International Telecommunication Union

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

G.113

(11/2007)

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

ITU-T G-SERIES RECOMMENDATIONS

TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

INTERNATIONAL TELEPHONE CONNECTIONS AND CIRCUITS	G.100–G.199
General definitions	G.100–G.109
General Recommendations on the transmission quality for an entire international telephone connection	**G.110–G.119**
General characteristics of national systems forming part of international connections	G.120–G.129
General characteristics of the 4-wire chain formed by the international circuits and national extension circuits	G.130–G.139
General characteristics of the 4-wire chain of international circuits; international transit	G.140–G.149
General characteristics of international telephone circuits and national extension circuits	G.150–G.159
Apparatus associated with long-distance telephone circuits	G.160–G.169
Transmission plan aspects of special circuits and connections using the international telephone connection network	G.170–G.179
Protection and restoration of transmission systems	G.180–G.189
Software tools for transmission systems	G.190–G.199
GENERAL CHARACTERISTICS COMMON TO ALL ANALOGUE CARRIER-TRANSMISSION SYSTEMS	G.200–G.299
INDIVIDUAL CHARACTERISTICS OF INTERNATIONAL CARRIER TELEPHONE SYSTEMS ON METALLIC LINES	G.300–G.399
GENERAL CHARACTERISTICS OF INTERNATIONAL CARRIER TELEPHONE SYSTEMS ON RADIO-RELAY OR SATELLITE LINKS AND INTERCONNECTION WITH METALLIC LINES	G.400–G.449
COORDINATION OF RADIOTELEPHONY AND LINE TELEPHONY	G.450–G.499
TRANSMISSION MEDIA AND OPTICAL SYSTEMS CHARACTERISTICS	G.600–G.699
DIGITAL TERMINAL EQUIPMENTS	G.700–G.799
DIGITAL NETWORKS	G.800–G.899
DIGITAL SECTIONS AND DIGITAL LINE SYSTEM	G.900–G.999
QUALITY OF SERVICE AND PERFORMANCE – GENERIC AND USER-RELATED ASPECTS	G.1000–G.1999
TRANSMISSION MEDIA CHARACTERISTICS	G.6000–G.6999
DATA OVER TRANSPORT – GENERIC ASPECTS	G.7000–G.7999
PACKET OVER TRANSPORT ASPECTS	G.8000–G.8999
ACCESS NETWORKS	G.9000–G.9999

For further details, please refer to the list of ITU-T Recommendations.

ITU-T Recommendation G.113

Transmission impairments due to speech processing

Summary

ITU-T Recommendation G.113 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 Recommendations G.107, G.108 and G.109. The impairment factor method, used by the E-model of ITU-T Recommendation 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). Appendix III provides guidelines regarding individual transmission impairment parameters other than *qdu* and equipment impairment factor, and Appendix IV gives provisional planning values for the wideband equipment impairment factor.

Source

ITU-T Recommendation G.113 was approved on 13 November 2007 by ITU-T Study Group 12 (2005-2008) under the ITU-T Recommendation A.8 procedure.

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications, information and communication technologies (ICTs). 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.

Compliance with this Recommendation is voluntary. However, the Recommendation may contain certain mandatory provisions (to ensure e.g., interoperability or applicability) and compliance with the Recommendation is achieved when all of these mandatory provisions are met. The words "shall" or some other obligatory language such as "must" and the negative equivalents are used to express requirements. The use of such words does not suggest that compliance with the Recommendation is required of any party.

INTELLECTUAL PROPERTY RIGHTS

ITU draws attention to the possibility that the practice or implementation of this Recommendation may involve the use of a claimed Intellectual Property Right. ITU takes no position concerning the evidence, validity or applicability of claimed Intellectual Property Rights, whether asserted by ITU members or others outside of the Recommendation development process.

As of the date of approval of this Recommendation, ITU had received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementers are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database at http://www.itu.int/ITU-T/ipr/.

CONTENTS

ITU-T Recommendation G.113

Transmission impairments due to speech processing

1 Scope

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], [ITU-T G.108] and [ITU-T 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.

The E-model reflects the combined perceptual effects of different types of impairments on the end-to-end speech transmission performance by using impairment factors. Impairment factors 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 degradation is called an "impairment factor" in a general sense, and "equipment impairment factor" when it is related to the degradations due to a specific piece of equipment.

The E-model is highly useful because it combines the effects of different impairment factors.

Network and service planners who are concerned with end-to-end speech transmission performance can use impairment factors with the E-model to assess the effects of introducing speech processing technologies.

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. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

[ITU-T G.107] ITU-T Recommendation G.107 (2005), *The E-model, a computational model for use in transmission planning*.

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

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

[ITU-T G.168] ITU-T Recommendation G.168 (2007), *Digital network echo cancellers*.

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

[ITU-T G.712] ITU-T Recommendation G.712 (2001), *Transmission performance characteristics of pulse code modulation channels*.

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

[ITU-T P.10] ITU-T Recommendation P.10/G.100 (2006), *Vocabulary for performance and quality of service*.

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

[ITU-T Q.551] ITU-T Recommendation Q.551 (2002), *Transmission characteristics of digital exchanges*.

[ITU-T Q.552] ITU-T Recommendation Q.552 (2001), *Transmission characteristics at 2-wire analogue interfaces of digital exchanges*.

[ITU-T Q.553] ITU-T Recommendation Q.553 (2001), *Transmission characteristics at 4-wire analogue interfaces of digital exchanges*.

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

[ETSI ETR 250] ETSI ETR 250 (1996), *Transmission and Multiplexing (TM); Speech communication quality from mouth to ear for 3,1 kHz handset telephony across networks*.

[ETSI TR 102 356] ETSI TR 102 356 V1.1.1 (2004), *Speech Processing, Transmission and Quality Aspects (STQ); Application and enhancements of the E-model (ETR 250); Overview of available documentation and ongoing work.*

3 Definitions

None.

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

GoB Good or Better

MOS Mean Opinion Score

PoW Poor or Worse

qdu Quantization distortion unit

5 Conventions

None.

6 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., the proliferation of digital mobile access networks and networks that transport voice calls as packets) this approach is no longer applicable.

Accordingly, the recommended planning approach is to use the impairment factor method (see clause 8) in conjunction with the E-model (see [ITU-T G.107]).

A fundamental 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] 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 of [ITU-T G.107].

7 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 [b-ITU-T P-series Supp.3], which states that 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.

8 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 contrast 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 9). 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].

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 (see 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], 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 an overview of documentation related to the E-model, see [ETSI TR 102 356].

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

9 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 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 9-1. Background information on these assignments is given in the notes associated with Table 9-1.

Table 9-1 – 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 Rec. 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 [b-ITU-T G-series Supp. 21].)

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 8.

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 9-1.

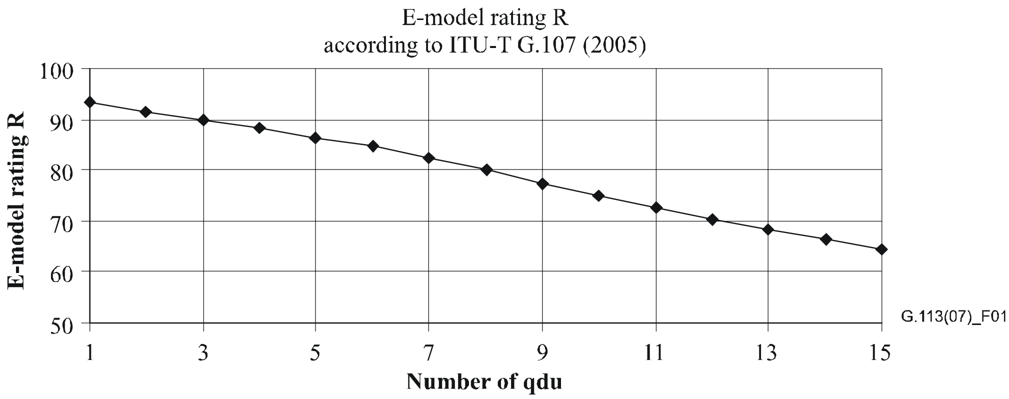

Figure 9-1 – Relationship between the number of qdu and E-model rating R

The graph in Figure 9-1 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 9-1. For coding laws other than PCM (A-law or μ-law) – e.g., according to ITU-T Recs 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.

Appendix I

Provisional planning values for the equipment impairment factor, *Ie,* and packet-loss robustness factor, *Bpl*

(This appendix does not form an integral part of this Recommendation)

This appendix provides up-to-date information on available values of the equipment impairment factor, *Ie*, and packet-loss robustness factor, *Bpl*, for codecs or codec families. It is intended to be updated regularly.

Table I.1 provides provisional planning values for the equipment impairment factor, *Ie*. These *Ie* values refer to non-error conditions without propagation errors, frame-erasures or packet loss. Subsequent tables deal with error and various loss conditions.

Table I.1 – Provisional planning values for the equipment impairment factor, *Ie*

Codec type	Reference	Operating rate [kbit/s]	*Ie* value
PCM (see Note)	G.711	64	0
ADPCM	G.726, G.727	40	2
	G.721, 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 9-1) needs to be considered as a separate input parameter to the E-model.			

Table I.2 – 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.3 provides provisional planning values for the equipment impairment factor, *Ie*, and for packet-loss robustness factor, *Bpl*, as specified in clause 3.5 of [ITU-T G.107].

Table I.3 – Provisional planning values for the equipment impairment factor, *Ie*, and for packet-loss robustness factor, *Bpl*

Codec	Packet size	PLC type	*Ie*	*Bpl*
G.723.1+VAD	30 ms	Native	15	16.1
G.729A+VAD	20 ms (2 frames)	Native	11	19.0
GSM-EFR	20 ms	Native	5	10.0
G.711	10 ms	None	0	4.3
G.711	10 ms	Appendix I of [ITU-T G.711]	0	25.1

The method to take account of packet loss is derived from conditions with random packet loss. This is the case where the probability of loss of a packet is independent of the probability of loss of any other packet. In systems with a jitter buffer (such as most VoIP applications), the applicable packet loss is measured at the output of the jitter buffer. ITU-T Rec. G.1020 proposes a de-jitter buffer emulation that may be used to estimate the packet discard to be expected at the output of a de-jitter buffer in case of network jitter. In general, users should be aware that:

- the assumption of packet loss independence is unsatisfactory for many real networks, for example, VoIP and mobile networks;
- jitter buffer implementations vary considerably, both between manufacturers and even between software revisions for a given device;
- proprietary codec implementations may have different robustness to packet loss from the values tabulated in the main body of this Recommendation.

However, for some coders, the subjective impairment due to burst packet loss can be reflected using the so-called burst ratio, *BurstR*, which partly captures the "burstiness" of a specific loss distribution (see equation 3-29 of [ITU-T G.107]).

$$BurstR = \frac{\text{Average length of observed bursts in an arrival sequence}}{\text{Average length of bursts expected for the network under "random" loss}}$$

when packet loss is random *BurstR* = 1; and when packet loss is bursty *BurstR* > 1.

Until further validation is provided, it is recommended that for bursty packet loss the *BurstR* approach of the E-model (equation 3-29 of [ITU-T G.107]) should be employed only for codecs with an efficient codec-state based PLC (i.e., with a packet loss robustness factor *Bpl* ≥ 16).

Two additional burst-loss cases with *Bpl* < 16 can currently be handled by using the provisional planning values of Table I.4, when loss ratios are low, i.e., for packet loss percentages *Ppl* ≤ 2%. The provided *Bpl* values are to be used with the packet loss model as specified in [ITU-T G.107], artificially setting *BurstR* = 1 in equation 3-29 of [ITU-T G.107] as in case of random packet loss.

Table I.4 – Provisional planning values for codecs under burst packet loss (to be applied for *Ppl* ≤ 2% with the random packet loss model, see [ITU-T G.107])

Codec	Packet size	PLC type	*BurstR*	*Ie*	*Bpl*
G.729E	20 ms	Native	4 (Note)	4	8.1
G.711	20 ms	Repeat 1/silence	4 (Note)	0	4.8
NOTE – Set *BurstR* = 1 in equation 3-29 of [ITU-T G.107].					

It has to be noted that the above *Ie* and *Bpl* values have been derived for a very specific sample of burst packet loss, and may not reflect the impairment due to burst packet loss in general.

Table I.5 provides examples for bursty packet loss conditions where all packets are lost in one burst. In this special loss-case, the values for the effective equipment impairment factor *Ie-eff* listed in Table I.5 should directly be used with equation 3-1 of [ITU-T G.107].

Table I.5 – Examples for burst packet loss (all packets lost in one burst)

Codec	n (lost packets)	Packet size	PLC type	*Ppl*	*BurstR*	*Ie-eff* (Note)
G.729E	6	20 ms	Native	1.5	5.91	9
G.729E	8	20 ms	Native	2	7.84	11
G.711	6	20 ms	Repeat 1/silence	1.5	5.91	7
G.711	8	20 ms	Repeat 1/silence	2	7.84	10
NOTE – To be used directly in equation 3-1 of [ITU-T G.107].						

Table I.6 provides additional descriptive information on various low bit-rate codecs.

Table I.6 – 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 GSM 06.10.

Table I.6 – Brief description of the low bit-rate codecs

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 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 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 Recommendation 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 Recommendation for speech coding at 40, 32, 24 and 16 kbit/s utilizing adaptive differential pulse code modulation (ADPCM).
G.728	ITU-T Recommendation 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 Recommendation 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

(This appendix does not form an integral part of this Recommendation)

II.1 Introduction

This appendix provides the most recent background material regarding the advantage factor, A. 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 user expectations of services according to the type of user 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] and [ETSI ETR 250]). 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 connection 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 of [ITU-T G.107].

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 of [ITU-T G.107] 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 that "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 [b-MÖLLER] from where this appendix has been extracted.

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 that 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 [b-WILKIE]. Over many studies it has been found that the number of actual users of an innovation develops in an S-shaped curve, see part a) 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 part b) of Figure II.1:

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 part c) 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 not only uses 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 part d) 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 degraded 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.

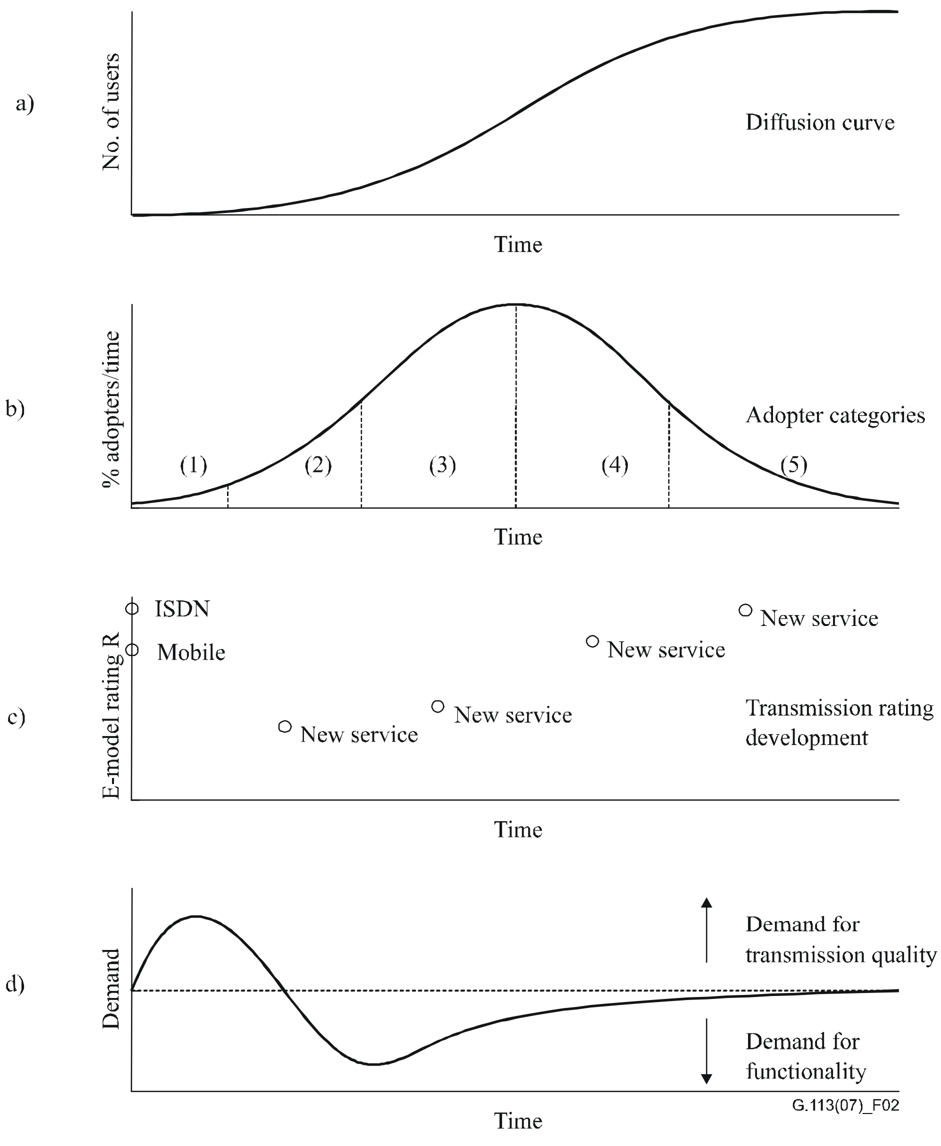

Figure II.1 – Diffusion, transmission quality and expectation for an innovation

Appendix III

Guidelines regarding individual transmission impairment parameters other than *qdu* and equipment impairment factor, *Ie*

(This appendix does not form an integral part of this Recommendation)

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 Rec. P.310 for narrow-band handset telephones; or ITU-T Rec. P.311 for wideband handset telephones; or ITU-T Rec. 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 Rec. 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 Rec. 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.

Appendix IV

Provisional planning values for the wideband equipment impairment factor, *Ie,wb*

(This appendix does not form an integral part of this Recommendation)

This appendix provides up-to-date information on available values of wideband equipment impairment factors, *Ie,wb,* for a number of codecs or codec families. It is intended to be updated regularly. These values are to be used on an extended transmission rating scale (*R*-scale) as it is defined in Appendix II of [ITU-T G.107].

Table IV.1 – Provisional planning values for the wideband equipment impairment factor, *Ie,wb*, for wideband codecs

Codec type	Reference	Operating rate [kbit/s]	*Ie,wb* value
ADPCM	ITU-T Rec. G.722	64	13
		56	20
		48	31
Modifies lapped transform coding	ITU-T Rec. G.722.1	32	13
		24	19
CELP	ITU-T Rec. G.722.2	23.85	8
		23.05	1
		19.85	3
		18.25	5
		15.85	7
		14.25	10
		12.65	13
		8.85	26
		6.6	41

Provisional planning values for the wideband equipment impairment factor, *Ie,wb*, for narrow-band codecs can be derived based on the extension of the *R*-scale by the following procedure.

Ie,wb values for NB codecs correspond to the sum of the *Ie* value defined for the NB case (see Appendix I) and the difference between the WB and the NB "direct" channel, the latter having a position of 93.2 on the *R*-scale (standard G.711 coding and normal noise floor):

$$Ie,wb = \sum_{codecs} Ie + (129 - 93.2) = \sum_{codecs} Ie + 35.8$$

The application of *Ie,wb* values and their potential additivity is for further study.

NOTE – Table IV.2 provides additional descriptive information on various low bit-rate wideband codecs.

Table IV.2 – Brief description of the low bit-rate codecs

G.722	ITU-T Recommendation for 7 kHz audio coding within 64 kbit/s using sub-band adaptive differential pulse code modulation (SB-ADPCM) within a bit-rate of 64 kbit/s.
G.722.1	ITU-T Recommendation for low-complexity coding at 24 and 32 kbit/s for hands-free operation in systems with low frame loss.
G.722.2	ITU-T Recommendation for wideband coding of speech at around 16 kbit/s using adaptive multi-rate wideband (AMR-WB).

Bibliography

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

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

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

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

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

SERIES OF ITU-T RECOMMENDATIONS

Series A	Organization of the work of ITU-T
Series D	General tariff principles
Series E	Overall network operation, telephone service, service operation and human factors
Series F	Non-telephone telecommunication services
Series G	**Transmission systems and media, digital systems and networks**
Series H	Audiovisual and multimedia systems
Series I	Integrated services digital network
Series J	Cable networks and transmission of television, sound programme and other multimedia signals
Series K	Protection against interference
Series L	Construction, installation and protection of cables and other elements of outside plant
Series M	Telecommunication management, including TMN and network maintenance
Series N	Maintenance: international sound programme and television transmission circuits
Series O	Specifications of measuring equipment
Series P	Telephone transmission quality, telephone installations, local line networks
Series Q	Switching and signalling
Series R	Telegraph transmission
Series S	Telegraph services terminal equipment
Series T	Terminals for telematic services
Series U	Telegraph switching
Series V	Data communication over the telephone network
Series X	Data networks, open system communications and security
Series Y	Global information infrastructure, Internet protocol aspects and next-generation networks
Series Z	Languages and general software aspects for telecommunication systems

Printed in Switzerland
Geneva, 2008