



The E-model

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E-Model Tutorial

Background

The E-Model ([ITU-T Rec. G.107](#) [1]) is a transmission planning tool that provides a prediction of the expected voice quality, as for a complete end-to-end (i.e. mouth-to-ear) telephone connection under conversational conditions. The E-Model takes into account impairments, in particular the impairment due to low bit-rate coding devices and one-way delay, as well as the "classical" telephony echo. It can be applied to assess the voice quality of wireline and wireless scenarios, based on circuit-switched and packet-switched connections.

The E-Model is not a true psycho-physical model, and cannot be used to accurately predict the absolute opinion of an individual user, the results are sufficiently accurate for transmission planning purposes. The E-model may also be useful for quality monitoring, as there are no agreed-upon methods and procedures for such an application of the E-model. The current methods for voice quality monitoring are described in [ITU-T Rec. P.561](#) and [ITU-T Rec. P.562](#), with the exception of the overall voice quality for Voice over IP, for which the E-model is not applicable.

The E-Model is based on modeling the results from a large number of subjective tests done in the past on a wide range of transmission scenarios. The output of the E-Model calculations is a scalar quality rating value known as the "Transmission Rating Factor, R". R can be transformed into other quality measures such as Mean Opinion Score (MOS-CQE [2]), Percentage Good or Better ((GoB) or Percentage Poor or Worse ((PoW). However, comparing these transformed measures with values of MOS, %GoB or %PoW from other sources, which may not have been obtained using the E-Model, is not recommended.

The E-Model is based on a mathematical algorithm, with which the individual transmission parameters are transformed into different impairment factors that are assumed to be additive on a psychological scale. The algorithm of the E-Model also takes into account the combination of impairments that occur simultaneously, as well as some masking effects. To the extent that impairments are present for which the E-Model is not applicable, E-model predictions may be inaccurate.

The relation between the different impairment factors and R is given by the equation:

$$R = R_0 - I_s - I_d - I_{e,eff} + A$$

where:

The term R_0 expresses the basic signal-to-noise ratio (received speech level relative to circuit and acoustic noise).

The term I_s represents all impairments that occur more or less simultaneously with the voice signal, such as: too low signal-to-noise ratio (SNR), non-optimum sidetone (STMR), quantization noise (qdu), etc.

The term I_d sums all impairments due to delay and echo effects.

The term $I_{e,eff}$ is an "effective equipment impairment factor", which represents impairments caused by low bit-rate coding devices and packet losses of random distribution. Values of I_e for specific codecs without packet loss are given in [ITU-T Rec. G.107](#). Values of I_e are transformed to $I_{e,eff}$ in case of random packet loss, using the E-model algorithm.

The term A is an "advantage factor", which allows for an "advantage of access" for certain systems relative to conventional systems for convenience. While all other impairment factors are subtracted from the basic signal-to-noise ratio R_0 , A is added to the basic signal-to-noise ratio to a certain amount. It can be used to take into account the fact that the user will tolerate some degradation in quality for the "advantage of access". Examples of such advantages are cordless and mobile systems or connectivity via multi-satellite hops.

Values of R fall in the range of $0 \leq R < 100$, with higher values indicating higher speech quality. Table 1, taken from [ITU-T Rec. G.107](#), defines the relationship between the E-Model Rating R to categories of speech transmission quality and to user satisfaction.

Table 1 - Definition of categories of speech transmission quality

Range of E-Model Rating R	Speech transmission quality category	User satisfaction
$90 \leq R < 100$	Best	Very satisfied
$80 \leq R < 90$	High	Satisfied

70 ≤ R < 80	Medium	Some users dissatisfied
60 ≤ R < 70	Low	Many users dissatisfied
50 ≤ R < 60	Poor	Nearly all users dissatisfied

NOTE 1 – Connections with E-Model Ratings R below 50 are not recommended.
NOTE 2 – Although the trend in transmission planning is to use E-Model Ratings R, equations to convert Ratings R into other metrics, e.g. %MOS, %GoB, PoW can be found in [ITU-T Rec. G.107 Annex B](#) [1].

Application of the E-Model

Figure 1 shows the basic reference configuration used by the E-model for estimating end to end speech quality, with the param

- SLR Send Loudness Rating
- RLR Receive Loudness Rating
- OLR Overall Loudness Rating¹
- STMR Sidetone Masking Rating²
- LSTR Listener Sidetone Rating²
- Ds D-value of telephone at send-side
- Dr D-value of telephone at receive-side²
- TELR Talker Echo Loudness Rating
- WEPL Weighted Echo Path Loss
- T Mean one way delay of the echo path
- Tr Roundtrip delay in a closed 4-wire loop
- Ta Absolute one-way delay in echo free connections
- qdu Number of quantization distortion units
- Ie Equipment impairment factor
- Ppl Random packet-loss probability
- Bpl Packet-loss robustness factor
- Nc Circuit noise referred to the 0 dBr-point
- Nfor Noise floor at the receive-side
- Ps Room noise at the send-side
- Pr Room noise at the receive-side
- A Advantage factor

The connection is basically divided into a "send side" (subscript _S) and a "receive side" (subscript _R) with a virtual centre referred to the user on the "receive side". The most important assumptions in the model is that the perceived quality is referred to the user on the "receive side", with this use conditions (talking and listening). This is important when inputting parameters relating to listening (loudness, equipment impairment, sidetone, delay, echo etc).

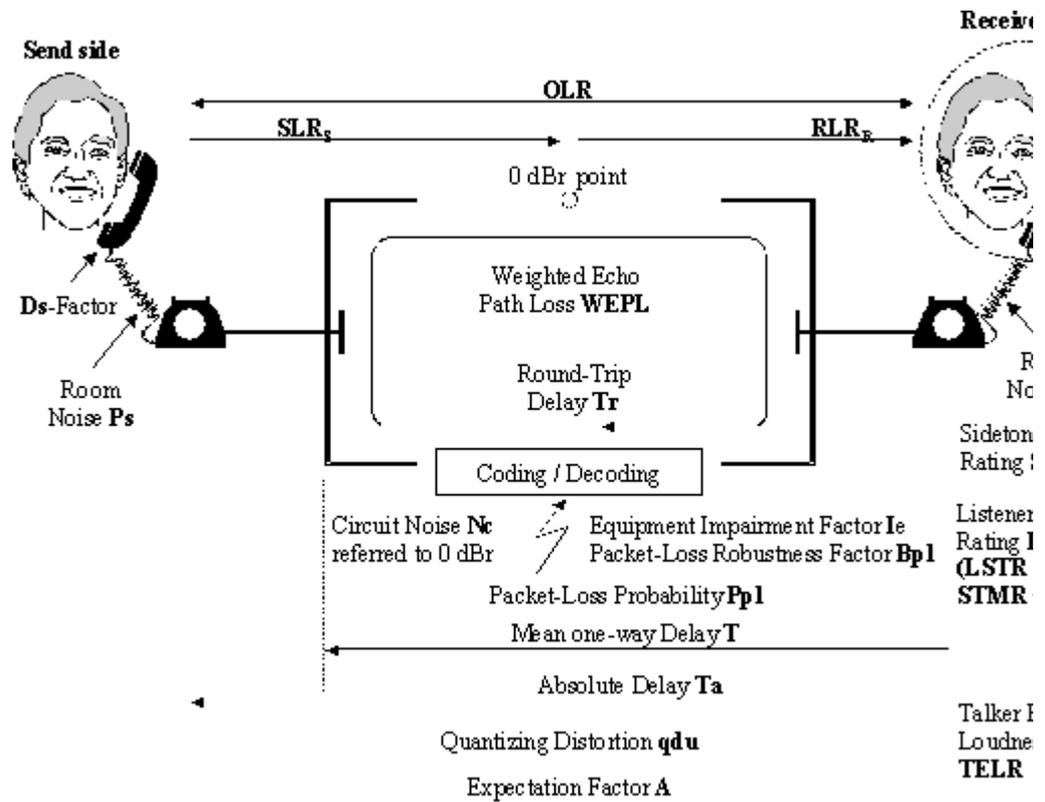


Figure 1 - Basic reference configuration of the E-Model

Values of SLR, RLR and Nc must be referred to a 0 dBr-point (i.e. any gains or losses taken into account). For the basic end to

To evaluate the impairment due to talker echo, the E-Model expects two parameters, the mean one-way echo path delay (T), and (TELR) of the echo path. It is very important to note that talker echo is referred to the user on the receive side. The value for TELR according to the basic formula:

$$TELR = SLR_R + EL + RLR_R$$

where: EL (Echo Loss), represents the effective echo return loss seen by the receive side.

To evaluate the impairment due to one-way delay in the absence of echo, two approaches may be used:

- i. Input the required value of Ta and set T = Tr = 0. This effectively turns "off" the echo algorithm and can be considered
- ii. It may be more realistic to input the required value of Ta and set T = Tr/2 = Ta, with TELR and WEPL set to their default values. This is a practical case of a near-ideal echo canceller.

Default values for all E-Model parameters and nominal parameter ranges are listed in Table 2. With these default values, the re

Table 2 - Default values and recommended ranges for the parameters

Parameter	Abbr.	Unit	Default value	Recommended range
Send Loudness Rating	SLR _S	dB	+8	0 to +12
Receive Loudness Rating	RLR _R	dB	+2	-5 to +10
Sidetone Masking Rating	STMR	dB	15	10 to 20
Listener Sidetone Rating	LSTR	dB	18	13 to 23
D-value of telephone, send side	D _s	-	3	-3 to 7
D-value of telephone receive side	D _r	-	3	-3 to 7
Talker Echo Loudness Rating	TELR	dB	65	5 to 75
Weighted Echo Path Loss	WEPL	dB	110	5 to 115
Mean one-way delay of the echo path	T	ms	0	0 to 100
Round trip delay in a 4-wire loop	Tr	ms	0	0 to 100
Absolute delay in echo free connections	Ta	ms	0	0 to 100
Number of Quantization distortion units	qdu	-	1	1 to 10
Equipment impairment factor	Ie	-	0	0 to 10
Packet-loss Robustness Factor	Bpl	-	1	1 to 10
Random Packet-loss Probability	Ppl	%	0	0 to 10
Circuit noise referred to 0 dBr-point	Nc	dBm0p	-70	-80 to -60

Noise floor at the receive Side	Nfor	dBmp	-64	
Room noise at the send side	Ps	dB(A)	35	35
Room noise at the receive side	Pr	dB(A)	35	35
Advantage factor	A	-	0	0
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 **Calculation Tool** that is provided here implements the basic E-Model calculation, based on the latest version [1].

The **Graphical tool** contains icons of the various elements of the network you wish to simulate. These icons have knowledge of element and feed the E-Model tool with appropriate values.

References

- [1] "The E-model, a computational model for use in transmission planning" **ITU-T Rec. G.107**
- [2] "Mean Opinion Score (MOS) terminology" **ITU-T Rec. P.800.1**
- [3] "Provisional planning values for the equipment impairment factor I_e and packet-loss robustness factor B_{pl} " **ITU-T Rec. G.108.1**
- [4] "Definition of categories of speech transmission quality" **ITU-T Rec. G.109**

Recommended bibliography

The following references provide more details on the E-Model and its application.

"Application of the E-Model - A Planning Guide" **ITU-T Rec. G.108**

"Guidance for assessing conversational speech transmission quality effects not covered by the E-Model" **ITU-T Rec. G.108.1**

"Transmission and Multiplexing (TM); Speech communication quality from mouth to ear for 3.1 kHz handset telephony across networks"

"The ETSI Computation Model: a Tool for Transmission Planning of Telephone Networks", N.O. Johannesson, IEEE Communications Magazine, 1998

"Voice Quality Recommendations for IP Telephony" TIA/EIA/TSB116 (March 2001) <http://www.tiaonline.org/standards/ip>.

¹ No direct input value; calculated as $OLR = SLR_S + RLR_R$.

² These parameters have a fixed relation by: $LSTR = STMR + Dr$.

³ The E-Model is not validated outside of these ranges