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

International telephone connections and circuits – General definitions

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

ITU-T Recommendation G.107

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ITU-T Recommendation G.107

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

Summary

This Recommendation 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 year 2000 revision, 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 year 2002 revision, the impairment due to random packet-loss has been included in a parametric way for different codecs. Since the 2003 version, an enhanced modelling of the quality in case of low talker sidetone levels is provided. The current revision enables more accurate quality predictions for codecs under (short-term) dependent packet loss.

Source

ITU-T Recommendation G.107 was approved on 1 March 2005 by ITU-T Study Group 12 (2005-2008) under the ITU-T Recommendation A.8 procedure.

FOREWORD

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CONTENTS

ITU-T Recommendation G.107

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

1 General

1.1 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 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).

This revision now includes packet-loss as a new parameter, and an enhancement of the talker sidetone modelling.

The E-model has not been fully verified by field surveys or laboratory tests for the very large number of possible combinations of input parameters. For many combinations of high importance to transmission planners, the E-model can be used with confidence, but for other parameter combinations, E-model predictions have been questioned and are currently under study. Accordingly, caution must be exercised when using the E-model for some conditions; for example, the E-model may give inaccurate results for combinations of certain types of impairments. Annex A provides further information in this regard.

1.2 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.

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

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

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

[4] ITU-T Recommendation G.113 (2001), *Transmission impairments due to speech processing*.

[5] ITU-T Recommendation G.113 Appendix I (2002), *Provisional planning values for the equipment impairment factor Ie and packet-loss robustness factor Bpl*.

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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.

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

[7] ITU-T Recommendation P.834 (2002), *Methodology for the derivation of equipment impairment factors from instrumental models*.

[8] ITU-T Recommendation P.862 (2001), *Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs*.

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

2.1 Introduction

The complexity of modern networks requires that for transmission planning the many transmission parameters are not only considered individually but also that their combination effects are 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 Rec. G.113 [4] gives guidance about specific impairments, including combinations effects based upon a simplification of the model. However, the output can also give nominal estimates of user reactions, for instance in the form of percentages finding the modelled connection "Good or Better" or "Poor or Worse", 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 Rec. G.108 [2]. In addition, the definition of categories of speech transmission quality can be found in ITU-T Rec. G.109 [3].

2.2 Source code

Annex C contains the source code in BASIC of the E-model described in this Recommendation. The purpose of this code is to ensure that users of the E-model are using consistent implementations of the formulae.

3 The 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 in 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.

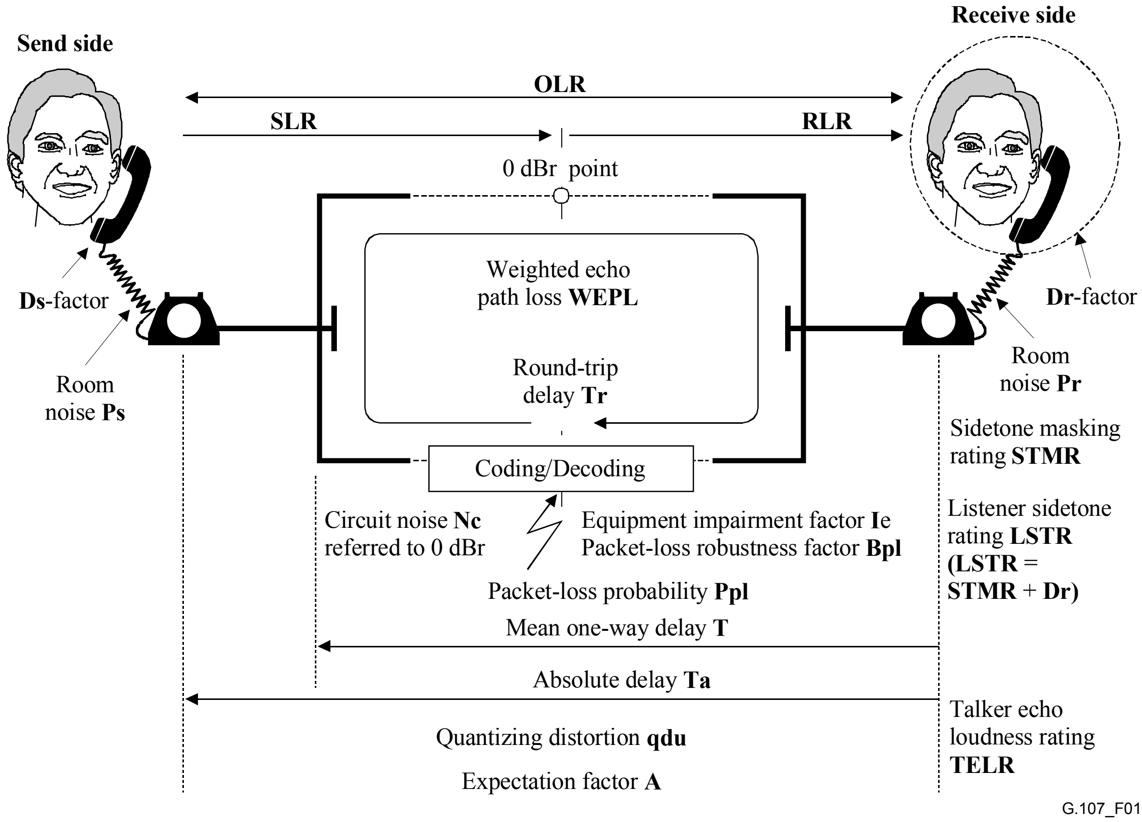

Figure 1/G.107 – Reference connection of the E-model

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 send side and receive side and may be of different amounts. The parameters SLR, 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 OLR (in any case the sum of SLR and RLR), number of qdu, equipment impairment factors *Ie* and advantage factor *A*, or referred only to the receive side, such as STMR, LSTR, WEPL (for calculation of Listener Echo) and TELR.

There are three different parameters associated with transmission time. The absolute delay *Ta* represents the total one-way delay between send side and receive side and is used to estimate the impairment due to too-long 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.

3.1 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 Bibliography, Supplement 3 to P-series]:

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:

$$R = Ro - Is - Id - Ie\text{-}eff + A \qquad (3\text{-}1)$$

Ro represents in principle the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise. The 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 packet-losses of random distribution. The advantage factor *A* allows for compensation of impairment factors when there are other advantages 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 formulae used in the E-model.

3.2 Basic signal-to-noise ratio, *Ro*

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

$$Ro = 15 - 1.5(SLR + No) \tag{3-2}$$

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

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

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:

$$Nos = Ps - SLR - Ds - 100 + 0.004(Ps - OLR - Ds - 14)^2 \tag{3-4}$$

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.

$$Nor = RLR - 121 + Pre + 0.008(Pre - 35)^2 \tag{3-5}$$

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

$$Pre = Pr + 10\log\left[1 + 10^{\frac{(10 - LSTR)}{10}}\right] \tag{3-6}$$

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

$$Nfo = Nfor + RLR \tag{3-7}$$

with *Nfor* usually set to −64 dBmp.

3.3 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:

$$Is = Iolr + Ist + Iq \tag{3-8}$$

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

$$Iolr = 20\left[\left\{1 + \left(\frac{Xolr}{8}\right)^8\right\}^{\frac{1}{8}} - \frac{Xolr}{8}\right] \tag{3-9}$$

where:

$$Xolr = OLR + 0.2(64 + No - RLR) \quad (3\text{-}10)$$

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

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

where:

$$STMRo = -10\log\left[10^{-\frac{STMR}{10}} + \mathrm{e}^{-\frac{T}{4}}10^{-\frac{TELR}{10}}\right] \quad (3\text{-}12)$$

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

$$Iq = 15\log\left[1 + 10^{Y} + 10^{Z}\right] \quad (3\text{-}13)$$

where:

$$Y = \frac{R_o - 100}{15} + \frac{46}{8.4} - \frac{G}{9} \quad (3\text{-}14)$$

$$Z = \frac{46}{30} - \frac{G}{40} \quad (3\text{-}15)$$

and:

$$G = 1.07 + 0.258Q + 0.0602Q^2 \quad (3\text{-}16)$$

$$Q = 37 - 15\log(\mathrm{qdu}) \quad (3\text{-}17)$$

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

3.4 Delay impairment factor, *Id*

Also *Id*, the impairment factor representing all impairments due to delay of voice signals is further subdivided into the three factors *Idte*, *Idle* and *Idd*:

$$Id = Idte + Idle + Idd \quad (3\text{-}18)$$

The factor *Idte* gives an estimate for the impairments due to Talker Echo:

$$Idte = \left[\frac{Roe - Re}{2} + \sqrt{\frac{(Roe - Re)^2}{4} + 100} - 1\right]\left(1 - \mathrm{e}^{-T}\right) \quad (3\text{-}19)$$

where:

$$Roe = -1.5(No - RLR) \quad (3\text{-}20)$$

$$Re = 80 + 2.5(TERV - 14) \quad (3\text{-}21)$$

$$TERV = TELR - 40\log\frac{1+\frac{T}{10}}{1+\frac{T}{150}} + 6e^{-0.3T^2} \qquad (3\text{-}22)$$

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 3-21 *TERV* is replaced by *TERVs*, where:

$$TERVs = TERV + \frac{Ist}{2} \qquad (3\text{-}23)$$

For 9 dB ≤ STMR ≤ 20 dB:

the above-given Equations 3-19 to 3-22 apply.

For STMR > 20 dB:

In Equation 3-18, *Idte* is replaced by *Idtes*, where:

$$Idtes = \sqrt{Idte^2 + Ist^2} \qquad (3\text{-}24)$$

The factor *Idle* represents impairments due to Listener Echo. The equations are:

$$Idle = \frac{Ro - Rle}{2} + \sqrt{\frac{(Ro - Rle)^2}{4} + 169} \qquad (3\text{-}25)$$

where:

$$Rle = 10.5(WEPL + 7)(Tr + 1)^{-0.25} \qquad (3\text{-}26)$$

The factor *Idd* represents the impairment caused by too-long absolute delay *Ta*, which occurs even with perfect echo cancelling.

For $Ta \leq 100$ ms:

$$Idd = 0$$

For $Ta > 100$ ms:

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

with:

$$X = \frac{\log\left(\frac{Ta}{100}\right)}{\log 2} \qquad (3\text{-}28)$$

3.5 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 test results as well as on network experience. Refer to Appendix I/G.113 [5] for the actually 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 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/G.113 for several codecs. With the Packet-loss Probability *Ppl*, *Ie-eff* is calculated using the formula:

$$Ie\text{-}eff = Ie + (95 - Ie) \cdot \frac{Ppl}{\frac{Ppl}{BurstR} + Bpl} \qquad (3\text{-}29)$$

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:

$$BurstR = \frac{1}{p+q} = \frac{Ppl/100}{p} = \frac{1 - Ppl/100}{q} \qquad (3\text{-}30)$$

As can be seen from Formula 3-29, the Effective Equipment Impairment Factor in case of *Ppl* = 0 (no packet-loss) is equal to the *Ie* value defined in Appendix I/G.113.

Please refer to Annex A/G.107 for the range of parameter-values the algorithm has been validated for.

3.6 Advantage factor, *A*

Due to the specific meaning of the advantage factor *A*, there is – consequently – no relation to all other transmission parameters. Some provisional values are given in Table 1.

Table 1/G.107 – 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

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

It should be noted that the values in Table 1, taken from ITU-T Rec. G.113 [4], are only provisional. The use of the factor *A* and its selected value in a specific application is up to the planner's decision. However, the values in Table 1 should be considered as absolute upper limits for *A*.

3.7 Default values

For all input parameters used in the algorithm of the E-model, the default values are listed in Table 2. It is strongly recommended to use these default values for all parameters which are not varied 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 2/G.107 – 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	(Note 2)
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	
Number of Quantization Distortion Units	qdu	–	1	1 ... 14	
Equipment Impairment Factor	Ie	–	0	0 ... 40	
Packet-loss Robustness Factor	Bpl	–	1	1 ... 40	(Note 3)
Random Packet-loss Probability	Ppl	%	0	0 ... 20	(Note 3)
Burst Ratio	BurstR	–	1	1 … 2	(Note 3)
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	
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.					

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

Due to the year 2000 revision the resulting rating *R* with all parameter values 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 of using the E-model

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 the 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 supposes that different kinds of impairments are additive on the scale of the transmission rating factor *R*. This feature has not been checked to a satisfying extent. Especially, 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, e.g., on the user group, and that the absolute values will change in 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 Rec. P.833 [6]. A new methodology for deriving equipment impairment factors from instrumental models such as P.862 [8] has been adopted as ITU-T Rec. P.834 [7].

– *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 (year 2000 revision) the Lombard effect (the fact that the speaker adopts his/her pronunciation and speaking level to the noise environment) is no longer disregarded. This has – in the 1998 version – lead 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 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 year 2000 revision of the E-model with all other parameters at their default values.

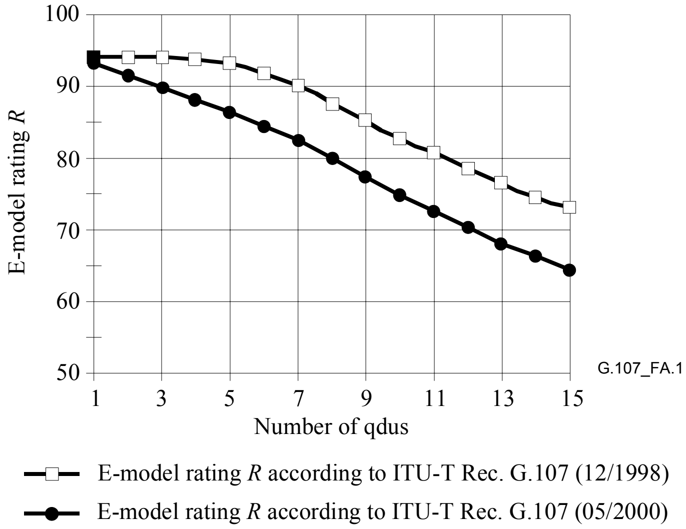

Figure A.1/G.107 – Relation between the number of qdu and E-model Rating *R*

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/G.113). As the aim is to reduce the amount of tabulated data for usage with the E-model, possibilities of replacing tabulated *Ies* for packet-loss by corresponding formulae were investigated. The chosen approach leads to results very similar to those previously defined as *Ie* for all codecs covered in the 2001 version of Appendix I/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 G.729(A) codec, but is assumed to be applicable also to the G.723.1, 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 higher Burst Ratios than 2.0, if 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 previous version of ITU-T Rec. G.107 (07/2002) 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 present revised version of the E-model algorithm, this observation is reflected by modifying the corresponding formula for *Ist* as a function of sidetone (*STMR*), see Equation 3-11.

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*, Equation 3-24. To remain consistent, the talker echo threshold from STMR > 15 dB (G.107, 07/2002) was extended to STMR > 20 dB (G.107, revised version). 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 (07/2002). The default value of *R* is 93.2 for both the previous and the current versions. The situation is depicted in Figure A.2.

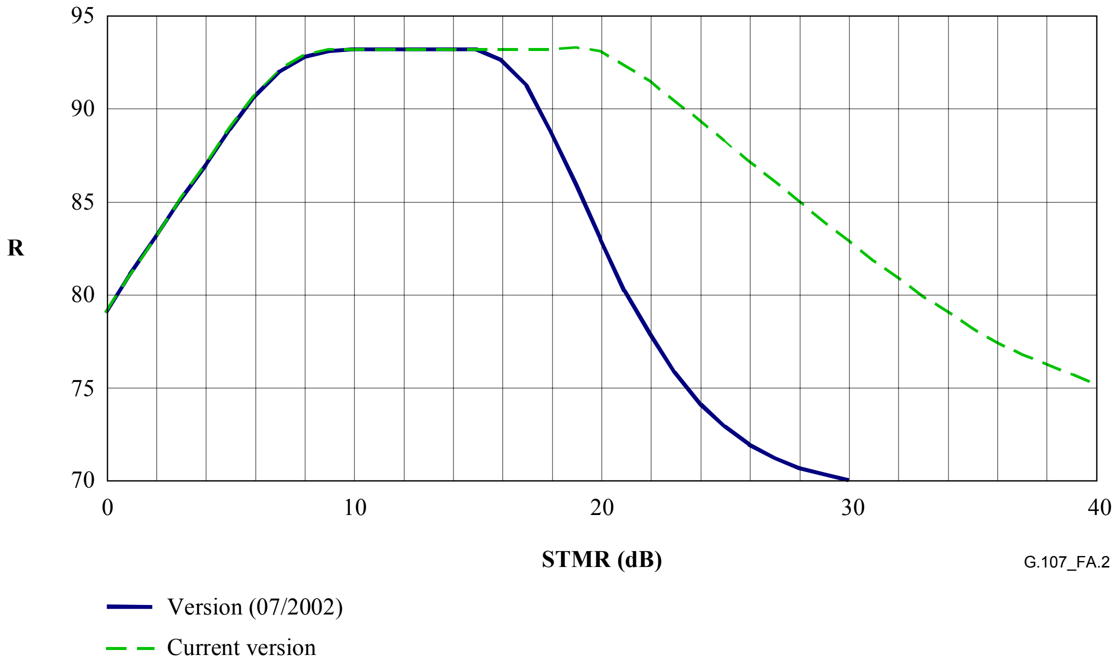

Figure A.2/G.107 – Comparison of *R* vs STMR for the current and the previous versions of the E-model algorithm

Annex B

Quality measures derived from the transmission rating factor *R*

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 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 formulae:

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 formula 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.

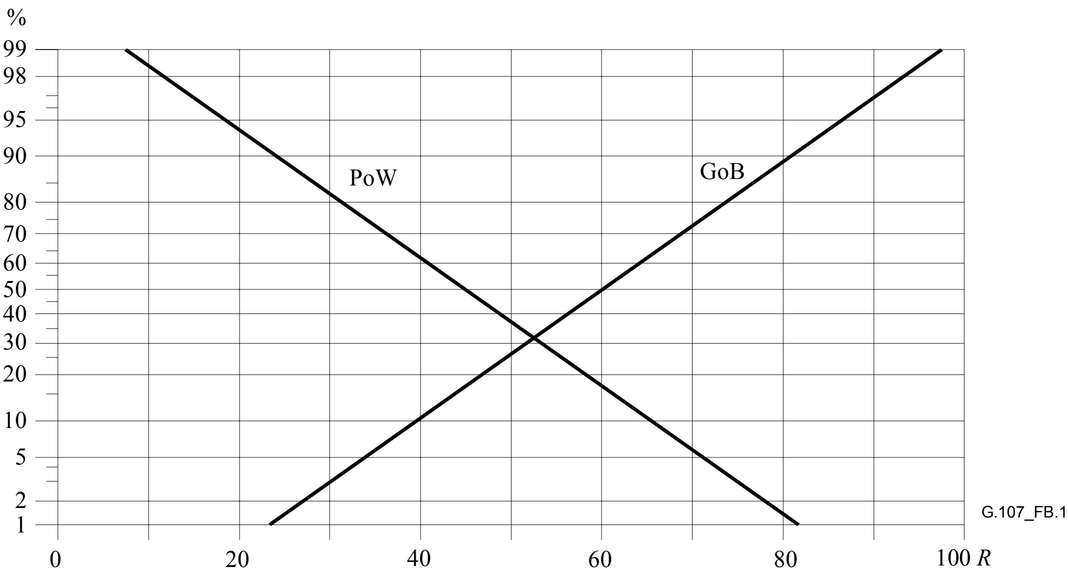

Figure B.1/G.107 – GoB (Good or Better) and PoW (Poor or Worse) as functions of rating factor *R*

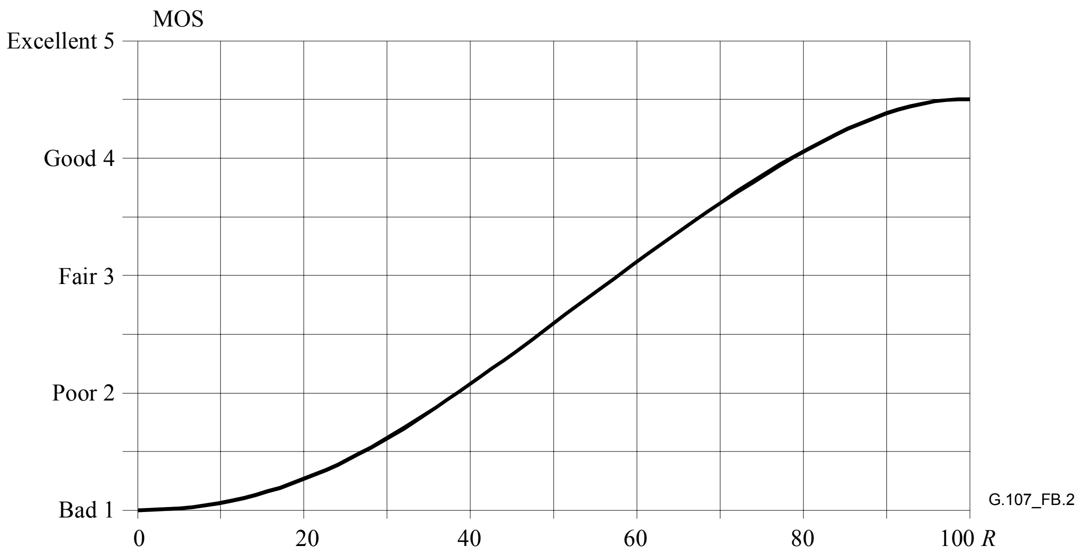

Figure B.2/G.107 – MOS_{CQE} as function of rating factor *R*

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/G.107 – 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/G.109 [3].

Annex C

Source code for G.107_5 in BASIC

```
1  CLS
2  PRINT "PROGRAM g107_4"
3  REM THIS VERSION IS CONFORM WITH THE ALGORITHM
4  REM DESCRIBED IN REC. G.107
5  REM PROGRAM WRITTEN BY N.O. JOHANNESSON
6  REM MODIFIED BY S. MOELLER, 1999; A. RAAKE, 2003, 2005
7  PRINT
8  PRINT "E-model, algorithm according to ITU-T Rec. G.107 (2003) Annex C,"
9  PRINT "for voice communication between side (S) and (R)."
10 PRINT
11 PRINT "The E-model gives an estimation of voice transmission quality"
12 PRINT "mouth-to-ear as perceived at Side (R). The model output is a"
13 PRINT "transmission rating factor R which can be transformed to give"
14 PRINT "estimates of customer opinion. Such estimates are only made for"
15 PRINT "transmission planning purposes and not for actual customer"
16 PRINT "opinion prediction. See ITU-T Rec. G.107 (2003) for more details."
17 PRINT
18 PRINT "Note: To continue program:"
19 PRINT "a) INPUT appropriate value as indicated on screen,"
20 PRINT "or"
21 PRINT "b) if no value is asked for, PRESS THE RETURN KEY."
22 PRINT
23 DIM X(100), Y(5, 100), Col(5), Pxo(5)
24 REM Set parameters at default values
25 GOSUB 29
26 REM Goto main menu
27 GOTO 199


29 REM Default Parameter Values, according to Tab.3/G.107 (lines 29-60)
30 SLR = 8
31 RLR = 2
32 STMRs = 15
33 Ds = 3
34 STMR = 15
35 Dr = 3
36 LSTRr = 18
37 TELR = 65
38 T = 0
39 WEPL = 110
40 Tr = 0
41 Ta = 0
42 Ie = 0
43 Bpl = 1
44 Ppl = 0
45 BurstR = 1
46 A = 0
47 Nc = -70
48 Ps = 35
49 Pr = 35
50 qdu = 1
55 Nfor = -64
60 RETURN


198 REM Main menu (lines 198-370)
199 PRINT
200 PRINT "Print current parameter values ", "=1"
210 PRINT "Input new parameter values     ", "=2"
```

```
220 PRINT "Compute table, one parameter   ", "=3"
230 PRINT "Set parameter at default values", "=4"
240 PRINT "Exit program                   ", "=5"
250 PRINT
260 INPUT Y1
270 CLS
280 IF Y1 = 1 THEN GOSUB 500
290 IF Y1 = 2 THEN GOSUB 1000
300 IF Y1 = 3 THEN GOSUB 2000
310 IF Y1 = 4 THEN GOSUB 30
320 IF Y1 = 5 THEN GOTO 9999
330 CLS
340 IF Y1 = 4 THEN PRINT , "Parameters set at default values !"
350 GOTO 199

500 REM SUB Print current parameter values (lines 500-700)
510 PRINT , "SLR="; SLR, "RLR="; RLR, "OLR= SLR + RLR="; SLR + RLR
520 PRINT , "Side (S): Ds="; Ds
530 PRINT , "Side (R): STMR="; STMR, "Dr="; Dr, "LSTR="; STMR + Dr
540 PRINT
550 PRINT , "TELR="; TELR, "Mean One-way Delay T ms="; T
560 PRINT , "WEPL="; WEPL, "Round-trip Delay Tr ms="; Tr
570 PRINT , "One-way Absolute Delay Ta ms="; Ta
580 PRINT
590 PRINT , "Noise Floor at Side (R) Nfor dBmp="; Nfor
600 PRINT , "Circuit Noise Nc dBm0p="; Nc
610 PRINT , "Room Noise, Side (S), Ps dB(A)="; Ps
620 PRINT , "Room Noise, Side (R), Pr dB(A)="; Pr
630 PRINT
640 PRINT , "qdu="; qdu
650 PRINT
660 PRINT , "Equipment Impairment Factor Ie="; Ie
661 PRINT
662 PRINT , "Packet-loss Robustness Factor Bpl="; Bpl
663 PRINT
664 PRINT , "Packet-loss Rate Ppl % ="; Ppl
665 PRINT
666 PRINT , "Burst Ratio ="; BurstR
667 PRINT
670 PRINT , "Advantage Factor A="; A
680 PRINT
690 INPUT C$
700 RETURN

1000 REM SUB Input Parameters (lines 1000-1270)
1020 CLS
1030 PRINT "Type designation of parameter for which the value is to be changed
!"
1031 PRINT
1032 PRINT "Note 1. New value of OLR is obtained indirectly, i.e. by new"
1033 PRINT "value of SLR or RLR. (OLR=SLR+RLR.)"
1034 PRINT
1035 PRINT "Note 2. New value of LSTR is obtained indirectly, i.e. by new"
1036 PRINT "value of STMR or Dr. (LSTR=STMR+Dr.)"
1037 PRINT
1040 INPUT "Parameter:"; A$
1050 INPUT "New Value="; Px
1060 PRINT A$; "="; Px
1070 IF ((A$ = "SLR") OR (A$ = "slr") OR (A$ = "Slr")) THEN SLR = Px
1080 IF ((A$ = "RLR") OR (A$ = "rlr") OR (A$ = "Rlr")) THEN RLR = Px
1090 IF ((A$ = "STMR") OR (A$ = "stmr") OR (A$ = "Stmr")) THEN STMR = Px
1100 IF ((A$ = "Dr") OR (A$ = "DR") OR (A$ = "dr")) THEN Dr = Px
1110 IF ((A$ = "Ds") OR (A$ = "DS") OR (A$ = "ds")) THEN Ds = Px
1120 IF ((A$ = "TELR") OR (A$ = "telr") OR (A$ = "Telr")) THEN TELR = Px
```

```
1130 IF ((A$ = "T") OR (A$ = "t")) THEN T = Px
1140 IF ((A$ = "WEPL") OR (A$ = "wepl") OR (A$ = "Wepl")) THEN WEPL = Px
1150 IF ((A$ = "Tr") OR (A$ = "TR") OR (A$ = "tr")) THEN Tr = Px
1160 IF ((A$ = "Ta") OR (A$ = "TA") OR (A$ = "ta")) THEN Ta = Px
1170 IF ((A$ = "Ie") OR (A$ = "IE") OR (A$ = "ie")) THEN Ie = Px
1171 IF ((A$ = "Bpl") OR (A$ = "BPL") OR (A$ = "bpl")) THEN Bpl = Px
1172 IF ((A$ = "Ppl") OR (A$ = "PPL") OR (A$ = "ppl")) THEN Ppl = Px
1173 IF ((A$ = "BurstR") OR (A$ = "BURSTR") OR (A$ = "burstr")) THEN BurstR = Px
1180 IF ((A$ = "A") OR (A$ = "a")) THEN A = Px
1190 IF ((A$ = "Nc") OR (A$ = "NC") OR (A$ = "nc")) THEN Nc = Px
1200 IF ((A$ = "Ps") OR (A$ = "PS") OR (A$ = "ps")) THEN Ps = Px
1210 IF ((A$ = "Pr") OR (A$ = "PR") OR (A$ = "pr")) THEN Pr = Px
1220 IF ((A$ = "qdu") OR (A$ = "QDU") OR (A$ = "Qdu")) THEN qdu = Px
1230 IF ((A$ = "Nfor") OR (A$ = "NFOR") OR (A$ = "nfor")) THEN Nfor = Px
1240 PRINT
1250 IF Y1 = 2 THEN INPUT "More parameters changed, Yes(1) or No(0)"; Ypar
1260 IF Ypar = 1 THEN GOTO 1020
1270 RETURN

2000 REM SUB Tabulate (lines 2000-3000)
2020 INPUT "Variable Parameter:"; A$
2030 PRINT "(To exit tabulation, put parameter value = 1000 !)"
2040 PRINT TAB(8); A$; TAB(18); "R"; TAB(28); "GOB %"; TAB(38); "POW %";
TAB(48); "MOS"
2050 INPUT Px
2060 IF Px = 1000 THEN GOTO 3000
2070 IF ((A$ = "SLR") OR (A$ = "slr") OR (A$ = "Slr")) THEN SLR = Px
2080 IF ((A$ = "RLR") OR (A$ = "rlr") OR (A$ = "Rlr")) THEN RLR = Px
2090 IF ((A$ = "STMR") OR (A$ = "stmr") OR (A$ = "Stmr")) THEN
2100   STMR = Px
2110   LSTR = STMR + Dr
2120 END IF
2130 IF ((A$ = "Dr") OR (A$ = "DR") OR (A$ = "dr")) THEN
2140   Dr = Px
2150   LSTR = STMR + Dr
2160 END IF
2170 IF ((A$ = "TELR") OR (A$ = "telr") OR (A$ = "Telr")) THEN TELR = Px
2180 IF ((A$ = "T") OR (A$ = "t")) THEN T = Px
2190 IF ((A$ = "WEPL") OR (A$ = "wepl") OR (A$ = "Wepl")) THEN WEPL = Px
2200 IF ((A$ = "Tr") OR (A$ = "TR") OR (A$ = "tr")) THEN Tr = Px
2210 IF ((A$ = "Ta") OR (A$ = "TA") OR (A$ = "ta")) THEN Ta = Px
2220 IF ((A$ = "Ie") OR (A$ = "IE") OR (A$ = "ie")) THEN Ie = Px
2221 IF ((A$ = "Bpl") OR (A$ = "BPL") OR (A$ = "bpl")) THEN Bpl = Px
2222 IF ((A$ = "Ppl") OR (A$ = "PPL") OR (A$ = "ppl")) THEN Ppl = Px
2223 IF ((A$ = "BurstR") OR (A$ = "BURSTR") OR (A$ = "burstr")) THEN BurstR = Px
2230 IF ((A$ = "A") OR (A$ = "a")) THEN A = Px
2240 IF ((A$ = "Nc") OR (A$ = "NC") OR (A$ = "nc")) THEN Nc = Px
2245 IF ((A$ = "Nfor") OR (A$ = "NFOR") OR (A$ = "nfor")) THEN Nfor = Px
2250 IF ((A$ = "Ps") OR (A$ = "PS") OR (A$ = "ps")) THEN Ps = Px
2260 IF ((A$ = "Pr") OR (A$ = "PR") OR (A$ = "pr")) THEN Pr = Px
2270 IF ((A$ = "qdu") OR (A$ = "QDU") OR (A$ = "Qdu")) THEN qdu = Px
2280 IF ((A$ = "Ie") OR (A$ = "IE") OR (A$ = "ie")) THEN Ie = Px
2290 IF ((A$ = "Ds") OR (A$ = "DS") OR (A$ = "ds")) THEN Ds = Px
2300 GOSUB 3500
2400 GOSUB 4000
2500 GOSUB 4100
2600 GOSUB 4200
2700 R = INT(R * 10 + .5) / 10
2800 PRINT TAB(8); Px; TAB(18); R; TAB(28); GOB; TAB(38); POW; TAB(48); MOS
2900 GOTO 2050
3000 RETURN

3500 REM Compute R (lines 3500-3880)
```

```
3509 REM Noise Summation, formulas (3) to (7)
3510 Nr1 = Ps – SLR – Ds – 100
3520 Nr1 = Nr1 + .004 * (Ps – SLR – RLR – Ds – 14) ^ 2
3530 LSTR = STMR + Dr
3540 Pro = Pr + 10 * LOG(1 + 10 ^ ((10 – LSTR) / 10)) / LOG(10)
3550 Pr1 = Pro + .008 * (Pro – 35) ^ 2
3560 Nr2 = Pr1 – 121 + RLR
3570 Nfo = Nfor + RLR
3580 No = 10 * LOG(10 ^ (Nr1 / 10) + 10 ^ (Nr2 / 10) + 10 ^ (Nc / 10) + 10 ^
(Nfo / 10)) / LOG(10)
3590 Nt = No – RLR

3599 REM Ro, formula (2)
3600 Ro = 15 – 1.5 * (SLR + No)

3609 REM Iolr, formulas (9) and (10)
3610 Xolr = SLR + RLR + .2 * (64 + Nt)
3620 Iolr = 20 * ((1 + (Xolr / 8) ^ 8) ^ (1 / 8) – Xolr / 8)

3629 REM Ist, formulas (11) and (12)
3630 STMRo = -10 * LOG(10 ^ (-STMR / 10) + 10 ^ (-TELR / 10) * EXP(-T / 4)) /
LOG(10)
3640 Ist = 12 * (1 + ((STMRo – 13) / 6) ^ 8) ^ (1 / 8)
3645 Ist = Ist – 28 * (1 + ((STMRo + 1) / 19.4) ^ 35) ^ (1 / 35)
3650 Ist = Ist – 13 * (1 + ((STMRo – 3) / 33) ^ 13) ^ (1 / 13) + 29

3659 REM Iq, formulas (13) to (17)
3660 IF qdu < 1 THEN qdu = 1
3670 Q = 37 – 15 * LOG(qdu) / LOG(10)
3680 G = 1.07 + .258 * Q + .0602 * Q ^ 2
3690 Iq = 15 * LOG(1 + 10 ^ ((Ro – 100) / 15) * 10 ^ (46 / 8.4 – G / 9) + 10 ^
(46 / 30 – G / 40)) / LOG(10)

3699 REM Is, formula (8)
3700 Isyn = Iolr + Ist + Iq

3709 REM TERV, formula (22)
3710 TERV = TELR + 6 * EXP(-.3 * T ^ 2) – 40 * LOG((1 + T / 10) / (1 + T / 150))
/ LOG(10)
3719 REM Modifications to satisfy formula (23)
3720 IF STMR < 9 THEN TERV = TERV + .5 * Ist

3729 REM Idte, formulas (19) to (21)
3730 Re = 80 + 2.5 * (TERV – 14)
3740 Roe = -1.5 * (No – RLR)
3750 Xdt = (Roe – Re) / 2
3760 Idte = Xdt + SQR(Xdt ^ 2 + 100)
3770 Idte = (Idte – 1) * (1 – EXP(-T))

3779 REM Modifications to satisfy formula (24)
3780 IF STMR > 20 THEN Idte = SQR(Idte ^ 2 + Ist ^ 2)

3789 REM Idle, formulas (25) and (26)
3790 Rle = 10.5 * (WEPL + 7) * (Tr + 1) ^ (-1 / 4)
3800 Xdl = (Ro – Rle) / 2
3810 Idle = Xdl + SQR(Xdl ^ 2 + 169)

3819 REM Idd, formulas (27) and (28)
3820 IF Ta < 100 THEN Idd = 0
3830 IF Ta = 100 THEN Idd = 0
3840 IF Ta > 100 THEN
       X = (LOG(Ta / 100)) / LOG(2)
       Idd = 25 * ((1 + X ^ 6) ^ (1 / 6) – 3 * (1 + (X / 3) ^ 6) ^ (1 / 6) + 2)
3850 END IF
```

```
3859 REM Id
3860 Id = Idte + Idle + Idd

3864 REM Inclusion of packet-loss: Ieef, formula (29)
3865 Ieef = Ie + (95 - Ie) * (Ppl / ((Ppl / BurstR) + Bpl))

3869 REM R, formula (1)
3870 R = Ro – Isyn – Id – Ieef + A
3880 RETURN

4000 REM Compute GOB, formula (B.2) (lines 4000-4050)
4010 Z# = (R – 60) / 16
4020 GOSUB 5000
4030 GOB = 100 * F#
4040 GOB = INT(GOB * 10 + .5) / 10
4050 RETURN

4100 REM Compute POW, formula (B.3) (lines 4100-4150)
4110 Z# = (R – 45) / 16
4120 GOSUB 5000
4130 POW = 100 * (1 – F#)
4140 POW = INT(POW * 10 + .5) / 10
4150 RETURN

4200 REM Compute MOS, formula (B.4) (lines 4200-4260)
4210 MOS = 1 + R * .035 + R * (R – 60) * (100 – R) * 7 * 10 ^ (-6)
4220 MOS = INT(MOS * 100 + .5) / 100
4230 IF R < 0 THEN MOS = 1
4240 IF MOS < 1 THEN MOS = 1
4250 IF R > 100 THEN MOS = 4.5
4260 RETURN

5000 REM Norm Distr F(Z), formula (B.1) (lines 5000-5130)
5010 S# = 0
5020 N% = 0
5030 H# = Z#
5040 S# = S# + H#
5050 H# = H# * (-1) * (Z#) ^ 2 * (2 * N% + 1) / ((N% + 1) * 2 * (2 * N% + 3))
5060 N% = N% + 1
5070 IF ABS(H#) < 10 ^ (-6) THEN GOTO 5090
5080 GOTO 5040
5090 S# = S# / (SQR(2 * 3.14159265#))
5100 F# = .5 + S#
5110 F# = INT(F# * 10 ^ 5 + .5) / 10 ^ 5
5120 REM PRINT "Z="; Z#, "F(Z)="; F#, "N="; N%
5130 RETURN
9999 END
```

Appendix I

Calculation of R from MOS_{CQE} values

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

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

with:

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

and:

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

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.

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