Planning and measuring the quality of telephone services, the E-model

VoIP is a very important service and, for VoIP, there exists a method to plan the network-level performance (delay and packet loss) into user-level quality (the Mean Opinion Score). This method is the E-model. With the E-model we can plan the network QoS of VoIP services in order to implement user-level Service Level Agreements.

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Example: quality of telephone services over IP

- The customer demand for Quality of Service is increasing, especially in the business segment
- Multimedia applications (for example VoIP or videoconference over IP) present stringent requirements on delay and packet loss
- With a best-Effort network, it is very difficult, if not impossible, to guarantee an adequate QoS for tyis type of services
- For example, the quality of VoIP is affected directly by the network performance, measured as packet delay adn packet loss
- In particular, the quality of the telephone service can be quantified by the MOS level (Mean Opinion Score)
- The MOS is a metric standardized by the ITU-T and it measures the satisfaction level of users
- A MOS level equal to 5 corresponds to completely satisfied users, while a MOS level equal to 1 corresponds to a population of unsatisfied users

Quality of telephone services: Mean Opinion Score

- The MOS is standardized in the ITU-T P.800 standard
- The scale of MOS values is:
  - Excellent: 5
  - Good: 4
  - Fair: 3
  - Poor: 2
  - Bad: 1
- Fractional values are admitted
- In order to measure the MOS level, the standard procedure specifies that a sample population must place calls through the tested system and each participant must score the quality of his/her conversation with a mark from 1 to 5
- Scores are then average to obtain the MOS value
- This procedure is expensive and time consuming
- Moreover, this procedure can be applied only to existing systems, and not in the project phase of a system

Quality of telephone services: Mean Opinion Score

- The average MOS value of classic PCM fixed telephony is 4.3
- This value is generally used as a reference to judge the MOS value of a telephone system
- The MOS level is affected, among other factors, by the adopted voice codec
- Each codec has an intrinsic MOS value and this value would be the MOS of a telephone coversation done with that codec, if all other conditions (noise, delay, echo, interference) are ideal
- In real life, the MOS value of a conversation place with a codec with an intrinsic MOS value equal to 3.9 will be lower than 3.9, as noise, delay, echo and interference are not null

Quality of telephone services: impairment factors

- There are several factors that can degrade the MOS value of a telephone conversation
  - Intrinsic MOS value of the voice codec
  - Impairment due to delay
  - Impairment due to signal degradation (distortion or, in the case of IP transport, packet loss)
  - Echo
- In the case of IP transport of telephone services, packet delay and packet loss are important impairment factors

Quality of telephone services: E-model

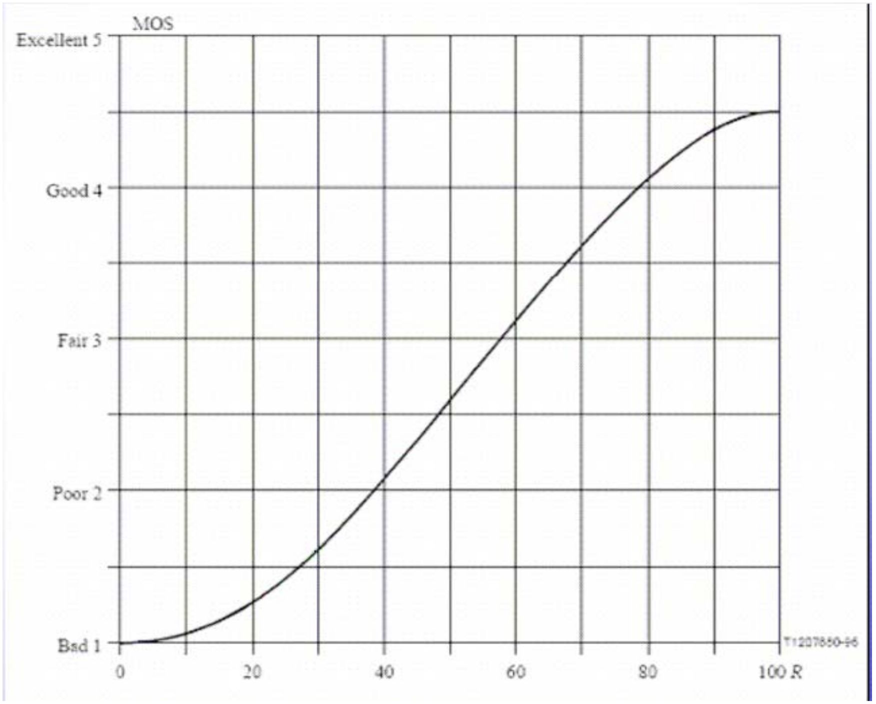
- In order to forecast the MOS value of a VoIP system, the ITU-T has standardized the E-model
- The E-model is a method to evaluate MOS (the quality that would be perceived by users) using as an input network quality performance indicators, such as packet delay and packet loss, voice codec ...
- With the E-model, it is possible to design the IP transport network in order to reach given values of packet delay and packet loss and, in turn, to use these values to obtain an estimation of MOS

Quality of telephone services: E-model

- The E-model is applied as follows
- The quality metric *R* is defined (ranging from 0 to 100)
- *R* is calculated and then it is remapped (with a non-linear function) on the MOS scale ranging from 1 to 5
- The *R* metric is defined as
  - ***R = Ro - Is - Id – Ie-eff + A***, where:
  - *Ro* is the basic signal/noise ratio
  - *Is* represents all impairments which occur more or less simultaneously with the voice signal, such as too loud speech level, quantization noise (*qdu*), etc.
  - *Id* sums all impairments due to delay and echo effects
  - *Ie-eff* is the equipment impairment factor (due to the codec). *Ie-eff* includes indirectly packet loss impairments
  - *A* is "advantage factor ", it represents an "advantage of access". While all other impairment factors are subtracted from the basic signal-to-noise ratio *Ro*, this value is added and thus compensates other impairments to a certain amount. It can be used to take into account the fact that the user will tolerate some decrease in transmission quality in exchange for the "advantage of access". Examples of such advantages are cordless and mobile systems or connections into hard-to-reach regions via multi satellite hops

Quality of telephone services: E-model

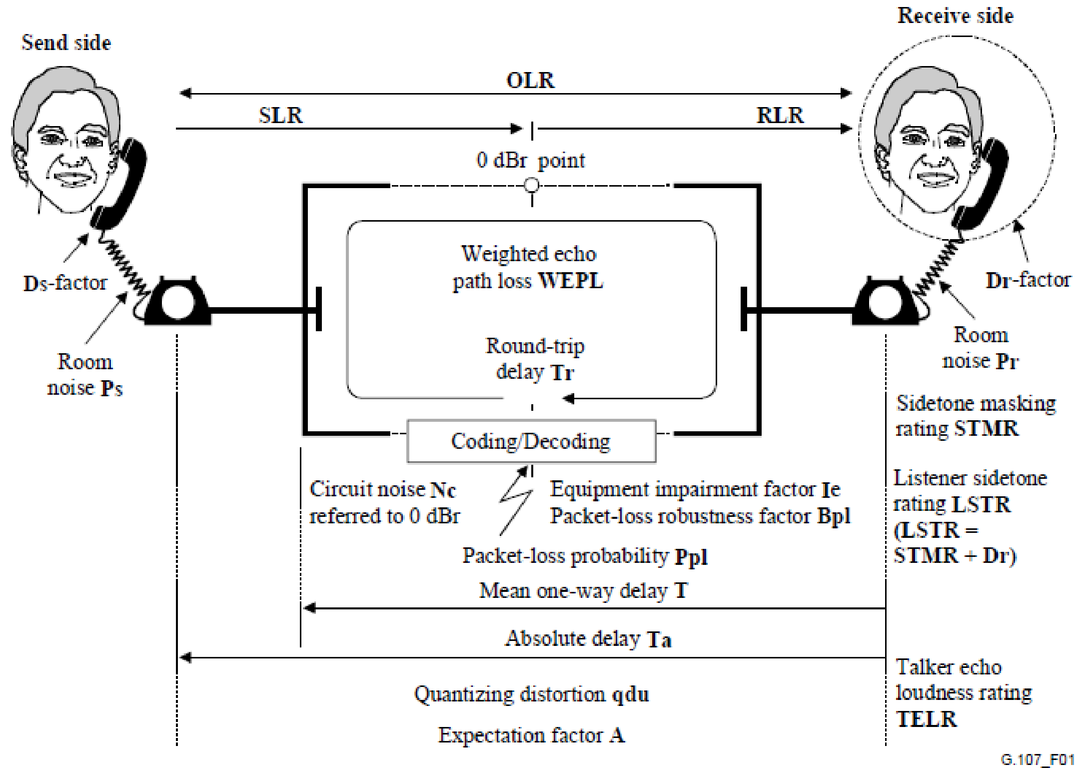
- Given the *R* value of a VoIP system, ranging from 0 to 100, the MOS value in a scale from 1 to 5 is obtained with the non linear function shown in the figure
- An approximated analytical expression of the depicted function is given in the box below



$$MOS = \begin{cases} R < 6{,}5 & 1 \\ 6{,}5 \le R \le 100 & 1 - \frac{7}{1000}R + \frac{7}{6250}R^2 - \frac{7}{1000000}R^3 \\ R > 100 & 4{,}5 \end{cases} \quad (4.9)$$

Reference connection of the E-model (OPTIONAL)

- Overall Loudness rating (OLR) is the sum of the Send Loudness Rating (SLR), of the telephone set at one end and the Receive Loudness Rating (RLR), of the other end set
  - OLR = SLR + RLR
- Impairments due to OLR may result from either too high or too low values of OLR
- The optimum value lies in the range from 6 to 10 dB
- The default value of OLR is 10 dB
- The default values of SLR is 8 dB
- The default value of RLR is 2 dB



Basic planning principles (OPTIONAL)

- Transmission planning based on the E-model provides a prediction of the expected quality – as perceived by the user – for an investigated connection
- Based on an end-to-end assessment for each transmission parameter (including the type and number of low bit-rate codecs) impairment values are derived
- This model accounts for low bit-rate coding devices as well as for impairments introduced by standard PCM coders and for impairments not directly related to digital processing (e.g., environmental noise)
- End-to-end speech transmission quality is expressed in terms of the E-model rating *R*, as a result of calculations with the E-model
- The E-model rating *R* can be transformed into other quality measures, such as mean opinion score (MOS) according to Annex B of [ITU-T G.107]

The impairment factor method (OPTIONAL)

- According to the impairment factor method, the fundamental principle of the E-model is based on a concept given in the description of the OPINE model [b-ITU-T P-series Supp.3], which states that transmission impairments can be transformed into "psychological factors"; and psychological factors on the psychological scale are additive.
- The impairment factor method allocates a value of impairment to each parameter and then allows the simple addition of these impairments to determine the overall impairment
- The result of any calculation with the E-model is the E-model rating *R*, which combines all transmission parameters relevant for the considered connection [ITU-T G.107]

Equipment impairment factor, Ie (OPTIONAL)

- The voice codec will contribute with distortions resulting in a decrease of the perceived speech transmission quality
- In contrast to the quantization distortion due to the standard 8-bit PCM coding (A-law or μ-law), these impairments cannot readily be quantified with a number of quantization distortion units *qdu* (see next four slides)
- The impairments introduced by different types of low bit-rate codecs are expressed by an "equipment impairment factor", *Ie*
- This factor should ideally cover all perceptively very diverse effects (distortion, sound degradation, degradation of voice quality, etc.) which can be associated with the codec used in the connection, except those already covered in another way by the E-model (e.g., overall attenuation, absolute delay)
- *Ie* values can be determined in auditory tests carried out according to a methodology given in [ITU-T P.833]

E-model inputs (OPTIONAL)

- Parameters currently used in planning
  - SLR Send Loudness Rating;
  - RLR Receive Loudness Rating;
  - OLR Overall Loudness Rating1;
  - TELR Talker Echo Loudness Rating;
  - T Mean one way delay of the echo path;
  - Ta Absolute delay in echo free connections;
  - qdu Number of quantization distortion units;
  - Ie Equipment impairment factor (low bit-rate codecs);
  - A Advantage factor
- These parameters are not used for planning, in general they are set to default values
  - STMR Sidetone Masking Rating2;
  - LSTR Listener Sidetone Rating2;
  - Ds D-value of telephone at send-side;
  - Dr D-value of telephone at receive-side2;
  - WEPL Weighted Echo Path Loss;
  - Tr Roundtrip delay in a closed 4-wire loop;
  - 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 note on the quantization distortion unit (qdu) (OPTIONAL)

- The quantization distortion due to the standard 8-bit PCM coding (A-law or μ-law), can be be quantified with a number of qdu
- A quantization distortion unit (qdu) was firstly defined in 1982 as equivalent to the distortion that results from a single encoding (A/D) and decoding (D/A) by an average G.711 codec
- Such a device has a signal/distortion ratio of 35 dB
- Conceptually, the number of qdus assigned to a particular PCM process should reflect the effect of only the quantization noise produced by the PCM process on speech
- In practice, the qdus must be determined from subjective measurements of real or simulated processes, where subjects will be exposed to not only the quantization noise but to other impairments produced by the digital process tested, including the departures from ideal frequency response in the anti-aliasing and reconstruction filters
- Formerly, the qdu was the basis for an end-to-end transmission planning of impairments due to digital processes, known as the "*14 qdu rule*"; this approach is no longer recommended by ITU-T

Basic signal-to-noise ratio, Ro – I (OPTIONAL)

- The basic signal-to-noise ratio *Ro* is defined by
  - $Ro = 15 - 1.5(SLR + No)$
- The term No [in dBm0p] is the power addition of different noise sources
  - $No = 10 \log (10^{Nc/10} + 10^{Nos/10} + 10^{Nor/10} + 10^{Nfo/10})$
- *Nc* [in dBm0p] is the sum of all circuit noise powers, all referred to the 0 dBr point
  - The default value of *Nc* is -70 and the permitted range is (-80, -40)
- *Nos* [in dBm0p] is the equivalent circuit noise at the 0 dBr point, caused by the room noise *Ps* at the send side
  - $Nos = Ps - SLR - Ds - 100 + 0.004(Ps - OLR - Ds - 14)^2$
- Where
  - $OLR = SLR + RLR$

Basic signal-to-noise ratio, Ro – II (OPTIONAL)

- 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
  - $Nor = RLR - 121 + Pre + 0.008(Pre - 35)^2$
- The term Pre [in dBm0p] is the "effective room noise" caused by the enhancement of Pr by the listener's sidetone path
  - $Pre = Pr + 10 \log (1 + 10^{(10 - LSTR)/10} + 10^{Nos/10} + 10^{Nor/10} + 10^{Nfo/10})$
- *Nfo* [in dBm0p] represents the "noise floor" at the receive side
  - $Nfo = Nfor + RLR$
  - with *Nfor* usually set to −64 dBmp

Simultaneous impairment factor, Is (I) (OPTIONAL)

- 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:
  - $Is = Iolr + Ist + Iq$
- Iolr represents the decrease in quality caused by too-low values of OLR and is given by:
  - $Iolr = 20\left(\left(\left(1+Xolr/8\right)^{8}\right)^{1/8}-Xolr/8\right)$
- Where Xolr is
  - $Xolr = OLR + 0.2\,(64 + No - RLR)$
- The factor Ist represents the impairment caused by non-optimum sidetone:
  - $Ist = 12\left(1+\left(\left(STMRo-13\right)/6\right)\right)^{8})^{1/8} - 28\left(1+\left(\left(STMRo+1\right)19.4\right)\right)^{35})^{1/35} - 13\left(1+\left(\left(STMRo-3\right)/33\right)\right)^{13})^{1/13}$

Simultaneous impairment factor, Is (II) (OPTIONAL)

- where
  - $STMRo = -10\log(10^{-STMR/10} + e^{-T/4}\,10^{-TELR/10})$
- The impairment factor Iq represents impairment caused by quantizing distortion:
  - $Iq = 15\log(1 + 10^{Y} + 10^{Z})$
- Where
  - $Y = (Ro - 100)/15 + 46/8.4 – G/9$
  - $Z = 46/30 – G/40$
  - $G = 1.07 + 0.258\,Q + 0.0602\,Q^2$
  - $Q = 37 – 15\log(qdu)$

Delay impairment factor, Id (I) (OPTIONAL)

- Also *Id*, the impairment factor representing all impairments due to delay of voice signals is further subdivided into the three factors:
  - (1) $Id = Idte + Idle + Idd$
- Where
  - (2) $Idte = ((Roe - Re)/2 + ((Roe - Re)^2/4 + 100)^{1/2} - 1)\,(1- e^{-T})$
  - Where
    - (3) $Roe = -1.5\,(No - RLR)$
    - (4) $Re = 80 + 2.5\,(TERV - 14)$
    - (5) $TERV = TELR - 40 \log((1+T/10) / (1+T/150)) + 6\, e^{-0.3T\wedge 2}$
- Particular cases
  - For $T < 1$ ms, Idte = 0
  - For *STMR* < 9 dB:
    - In (4) *TERV* is replaced by $TERVe = TERV + Ist/2$
  - For 9 dB <= *STMR* <= 20 dB, equations (1) to (5) apply
  - For *STMR* < 20 dB in (1) Idte is replaces by $Idtes = (Idte^2 + Ist^2)^{1/2}$

Delay impairment factor, Id (II) (OPTIONAL)

- The factor Idle represents impairments due to Listener Echo:
  - $Idle = ((Ro - Rle)/2 + ((Ro - Rle)^2/4 + 169)^{1/2})$
- Where
  - $Rle = 10.5\ (WEPL + 7)\ (Tr + 1)^{-0.25}$
- The factor *Idd* represents the impairment caused by too-long absolute delay *Ta*, which occurs even with perfect echo cancelling
  - For *Ta* <= 100 ms:
    – $Idd = 0$
  - For *Ta* > 100 ms
    – $Idd = 25\ ((1+X^6)^{1/6} - 3(1+(X/3)^6)^{1/6} + 2)$
    – Where
    – $X = \log(Ta / 100) / \log 2$

Equipment impairment factor, Ie (I) (OPTIONAL)

- 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
- The actually recommended values of *Ie* are listed in Appendix I/G.113
- 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

Equipment impairment factor, Ie (II) (OPTIONAL)

- With the Packet-loss Probability *Ppl*, *Ie-eff* is calculated using the formula:
  - *Ie-eff* = *Ie* + (95 - *Ie*) *Ppl* (*Ppl* / *BurstR* + *Ppl*)
- Where *BurstR* is the so-called Burst Ratio, which is defined as:
  - *BurstR* = Ave_observed_bursts / Ave_random_bursts
  - Where
    - Ave_observed_bursts is the average length of observed bursts in an arrival sequence
    - Ave_random_bursts is the 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* = 1/(*p*+*q*) = *Ppl* /(100 *p*)

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 the following table

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

Default values and permitted ranges

Table 2/G.107 – Default values and permitted ranges for the parameters

Parameter	Abbr.	Unit	Default value	Permitted range
Send Loudness Rating	SLR	dB	+8	0 ... +18
Receive Loudness Rating	RLR	dB	+2	–5 ... +14
Sidetone Masking Rating	STMR	dB	15	10 ... 20
Listener Sidetone Rating	LSTR	dB	18	13 ... 23
D-Value of Telephone, Send Side	Ds	–	3	–3 ... +3
D-Value of Telephone Receive Side	Dr	–	3	–3 ... +3
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
Random Packet-loss Probability	Ppl	%	0	0 ... 20
Burst Ratio	BurstR	–	1	1 … 2
Circuit Noise referred to 0 dBr-point	Nc	dBm0p	–70	–80 ... –40
Noise Floor at the Receive Side	Nfor	dBmp	–64	–
Room Noise at the Send Side	Ps	dB(A)	35	35 ... 85
Room Noise at the Receive Side	Pr	dB(A)	35	35 ... 85

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

Codec type	Reference	Operating rate [kbit/s]	*Ie* value
PCM (see Note)	G.711	64	0
ADPCM	G.726, G.727	40	2
	G.721, G.726, G.727	32	7
	G.726, G.727	24	25
	G.726, G.727	16	50
LD-CELP	G.728	16	7
		12.8	20
CS-ACELP	G.729	8	10
	G.729-A + VAD	8	11
VSELP	IS-54	8	20
ACELP	IS-641	7.4	10
QCELP	IS-96a	8	21
RCELP	IS-127	8	6
VSELP	Japanese PDC	6.7	24
RPE-LTP	GSM 06.10, full-rate	13	20
VSELP	GSM 06.20, half-rate	5.6	23
ACELP	GSM 06.60, enhanced full rate	12.2	5
ACELP	G.723.1	5.3	19
MP-MLQ	G.723.1	6.3	15

G.711 codec

- G.711 is the traditional PCM coded
- It is a waveform, codec, with a sampling rate of 8 kHz,and non linear quantization with 256 levels and A o μ law
- The bit rate is equal to 64 kbit/s
- The intrinsic MOS is greater than 4.3

ADPCM codecs

- ADPCM means adaptive differential PCM
- Instead of coding the amplitude of each sample, the difference between consecutive samples is coded
- For example, G.721 and G.726 are ADPCM codecs with speed equal to 16, 24, 32, 40 kbit/s
- A 32 kbit/s ADPCM codec has an intrinsic MOS value of about 4.0

CELP codecs

- CELP means code-excited linear predictive
- These codecs implement a complex mathematical model of the vocal segment of human beings
- The codec sends the parameters of the model rather than the amplitede of the samples of the speech waveform
- CELP codecs:
  - are complex
  - have in general a greater delay than waveform codecs
  - tend to have a lower MOS
  - Produce a bit stream with a significantly smaller rate than waveform codecs
- For example, the G.729 codec operated with a coding delay of 15 ms
- The bit rate is equal to 8 kbit/s with an intrinsic MOS of about 3.0
- The G.729 codec is more sensitive to delay and loss than waveform codecs