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

International telephone connections and circuits – Transmission planning and the E-model

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The E-model: a computational model for use in transmission planning

Recommendation ITU-T G.107

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Recommendation ITU-T G.107

The E-model: a computational model for use in transmission planning

Summary

Recommendation ITU-T G.107 gives the algorithm for the so-called E-model as the common ITU-T transmission rating model. This computational model can be useful to transmission planners, to help ensure that users will be satisfied with end-to-end transmission performance. The primary output of the model is a scalar rating of transmission quality. A major feature of this model is the use of transmission impairment factors that reflect the effects of modern signal processing devices.

In the 2000 version of this Recommendation, an enhanced version of the E-model was provided in order to better take into account the effects of room noise at the send side and quantizing distortion. With the 2002 version, the impairment due to random packet loss was included in a parametric way for different codecs. Since the 2003 version, an enhanced modelling of quality in the case of low talker sidetone levels is provided. The 2005 version enabled more accurate quality predictions for codecs under (short-term) dependent packet loss. The 2009 version included an Appendix II describing a provisional impairment factor framework for wideband speech transmission. In 2011, this appendix was updated in favour of a new Recommendation ITU-T G.107.1. In the current version, the model has been extended to provide an assessment of delay impairments that is also better tailored to less delay-sensitive use cases. Carrier-grade or enterprise-grade telephony systems must be assessed with the default delay sensitivity parameters, while communication systems with low or very low interactivity requirements can be assessed with different parameters. A reference implementation is given in Appendix III.

History

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1.0	ITU-T G.107	1998-12-03	12	11.1002/1000/4541
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5.1	ITU-T G.107 (2005) Amd. 1	2006-06-13	12	11.1002/1000/8864
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FOREWORD

NOTE

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Recommendation ITU-T G.107

The E-model: a computational model for use in transmission planning

1 Scope

This Recommendation describes a computational model, known as the E-model, that has proven useful as a transmission planning tool for assessing the combined effects of variations in several transmission parameters that affect the conversational[1] quality of 3.1 kHz handset telephony. This computational model can be used, for example, by transmission planners to help ensure that users will be satisfied with end-to-end transmission performance whilst avoiding over-engineering of networks. It must be emphasized that the primary output from the model is the "rating factor" R but this can be transformed to give estimates of customer opinion. Such estimates are only made for transmission planning purposes and not for actual customer opinion prediction (for which there is no agreed-upon model recommended by the ITU-T).

The E-model can be used with confidence for many combinations of high importance to transmission planners, but for some parameter combinations of high importance, E-model predictions have been questioned and are currently under study. Annex A provides further information in this regard.

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.107.1] Recommendation ITU-T G.107.1 (2015), *Wideband E-model*.
<http://www.itu.int/rec/T-REC-G.107.1>

[ITU-T G.108] Recommendation ITU-T G.108 (1999), *Application of the E-model: A planning guide*.
<http://www.itu.int/rec/T-REC-G.108>

[ITU-T G.109] Recommendation ITU-T G.109 (1999), *Definition of categories of speech transmission quality*.
<http://www.itu.int/rec/T-REC-G.109>

[ITU-T G.113] Recommendation ITU-T G.113 (2007), *Transmission impairments due to speech processing*.
<http://www.itu.int/rec/T-REC-G.113>

[ITU-T P.833] Recommendation ITU-T P.833 (2001), *Methodology for derivation of equipment impairment factors from subjective listening-only tests*.
<http://www.itu.int/rec/T-REC-P.833>

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[1] Conversational quality in this context refers to transmission characteristics, e.g., long transmission times, effects of talker echoes, etc. However, the E-model, as described in this Recommendation, is not intended to model transmission impairments during double talk situations.

[ITU-T P.834] Recommendation ITU-T P.834 (2015), *Methodology for the derivation of equipment impairment factors from instrumental models*.
<http://www.itu.int/rec/T-REC-P.834>

[ITU-T P.863] Recommendation ITU-T P.863 (2014), *Perceptual objective listening quality assessment*.
<http://www.itu.int/rec/T-REC-P.863>

3 Definitions

3.1 Terms defined elsewhere

None.

3.2 Terms defined in this Recommendation

None.

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

BurstR	Burst Ratio
GoB	Good or Better
LSTR	Listener Sidetone Rating
MOS	Mean Opinion Score
MNRU	Modulated Noise Reference Unit
OLR	Overall Loudness Rating
OPINE	Overall Performance Index model for Network Evaluation
PLC	Packet Loss Concealment
PoW	Poor or Worse
qdu	quantization distortion unit
RLR	Receive Loudness Rating
SLR	Send Loudness Rating
STMR	Sidetone Masking Rating
TELR	Talker Echo Loudness Rating
WEPL	Weighted Echo Path Loss

5 Conventions

None.

6 The E-model, a computational model for use in transmission planning

6.1 Introduction

The complexity of modern networks requires not only that, for transmission planning, each of the many transmission parameters be considered individually but also that their combined effects be taken into account. This can be done by "expert, informed guessing", but a more systematic approach is desirable, such as by using a computational model. The output from the model described here is a scalar quality rating value, *R*, which varies directly with the overall conversational quality. [ITU-T G.113] gives guidance about specific impairments, including combined effects based upon a simplification of the model. However, the output can also give nominal estimates of the user reactions, for instance in the form of percentages finding the modelled connection good or better (GoB) or poor or worse (PoW), as described in Annex B. Furthermore, detailed guidance on the proper application of the E-model, as described in this Recommendation, is provided in [ITU-T G.108]. In addition, the definition of categories of speech transmission quality can be found in [ITU-T G.109].

7 Structure and basic algorithms of the E-model

The E-model is based on the equipment impairment factor method, following previous transmission rating models. It was developed by an ETSI ad hoc group called "Voice Transmission Quality from Mouth to Ear".

The reference connection, as shown in Figure 1, is split into a send side and a receive side. The model estimates the conversational quality from mouth to ear as perceived by the user at the receive side, both as listener and talker.

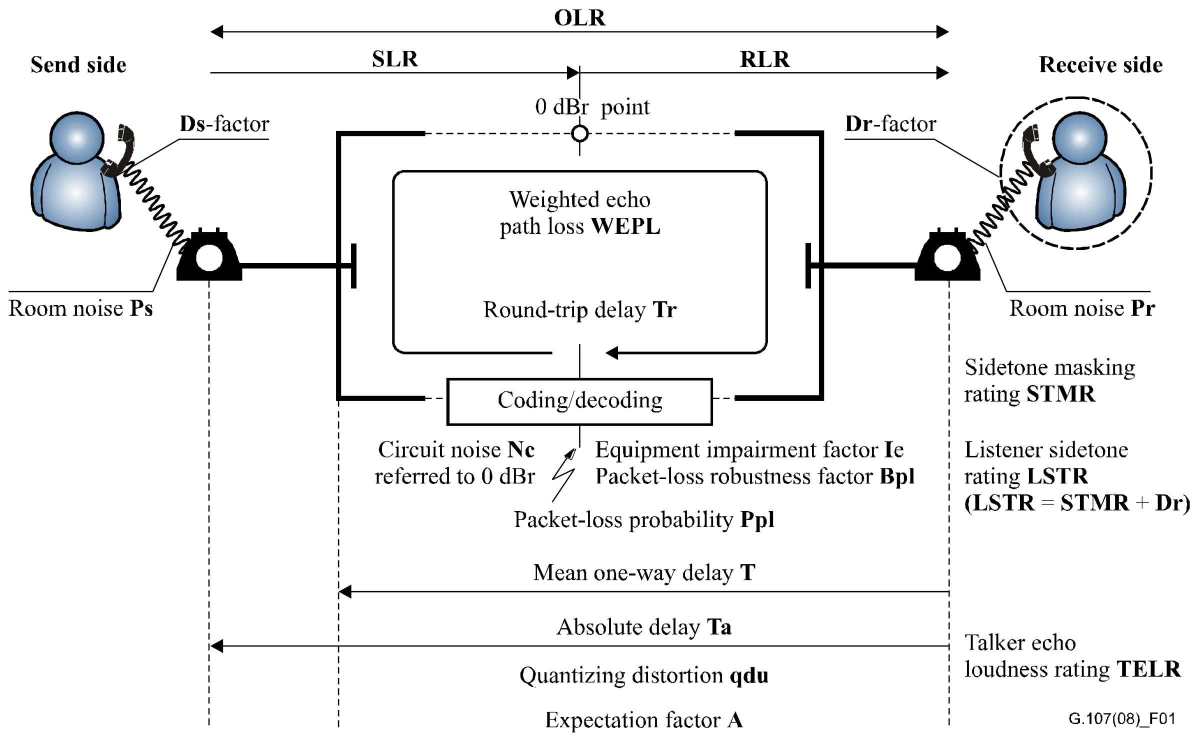

Figure 1 – Reference connection of the E-model

The transmission parameters used as an input to the computation model are shown in Figure 1. Values for room noise and for the *D*-factors are handled separately in the algorithm for the send side and receive side and may be of different amounts. The parameters send loudness rating (SLR), receive loudness rating (RLR) and circuit noise Nc are referred to a defined 0 dBr point. All other input parameters are either considered as values for the overall connection, such as overall loudness rating (OLR), i.e., the sum of SLR and RLR, number of qdu, equipment impairment factors *Ie* and advantage factor *A*, or referred to only for the receive side, such as sidetone masking rating (STMR), listener sidetone rating (LSTR), weighted echo path loss (WEPL) used for the calculation of listener echo and talker echo loudness rating (TELR).

There are three different parameters associated with transmission time. The absolute delay *Ta* represents the total one-way delay between the send side and receive side and is used to estimate the impairment due to excessive delay. The parameter mean one-way delay *T* represents the delay between the receive side (in talking state) and the point in a connection where a signal coupling occurs as a source of echo. The round-trip delay *Tr* only represents the delay in a 4-wire loop, where the "double reflected" signal will cause impairments due to listener echo.

7.1 Calculation of the transmission rating factor, R

According to the equipment impairment factor method, the fundamental principle of the E-model is based on a concept given in the description of the OPINE model, see [b-ITU-T P-Sup.3].

Psychological factors on the psychological scale are additive.

The result of any calculation with the E-model in a first step is a transmission rating factor *R*, which combines all transmission parameters relevant for the considered connection. This rating factor *R* is composed of:

$$R = Ro - Is - Id - Ie\text{-}eff + A \tag{7-1}$$

Ro represents in principle the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise. Factor *Is* is a combination of all impairments which occur more or less simultaneously with the voice signal. Factor *Id* represents the impairments caused by delay and the effective equipment impairment factor *Ie-eff* represents impairments caused by low bit-rate codecs. It also includes impairment due to randomly distributed packet losses. The advantage factor *A* allows for compensation of impairment factors when the user benefits from other types of access to the user. The term *Ro* and the *Is* and *Id* values are subdivided into further specific impairment values. The following clauses give the equations used in the E-model.

7.2 Basic signal-to-noise ratio, Ro

The basic signal-to-noise ratio *Ro* is defined by:

$$Ro = 15 - 1.5(SLR + No) \tag{7-2}$$

The term *No* (in [dBm0p]) is the power addition of different noise sources:

$$No = 10\log\left[10^{\frac{Nc}{10}} + 10^{\frac{Nos}{10}} + 10^{\frac{Nor}{10}} + 10^{\frac{Nfo}{10}}\right] \tag{7-3}$$

Nc (in [dBm0p]) is the sum of all circuit noise powers, all referred to the 0 dBr point.

Nos (in [dBm0p]) is the equivalent circuit noise at the 0 dBr point, caused by the room noise *Ps* at the send side:

$$Nos = Ps - SLR - Ds - 100 + 0.004(Ps - OLR - Ds - 14)^2 \tag{7-4}$$

where *OLR* = *SLR* + *RLR*. In the same way, the room noise *Pr* at the receive side is transferred into an equivalent circuit noise *Nor* (in [dBm0p]) at the 0 dBr point.

$$Nor = RLR - 121 + Pre + 0.008(Pre - 35)^2 \tag{7-5}$$

The term *Pre* (in [dBm0p]) is the "effective room noise" caused by the enhancement of *Pr* by the listener's sidetone path:

$$Pre = Pr + 10\log\left[1 + 10^{\frac{(10 - LSTR)}{10}}\right] \tag{7-6}$$

Nfo (in [dBm0p]) represents the "noise floor" at the receive side,

$$Nfo = Nfor + RLR \tag{7-7}$$

with *Nfor* usually set to −64 dBmp.

7.3 Simultaneous impairment factor, Is

The factor *Is* is the sum of all impairments which may occur more or less simultaneously with the voice transmission. The factor *Is* is divided into three further specific impairment factors:

$$Is = Iolr + Ist + Iq \tag{7-8}$$

Iolr represents the decrease in quality caused by too-low values of OLR and is given by:

$$Iolr = 20\left[\left\{1+\left(\frac{Xolr}{8}\right)^8\right\}^{\frac{1}{8}} - \frac{Xolr}{8}\right] \tag{7-9}$$

where:

$$Xolr = OLR + 0.2(64 + No - RLR) \tag{7-10}$$

The factor *Ist* represents the impairment caused by non-optimum sidetone:

$$Ist = 12\left[1+\left(\frac{STMRo-13}{6}\right)^8\right]^{\frac{1}{8}} - 28\left[1+\left(\frac{STMRo+1}{19.4}\right)^{35}\right]^{\frac{1}{35}} - 13\left[1+\left(\frac{STMRo-3}{33}\right)^{13}\right]^{\frac{1}{13}} + 29 \tag{7-11}$$

where:

$$STMRo = -10\log\left[10^{-\frac{STMR}{10}} + \mathrm{e}^{-\frac{T}{4}} 10^{-\frac{TELR}{10}}\right] \tag{7-12}$$

The impairment factor *Iq* represents impairment caused by quantizing distortion:

$$Iq = 15\log\left[1+10^{Y}+10^{Z}\right] \tag{7-13}$$

where:

$$Y = \frac{R_o - 100}{15} + \frac{46}{8.4} - \frac{G}{9} \tag{7-14}$$

$$Z = \frac{46}{30} - \frac{G}{40} \tag{7-15}$$

and:

$$G = 1.07 + 0.258Q + 0.0602Q^2 \tag{7-16}$$

$$Q = 37 - 15\log(\mathrm{qdu}) \tag{7-17}$$

In this equation, qdu means the number of qdu for the whole connection between the send side and the receive side.

NOTE – If an impairment factor *Ie* is used for a piece of equipment, then the qdu value for that same piece of equipment must not be used.

7.4 Delay impairment factor, Id

The impairment factor *Id* representing all impairments due to delay of voice signals is also further divided into three factors: *Idte*, *Idle* and *Idd*, where

$$Id = Idte + Idle + Idd \tag{7-18}$$

The factor *Idte* gives an estimate for the impairments due to the talker echo:

$$Idte = \left[\frac{Roe - Re}{2} + \sqrt{\frac{(Roe - Re)^2}{4} + 100} - 1 \right] \left(1 - e^{-T}\right) \tag{7-19}$$

where:

$$Roe = -1.5(No - RLR) \tag{7-20}$$

$$Re = 80 + 2.5(TERV - 14) \tag{7-21}$$

$$TERV = TELR - 40 \log \frac{1 + \frac{T}{10}}{1 + \frac{T}{150}} + 6e^{-0.3T^2} \tag{7-22}$$

For values of $T < 1$ ms, the talker echo should be considered as sidetone, i.e., $Idte = 0$. The computation algorithm furthermore combines the influence of STMR to talker echo. Taking into account that low values of STMR may have some masking effects on the talker echo and, for very high values of STMR, the talker echo may become more noticeable, the terms *TERV* and *Idte* are adjusted as follows:

For STMR < 9 dB:

In Equation 7-21, *TERV* is replaced by *TERVs*, where:

$$TERVs = TERV + \frac{Ist}{2} \tag{7-23}$$

For 9 dB ≤ STMR ≤ 20 dB:

Equations 7-19 to 7-22 apply.

For STMR > 20 dB:

In Equation 7-18, *Idte* is replaced by *Idtes*, where:

$$Idtes = \sqrt{Idte^2 + Ist^2} \tag{7-24}$$

The factor *Idle* represents impairments due to listener echo. The equations are:

$$Idle = \frac{Ro - Rle}{2} + \sqrt{\frac{(Ro - Rle)^2}{4} + 169} \tag{7-25}$$

where:

$$Rle = 10.5(WEPL + 7)(Tr + 1)^{-0.25} \tag{7-26}$$

The factor *Idd* represents the impairment caused by too-long absolute delay *Ta*, which occurs even with perfect echo cancelling. Only when the effect due to pure delay is attributed to the service, is the effect, to some extent, reflected in the respective speech quality evaluation by the user. It is noted that the effect due to pure delay may instead be attributed to the other conversation partner, or may be attributed to a conversational difficulty; for example if the conversation partners do not know each other, or delay may simply reduce the efficiency of the communication. In the current version of the E-model, different types of conversations and/or users are considered, in terms of their interactivity and delay-sensitivity. As a consequence, the predictions can be tailored to the delay sensitivity requirements of the user group.

For $Ta \le mT$:

$$Idd = 0$$

For $Ta > mT$:

$$Idd = 25\left\{\left(1 + X^{6 \cdot sT}\right)^{\frac{1}{6 \cdot sT}} - 3\left(1 + \left[\frac{X}{3}\right]^{6 \cdot sT}\right)^{\frac{1}{6 \cdot sT}} + 2\right\} \qquad (7\text{-}27)$$

with:

$$X = \frac{\log\left(\frac{Ta}{mT}\right)}{\log 2} \qquad (7\text{-}28)$$

The following fixed settings of delay sensitivity, *sT* and minimum perceivable delay, *mT*, are recommended, reflecting different use cases and user groups. Two aspects are addressed by these settings:

– The interactivity of the conversation and the sensitivity of the users to the delay-effect.

– The application scenario, that is, whether a given call is being made in a business context or in an everyday situation. Even if users may not notice the delay, it may be very critical for the efficiency or even effectiveness of a given call, for example in a business context.

As a consequence, in the case where it is uncertain what user group or what application scenario is being addressed with the planned service, it is recommended that the default class is used. Any case of non-default value usage should be explicitly mentioned when reporting results.

Based on these considerations, the following settings, shown in Table 1, are recommended for *sT* and *mT*:

Table 1 – Delay-sensitivity classes for different use cases

Class of delay-sensitivity	***sT***	***mT* (ms)**	**Use case**
Default	1	100	– Applicable to all types of telephone conversations. – Must be used for: • Carrier-grade fixed or mobile telephony; • Enterprise-grade fixed or mobile telephony; • When targeted user group and delay requirements are unknown.
Low	0.55	120	– Applicable only in cases where it is known that users have low sensitivity to delay, e.g., non time-sensitive conversation scenarios.
Very low	0.4	150	– Applicable only in cases where it is known that users have very low sensitivity to delay, e.g., in primarily non-interactive cases, such as mainly listening to a conversation or to a lecture.

7.5 Equipment impairment factor, Ie

The values for the equipment impairment factor *Ie* of elements using low bit-rate codecs are not related to other input parameters. They depend on subjective mean opinion score (MOS) test results as well as on network experience. Refer to Appendix I of [ITU-T G.113] for the currently recommended values of *Ie*.

Specific impairment factor values for codec operation under random[2] packet-loss have formerly been treated using tabulated, packet-loss dependent *Ie*-values. Now, the packet-loss robustness factor *Bpl* is defined as a codec-specific value. The packet-loss dependent effective equipment impairment factor *Ie-eff* is derived using the codec-specific value for the equipment impairment factor at zero packet-loss *Ie* and the packet-loss robustness factor *Bpl*, both listed in Appendix I of [ITU-T G.113] for several codecs. With the packet-loss probability *Ppl*, *Ie-eff* is calculated using the equation:

$$Ie\text{-}eff = Ie + (95 - Ie) \cdot \frac{Ppl}{\frac{Ppl}{BurstR} + Bpl} \tag{7-29}$$

BurstR is the so-called burst ratio, which is defined as:

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

When packet loss is random (i.e., independent) *BurstR* = 1; and

when packet loss is bursty (i.e., dependent) *BurstR* > 1.

For example, for packet loss distributions corresponding to a 2-state Markov model with transition probabilities *p* between a "found" and a "loss" state, and *q* between the "loss" and the "found" state, the burst ratio can be calculated as:

$$BurstR = \frac{1}{p+q} = \frac{Ppl/100}{p} = \frac{1 - Ppl/100}{q} \tag{7-30}$$

As can be seen from Equation 7-29, the effective equipment impairment factor in the case of *Ppl* = 0 (no packet-loss) is equal to the *Ie* value defined in Appendix I of [ITU-T G.113].

Please refer to Annex A for the range of parameter values the algorithm has been validated for.

Ie-eff should be derived by using the *Ie* and *Bpl* values if they are provided in [ITU-T G.113] and the packet-loss burstiness can be defined by observing packet-loss characteristics. If, for practical reasons, it is difficult to observe the packet-loss rate (*Ppl*) and/or packet-loss characteristics to derive the burstiness parameter (*BurstR*), the [ITU-T P.834] approach can be used to directly derive *Ie-eff*.

If *Ie* is derived directly by using the instrumental method recommended in [ITU-T P.834], it already reflects the effect of packet loss introduced in the preparation of speech materials under test. Therefore, Equation 7-29 should not be used. Rather, the *Ie* value derived by [ITU-T P.834] in *Ie-eff* in Equation 7-1 should be used.

7.6 Advantage factor, A

Due to the specific meaning of the advantage factor *A*, there is consequently no relation to any of the other transmission parameters. Some provisional values are given in Table 2.

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[2] The probability of losing a packet is regarded as independent of the reception state (received/lost) of the previous packet.

Table 2 – Provisional examples for the advantage factor *A*

Communication system example	Maximum value of A
Conventional (wirebound)	0
Mobility by cellular networks in a building	5
Mobility in a geographical area or moving in a vehicle	10
Access to hard-to-reach locations, e.g., via multi-hop satellite connections	20

It should be noted that the values in Table 2 are only provisional. The use of factor *A* and its selected value in a specific application is the planner's decision. However, the values in Table 2 should be considered as absolute upper limits for *A*. Additional background information on the advantage factor *A* can be found in Appendix II to [ITU-T G.113].

7.7 Default values

The default values for all input parameters used in the algorithm of the E-model, are listed in Table 3. It is strongly recommended to use these default values for all parameters that do not vary during planning calculation. If all parameters are set to the default values, the calculation results in a very high quality with a rating factor of $R = 93.2$.

Table 3 – Default values and permitted ranges for the parameters

Parameter	Abbr.	Unit	Default value	Permitted range	Remark
Send loudness rating	SLR	dB	+8	0 ... +18	(Note 1)
Receive loudness rating	RLR	dB	+2	–5 ... +14	(Note 1)
Sidetone masking rating	STMR	dB	15	10 ... 20	(Notes 2, 4)
Listener sidetone rating	LSTR	dB	18	13 ... 23	(Note 2)
D-Value of telephone, send side	Ds	–	3	–3 ... +3	(Note 2)
D-Value of telephone, receive side	Dr	–	3	–3 ... +3	(Note 2)
Talker echo loudness rating	TELR	dB	65	5 ... 65	
Weighted echo path loss	WEPL	dB	110	5 ... 110	
Mean one-way delay of the echo path	T	ms	0	0 ... 500	
Round-trip delay in a 4-wire loop	Tr	ms	0	0 ... 1000	
Absolute delay in echo-free connections	Ta	ms	0	0 ... 500	
Delay sensitivity	sT	–	1	0.4 … 1	(Note 7)
Minimum perceivable delay	mT	ms	100	20 … 150	(Note 7)
Number of quantization distortion units	qdu	–	1	1 ... 14	
Equipment impairment factor	Ie	–	0	0 ... 40	(Note 5)
Packet-loss robustness factor	Bpl	–	4.3	4.3 ... 40	(Notes 3, 5)
Random packet-loss probability	Ppl	%	0	0 ... 20	(Notes 3, 5)
Burst ratio	BurstR	–	1	1 … 8	(Notes 3, 6)
Circuit noise referred to 0 dBr-point	Nc	dBm0p	–70	–80 ... –40	
Noise floor at the receive side	Nfor	dBmp	–64	–	(Note 3)
Room noise at the send side	Ps	dB(A)	35	35 ... 85	

Table 3 – Default values and permitted ranges for the parameters

Parameter	Abbr.	Unit	Default value	Permitted range	Remark
Room noise at the receive side	Pr	dB(A)	35	35 ... 85	
Advantage factor	A	–	0	0 ... 20	

NOTE 1 – Total values between microphone or receiver and 0 dBr-point.

NOTE 2 – Fixed relation: LSTR = STMR + D.

NOTE 3 – Currently under study.

NOTE 4 – Equation 7-24 provides also predictions for STMR > 20 dB. However, such values can hardly be measured in a reliable way because the measurement device will mainly cover the acoustic coupling, and not the electrical one.

NOTE 5 – If Ppl > 0%, then the Bpl must match the codec, packet size and packet loss concealment (PLC) assumed.

NOTE 6 – E-model predictions for values of BurstR > 2 are only valid if the packet loss percentage is Ppl < 2%.

NOTE 7 – Only predefined settings are allowed for sT and mT, reflecting specific use cases, see main body of the Recommendation.

The 2000 version of this Recommendation provided an enhanced version of the E-model algorithm (see Annex A).

Due to the changes made in the 2000 version of this Recommendation, the resulting rating R with all parameter values set to default has slightly changed (from $R = 94.2$ to $R = 93.2$). For practical planning purposes, however, this slight deviation should be considered insignificant.

Annex A

Conditions for using the E-model

(This annex forms an integral part of this Recommendation.)

NOTE – The assessment and enhancement of the E-model algorithm are for further study. New results will be included as soon as they become available.

A.1 Examples of conditions where caution must be exercised when using the E-model

– *The overall level of equipment impairment factors*

Some experimental investigations suggest that the general tendency of the equipment impairment factors is too pessimistic, so that a hidden security margin may be incorporated.

– *The overall additivity property of the model*

The E-model assumes that different kinds of impairments are additive on the scale of the transmission rating factor *R*. This feature has not been checked to a satisfactory extent. More specifically, very few investigations are available regarding the interaction of low bit-rate codecs with other kinds of impairments, e.g., with room noise. Additionally, the order effects when tandeming several low bit-rate codecs remain uncertain.

– *The coverage of talker sidetone*

Some experiments show that the E-model disregards some masking effects occurring for talker sidetone, namely in conjunction with circuit noise, room noise at receive side and low delay talker echo (< 10 ms).

– *The advantage factor A*

Up to now it has not been clarified under which conditions the given values for the advantage factor should be applied. It is expected that these values may depend for example on the user group, and that the absolute values will change in the long term.

– *Derivation methodology for new equipment impairment factors*

A new methodology for deriving equipment impairment factors from subjective listening quality tests has been adopted as [ITU-T P.833]. A new methodology for deriving equipment impairment factors from instrumental models such as [ITU-T P.863] has been adopted as [ITU-T P.834].

– *Predictions for different types of room noise and different frequency shapes in the communication channel, in the sidetone path and in the echo path*

The E-model regards the effect of room noise only by means of an A-weighted level. The actual opinion on the speech communication quality may depend even on the type and disturbance of the environmental noise. The frequency characteristics of the communication channel, of the sidetone and of the echo path are not explicitly regarded by the E-model, but only implicitly by means of loudness ratings. However, they may affect the perceived transmission quality.

A.2 Conditions for which the performance of the E-model has been improved by updating from the earlier version

– *The effect of room noise at send side*

With the present enhanced E-model algorithm (2000 version), the Lombard effect (the fact that the speaker adapts his/her pronunciation and speaking level according to the noise environment) is no longer disregarded. This had, in the 1998 version, led to too pessimistic E-model predictions for high room noise levels *Pr*.

– *Predictions for quantizing distortion*

In case of the 1998 version of the E-model, subjective test results for modulated noise reference unit (MNRU) reference conditions were very often more pessimistic than E-model predictions. The graphs in Figure A.1 have been derived from the 1998 version and the 2000 version of the E-model with all other parameters at their default values.

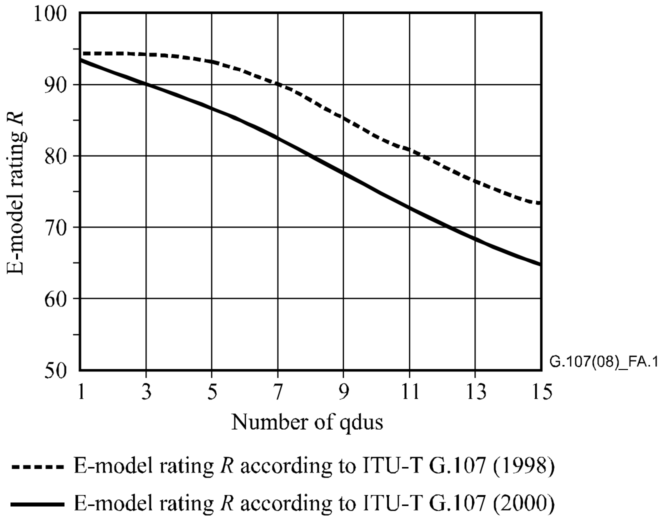

Figure A.1 – Relation between the number of qdu and E-model rating *R*

With respect to the slightly enhanced algorithm of the E-model, as given in this Recommendation, the relation between the parameter qdu and the E-model rating *R* has been changed in order to align the algorithm better with the available subjective test results.

– *Predictions for codec performance under random packet loss*

Impairments due to codecs under packet-loss conditions were formerly handled by using codec-dependent tabulated equipment impairment factors for different packet-loss rates (in former versions of Appendix I of [ITU-T G.113]). As the aim is to reduce the amount of tabulated data for usage with the E-model, the possibilities of replacing tabulated *Ies* for packet loss with corresponding equations were investigated. The approach selected leads to results very similar to those previously defined as *Ie* for all codecs covered in the 2001 version of Appendix I of [ITU-T G.113].

– *Predictions for codec performance under dependent packet loss*

With this version of the algorithm, loss distributions characterized by medium (short-term) loss dependencies (as opposed to long-term loss dependencies) have been integrated in the E-model. Up to now, the included approach has been evaluated only for the ITU-T G.729(A) codec, but is assumed to be applicable also to the ITU-T G.723.1 codec, and supposedly other codecs. Pending further verification, the algorithm should not be used with burst ratios higher than *BurstR* = 2.0. The model can also be applied to burst ratios higher than 2.0, if the packet loss percentages *Ppl* are lower than 2%.

– *The effect of talker sidetone*

Estimates of voice quality as a function of STMR for values > 15 dB, as provided by the 2002 version of this Recommendation, were too pessimistic and did not accurately match the results obtained in auditory tests. This proved to be especially important for telephones in North America that are typically specified to have nominal values of STMR from 16 to 18 dB.

In the current revised version of the E-model algorithm, this observation is reflected by modifying the corresponding equation for *Ist* as a function of sidetone (*STMR*), see Equation 7-11.

As mentioned in the main body of this Recommendation, talker echo may become more noticeable for quiet values of STMR. This is addressed by switching from *Idte* to *Idtes*, see Equation 7-24. To remain consistent, the talker echo threshold from STMR > 15 dB (2002 version of this Recommendation) was extended to STMR > 20 dB (the current version of this Recommendation). The modifications have no impact for values of STMR < 15 dB. Consequently, the quality prediction for the transmission rating factor *R* for the default settings (*STMR* = 15 dB) does not differ from that predicted by the previous model version (2002). The default value of *R* is 93.2 for both the previous and the current versions. The situation is depicted in Figure A.2.

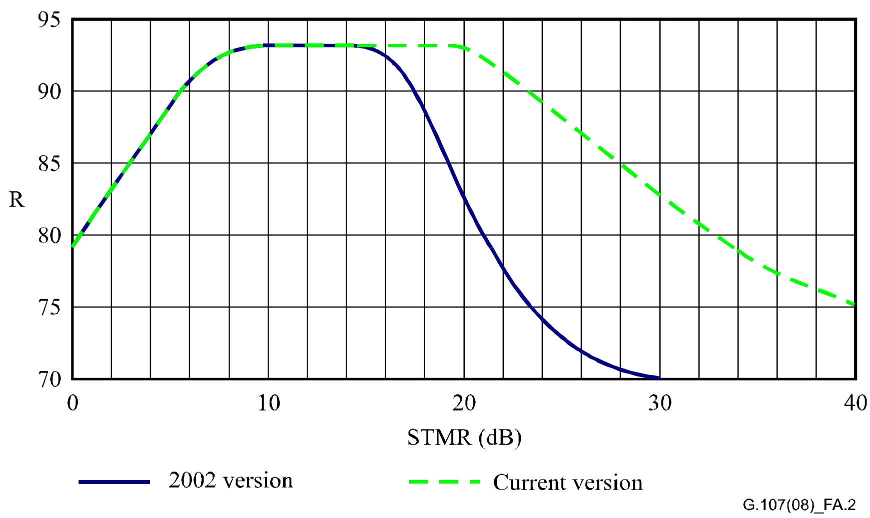

Figure A.2 – Comparison of *R* versus STMR for the current and the previous versions of the E-model algorithm

– *The effect of pure delay*

Based on several conversation tests, it has been shown that even long delay values may not affect the perceived speech quality, that is, in terms of the attribution of the delay effect to the system. In such cases, the predictions by previous versions of the E-model may be more pessimistic than actual user opinion. As a consequence, in specific cases speech quality predictions may be sought that are better tailored to some less stringent delay requirements.

As a consequence, in the current revised version of the E-model, the delay sensitivity sT and minimum perceivable delay mT have been introduced. With an appropriate choice of these two parameters, both default and less delay-sensitive calls can be addressed. It is advisable to use these new parameters carefully according to the expected delay class according to Table 1.

Annex B

Quality measures derived from the transmission rating factor *R*

(This annex forms an integral part of this Recommendation.)

The transmission rating factor *R* can lie in the range from 0 to 100, where *R* = 0 represents an extremely bad quality and *R* = 100 represents a very high quality. The E-model provides a statistical estimation of quality measures. The percentages for a judgement of good or better (GoB) or poor or worse (PoW) are obtained from the *R*-factor by means of the Gaussian error function:

$$E(x) = \frac{1}{\sqrt{2\pi}} \int_{-\infty}^{x} e^{-\frac{t^2}{2}} dt \qquad \text{(B-1)}$$

The equations are:

$$GoB = 100E\left(\frac{R-60}{16}\right)\% \qquad \text{(B-2)}$$

$$PoW = 100E\left(\frac{45-R}{16}\right)\% \qquad \text{(B-3)}$$

An estimated mean opinion score (MOS_{CQE}) for the conversational situation in the scale 1-5 can be obtained from the *R*-factor using the equations:

For $R < 0$: $MOS_{CQE} = 1$

For $0 < R < 100$: $MOS_{CQE} = 1 + 0.035R + R(R-60)(100-R)7 \cdot 10^{-6}$ (B-4)

For $R > 100$: $MOS_{CQE} = 4.5$

This equation can be inverted in the range $6.5 \le R \le 100$ to calculate *R* from MOS_{CQE}, see Appendix I. GoB, PoW and MOS_{CQE} as functions of *R* are depicted in Figures B.1 and B.2, respectively.

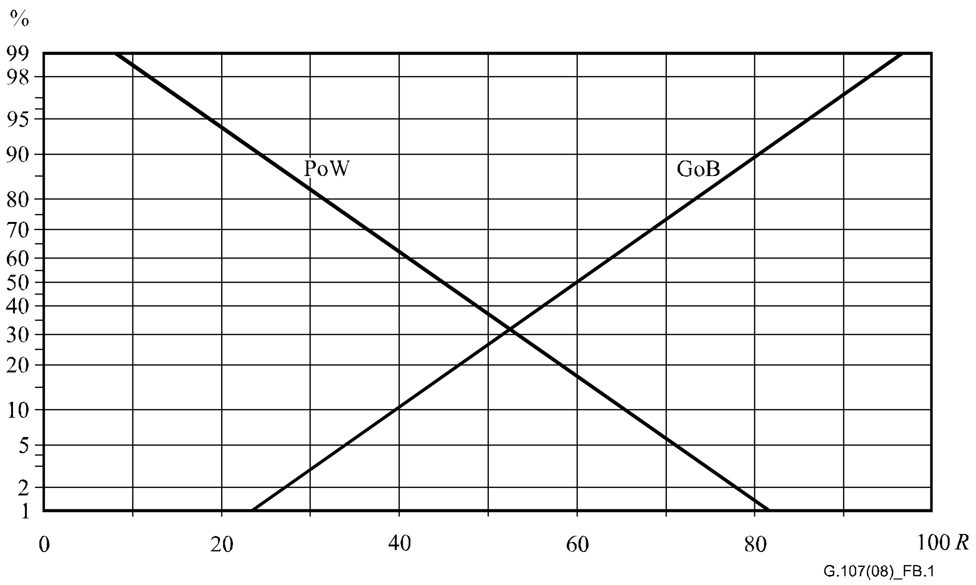

Figure B.1 – Good or better (GoB) and poor or worse (PoW) as functions of rating factor *R*

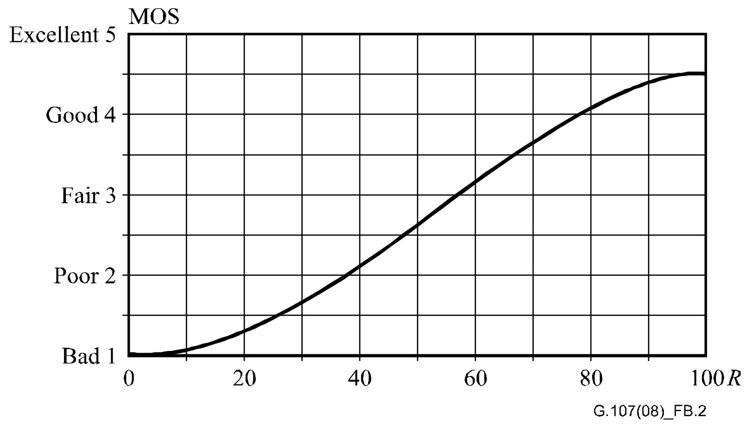

Figure B.2 – MOS_{CQE} as function of rating factor *R*

In some cases, transmission planners may not be familiar with the use of quality measures such as the *R* rating factor obtained from planning calculations, and thus provisional guidance for interpreting calculated *R* factors for planning purposes is given in Table B.1[3]. This table also contains equivalent transformed values of *R* into estimated conversational MOS_{CQE}, GoB and PoW.

Table B.1 – Provisional guide for the relation between *R*-value and user satisfaction

***R*-value (lower limit)**	**MOS_{CQE} (lower limit)**	**GoB (%) (lower limit)**	**PoW (%) (upper limit)**	**User satisfaction**
90	4.34	97	~0	Very satisfied
80	4.03	89	~0	Satisfied
70	3.60	73	6	Some users dissatisfied
60	3.10	50	17	Many users dissatisfied
50	2.58	27	38	Nearly all users dissatisfied

3 The source of Table B.1 is Table 1 of [ITU-T G.109].

Appendix I

Calculation of *R* from MOS_{CQE} values

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

In the range $6.5 \le R \le 100$, *R* can be calculated from MOS_{CQE} using the formula:

$$R = \frac{20}{3}\left(8 - \sqrt{226}\cos\left(h + \frac{\pi}{3}\right)\right) \qquad \text{(I-1)}$$

with:

$$h = \tfrac{1}{3}\operatorname{arctan2}\left(18566 - 6750 MOS_{CQE}, 15\sqrt{-903522 + 1113960 MOS_{CQE} - 202500 MOS_{CQE}{}^{2}}\right) \qquad \text{(I-2)}$$

and:

$$\operatorname{arctan2}(x,y) = \begin{cases} \arctan\left(\dfrac{y}{x}\right) & \text{for } x \ge 0 \\ \pi - \arctan\left(\dfrac{y}{-x}\right) & \text{for } x < 0 \end{cases} \qquad \text{(I-3)}$$

The function arctan2 (*x*, *y*) is implemented in ANSI C as the function atan2 (*y*, *x*). Users should note that the order of the two parameters differs in this case.

Appendix II

Provisional impairment factor framework for wideband speech transmission

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

The contents of this appendix have been moved to [ITU-T G.107.1].

Appendix III

Reference implementation of the E-model in ITU-T G.107

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

The reference implementation of the E-model is no longer provided in the traditional qbasic listing (which was provided in previous versions of this Recommendation), since it was no longer found to be appropriate. Therefore, the ITU-T provides a new implementation of the E-model in PHP4 scripting language.

The associated electronic attachment (which is available free of charge on the ITU publications website at http://www.itu.int/rec/T-REC-G.107-201112-S/en) contains the following files in the Software folder:

– General instructions: readme.txt
– PHP4 script: calc.php
– 3 required Java scripts:
  - wz_tooltip.js
  - tip_centerwindow.js
  - tip_followscroll.js
– Copyright notice

In order to run this software on any server outside ITU, a licence must be obtained from the organization indicated in the software Copyright notice.

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Appendix IV

Use of the E-model in conjunction with noise reduction or echo canceller systems in the network or the terminal equipment

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

Modern networks or terminals frequently contain devices for echo cancellation and/or noise reduction. Echo cancellers are expected to significantly reduce the echo and the amount of residual echo may be considered in the same way the standard E-model does, i.e., via the residual talker echo loudness rating *TELR* and the mean one-way delay *T* of the residual echo path. However, the echo attenuation may also vary over time (so-called "echo pumping"), and the cancellation process may lead to degradations of the transmitted speech signal. Degradations of the transmitted speech signal may also result from imperfect noise reduction, e.g., when parts of the speech spectrum are subtracted by the noise reduction algorithm. Such a degradation due to imperfect noise reduction is also not covered by the current version of the E-model.

In order to assess these effects in a more elaborated way, it is proposed to go through all the steps of the following provisional procedure which apply to the given scenario (i.e., noise reduction, echo cancellation, or both), see also [b-Möller]:

1) The residual noise resulting from imperfect background noise reduction may occur either during speech intervals or during pauses; parameters describing these two situations are defined in [b-ITU-T G.160], namely *SNRI* (the SNR improvement during speech in dB) and *TNLR* (the total noise level reduction in dB). It is proposed to use a weighting of half and half (corresponding to roughly 50% speech activity) for the speech and silence parts and change Equation 7-4 as follows:

$$\begin{aligned} Nos \; = \; & Ps - SLR - Ds - 0.5(SNRI + TNLR) - 100 \\ & + 0.004(Ps - OLR - Ds - 14)^2 \end{aligned} \tag{IV-1}$$

   This amendment is meant to capture the effect of residual noise of the noise reduction mechanism.

   In the case that a non-white background noise is assumed, the factor *Ds* of Equation IV-1 might depend on the noise type used for its measurement; see the ITU-T Handbook on Telephonometry [b-ITU-T HB Teleph]. In that case, it is suggested to use noise of the same type as it is assumed to occur in the background.

2) The effects of speech degradation from imperfect noise reduction can be captured by estimating via an additional equipment impairment factor *Ie,nr* reflecting the noise reduction equipment. Such an additional equipment impairment factor should ideally be derived with the help of auditory listening-only tests carried out in accordance with [b-ITU-T P.835]. As an alternative, the S-MOS scores might also be estimated with the objective model of [b-ETSI EG 202]. Provided that such S-MOS scores are available for (1) the connection of the noise-reduced case and (2) for a noise-free connection without the noise-reduction system applied, *Ie,nr* can be calculated as:

$$Ie,nr = \min\left(R(\text{S-MOS2}) - R(\text{S-MOS1}), 0\right) \tag{IV-2}$$

   In this equation, the transformation from S-MOS to *R* is performed using the relationship between MOS and *R* given in the E-model. The resulting *Ie,nr* scores are preferably normalized following the procedure of [ITU-T P.834].

3) The effects of residual echo are taken into account in the standard way of the E-model, i.e., via the talker-echo impairment factor *Idte*; the frequency-dependent attenuation of the residual echo path has to be used for the calculation of *TELR* at this stage.

4) The effects of speech degradation from imperfect echo cancellation can be estimated via an additional equipment impairment factor *Ie,ec* which is calculated with the help of the procedure of [ITU-T P.834], using the instrumental model of [ITU-T P.863]. The calculation is performed as in step 2 and it is also preferable to normalize the obtained raw *Ie,ec* score with the help of the procedure of [ITU-T P.834].

   Please note that [ITU-T P.863] is not intended to be used with talker echo as a test factor, so applying it to derive impairment factors for echo cancellers should be exercised with care.

5) Both *Ie,nr* and *Ie,ec* are added to the effective equipment impairment factor *Ie,eff* before calculating the overall transmission rating *R*.

6) The effects of delay are captured in the usual way via *Idd*.

The proposed methodology is only provisional, as it has not been thoroughly validated. However, it is assumed that it results in better estimations than by not considering the mentioned effects of noise reduction and echo canceller equipment. The following effects are not yet covered by the methodology and are thus for further study in ITU-T:

– The effect of time-varying echo paths.

– The effects of variable background noise transmission due to the echo canceller.

– The reduction of the double-talk capability due to the echo canceller.

– The degradation resulting from the acoustic characteristics of the terminal to which noise reduction and/or echo cancellers might be integrated.

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