similar systems, which may result in quantization distortion that slightly exceeds the recommended limit of 4 qdus.

5.7 Limitations of the planning rule

It is assumed that for estimating the transmission impairment due to the presence of unintegrated digital processes in international telephone connections, the units of quantizing distortion can be added to determine the overall level of impairment.

For international telephone that include tandem digital processes in an all-digital environment, adding the individual units of quantizing distortion might not accurately reflect the accumulated quantizing distortion (and, consequently, the accumulated units of transmission impairment). This could be the case since the individual amounts of quantizing distortion power produced by the individual digital processes might not be uncorrelated and, therefore, the addition of individual units of quantizing distortion might, under some circumstances, indicate totals that could be different from those actually in effect. This is explained in some detail in Supplement No. 21, *Red Book*, Fascicle III.1.

Although the 5 + 4 + 5 = 14 qdus rule given in 5.6 might under some conditions provide only approximate results, the rule, nevertheless, is considered to be suitable for most planning purposes particularly in cases involving unintegrated digital processes. Examples of tandem digital processes which are explicitly taken into account in Table 1 are A-μ-A code conversion, μ-A-μ code conversion, and PCM-ADPCM-PCM conversion.

6 The equipment impairment factor method

As described in 4.9, this is the second method for assessing digital impairments.

The equipment impairment factor method is based on the principle that transmission impairments can be transformed into “psychological factors”; and that psychological factors on the psychological scale are additive.

The equipment impairment factor method allocates a value of distortion to each network element and then allows the simple addition of these impairments to determine the overall equipment impairments, Itot. The Expectation Factor A is then subtracted from this number to generate a Calculated Planning Impairment Factor, Icpif.

The equipment impairment factor method is recommended for use where the digital processing is performed using non-waveform coders. For example, the following coders come under this category: LD-CELP, VSELP, RPE-LTP, and CELP+.

NOTE – This method can also be used for waveform coders, as long as an equipment impairment factor has been allocated to the digital process or indirectly by using the qdus allocated.

The equipment impairment factor distortion allocation for digital processes can only be made by using subjective mean opinion score test results and use of the formulas developed and presented later in this Recommendation. Annex E discusses this methodology in detail. Therefore, the equipment impairment factor should, in general, be determined by subjective measurement methods, such as those found in Recommendation P.800 and P.830. Although planning rules should be applicable to all signals transmitted in the voice-frequency band, this particular method is only applicable to voice transmission.

Considerations concerning the combination of effects of the transmission impairments can be found in clause 7, while considerations concerning the expectation factor can be found in clause 8. Considerations concerning the Calculated Planning Impairment Factor can be found in clause 9.

TABLE 1/G.113

Planning values for quantizing distorsion
(Speech service only; see clause 10 for voiceband data considerations)
(see Notes 1, 11, and 12)

Digital process	Quantizing distortion units (qdu)	Notes
Processes involving A/D conversion		
8-bit PCM codec-pair (according to Recommendation G.711, A- or μ-law)	1	2, 3
7-bit PCM codec-pair (A- or μ-law)	3	3, 4, 5
Transmultiplexer pair based on 8-bit PCM, A- or μ-law (according to Recommendation G.792)	1	3
32 kbit/s ADPCM (with adaptive predictor) (combination of an 8-bit PCM codec pair and a PCM-ADPCM-PCM tandem conversion according to Recommendations G.721/G.726/G.727)	3,5	6
Purely digital processes		
Digital loss pad (8-bit PCM, A- or μ-law)	0,7	7
A/μ-law or μ/A-law converter (according to Recommendation G.711)	0,5	10
A/μ/A-law tandem conversion	0,5	
μ/A/μ-law tandem conversion	0,25	
PCM to ADPCM to PCM conversion (according to Recommendations G.721/G.726/G.727)	2,5	8, 9
8-7-8 bit transcoding (A- or μ-law)	3	9

NOTES

1 As a general remark, the number of units of quantizing distortion entered for the different digital processes is that value which has been derived at a mean Gaussian signal level of about –20 dBm0. The cases dealt with in Supplement No. 21 [3] are in accordance with this approach.

2 By definition.

3 For general planning purposes, half the value indicated may be assigned to either of the send or receive parts.

4 This system is not recommended by the ITU-T but is in use by some Administrations in their national networks.

5 The impairment indicated for this process is based on subjective tests.

6 Recommendations G.726 and G.727 perform equivalently at corresponding bit rates, including 24 and 40 kbit/s. However, qdu values cannot be assigned for 24 and 40 kbit/s operation, at this time. For evaluation of ADPCM codecs in the context of overall circuit quality, it appears the "Equipment Impairment Factor" method described in clause 6 gives a more accurate description of their subjective effects on speech quality.

7 The impairment indicated is about the same for all digital pad values in the range 1-8 dB. One exception is the 6 dB A-law pad which introduces negligible impairment for signals down to about –30 dBm0 and thus attracts 0 units for quantizing distortion.

8 The value of 2.5 units was derived by subtracting the value for an 8-bit PCM codec pair from the 3.5 qdus determined subjectively for the combination of an 8-bit PCM code pair and a PCM/ADPCM/PCM conversion. Multiple synchronous digital conversions, such as PCM/ADPCM, PCM/ADPCM/PCM, are assigned a value of 2.5 qdus. For evaluation of ADPCM codecs in the context of overall circuit quality, it appears that the "Equipment Impairment Factor" method described in clause 6 gives a more accurate description of their subjective effects on speech quality.

9 This process might be used in a digital speech interpolation system.

10 The qdu contributions made by coding law converters (e.g. μ-law to A-law) are assigned to the international part.

11 The qdu assignments to these digital processes reflect, to the extent possible, only the effect of quantization distortion on speech performance. Other impairments, such as circuit noise, echo and attenuation distortion also affect speech performance. The effect of these other impairments must therefore be taken into account in the planning process.

12 The qdu impairments in this table are derived under the assumption of negligible bit error.

7 Consideration of the combination effects of transmission impairments

The previous clause describes the equipment impairment factor method that is to be used when non-waveform digital processes, i.e. 16 kbit/s or lower encoders, occur in a connection. The equipment impairment factor method is also applicable for all types of networks, e.g. analogue, mixed analogue/digital, whether or not they contain low-bit rate non-waveform coders.

7.1 General

Clause 4 gives an overview of individual parameter limits for acceptable performance. However, there is also a need to consider the *combination effects* of those impairments that can occur simultaneously in a connection. The "worst case" where all impairments lie at their upper individually permissible limits would surely result in a rather bad transmission quality. On the other hand, there are also cases where one impairment masks the effect of another. Furthermore, it is desirable to be able to "trade" impairments so that a connection, where most impairments are quite small, could be allowed to include one larger impairment and still be considered sufficiently good.

There exist a number of computation models by which such an evaluation can be made, at least for some of the transmission parameters, see for instance Annex A/P.11 and Supplement No. 3 to the P-Series Recommendations [1]. The methodology described in what follows is a much simplified version of the provisional ETSI computation model, see Bibliography, which encompasses the relevant parameters dealt with in this Recommendation. (The simplified method should suffice to give adequate planning information in most cases. The complete ETSI model includes the effect of additional impairments as well as some mutual masking effects. The mathematical treatment is much more elaborate and presents predictions of users' opinions.)

In this context it is worth noticing the general fact that "the user's perception of quality of a product or service is determined by the degree to which the user's expectations are fulfilled or exceeded". Thus, for speech transmission quality, the transmission planner (at present) should consider two categories, namely:

1) PSTN quality – Average quality of long distance Public Switched Telephone Network connections, i.e. good intelligibility, good speaker identification, naturalness, only minor disturbing impairments.

2) Communication quality – Quality as found in many mobile systems, characterized by good intelligibility, speaker identity maintained, but some loss in quality when directly compared to PSTN quality.

For each category, the user may find an offered system performance quite satisfactory, i.e. having a "good quality" with regard to his needs and expectations. It is only in exceptional cases that it is relevant for the user to make relative quality comparisons between the two categories.

7.2 Impairment factors and the total impairment value

The total impairment value *Itot* is the sum of individual impairment factors.

$$Itot = Io + Iq + Idte + Idd + Ie \qquad (7.1)$$

Io represents impairments caused by non-optimum overall loudness rating OLR and/or high circuit noise.

Iq represents impairment caused by PCM type quantizing distortion.

NOTE 1 – *Io* and *Iq* are caused by impairments occurring simultaneously with speech.

Idte represents impairments caused by talker echo.

Idd speech communication difficulties caused by long one-way transmission times.

NOTE 2 – *Idte* and *Idd* are caused by impairments which appear delayed with regard to the voice signal.

Ie represents transmission impairments caused by special equipment in the connection, in particular non-waveform low-bit-rate codecs.

NOTE 3 – It is possible for the equipment impairment factors to be asymmetric and thus for *Itot* to also be asymmetric.

7.3 Planning values for impairment factors

7.3.1 The impairment factor *Io*

For normal ranges of Overall Loudness Ratings (OLRs) and circuit noise Nc dBm0p (at the 0 dBr reference point nearest the receiving end), the following relation holds approximately:

$$Io = Ilr(OLR) + In(Nc) \tag{7.2}$$

Ilr and *In* are given in Tables 2 and 3, respectively.

TABLE 2/G.113

Relationship between OLR and *IIr*

OLR (dB)	*Ilr* (eif)
5....10	0
15	7
20	14
25	21

TABLE 3/G.113

Relationship between Nc and *In*

Nc (dBm0p)	*In* (eif)
≤60	0
–60	4
–50	15
–40	30

7.3.2 The impairment factor *Iq*

Table 4 gives *Iq*, applicable for PCM processes, as a function of quantization distortion units qdu for normal, low values of circuit noise. (If the circuit noise is appreciably higher, a certain masking takes place so that the value of *Iq* becomes lower. However, this effect can in general be ignored.)

NOTE – In Table 4 only qdus from PCM processes are to be included. Impairments caused by ADPCM codecs should be considered in the form of Equipment Impairment Factors, see Table 7.

7.3.3 The impairment factor *Idte*

Talker echo can, to some extent, be somewhat masked by a high circuit noise or a very strong sidetone. However, for normal, modern connections these effects can be ignored. To evaluate *Idte,* the "1% curves" in Figure 2/G.131, for fully digital connections, are taken as a reference. For a given mean one-way delay T, the actual talker echo loudness rating TELR is compared with the value read from the curve, TELRc. The impairment factor *Idte* is then obtained from Table 5.

TABLE 4/G.113

Relationship between qdus and *Iq*

qdu	*Iq* (eif)
0	0
2	0
4	0
6	2
8	7
10	11
15	20
20	28

TABLE 5/G.113

Relationship between residual talker echo and *Idte*

(TELR – TELRc) (dB)	*Idte* (eif)
15	0
10	3
5	8
0	17
–5	30
–10	40
–15	50

NOTES

1 The “1% curves” in Figure 2/G.131 represent a fixed amount of impairment and compare closely with Table 5.

2 Echo cancellers will reduce residual talker echo and this must be taken into account.

7.3.4 The impairment factor *Idd*

The amount of impairment caused by a long one-way mean delay Ta ms depends to a large extent on the rate of interactivity of the communicating parties, see Recommendation G.114. The values in Table 6 represent a combination of the impairments resulting from “general quality” and “interruptability”.

Note that the values in Table 6 apply even if there is a very good talker echo cancellation. If this is less than perfect, impairment from each has to be considered also.

7.3.5 The impairment factor *Ie*

The impairment factor *Ie* applies for complex speech processing devices, in particular low-bit-rate codecs. Each type of codec is represented by a specific value *K* for its corresponding impairment. When tandeming several codecs, whether of the same or different types, the total equipment impairment factor is obtained as the sum of the individual *K*-values.

$$Ie = \Sigma\ Ki \qquad (7.3)$$

Table 7 gives the *K*-values for some low-bit-rate codecs.

TABLE 6/G.113

Relationship between one-way delay and *Idd*

Ta (ms)	*Idd* (eif)
150	0
200	3
250	10
300	15
400	25
500	30
600	35
800	40
> 800	40

TABLE 7/G.113

Planning values for equipment impairment factor contribution *Ie*

(Speech service only; see clause 10 for voiceband data considerations)

Codec type	Operating Rate (kbit/s)	K (eif) (Note)
APDCM (Rec. G.726, Rec. G.727)	40	2
	32	7
	24	25
	16	50
LD-CELP (Rec. G.728)	16	7
	12.8	20
VSELP (IS 54, USA)	8	20
RPE-LTP (GSM)	13	20
NOTE – $Ie = \Sigma Ki$.		

Annex E gives some information about how equipment impairment factors can be determined by means of subjective tests.

8 Considerations Concerning the Expectation Factor *A*

The Expectation Factor *A* represents an “advantage of access” that certain systems have over conventional PSTNs. The overall transmission quality as perceived by the user is influenced by the ease or difficulty to establish a connection. In certain cases, wireless systems have an advantage in that they allow spatial flexibility in the provision of service and as a result, the user may discount the subjective impairments resulting from the speech transmission effects associated with wireless systems. Examples are mobile telephony and multi-hop satellite connections to hard-to-reach regions. However, the expectation factor may be asymmetric. For example, for a call from a mobile subscriber to a PSTN subscriber, the PSTN subscriber may expect PSTN quality while the mobile subscriber may expect mobile quality.

The expectation factor *A* relates to the users' quality expectation with regard to the speech communication service utilized. It is the responsibility of the planner to assign a value for the Expectation Factor. Table 8 provides provisional maximum values of *A* for various access arrangements. (Note that users' expectations may change with time!) Caution is suggested in the use of the Expectation Factor as it is provisional and its applicability in all operating arrangements has not been validated.

TABLE 8/G.113

Relationship between type of access and Expectation Factor *A*

Communication system example	*A Max* (eif) (Note 1)
Conventional (wirebound)	0
Private Networks	(Note 2)
Mobility allowed within a building	5
Mobility allowed within a geographical area or within a vehicle	10
PSTN access from hard-to-reach locations, e.g. via multi-hop satellite connections	20

NOTES

1 The planner is responsible to assign an expectation factor planning value. It is provisionally recommended that the assigned value be not greater than the value shown in this table.

2 The need for an expectation value for private networks is an item for further study.

9 Considerations associated with the Calculated Planning Impairment Factor Icpif

The Calculated Planning Impairment Factor Icpif is obtained by use of the following equation:

$$Icpif = Itot - A \tag{9.1}$$

It should be noted that when the expectation value is zero then Icpif = Itot and for this case the performance becomes only a function of the equipment impairment factors. This is true for those situations where only conventional wirebound accesses are used.

Table 9 lists (provisional) Recommendations for limits for the total impairment value *Icpif* with regard to different quality levels.

TABLE 9/G.113

Quality levels as function of the total impairment value *Icpif*

Upper limit for *Icpif*	Speech communication quality
5	Very good
10	Good
20	Adequate
30	Limiting case
45	Exceptional limiting case
55	Customers likely to react strongly (complaints, change of network operator)

NOTE – *Itot*, in the equation *Icpif* = *Itot* – *A*, is very near in numerical value to the decrease in *R*-rating, caused by similar impairments, of the Bellcore Transmission Rating model, described in Supplement No. 3 to the P-Series Recommendations.

The following examples are presented to show how a planner may use the concept of *Icpif*.

Example 1

For this case it is assumed that mobility within a building is to be designed into the telecommunications system. If we assume that the system has the following impairment values:

$$Io = 0;\ Iq = 0;\ Idte = 0;\ Idd = 3;\ Ie = 7$$

then $Itot = 10$.

The planner may wish to assign the maximum expectation factor to this arrangement allowed by Table 8, i.e. 5.

The number for $Icpif = Itot - A = 10 - 5 = 5$.

If Table 9 is observed, then it would suggest that the performance of this system, when a call is placed between the user with mobility to a user connected by landline would be considered "very good" by the user with a mobile terminal and "good" by a wireline customer.

Example 2

For this case it is assumed that mobility within a geographical area is to be designed into the telecommunications system. If we assume that the system has the following impairment values:

$$Io = 7;\ Iq = 0;\ Idte = 0;\ Idd = 0;\ Ie = 20 \text{ (Digital mobile)}$$

then $Itot = 27$.

The planner may wish to assign the maximum expectation factor to this arrangement allowed by Table 8, i.e. 10.

The number for $Icpif = Itot - A = 27 - 10 = 17$.

If Table 9 is observed, then it would suggest that the performance of this system, when a call is placed between the user with geographic mobility to a user connected by landline would be considered "adequate" by the user with a mobile terminal and "limiting case" by a wireline customer.

Example 3

For this case it is assumed that the ability to reach a remote geographical area, via two satellite hops, is to be designed into the telecommunications system. If we assume that the system has the following impairment values:

$$Io = 11;\ Iq = 0;\ Idte = 0;\ Idd = 30;\ Ie = 7$$

then $Itot = 48$.

The planner may wish to assign the maximum expectation factor to this arrangement allowed by Table 8, i.e. 20.

The number for $Icpif = Itot - A = 48 - 20 = 28$.

If Table 9 is observed, then it would suggest that the performance of this system, when a call is placed between the user in a remote geographic area to a user connected by landline would be considered "limiting" by the user at the remote location and would be likely to get a strong reaction from a wireline customer unaware of the far end communication system.

10 Effect of transmission impairments on voiceband data performance

Just as speech quality is affected by the transmission impairments found on telephone connections, so too is voiceband data quality. Many different impairments are present on a connection; some are steady-state impairments (e.g. loss, noise, quantization distortion, phase jitter, harmonic and intermodulation distortions, envelope delay distortion, echo, and attenuation distortion) while others are transient (e.g. impulse noise, phase or gain hits, and dropouts) and may tend to occur infrequently. Both steady-state and transient impairments can affect speech and voiceband data. However, the transient impairments usually have a bigger impact on data than on speech. This is also true of some of the steady-state impairments, e.g. phase jitter and envelope delay distortion. Because of this, planning rules for connections carrying speech usually concentrate on controlling the steady-state impairments, and less attention is paid to the transient impairments. If new planning rules are to be created with the intent of controlling the buildup of the impairments that are important to voiceband data, then these new rules will have to treat the transient as well as the steady-state impairments.

The extent to which certain impairments affect voiceband data depend upon the modem speed, modulation use and other characteristics such as whether the modem contains an equalizer to correct for envelope delay distortion. Low-speed modems, operating at 1200 bit/s or less can usually tolerate a poorer SNR than higher speed modems. They also tend to be less sensitive to envelope delay distortion than the higher speed modems. Modems operating at 4800 bit/s and higher will usually contain an envelope delay distortion equalizer to minimize the effect of envelope delay distortion on the performance. Transients affect all modems, to a greater or lesser extent depending on many factors.

Two other factors influencing how impairments impact on voiceband data performance are:

a) whether error detection and/or correction techniques are employed; and

b) how the information to be sent is encoded.

If error correction is not used, then error causing impairments will cause errors in the output data. However, if error correction is used, then the impact of error causing impairments will only reduce the data throughput rate. Depending on how customer information is coded, errors can have more or less serious effects. For example, the loss of a letter in a word, because of a bit error in the 8 bits representing the letters of the alphabet, is probably less important than an error in the 8 bits used to convey information about the size, shape or location of a graphical symbol in an image.

Bit compression techniques such as ADPCM (according to Recommendation G.726) have a very significant effect on high speed (≥ 4800 bit/s) modem performance.

Annex C gives results of studies using 32 kbit/s ADPCM. These results demonstrate the need for using higher encoding rates, such as 40 kbit/s ADPCM, for 9.6 kbit/s voiceband data (VBD) transmission. Modern Digital Circuit Multiplication Equipment (DCME, see Recommendation G.763) uses signal classifiers to detect higher speed VBD and encode it using 40 kbit/s ADPCM.

From the point of view of developing a simple planning rule which can be used to assess the effects of digital processes on voiceband data performance, several points are important:

1) Impairments (especially transients) other than those customarily measured for speech performance are important for measuring voiceband data performance.

2) A simple measure of the steady-state impairments (e.g. signal-to-total noise ratio) may not prove to be a satisfactory basis for a voiceband data planning rule. A planning rule may have to take the transient impairments into account.

3) Modem type and speed must be taken into account. Thus, unlike the planning rules for speech, rules for voiceband data may turn out to be modem-specific.

4) The type of data service may influence the extent to which certain kinds of data errors and, thus, certain impairments are important. Therefore the planning rules may have to be service-specific.

5) Only an objective measurement method taking these first four points into account is likely to provide a successful basis for deriving useful planning rules.

6) Such a measurement method does not exist at present.

Therefore, until much more progress has been made in determining what impairments affect voiceband data performance, how to measure these impairments, what levels of these impairments are important, and how the differences in modem type, speed and other characteristics can be accounted for, this Recommendation must be limited in its application to speech services only.

References

[1] ITU-T Recommendation P.11 (1993), *Effect of transmission impairments*.

[2] ITU-T Recommendation P.31 (1993), *Transmission characteristics for digital telephones.*

[3] Supplement 21, *The Use of Quantization Distortion Units in the Planning of International Connections*, page 326, Fascicle III.1, *Red Book*, Geneva, 1985.

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ETSI draft prETR "VTQM-E", Version: 2.1, 1995: *Speech Communication Quality from Mouth to Ear of 3.1 kHz Handset Telephony across Networks.*

Annex A

Information for planning purposes concerning attenuation distortion and group-delay distortion introduced by circuits and exchanges in the switched telephone network

(This annex forms an integral part of this Recommendation)

In recognition that analogue carrier systems and channel banks have been replaced by their digital counterparts, the information of Annex A/G.113 (Melbourne Nov. 1988, *Blue Book*) has not been provided here. However, if this information is required for planning purposes, the reader is requested to refer to the annex in the *Blue Book*. However, for the information of the reader, it should be noted that G.234 8-channel terminating equipments and G.235 16-channel terminal equipments have been cancelled as they are considered obsolete.

Annex B

Effect of transmission impairments on voiceband data

(This annex forms an integral part of this Recommendation)

B.1 Introduction

The present transmission plan for international connections provides guidance for the control of transmission performance, primarily to permit satisfactory transmission of speech signals. The significant impairments and their effect on speech signals are described in Recommendation P.11. These impairments include loudness loss, circuit noise, sidetone loudness loss, room noise, attenuation distortion, talker echo, listener echo, quantizing distortion and phase jitter. Other Recommendations involving data performance on leased circuits include Recommendations H.12, M.1020 and M.1025.

The use of international connections for the transmission of non-speech signals such as voiceband data creates the need for increasing the scope of the transmission plan to include guidance on the control of additional impairments. The significant impairments for voiceband data include impulse noise, envelope delay distortion, phase jitter, non-linear distortion, tone-to-noise ratio, frequency shift, gain transients and phase transients. The following subclauses provide information on these impairments based on AT&T's experience. All the parameter values quoted are illustrative minimum end-to-end performance objectives of the *pre-divested* AT&T public switched network. Typical values obtained on the network are much better than the minimum objectives. These minimum values are considered to be consistent with satisfactory modem performance at speeds up to 4.8 kbit/s. More stringent minimum objectives are considered necessary for satisfactory performance at higher speeds such as 9.6 kbit/s. The parameter values shown are for illustration only and do not represent a proposed Recommendation.

B.2 Impulse noise

Impulse noise is defined as any excursion of the noise waveform on a channel which exceeds a specified level threshold. Impulse noise is evaluated on channels by counting the number of excursions during a predetermined time interval. In order to minimize contributions due to thermal noise, the minimum threshold is normally set 12 to 18 dB above the r.m.s. value of the noise. The impulse noise level is designated to be that threshold at which the average counting rate is equal to one per minute.

The measuring instruments used to count noise impulses may employ either electromechanical or electronic counters. In some sets, the maximum counting rate is controlled to be seven per second.

The contribution of impulse noise to error rate becomes significant when the noise peaks reach a level 3 to 12 dB below the r.m.s. data signal level depending upon: the type of modulation used by the data modems, the speed of transmission in bits per second, and the magnitudes of other transmission impairments on the channel. The minimum impulse noise objective is that no more than 15 counts in 15 minutes are to be tallied at a level above threshold which is 6 dB below the received data level. Control is exercised through engineering rules and limits on measured impulse noise levels.

Since most impulse noise originates as transients from the operation of relays and other switching equipment, engineering rules and mitigative measures are aimed at shielding low-level carrier signals from the radiation associated with these transients.

B.3 Envelope delay (group delay)

Envelope delay is defined as the derivative with respect to frequency of the phase characteristic of the channel. Measuring this derivative is impractical, so it is approximated by a difference measurement. There are numerous envelope delay measuring sets in use employing various frequency widths for this difference measurement. The AT&T standard is 166-2/3 Hz. In test results, these differences show up as varying resolution of ripples in the envelope delay characteristic. Narrow frequency widths yield higher resolution but reduced accuracy.

The frequency of minimum envelope delay in telecommunication channels is usually in the vicinity of 1800 Hz. Therefore, envelope delay measurements are usually normalized to zero at 1800 Hz. Departure from zero at other frequencies is referred to as envelope delay distortion. Envelope delay distortion gives rise to inter-symbol interference in data transmission which causes errors and increased sensitivity to background noise.

In the network, envelope delay is controlled primarily in the design of channel bank filters and other apparatus. Typical minimum objectives for envelope delay distortion are 800 μsec maximum in the band from 1004 to 2404 Hz and 2600 μsec maximum in the band 604 to 2804 Hz.

B.4 Phase jitter

Phase jitter is defined as unwanted angular modulation of a transmitted signal. Its most commonly observed property is that it perturbs the zero crossings of a signal. Since noise also perturbs the zero crossings of a signal, it usually causes readings on a phase jitter measuring set even though no incidental modulation may be present.

Phase jitter impairs data transmission by reducing data receiver margin to other impairments. Phase jitter is controlled by the design of transmission equipment. Although specific sources of phase jitter, such as primary carrier frequency supplies, have been located in the field, the corrective techniques have usually required design changes in specific equipment. The end-to-end minimum objective for phase jitter is 10 degrees peak-to-peak for the frequency band of 20 to 300 Hz and 15 degrees peak-to-peak for the band of 4 to 300 Hz.

B.5 Non-linear distortion

Non-linear elements in transmission equipment give rise to harmonic and intermodulation distortion which are more generally referred to as non-linear distortion. Non-linear distortion measurements are made usually in terms of intermodulation distortion measurements.

Non-linear distortion can be broadly defined as the generation of signal components from the transmitted signal that add to the transmitted signal usually in an undesired manner. The non-linear distortion of concern here is that found within an individual voice channel. It should not be confused with the intermodulation noise caused by non-linearities in the multiplex equipment and line amplifiers of a frequency division multiplex system. Although these non-linearities can contribute to the non-linear distortion at voice frequencies, their contribution is usually negligible.

Non-linear distortion is commonly measured and identified by the effect it has on certain signals. For example, if the signal is a tone having frequency A, the non-linear distortion appears as harmonics of the input, i.e. it appears as tones at $2A$, $3A$, etc. Since most of the distortion product energy usually occurs as the second and third harmonics, distortion is often quantified by measuring the power of each of these harmonics and is called second and third harmonic distortion. If the amount of non-linear distortion is measured by the power sum of all the harmonics, the result is called total harmonic distortion. These distortion powers are not meaningful unless the power of the wanted signal (the fundamental) is known, so measurements are usually referred to the power of the fundamental and termed second, third, or total harmonic distortion.

Historically, two different methods of measuring non-linear distortion on voiceband channels have been used: the single-tone method and the 4-tone method. However, the single-tone method is no longer used.

For the 4-tone method, four equal level tones are transmitted as two sets of tones at a composite signal power of data level (–13 dBm0). One set consists of tones at 856 and 863 Hz (a 7-Hz spacing). A second set uses frequencies of 1374 and 1385 Hz (an 11-Hz spacing). The frequency spacing within each set of tones is not critical but should be different for each set. Let these four tones be called A_1, A_2, B_1 and B_2. The second order products ($A + B$) fall at $A_1 + B_1$, $A_1 + B_2$, $A_2 + B_1$ and $A_2 + B_2$. If the spacing between A_1 and A_2 is the same as that between B_1 and B_2 then $A_1 + B_2 = A_2 + B_1$ and these two components will add on a voltage basis and give an erroneous reading.

The third order products ($2B – A$) fall at $2B_1 – A_1$, $2B_1 – A_2$, $2B_2 – A_1$, $2B_2 – A_2$, $B_1 + B_2 – A_1$ and $B_1 + B_2 – A_2$. The receiver uses 50-Hz wide filters to select the $A + B$, $B – A$, and $2B – A$ products. R_2 is the ratio of the power of the received composite fundamentals to the power average of the $A + B$ and $B – A$ products. R_3 is the ratio of received composite fundamentals to the $2B – A$ products.

An advantage of the 4-tone method, the method currently used in AT&T, is that the 4-tone test signal has an amplitude density function quite similar to that of a data signal. However, because of the relatively wide (50 Hz) passband of the receiver filters, the measurements with the 4-tone method are more affected by circuit noise.

The intermodulation products arising from non-linear distortion add to the wanted signal and interfere with it much as noise does. The intermodulation products are more damaging than noise, however, and the ratio of fundamental to second- or third-order products should be in the range of 25 to 38 dB, depending upon the type of data transmission, for satisfactory operation.

Non-linear distortion is controlled primarily in the design of equipment. However, such things as aging vacuum tubes in older equipment and poor alignment of PCM channel banks can cause this distortion to increase over its design limits. The overall customer-to-customer minimum objective for non-linear distortion using the 4-tone method of measurement is 27 dB minimum for R_2 and 32 dB minimum for R_3.

B.6 Tone-to-noise ratio

For voice transmission, the noise that is heard during the quiet intervals of speech is most important and this is what the standard message circuit noise measurement evaluates. For data transmission, the noise on the channel during active transmission and corresponding signal-to-noise ratio is important. In systems using compandors or quantizers, the noise increases during active transmission. In order to measure this noise, a –16, –13, or –10 dBm0 tone is transmitted from the far end of the channel under test and then filtered out ahead of the noise measuring set. The filter used to remove the tone is a narrow notch filter centered at the frequency of the tone. This type of measurement is also referred to as noise-with-tone. Test equipment is now available which uses 1004 Hz as the tone for this measurement.

Noise, of course, can cause errors in data transmission and a tone signal-to-noise ratio objective of at least 24 dB should be maintained for satisfactory performance. Noise is controlled in the design of transmission equipment, in the engineering of transmission systems (by such factors as repeater spacing), and in the maintenance of these systems.

B.7 Frequency shift

When a tone experiences a change in frequency as it is transmitted over a channel, the channel is said to have frequency shift or offset. Frequency shift can be measured by using frequency counters at both ends of a channel. When the input frequency differs from the output frequency, the difference is the frequency shift on the channel.

In modern telecommunication equipment, the frequency shift, if any at all, is usually on the order of 1 Hz or less. Some older carrier systems may have substantial amounts of offset, e.g. 15 to 20 Hz.

Frequency shift is important in systems which use narrow-band receiving filters such as telegraph multiplexers and remote meter reading equipment. When systems using these types of transmission experience frequency shift, the received signals fall outside the bandwidth of the filters. Frequency shift can occur on facilities which use single sideband suppressed carrier transmission. Within AT&T, frequency shift is controlled by means of the frequency synchronization network. The minimum objective for frequency shift is ± 5 Hz.

B.8 Gain and phase transients

Gain and phase changes that occur very rapidly may be encountered on telecommunication channels. Some of the more common causes of these phenomena are automatic switching to standby facilities or carrier supplies, patching out working facilities to perform routine maintenance, fades or path changes in microwave facilities, and noise transients coupled into carrier frequency sources. The channel gain and phase (or frequency) shift may return to its original value in a short time or remain at the new values indefinitely.

Gain changes are typically detected by changes in an automatic gain control circuit and phase changes by means of a phase locked loop. In order to provide protection against the test set detectors falsely operating on peaks of uncorrelated noise (impulse noise), a guard interval of 4 ms is designed into the gain or phase peak indicating instrument. Unfortunately, such a guard interval will also effectively make out true phase hits shorter than 4 ms that are not also accompanied by a peak amplitude excursion. The risk is considered justified at this time when one compares the known relative frequencies of occurrence of phase jumps to those of impulse noise.

Instrument used to measure gain and phase hits, as the rapid gain and phase changes are usually called, do so by monitoring the magnitude and phase of a sinusoidal tone. Hits are recorded and accumulated on counters with adjustable threshold levels. Gain hit counters typically accumulate events exceeding thresholds of 2, 3, 4 and 6 dB although they do not distinguish an increase from a decrease of magnitude. Similarly, phase hit counters accumulate changes at thresholds from 5 to 45 degrees in 5-degree steps. They respond to any hits equal to or in excess of the selected threshold. A switch which removes the impulse noise blanking feature under the user’s discretion may be desirable when impulse phase hit activity is suspected. The wide variety in hit waveforms, the effect of noise on measurements, and the allowable tolerances in thresholds and measurement circuitry, will generally contribute to different hit counts even on instruments of identical design. This variability will lead to some confusing among those testing with hit counters of different manufacturers. An alternative specification of the entire hit counting circuitry is under further investigation by the Institute of Electrical and Electronic Engineers.

Gain hits begin to cause errors in high-speed data transmission when their magnitude is on the order of 2 to 3 dB. Phase hits begin to cause errors when their magnitude is about 20 to 25 degrees. The end-to-end minimum objective for gain hits is to have no more than eight gain hits exceeding 3 dB in 15 minutes; the minimum objective for phase hits is to have no more than eight phase hits in 15 minutes at a threshold of 20 degrees. A dropout is defined as a decrease in level greater than or equal to 12 dB lasting at least 4 ms. The minimum objective for dropouts is to have no more than two dropouts per hour.

Annex C

Adaptive Differential Pulse Code Modulation (ADPCM) performance impact on voiceband data

(This annex forms an integral part of this Recommendation)

Abstract

This annex is mainly based on an AT&T Bell Laboratories paper given at the "IEEE Global Telecommunications Conference", 2-5 December 1985. It is provided to support this Recommendation as applied to voiceband data performance. The results indicate that, assigning a data qdu value to equipment using 32 kbit/s ADPCM (see Recommendation G.726) would be a difficult task since the performance is strongly dependent on the modem speed and type.

The annex reports on the results of a collection of empirical tests of high speed voiceband data modem error performance through channels containing asynchronously tandemed 32 kbit/s ADPCM (see Recommendation G.726) systems interspersed with simulated analogue impairments. A representative sample of 4.8 kbit/s transmission, and two 9.6 kbit/s devices were tested: an experimental design of Recommendation V.32 operating at 9.6 kbit/s for a full duplex modem, and another currently available 9.6 kbit/s product (similar to a V.29 modem). The results of the testing indicate that 4.8 kbit/s voiceband data transmission will perform adequately through asynchronous tandemed ADPCM systems, but that 9.6 kbit/s transmission is limited and, with certain modems, unacceptable under the same conditions.

C.1 Introduction

It is possible to use Adaptive Differential Pulse Code Modulation (ADPCM) at bit rates lower than 64 kbit/s per channel with, in many cases, less than proportional decrease in analogue transmission performance. Therefore, the use of a 32 kbit/s ADPCM algorithm on voice grade channels would essentially double the channel capacity of the associated facilities.

With the potential economic benefit due to increased capacity also comes the expectation of ensuing degradation of individual channel performance. Our results show that high speed voiceband data (e.g. 4.8 kbit/s or greater) would incur significant performance penalties with this new technology in place.

In this annex we report on the results of a collection of empirical tests of high speed voiceband data modem error performance through channels containing concatenated 32 kbit/s ADPCM (see Recommendation G.726) systems [1] interspersed with simulated analogue impairments. The channel configurations are designed to be representative of actual topologies possible on the public switched network with ADPCM systems in place. Asynchronously tandemed[1) ADPCM hardware contained in these test channels range in number from zero to seven while the interspersed analogue impairments are obtained by allocating parameters from impairment distributions measured in the end office connections study (EOCS) [2], loop studying 1970 [3], and 1980 loop surveys. We also tested performance using connections with asynchronously tandemed 64 kbit/s PCM systems, implemented in D4 channel banks, to compare with ADPCM configurations that showed particularly poor performance, so that it could be determined whether the ADPCM algorithm or simply the PCM coding was at root.

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1) Asynchronous tandeming takes place when a previously ADPCM coded signal is decoded to its analogue version and then recoded in a subsequent ADPCM system.

Modems used for the testing were of the high speed type. We tested a representative sample of 4.8 kbit/s transmission (V.29 type), and two 9.6 kbit/s modems: an experimental design of the V.32 modem standard for a full duplex modem, and another currently available device (V.29 type). All of these devices are 2-2 wire modems which are, or will be, marketed for use on the public switched network.

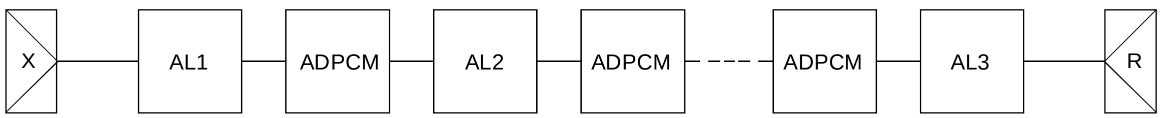
The results of our testing indicate that 4.8 kbit/s voiceband data transmission will perform adequately through multiple asynchronous tandeming of ADPCM systems, but that 9.6 kbit/s transmission is limited and, with certain modems, unacceptable under the same configurations.

C.2 Test condition architecture

It is known that ADPCM algorithm precision is to a great extent dependent on the nature of the signal which is to be encoded and transmitted. Signals with little or no stochastic components, such as pure tones, traverse these systems very well, with little or no distortion. On the other hand, high speed voiceband data signals which inherently have a large stochastic component and substantial bandwidth are significantly affected by ADPCM coding. Due to this, our test condition architecture examines these high speed modem types. We have furthermore tried to efficiently limit the quantity of testing required by using a universal architecture template for all our studies.

C.2.1 4.8 kbit/s half-duplex

Figure C.1 shows the test configuration architecture for 4.8 kbit/s half-duplex testing. The configuration is shown terminated on both ends with modems. The sequence of additional apparatus on the chart begins from the left with simulated analogue impairments (AL1) representative of analogue loop and Access Trunk (AT). Then the long haul segment consists of an ADPCM system, one 500-mile equivalent L-carrier analogue link (AL2) followed by from 1 to 6 ADPCM's respectively. This structure is representative of an interexchange portion consisting of multiple links and models the segment as if all analogue impairments occur early in the segment. Although this placement of the analogue impairments is somewhat conservative, it is counterbalanced by the fact that the impairments are those of a single L-carrier link and is a good approximation of reality given the constraint of using a single impairment simulator for the long haul part. Finally egress to the receiver proceeds through another analogue impairment simulator (AL3) representative of analogue trunk and loop. Interspersing analogue impairments with ADPCMs in this manner for the connection is more representative of actual network topologies and applications than simply lumping all analogue impairments in one place.



Modem	Access	LH			Egress	Modem
	Loop & AT	ADPCM	1 LMX	n (ADPCM)	AT & Loop	# tests
4.8 kbit/s	85	None	85	None	85	1
	None	ADPCM	None	1-6 ADPCM	None	6
	None	PCM	None	1-6 PCM	None	6 AN
	μ, 85	ADPCM	μ, 85	1-6 ADPCM	μ, 85	48
	μ, 85	PCM	μ, 85	1-6 PCM	μ, 85	48 AN

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FIGURE C.1/G.113

Test condition architecture for 4.8 kbit/s modem

It is clearly necessary to determine, for this configuration, the type and actual values of the analogue impairments to be dialed into simulators AL1, AL2 and AL3. Using a network performance modelling tool, the results of the End Office Connections Study (EOCS), and the assumption that high speed data customers connect to the network via data jacks, we derived the end-to-end mean (M) and 85th percent conditions of the major subset of impairments for switched network channels. Note that although we refer to the channel with each impairment at the 85 percent level as the 85th percentile channel, in fact it is somewhat worse because all impairments at 85% in one channel simultaneously would actually appear less than 15% of the time. Nevertheless, we then allocated these end-to-end values to the analogue impairment simulators. The results of this allocation, the impairment types, and the end-to-end values are shown in Table C.1. The values designated are allocated from the end-to-end mean (M), while the values designated "85" are allocated from the 85% end-to-end impairment values. The discussion of Figure C.1 can now be completed by describing the various values of analogue impairments as well as type and number of digital equipment present. The first configuration shows no ADPCMs but contains the allocated impairments from the 85th percent channel. Next, for additional reference, we tested six channels containing from 2 to 7 ADPCMs only, with no analogue impairments. Another six channels were to be tested as necessary with only PCM devices asynchronously tandemed, if and only if the previous corresponding ADPCM tests showed poor performance. Finally, the important tests with both analogue impairments allocated to the simulators from the mean (μ) and 85th percent channel with from 2 to 7 ADPCMs (or PCMs as necessary) were performed.

C.2.2 9.6 kbit/s full and half-duplex

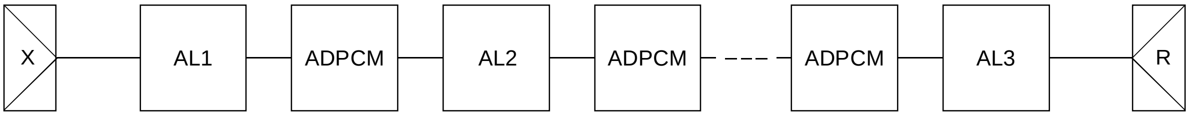
Here the test configuration architecture template is shown with a chart in Figure C.2. An experimental implementation of the V.32 modem standard for a 9.6 kbit/s full-duplex modem was tested under identical values of analogue impairments as those used for the 4.8 kbit/s modem. Although the channel segments have the same representation in the template, we only tested from 1 to 3 ADPCMs in the long haul segment. The simulated full-duplex operation was tested with the opposite channel excited with data, a signal-to-listener echo ratio of 12 dB, and a listener echo delay of 25 ms, in line with tests previously reported [4]. For these tests, Table C.1 again has the relevant values for the analogue impairment simulators.

Also shown are three tests of another 9.6 kbit/s half-duplex modem with ADPCMs only. This modem is specifically designed for use on the public switched network and represents expected performance of the most currently available 9.6 kbit/s technology.

TABLE C.1/G.113

EOCS derived test conditions

	AL1	AL2	AL3	E-E
Impairment	μ/85	μ/85	μ/85	M/85
Loss (dB)	11.0/11.4	1.1/1.7	11.0/11.4	23.0/24.5
C-notch noise (dBmC)	32.0/35.6	37.5/38.5	24.0/27.6	29.4/31.0
Slope (dB)	1.5/3.0	0.0/0.2	1.5/3.0	2.9/6.1
Env. delay distortion (μs)	226/388	632/755	226/388	1084/1535
2nd intermod. (dB)	66.0/50.2	58.4/53.8	66.0/50.2	52.7/46.3
3rd intermod. (dB)	74.0/53.0	56.9/50.3	74.0/53.0	51.7/44.3
Phase jitter (p-p)	0.5/0.7	1.9/3.7	0.5/0.7	3.5/5.1
Level (dBm)				–27.0/28.5
S/N (dB)				31.6/28.5



Modem	Access	LH			Egress	Modem
	Loop & AT	ADPCM	1 LMX	n (ADPCM)	AT & Loop	# tests
V.32	85	None	85	None	85	1
9.6 kbit/s	None	ADPCM	None	1-3 ADPCM	None	3
V.32	None	PCM	None	1-3 PCM	None	3 AN
	μ, 85	ADPCM	μ, 85	1-3 ADPCM	μ, 85	24
	μ, 85	PCM	μ, 85	1-3 PCM	μ, 85	24 AN

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FIGURE C.2/G.113

Test condition architecture for 9.6 kbit/s modems

C.2.3 4.8 kbit/s ADPCM performance

For 4.8 kbit/s transmission, the salient results are shown in Figure C.3. We have plotted four curves on the axes: two 1000-bit Block Error Rates (BLER) and two bit error rates (BER), one each for the mean and 85% EOCS channels. The abscissa counts the number of asynchronously tandemed ADPCMs in the connection. Due to the architecture of the tests these are enumerated as $1 + n$. The "1" represents the ADPCM between AL1 and AL2 while n is the number of ADPCM systems between AL2 and AL3.

We see clearly from the graphs that all the error performance measures degrade as the number of asynchronously tandemed ADPCMs increases, and that performance on the 85% channel, containing worse values of analogue impairments, is inferior to the mean channel results. We assume an acceptance limit for modem accuracy behaviour of a BER < 10^{-5} on 85% of channels and a BLER < 10^{-2} on 85% channels. Hence, if we focus on the 85% channel from EOCS, we see that 4.8 kbit/s performance will be at acceptable limits if the number of ADPCMs is between 4 and 5 for BLER and between 3 and 4 for BER. More recent results imply that for some modems the BER criteria is marginal with 3 in tandem and only 2 would be acceptable. We know of course that the BER criterion is stricter than the BLER limit because bit errors represent a greater burst phenomenon which is to a large extent ameliorated by the use of block transmission implemented with an error detection/correction protocol. Nevertheless, we tested and we present both results because customer data communication applications will dictate which measure is more relevant.

C.2.4 V.32 modem-ADPCM performance

The outcome of tests on the experimental testbed representing a 9.6 kbit/s device conform to Recommendation V.32 is shown in Figure C.4. Note that we have again plotted four performance curves. As before, performance of the 85th percent channel is inferior to that of the mean channel. If we now focus on the 85th percent channel BLER, we see that the acceptable performance limit occurs between 2 and 3 asynchronously tandemed ADPCMs, while for BER the number is somewhere between 0 and 1. Which performance measure is appropriate depends on customer application. We are here observing that a larger stochastic component of the data signal implies poorer error performance of the modem. In this case the use of 9.6 kbit/s shows a definite degradation in performance over the same topology with 4.8 kbit/s devices.

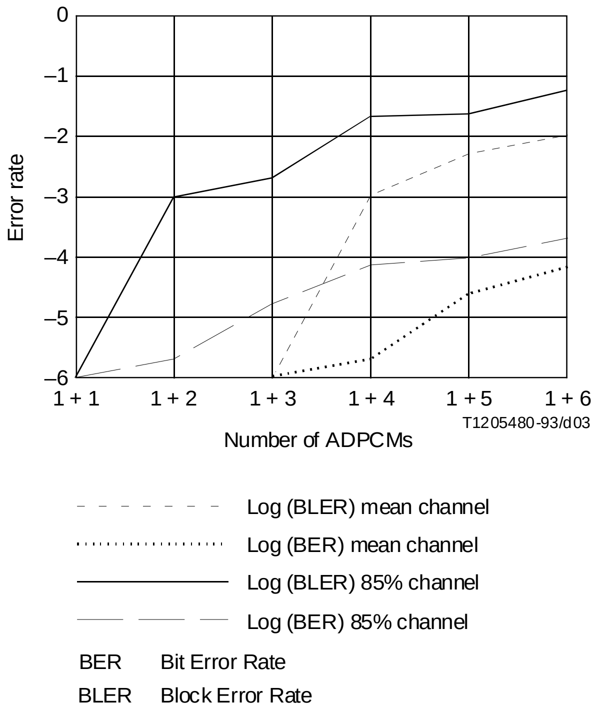

FIGURE C.3/G.113

ADPCM performance (mean and 85% channels) with a 4.8 kbit/s modem

It is also interesting to see if changing the position of segments with poorer impairment values effects modem performance. Figure C.5 shows a graph of three BLER curves for V.32 modems where we have taken the allocated 85th percent segment first on access, then on the long-haul part, and finally on the egress of the test channel, the other segments being at the allocated mean values of impairments. First, note that these curves fall between the full 85th percent channel and the mean channel in performance. Next, note that there does appear to be a mild dependence on the location of the more severe impairment values. Worse impairments close to the transmitter appear to have a more destructive effective on modem BLER performance than if they appear closer to the receiver. This means that analogue impairments on access are probably more significant in affecting modem error rates than those in the long-haul network or egress. The observed effect is mild, however, probably because the impairment values of the allocated 85th percent segments are really not much poorer than those for the allocated mean segments.

C.2.5 9.6 kbit/s – ADPCM performance

As a final test of modem performance, we have subjected another 9.6 kbit/s device, utilizing more traditional technology, to a sequence of asynchronously tandemed ADPCMs. This modem is a 2-wire device advertised by the vendor for use on the public switched network at signalling rates to 9.6 kbit/s. We have tested the device performance with no analogue impairments at all in the test channel. During the course of the empirical determination, it was discovered that the modem start sequence and the ADPCM algorithm interacted to prevent commencement of communication between transmitter and receiver. It was therefore necessary to test by allowing modem training to occur on an ordinary PCM channel after which ADPCMs were cut in to observe performance. Similar availability problems would also probably occur for any speed modem whose start-up training sequence is similar to that of this 9.6 kbit/s product.

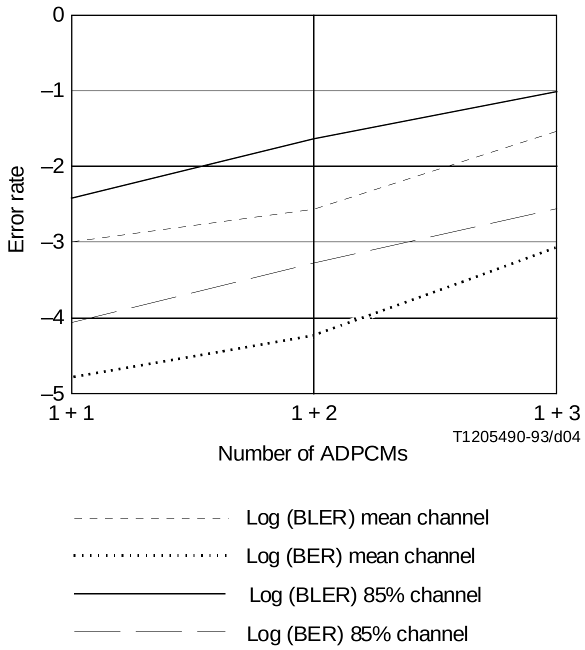

FIGURE C.4/G.113

ADPCM performance (mean and 85% channels) with a V.32 modem

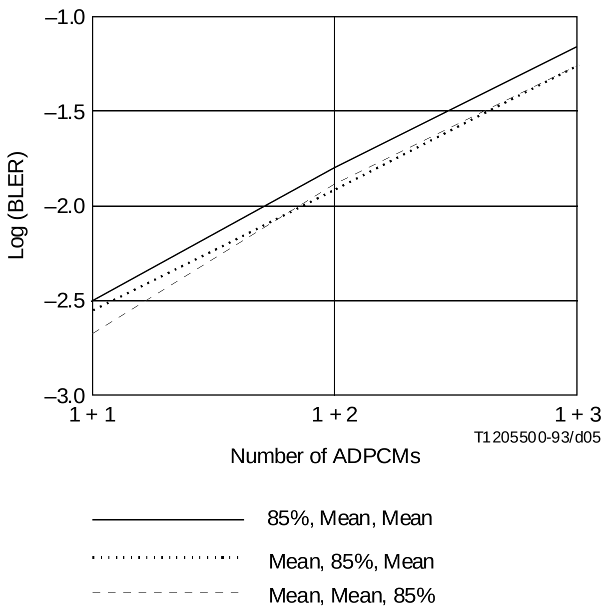

FIGURE C.5/G.113

ADPCM performance (impairment position study) with a V.32 modem

Figure C.6 shows the performance results for this modem. Without analogue impairments the number of ADPCMs may simply be enumerated sequentially. The BLER outcome indicates that between 0 and 1 ADPCM encoding is all that can meet our performance criterion. For BER it appears, again by our normal criterion, that ADPCM is incompatible with proper operation of the modem. Since it is expected that many modem vendors will, or have already, announced high speed 2-wire devices for use on the public switched network, the presence of ADPCM on these channels is likely to cause performance problems for those devices which are similar to the one tested for training, modulation and detection.

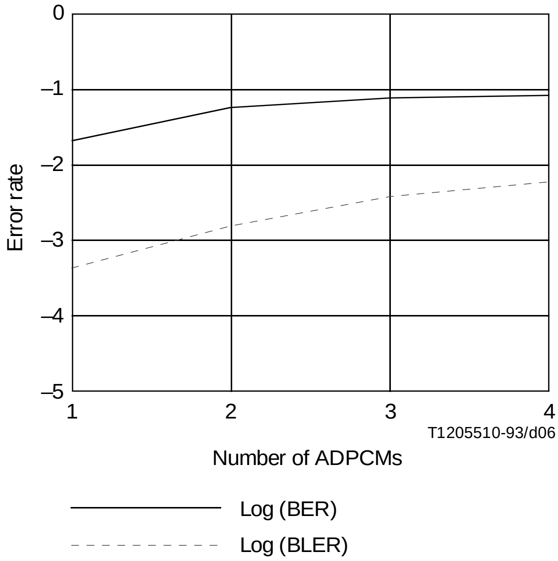

FIGURE C.6/G.113

9.6 kbit/s ADPCM performance (no analogue impairments) with a 9.6 kbit/s modem

C.3 Conclusions

In this annex we have reported on the architecture, laboratory apparatus, and results of a collection of empirical tests of high speed voiceband data modem error performance through channels containing asynchronously tandemed ADPCM systems interspersed with simulated analogue impairments. The results are compactly displayed in Table C.2 which shows that communication at 4.8 kbit/s may proceed through more asynchronous tandemed ADPCMs than in the case of using 9.6 kbit/s devices. Furthermore, communication at 9.6 kbit/s can be unacceptable when a BER criterion is applied, but sometimes acceptable when a BLER criterion is applicable. Clearly the appropriate criterion depends on the data communication user’s application.

TABLE C.2/G.113

Number of allowed ADPCMs on EOCS 85% channel

Modem	BER = 10^{-5}	BLER = 10^{-2}
4.8 kbit/s (V.29)	3/4[a)]	4/5
V.32	0/1	2/3
9.6 kbit/s	0	0/1
[a)] More recent results imply the range is 2/4.		

Bibliography

[1] Draft-Proposed American National Standard: 32 kbit/s ADPCM Algorithm and Line Format, Committee T1, Subcommittee T1Y1, *Document No. T1Y1, LB 85-01*, 28 March 1985.

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[3] MANHIRE (L. M.): Physical and Transmission Characteristics of Customer Loop Plant, *BSTJ*, 57, 1/70.

[4] KALB (M.), MORTON (C. H.), SHYNK (JU. J.): DATACAL – A Voiceband Data Communication Connection Performance Model, *Proc. of the Second International Network Planning Symposium*, *University of Sussex*, Brighton, UK, 21-25 March 1983.

Annex D

Compatibility of speech coding algorithms and voiceband data

(This annex forms an integral part of this Recommendation)

As described in Annexes B and C, digital speech coders have an impact on voiceband data. The following list provides some approximate information concerning the capability of various speech coding algorithms to support voiceband data signals.

1) The 16 kbit/s LD-CELP (see Recommendation G.728) algorithm supports voiceband data only up to 2400 bit/s.

2) The 32 kbit/s ADPCM (see Recommendation G.726) supports voiceband data up to 4800 bit/s.

3) The 40 kbit/s ADPCM (see Recommendation G.726) supports voiceband data up to 9600 bit/s, with 14 400 bit/s supported for only non-tandem connections.

This list will be expanded in the future. The list is intended for general guidance only. Annex C provides more detail regarding item 2), and additional data regarding item 3) can be found in [1].

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[1] DIMOLITSAS (S.), CORCORAN (F. L.), ONUFRY (M.), SUYDERHOUD (H. G.): Evaluation of ADPCM Coders for Digital Circuit Multiplication Equipment, *COMSAT Technical Review*, Vol. 17, No. 2, pp. 323-345, Fall 1987.

Annex E

Methodology for derivation of impairment factors

(This annex forms an integral part of this Recommendation)

The impairment factor may be determined once a Mean Opinion Score (MOS) has been generated. Guidance on how to perform subjective tests to generate an MOS may be found in various ITU-T Recommendations. Specifically, Recommendation P.80 "Methods for subjective determination of transmission quality" provides general information while Recommendation P.83 "Subjective performance assessment of telephone-band and wideband digital codecs" deals with digital codecs, and Recommendation P.84 "Subjective listening test method for evaluating digital circuit multiplication and packetized voice systems" deals with digital circuit multiplication equipment.

Once the MOS figure is available, it is possible to derive the impairment factor for a particular equipment, in particular for low-bit-rate encoders by adopting the following procedure.

It is advisable to evaluate codec tests from several different experiments. Doing this, it is necessary to normalize the MOS figures by the so called "Equivalent Q" method. This method is based on comparisons with a Modulated Noise Reference Unit (MNRU) created quantization distortion attenuation, characterized by the unit Q dB.

In the "Equivalent Q" method the relationship between the MNRU MOS and Q can be shown to have, with good accuracy, the following general relation:

$$MOS = 1 + A + B \cdot tanh\left[\frac{Q - Q_m}{C}\right] \qquad \text{(E.1)}$$

where A, B, C, and Q_m are constants, obtained by curve fitting to each set of measured reference MOS(Q) values, i.e. the subjective test team gives MOS figures for a number of quantizing distortions introduced by Q values of the MNRU. (Note that the constants are in general different for each reference curve.)

The MOS figure for the codec under test is then transformed into an "Equivalent Q"-value using the relationship:

$$Q = Q_m + \frac{C}{2} \cdot \ln \frac{B - A - 1 + MOS}{B + A + 1 - MOS} \qquad \text{(E.2)}$$

The corresponding *Ie* value is obtained by the relationships:

$$Ie = Iq = 15 \cdot \log\left[1 + 10^{(R_0 - 100)/15} \cdot 10^{(46 - G)/10}\right] \qquad \text{(E.3)}$$

where $R_0 = 95$, and:

$$G = 1.07 + 0.258 \cdot Q + 0.0602 \cdot Q^2 \qquad \text{(E.4)}$$

Each set of subjective tests will give a particular value of the impairment factor *Ie*. However, in general it has turned out by analysis of the results that, for transmission planning purposes, each type of codec can be associated with a specific impairment factor K, and that these factors can be added to obtain the total impairment factor when codecs are connected in tandem.

Bibliography

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