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SERIES G: TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

International telephone connections and circuits – General Recommendations on the transmission quality for an entire international telephone connection

Transmission impairments due to speech processing

ITU-T Recommendation G.113

(Formerly CCITT Recommendation)

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ITU-T Recommendation G.113

Transmission impairments due to speech processing

Summary

This Recommendation provides guidance regarding transmission impairments introduced by digital speech processing systems. The information provided is for use in conjunction with the transmission planning approach described in ITU-T G.107, G.108 and G.109. The Impairment Factor method, used by the E-model of ITU-T G.107, is now recommended. The earlier method that used Quantization Distortion Units is no longer recommended. Updated Impairment Factor values for various digital processing systems are provided in Appendix I. Appendix II contains guidance on how an Advantage Factor can be used to reflect the variation in user expectation of quality for different communications systems (e.g. mobile).

Source

ITU-T Recommendation G.113 was revised by ITU-T Study Group 12 (2001-2004) and approved under the WTSA Resolution 1 procedure on 23 February 2001.

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications. The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. 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 Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

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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As of the date of approval of this Recommendation, ITU had not received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementors are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database.

CONTENTS

ITU-T Recommendation G.113

Transmission impairments due to speech processing

1 Introduction

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 – D/A conversions and the impairment impact of waveform type codecs was provided in the form of planning rules. This method has formerly been referred to as 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 was revised to remove text that became obsolete due to major changes in the approach ITU-T recommends for transmission planning, e.g. in ITU-T G.107, G.108 and G.109. The ITU-T no longer recommends the Quantization Distortion Method for end-to-end transmission planning of speech performance. However, the concept of Quantization Distortion Units retains its validity for processes involving Pulse Code Modulation according to ITU-T G.711. Guidance is provided on the newly recommended principle for transmission planning, i.e. the Impairment Factor Method, which the algorithm of the E-Model (ITU-T G.107) is based upon. The Impairment Factor Method allows various transmission impairments to be evaluated during transmission planning.

The E-Model reflects the effects of different types of impairments on the end-to-end speech transmission performance. All perceptively different effects are transformed onto a so-called "psychological scale", i.e. the scale of impairment factors or the transmission rating scale. Impairment factors thus represent the (degrading) contribution of one instrumentally measurable attribute of the connection (e.g. attenuation, loss), or of a complete piece of equipment (e.g. a low-bit rate coding and decoding process) on the overall quality as experienced by the user. This contribution is called an "impairment factor" in a general sense, and "equipment impairment factor" when it is related to the (perceptively not coherent) degradations due to a specific piece of equipment. The E-Model combines the different impairment factors by applying the so-called "impairment factor principle", a basic principle used in transmission planning:

> "Transmission impairments can be transformed into so-called "psychological factors". These "psychological factors" are additive on a "psychological scale"."

The impairment factor principle thus allows various transmission impairments to be evaluated in the transmission planning phase.

This Recommendation is intended to provide guidance to network and service planners who are concerned with end-to-end speech transmission performance. Information relating to transmission impairments found in analogue, analogue/digital, un-integrated network, integrated digital network and integrated services digital network connections is presented in this Recommendation either directly or by reference to other documents.

The current regulatory operating environment in certain countries makes allowance for other networks – irrespective whether they are circuit switched or packet switched – to interconnect with the PSTN. Examples are Private Networks, Digital Cellular Networks and the Internet; furthermore, customers are entitled 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.

This Recommendation provides guidance concerning impairments that affect a modern telephone connection with regards to the end-to-end speech transmission quality.

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.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published.

[1] ITU-T G.100 (2001), *Definitions used in Recommendations on general characteristics of international telephone connections and circuits*.

[2] ITU-T G.107 (2000), *The E-Model, a computational model for use in transmission planning*.

[3] ITU-T G.108 (1999), *Application of the E-Model*: *A planning guide*.

[4] ITU-T G.109 (1999), *Definition of categories of speech transmission quality*.

[5] ITU-T G.168 (2000), *Digital network echo cancellers*.

[6] ITU-T G.711 (1988), *Pulse code modulation (PCM) of voice frequencies*.

[7] ITU-T G.712 (1996), *Transmission performance characteristics of pulse code modulation channels*.

[8] ITU-T O.132 (1988), *Quantizing distortion measuring equipment using a sinusoidal test signal*.

[9] ITU-T P.833 (2001), *Methodology for derivation of equipment impairment factors from subjective listening-only tests*.

[10] ITU-T Q.551 (1996), *Transmission characteristics of digital exchanges*.

[11] ITU-T Q.552 (1996), *Transmission characteristics at 2-wire analogue interfaces of digital exchanges*.

[12] ITU-T Q.553 (1996), *Transmission characteristics at 4-wire analogue interfaces of digital exchanges*.

[13] ITU-T Q.554 (1996), *Transmission characteristics at digital interfaces of digital exchanges*.

[14] ETSI ETR 250 (1996), *Transmission and Multiplexing (TM); Speech communication quality from mouth to ear for 3.1 kHz handset telephony across networks*.

3 Basic planning principle

The rapidly changing scenario in the field of multiple interconnected networks with increasing size and complexity – in combination with new technologies and the constraint for more economical solutions – requires more flexibility with respect to transmission planning.

In general, the quality of speech transmission via telephone channels is based on a subjective judgment by the users at both ends. Therefore, traditionally transmission was – in principle – derived from an end-to-end consideration in conjunction with a partitioning of all relevant parameters between different networks or parts of a network. As networks become more complex (e.g. North

American standards for internetworking are modernized and European countries are moving into liberalization) this approach is no longer applicable.

In conjunction with increasing liberalization in many countries, the responsibility for a sufficient speech transmission quality is now shifted to the operator of the terminating (e.g. private) network. However, planning of such networks with respect to speech transmission quality needs knowledge and experience in the field of transmission parameters and their influence to quality. In recognition thereof, the ITU-T recommends an appropriate planning method, the impairment factor method (see clause 5) in conjunction with the E-Model (see ITU-T G.107 [2]).

It should be noted that the preferable purpose of network planning is to control the summation of transmission impairments, caused by the different network elements in all possible configurations. It is not the task of network planning to limit the transmission impairment of a specific network element. Unless indicated otherwise, it is assumed that transmission, switching and terminal elements in general are designed to meet all relevant requirements as given in ITU-T Recommendations and in international or national standards applicable for this type of element.

In auditory tests – using human subjects – estimates of the integral quality, thus covering different quality dimensions, are often expressed in terms of MOS, %GoB or %PoW. During transmission planning, however, it is not practical to perform subjective tests. Therefore, a method must be provided which enables the planner to combine by calculation all existing transmission impairments in the given connection to a total value of impairment. This calculation must be performed by using an algorithm based on subjective testing. In telephone connections consisting of a variety of network elements, different transmission parameters may also simultaneously contribute to the total impairment. Therefore, the planning method used must also incorporate combination effects. For all configurations, the planning of speech transmission quality should be based on an end-to-end consideration rather than on a specification of individual objective parameter limits.

For the calculation of the different impairment values, particularly if the combined effect from the presence of more than one parameter needs to be considered, computational models are used for planning purposes. Several such "Rating-Models" have been developed and were contained and described in former ITU-T publications, which are no longer recommended for application and which today have bibliographic status, only. Today, ITU-T G.107 [2] gives the algorithm for the so-called E-Model as the common ITU-T Transmission Rating Model.

Transmission planning based on the E-Model – as recommended – provides a prediction of the expected quality – as perceived by the user – for an investigated connection. Based on an end-to-end assessment for each transmission parameter (including the type and number of low bit rate codecs) impairment values are derived. This model accounts for low bit rate coding devices as well as for impairments introduced by standard PCM coders and for impairments not directly related to digital processing (e.g. environmental noise).

The basic planning principle – as recommended – deviates from previous planning methods for network interconnection scenarios. End-to-end speech transmission quality is now expressed in terms of the E-Model Rating R, as a result of calculations with the E-Model. The E-Model Rating R can be transformed into other quality measures, which have been used in transmission planning before, such as Mean Opinion Score (MOS), Percentage Good or Better (%GoB) or Percentage Poor or Worse (%PoW), according to Annex B/G.107 [2].

4 The impairment factor method

According to the impairment factor method, the fundamental principle of the E-Model is based on a concept given in the description of the OPINE model [see Bibliography, ITU-T P-series Supplement 3]:

> "Transmission impairments can be transformed into "psychological factors"; and psychological factors on the psychological scale are additive."

The impairment factor method allocates a value of impairment to each parameter and then allows the simple addition of these impairments to determine the overall impairment. It should be noted that the impairment factor allocation can only be made by using subjective mean opinion score test results.

The result of any calculation with the E-Model is the E-Model Rating R, which combines all transmission parameters relevant for the considered connection. ITU-T G.107 provides details on how this E-Model Rating R is composed.

5 Equipment impairment factor, *Ie*

Modern coding laws, like those associated with low bit rate codecs as described in the G.720 series of ITU-T Recommendations or the GSM Standards, as well as ADPCM with different operating bit rates, will contribute with distortions resulting in a decrease of the perceived speech transmission quality. In contrary to the quantization distortion due to the standard 8-bit PCM coding (A-law or μ-law), these impairments cannot readily be quantified with a number of qdu (see clause 6). Following the considerations of clause 1, the impairments introduced by different types of low-bit rate codecs are expressed by an "Equipment Impairment Factor", *Ie*. This factor should ideally cover all perceptively very diverse effects (distortion, sound degradation, degradation of voice quality, etc.) which can be associated with the codec used in the connection, except those already covered in another way by the E-Model (e.g. overall attenuation, absolute delay). *Ie* values can be determined in auditory tests carried out according to a methodology given in ITU-T P.833 (2001).

For the purpose of end-to-end transmission planning with the E-Model, each codec can be assigned an equipment impairment factor which forms an input parameter to the E-Model. For asynchronously working tandems of different types of codecs, or of several codecs of the same type, the underlying additivity of the E-Model (cf. the "impairment factor principle" cited in clause 1) is assumed to be valid as well, i.e. the respective equipment impairment factors are added on the scale of the transmission rating R.

In Annex G of ETSI ETR 250 [14], comparisons have been made between subjective and predicted MOS values for various combinations of codecs. In general, the agreement is quite good, better than when the qdu method is applied.

For up-to-date information on values for the equipment impairment factor, Appendix I provides guidance; Appendix I is intended to be updated regularly.

6 Quantization distortion unit (qdu)

The qdu concept is not applicable for low bit rate codecs.

A Quantization Distortion Unit (qdu) was defined in 1982 as equivalent to the distortion that results from a single encoding (A/D) and decoding (D/A) by an average G.711 codec. Such a device has a Signal/Distortion ratio of 35 dB when measured according to ITU-T O.132.

Conceptually the number of qdus assigned to a particular PCM process should reflect the effect of only the quantization noise produced by the PCM 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 to other impairments produced by the digital process tested, including the departures from ideal frequency response in the anti-aliaising and reconstruction filters.

The qdu was the traditional parameter used for evaluating digital transmission impairments and this parameter is still useful for the characterization of transmission, network and terminal elements that do incorporate pure PCM processes according to ITU-T G.711.

Formerly, the qdu was the basis for an end-to-end transmission planning of impairments due to digital processes, known as the "14 qdu rule"; this approach is no longer recommended by the

ITU-T. Nevertheless, today the qdu functions rather as an input parameter for E-Model calculations of end-to-end speech transmission performance.

The units of quantization distortion (qdu) assigned to a number of PCM processes are given in Table 1. Background information on these assignments is given in the Notes associated with Table 1.

Table 1/G.113 – Planning values for quantization distortion (see Notes 1, 7 and 8)

PCM process	Quantization distortion units (qdu)	Notes
8-bit PCM codec-pair (according to ITU-T G.711, A- or μ-law)	1	(2, 3)
Transmultiplexer pair based on 8-bit PCM, A- or μ-law (according to ITU-T G.792)	1	(3)
Digital loss pad (A- or μ-law)	0.7	(4)
A/μ-law or μ/A-law converter (according to ITU-T G.711)	0.5	(5)
A/μ/A-law tandem conversion	0.5	
μ/A/μ-law tandem conversion	0.25	
digital echo cancellers (ITU-T G.168)	0.7	(6)

NOTE 1 – As a general remark, the number of units of quantization distortion entered for the different digital processes is that value which has been derived at a mean Gaussian signal level of about –20 dBm0. (See ITU-T G-series Supplement 21 [2].)

NOTE 2 – By definition.

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

NOTE 4 – 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 quantization distortion.

NOTE 5 – The qdu contributions made by coding law converters (e.g. μ-law to A-law) are assigned to the international part.

NOTE 6 – The assignment of a specific value for the Equipment Impairment Factor *Ie* for digital echo cancellers is for further study.

NOTE 7 – To the extent possible the qdu assignments to these digital processes reflect the effect of quantization distortion on speech performance only. 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.

NOTE 8 – The qdu impairments in this table are derived under the assumption of negligible bit error.

It should be noted that low bit rate codecs and 32 kbit/s ADPCM should not be characterized by qdus but rather the Equipment Impairment Factor, *Ie*, see clause 5.

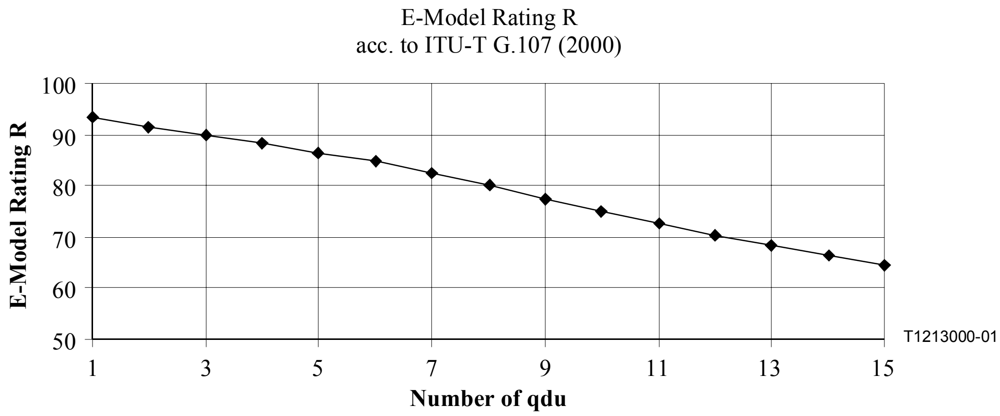

Figure 1/G.113 – Relation between the number of qdu and E-Model Rating R

With the increasing use of digital transmission and connection elements in private and public networks, the importance of quantization noise will decrease. However, quantization distortion can be ignored in planning only if fully bit-transparent routing can be assumed. Whenever mixed digital/analogue elements are present in a connection, the resulting number of qdu needs to be subject to planning. The influence of the number of qdus in a connection to the E-Model Rating R is shown in Figure 1.

The graph in Figure 1 has been derived from the E-Model with all other parameters at their default values. Because the E-Model calculations always include the number of qdu as an input parameter, it is recommended that the correct number of qdu of the connection be determined and used as an input to the model instead of the default value (1 qdu).

The parameter qdu in transmission planning applies not only to A/D-D/A conversions but also to other processes influencing the digital bit-stream. Those processes are, for example, the insertion of digital loss or gain, signal addition in conference circuits, use of digital echo cancellers, as can be seen in Table 1. For coding laws other than PCM (A-law or μ-law) – e.g. according to ITU-T G.726, G.727 or G.728 – the parameter qdu is, for transmission planning, replaced by the equipment impairment factor, *Ie*.

It is anticipated that as more practical experience is gained in the use of the impairment factor method, the qdu method will no longer be recommended for PCM.

Bibliography

[1] ITU-T P-series Supplement 3 (1993), *Models for predicting transmission quality from objective measurements*. (Withdrawn in 1997.)

[2] ITU-T G-series Supplement 21 (1984), *The use of quantization distortion units in the planning of international connections*. (Withdrawn in 1998.)

[3] ITU-T G-series Supplement 24 (1984), *Consideration concerning quantizing distortion units of some digital devices that process encoded signals*. (Withdrawn in 1998.)

ANNEX A

A comparison of the qdu approach versus the *Ie*-value assignment for ADPCM (32 kbit/s)

Regarding ADPCM 32 kbit/s an interesting comparison can be made:

The qdu methodology is also implemented in the E-Model. Formerly, i.e. before the Impairment Factor Method approach was available, one ADPCM 32 kbit/s codec had been considered to have qdu = 3.5 whereas today a value of the Equipment Impairment Factor, *Ie* = 7 has been assigned to that codec.

Figure A.1 shows the resulting values in terms of the E-Model Rating R for up to four ADPCM 32 kbit/s codecs in tandem for the case that all other input values for the E-Model are default. Figure A.1 gives a comparison of the qdu approach versus the *Ie*-value approach for both, the initial algorithm of the E-Model as per ITU-T G.107 (12/1998) as well as for the enhanced version of the algorithm as per ITU-T G.107 (05/2000).

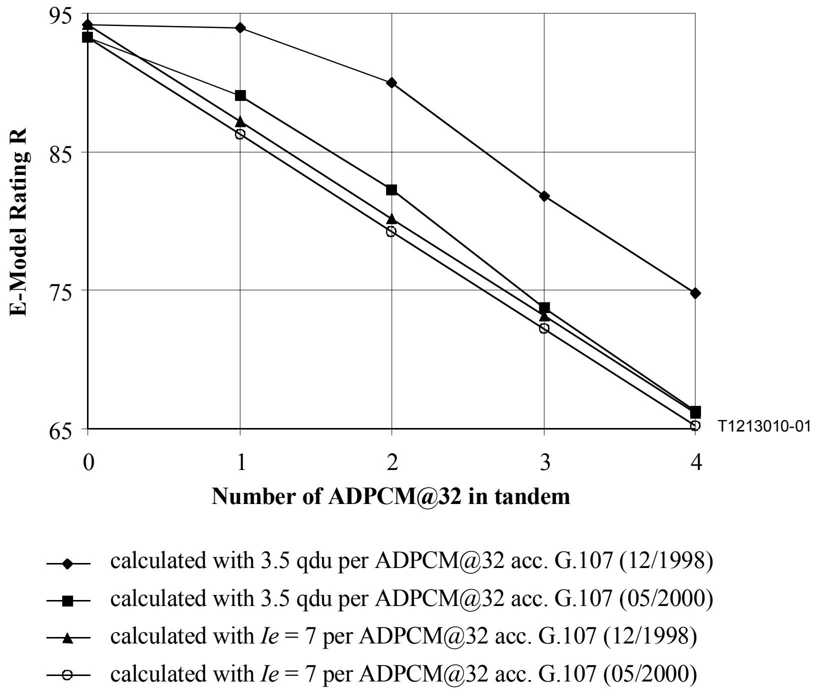

Figure A.1/G.113 – Comparison of E-Model Ratings R for ADPCM (32 kbit/s) tandemings using either *Ie* or qdu values

As can be seen, the enhancement of the E-Model algorithm has nearly aligned both approaches. Nevertheless, it should be noted – as stated before in this Recommendation – the application of the concept of qdu is no longer recommended for coding processes other than PCM according to ITU-T G.711.

APPENDIX I

Provisional planning values for the equipment impairment factor *Ie*

This appendix provides up-to-date information on available values of the Equipment Impairment Factor, *Ie*. It is intended to be updated regularly.

Table I.1 of *Ie* values refers to non-error conditions. For propagation errors and frame-erasures or packet loss, no definite values are available which would be valid for more than one codec or codec family. In order to help the transmission planner, examples of *Ie* values under conditions of packet loss are given in Tables I.2 and I.3, and for propagation error patterns EP1 and EP2 in Table I.4. These values are provisional only as they were determined in single or a few experiments. In Table I.5, a brief description of the codecs is provided for information.

Table I.1/G.113 – Provisional planning values for the equipment impairment factor *Ie*

Codec type	Reference	Operating rate kbit/s	*Ie* value
PCM (Note)	G.711	64	0
ADPCM	G.726, G.727	40	2
	G.721(1988), G.726, G.727	32	7
	G.726, G.727	24	25
	G.726, G.727	16	50
LD-CELP	G.728	16	7
		12.8	20
CS-ACELP	G.729	8	10
	G.729-A + VAD	8	11
VSELP	IS-54	8	20
ACELP	IS-641	7.4	10
QCELP	IS-96a	8	21
RCELP	IS-127	8	6
VSELP	Japanese PDC	6.7	24
RPE-LTP	GSM 06.10, Full-rate	13	20
VSELP	GSM 06.20, Half-rate	5.6	23
ACELP	GSM 06.60, Enhanced Full Rate	12.2	5
ACELP	G.723.1	5.3	19
MP-MLQ	G.723.1	6.3	15
NOTE – For every PCM process the number of Quantization Distortion Units, qdu (which should be determined according to Table 1) needs to be considered as a separate input parameter to the E-Model.			

Table I.2/G.113 – Provisional planning values for the equipment impairment factor *Ie* under conditions of random packet loss, codecs G.729-A + VAD, G.723.1-A + VAD and GSM EFR

% Packet Loss	G.729-A + VAD	G.723.1-A + VAD 6.3 kbit/s	GSM EFR
0	11	15	5
0.5	13	17	(Note 2)
1	15	19	16
1.5	17	22	(Note 2)
2	19	24	21
3	23	27	26
4	26	32	(Note 2)
5	(Note 2)	(Note 2)	33
8	36	41	(Note 2)
16	49	55	(Note 2)

NOTE 1 – Number of frames per packet:
- G.729-A + VAD: 2;
- G.723.1-A + VAD: 1;
- GSM EFR: 1.

NOTE 2 – No values have been available for these conditions.

Table I.3/G.113 – Provisional planning values for the equipment impairment factor *Ie* under conditions of packet loss, codecs G.711 without and with Packet Loss Concealment (PLC)

Packet Loss %	G.711 w/o PLC	G.711 w/ PLC	
		Random Packet Loss	Bursty Packet Loss
0	0	0	0
1	25	5	5
2	35	7	7
3	45	10	10
5	55	15	30
7	(Note 2)	20	35
10	(Note 2)	25	40
15	(Note 2)	35	45
20	(Note 2)	45	50

NOTE 1 – Speech packet length: 10 ms.

NOTE 2 – No values have been available for these conditions.

Table I.4/G.113 – Provisional planning values for the equipment impairment factor *Ie* under propagation error conditions, GSM codecs

Codec type	Error pattern	*Ie* Range
GSM-HR	EP1	25...32
	EP2	31...42
GSM-FR	EP1	32...39
	EP2	40...45
GSM-EFR	EP1	15...22
	EP2	26...35
NOTE 1 – The range given results from the difficulties in deriving exact impairment factor values for these conditions. NOTE 2 – EP1 is equivalent to 10 dB C/I, EP2 is equivalent to 7 dB C/I. C/I is the carrier to interference ratio.		

Table I.5/G.113 – Brief description of the low bit rate codecs

IS-54	First generation digital TDMA cellular system in North America utilizing Vector Sum Excited Linear Prediction (**VSELP**) coding at a net bit rate of 7.95 kbit/s (plus 5.05 kbit/s FEC).
IS-96a	First generation digital CDMA cellular system in North America utilizing Qualcomm Code-Excited Linear Prediction (**QCELP**) coding at a variable net bit rate of 8, 4, and 2 kbit/s.
IS-127	Second generation digital CDMA cellular system in North America utilizing Residual Code-Excited Linear Prediction (**RCELP**) coding at a variable net bit rate of 8, 4, and 2 kbit/s.
IS-641	Second generation digital TDMA cellular system in North America utilizing Algebraic Code-Excited Linear Prediction (**ACELP**) coding at a net bit rate of 7.4 kbit/s (plus 5.6 kbit/s FEC).
GSM-FR	First generation digital European Global System for Mobile Communications (**GSM**) cellular system utilizing Regular Pulse Excitation Long Term Prediction (**RPE-LTP**) coding at a net bit rate of 13 kbit/s (plus 9.8 kbit/s FEC). Defined in ETSI standard GSM 06.10.
GSM-HR	Half-rate version of the voice codec for the GSM system utilizing Vector Sum Excited Linear Prediction (**VSELP**) coding at a net bit rate of 5.6 kbit/s. Defined in ETSI Standard GSM 06.20.
GSM-EFR	Second generation speech codec of the digital European Global System for Mobile Communications (GSM) cellular system utilizing Algebraic Code-Excited Linear Prediction (**ACELP**) coding at a net bit rate of 12.2 kbit/s (plus 10.6 kbit/s FEC). Defined in ETSI standard GSM 06.60.
PDC	First generation digital Japanese Personal Digital Communication (PDC) system utilizing a Japanese version of Vector Sum Excited Linear Prediction (**JVSELP**) coding at a net bit rate of 6.7 kbit/s (plus 4.5 kbit/s FEC).
G.723.1	ITU-T standard for speech coding in PSTN videophones utilizing Algebraic Code-Excited Linear Prediction (**ACELP**) coding at 5.3 kbit/s and Multipulse Maximum Likelihood Quantization (**MP-MLQ**) at 6.3 kbit/s.
G.726	ITU-T standard for speech coding at 40, 32, 24, and 16 kbit/s utilizing Adaptive Differential Pulse Code Modulation (**ADPCM**).

Table I.5/G.113 – Brief description of the low bit rate codecs *(concluded)*

G.728	ITU-T standard for speech coding at 16 kbit/s utilizing Low-Delay Code-Excited Linear Prediction Coding (**LD-CELP**). This algorithm also has 12.8 and 9.6 kbit/s bit rate extensions.
G.729	ITU-T standard for speech coding at 8 kbit/s utilizing Conjugate Structure Algebraic Code-Excited Linear Prediction Coding (**CS-ACELP**).

APPENDIX II

Considerations concerning the Advantage Factor A

II.1 Introduction

This appendix provides regarding the Advantage Factor, A, most recent background material. The Advantage Factor does not really deal with codec or signal processing distortion but rather with the relative ponderation of functionality and transmission quality in users' expectation of services according to the type of users and the time.

The "Advantage Factor A" represents an "Advantage of Access", introduced into transmission planning for the first time via the E-Model (ITU-T G.107 [2] and ETSI ETR 250 [14]). This factor enables the planner to take into account the fact that customers may accept some decrease in quality for access advantage: e.g. mobility or connections into hard-to-reach regions. This value can be used directly in conjunction with all other impairment values and as an input parameter to the E-Model. Provisional A values are listed in Table 1/G.107 [2].

These values are provisional since they have not been confirmed by subjective investigations to date. Therefore, the Advantage Factor A should be used with care and with respect to the specific situation of the user. The use of the advantage factor in transmission planning of networks and the selected values are subject to the planner's decision; however, the values in Table 1/G.107 [2] should be considered as the maximum upper limit for A.

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.

NOTE – In other documents the term "Expectation Factor" has frequently been used to express the same issue which the "Advantage Factor, A" stands for.

II.2 Components of user expectation

The Advantage Factor, A, is a new feature of the E-Model with respect to its precursors. It should represent the so-called "advantage of access" that certain systems have over conventional wirebound communication systems. Until now, only provisional values are provided for cordless and mobile systems, and for multi-hop satellite connections to hard-to-reach regions.

With the introduction of VoIP, communication systems may more frequently be operated from a computer terminal instead of a conventional – handset, headset or hands-free operated – terminal. It is, therefore, useful to have a look at the so-called "expectation" in telephone calls originated from a computer terminal. The influencing dimensions are discussed on a theoretical basis, and experimental data comparing perceived "overall quality" are presented. The work is discussed in more detail in Möller [1] where this appendix has been extracted from.

The term "expectation" is a rather diffuse one, and it is not used in a unified way in telephony. Often it is used in the sense of "advantage of access" which a user of a special system or service can experience over a corresponding standard system. An example is a mobile communication system: the user can make calls from nearly all locations (provided that an adequate coverage of the geographical area is present), and he/she is reachable for urgent calls or in cases of emergency. This advantage is not linked to characteristics of the transmission, but to the special system/service. In turn, the user may tolerate some degradations inherent in the system which would not be observable for the equivalent standard system. In this case the term "expectation" is used as a measure of the trade-off between tolerable transmission impairments and the inherent advantage of the system.

In a more analytic picture there are at least three components which have an influence on the expectation defined above. These are:

- the relation to the user's general experience with telephone connections (memory);
- the expectation resulting from the price of the connection or service (especially for new services there is no general equivalence "high price = high speech transmission quality"); and
- the appropriateness of reaching specific goals, i.e. the call motivation; the appropriateness will be different for announcements or purely informative calls rather than, e.g. for private calls.

It becomes obvious that handling expectation as a single cause-and-effect parameter is too simplified. As the importance of expectation will be very high for new types of services or systems (when it is not yet stabilized), it is worth describing in some detail what happens when a new product is put on the market.

The development of expectation in a new product (an innovation) can be analysed with the help of the diffusion theory, which is generally accepted for describing consumer behaviour on the introduction of an innovation. Details of this theory can, e.g. be found in Wilkie, 1994 [2]. Over many studies it has been found that the number of actual users of an innovation develops in an S-shaped curve, see the first diagram of Figure II.1. The time it takes to diffuse a product depends on many factors, so no scaling can be given. Different people proceed through the adoption process at different points in time. According to the adoption time, users can be divided into 5 classes, see Figure II.1, second diagram:

1) *Innovators*: A very small group of persons who are very quick to purchase a new product or use a service. They are very willing to accept new technologies. Innovators have been found to be people with a higher income level, higher occupational status, and they are more socially mobile than other groups. Interestingly, they are not well integrated into social groups, so they do not rely on other's opinions as to whether products suit their own purposes.
2) *Early adaptors*: A somewhat larger group following the innovators. They are still quick to purchase a product or use a service, but are much more integrated in their respective social group and believe in group norms. This is an aspect which seems to be apparent, e.g. for the early adaptors of mobile telephones.
3) The *early majority*: These people enter the market next, but they are much less willing to take risks. About one third of all adaptors belong to this group.

4) The *late majority*: This group enters the market when "newness" declines, so they are not really purchasing a new product or using a new service. They are less influenced by their corresponding social group behaviour and can be more easily influenced by advertisements.

5) The *laggards*: They enter the market when an innovation is already well accepted.

Starting off from these results of diffusion theory, a trade-off between transmission quality and user demands can be seen as the origin of an advantage or disadvantage which the user of a special system or service experiences with respect to a normal, standard system or service. In the third diagram of Figure II.1, the potential transmission ratings for a new service are given as an example in order to represent a measure of the transmission quality. It can be seen that there is first a clear decrease in transmission quality for the new service in relation to conventional wirebound (ISDN) or mobile (GSM) systems.

With the introduction of higher quality for this new service the difference decreases. The drop in quality is expected to lead first to a higher demand for transmission quality. In this phase a user cannot directly experience an access advantage, as long as he/she does not experience an advantage in functionality. An increase in functionality is expected to start when the user does not only use the new system to replace a conventional one, but when he/she also starts to use it for other, different purposes or in different situations.

With the increase in functionality the demand for transmission quality decreases. During this phase the user may accept a lower transmission quality, because it is the functionality gain which is regarded as more important. When he/she has become used to the increased functionality, the demand for transmission quality may slowly increase and an overall balance will be reached. This hypothetical behaviour is depicted in the lower diagram of Figure II.1.

The importance a user attributes to functionality or transmission quality depends on the type of user. An innovator or early adaptor is more likely to appreciate the increase in functionality and will more easily accept the degradation in the transmission quality. A user of the early or late majority may be more affected by a bad transmission quality. On the other hand, these users will enter to the market later, when transmission quality has improved. Also functionality will increase when a larger number of users makes this demand.

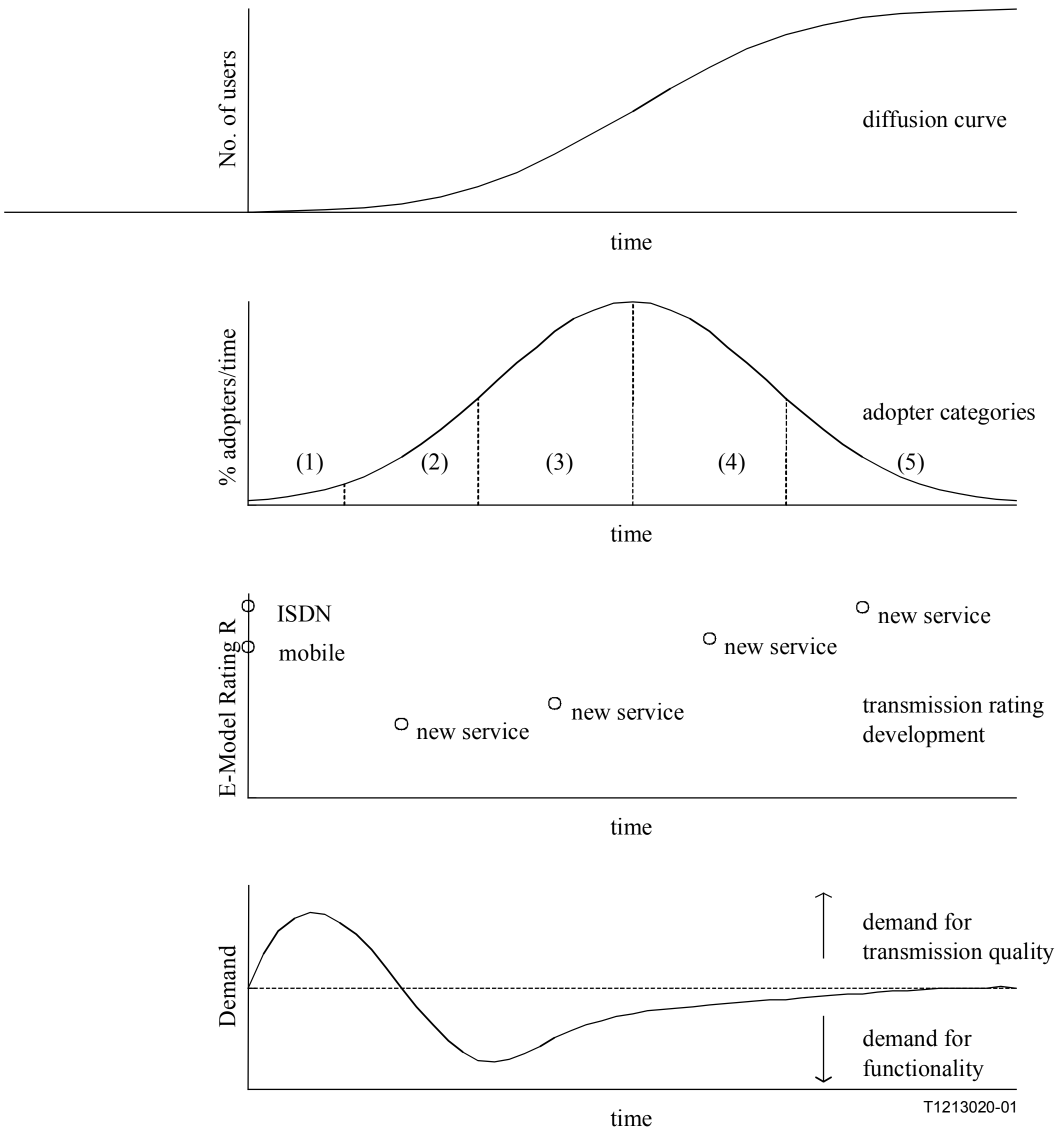

Figure II.1/G.113 – Diffusion, transmission quality and expectation for an innovation

Bibliography

[1] MÖLLER (S.): Assessment and Prediction of Speech Quality in Telecommunications, *Kluwer Academic Publishers*, USA-Boston, 1991.

[2] WILKIE (W.L.): Consumer Behaviour, *John Wiley & Sons Inc.*, USA-New York, NY, 1994.

APPENDIX III

Guidelines regarding individual transmission impairment parameters other than qdu and Equipment Impairment Factor *Ie*

This appendix provides information on impairments other than those caused by digital speech processing. It is provided for guidance, because transmission quality is also effected by these impairments.

III.1 Attenuation distortion

The attenuation distortion of an end-to-end telephone connection depends on the filtering with respect to the conversion from analogue to digital and vice versa as well as on the electro-acoustical properties of the terminal.

All-digital connections with analogue access interfaces should meet the attenuation distortion requirements as given in ITU-T G.712 or the Q.550 series of ITU-T Recommendations, respectively.

On all-digital connections which utilize digital telephone sets and all-digital facilities, the attenuation response should meet the attenuation distortion requirements of ITU-T P.310 for narrow-band handset telephones; or ITU-T P.311 for wideband handset telephones, or ITU-T P.341 for wideband hands-free telephones.

III.2 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. ITU-T G.712 provides guidance in this matter.

III.3 Talker echo

For modern network environments this is one of the key parameters, since 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. ITU-T G.131 gives guidance in this matter.

III.4 One-way transmission time

For modern network environments this is one of the key parameters, in which the contribution of speech processing is not negligible. ITU-T G.114 provides guidance in this matter.

III.5 Effect of random bit errors

As a general guideline, if the BER is $<10^{-6}$ then voiceband services are not significantly impacted, although for some coding schemes Appendix I may provide guidance in this matter.

III.6 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 transmission quality in the presence of burst errors can be derived from subjective evaluations; although for some coding schemes Appendix I may provide guidance in this matter.

III.7 Effect of syllable speech clipping

Syllable speech clipping (i.e. in the time domain) in DCME, PCME, or wireless accesses will affect speech transmission 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 transmission quality in the presence of speech clipping can be derived from subjective evaluations.

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