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Technical Report

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Design Guide;
Part 7: Design Guide for Elements of a TIPHON connection from an end-to-end speech transmission performance point of view

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Foreword

This Technical Report (TR) has been produced by ETSI Project Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON).

The present document is part 7 of a TIPHON Working Group 5 multi-part deliverable covering TIPHON Quality of Service (QoS) classification. The structure of this work is illustrated in figure 1:

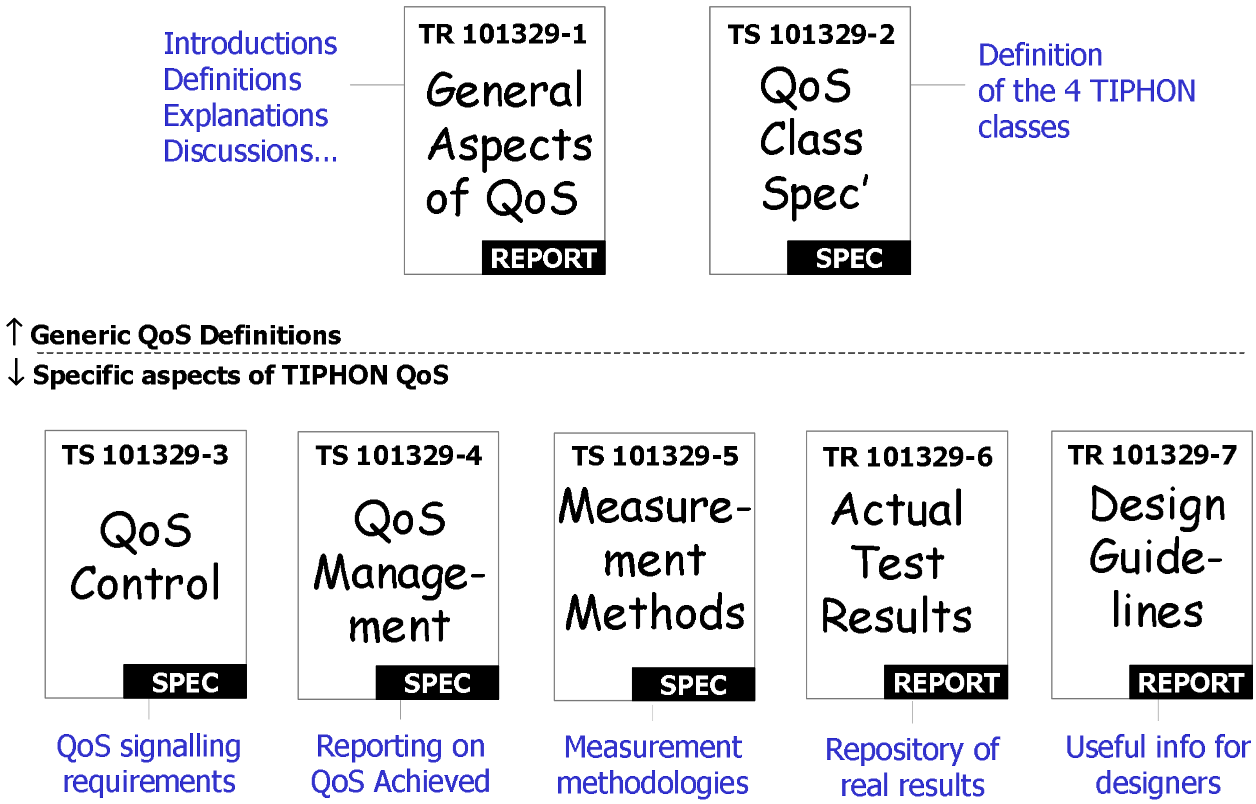


Figure 1: TIPHON WG5 QoS Documentation StructureFigure 1: Structure of TIPHON QoS Documentation

For a concise understanding of the guidance provided in the present document it is strongly recommended that the reader be aware of the content of the most recent version of TS 101 329-6 which is a repository of real results.

1 Scope

The present document provides guidance to TIPHON equipment manufacturers and network designers with respect to the consideration of speech performance issues in the practical design phases.

2 References

For the purposes of this Technical Report (TR) the following references apply:

[1] ETSI TS 101 329-2: "Telecommunications and Internet Protocol Harmonization over Networks (TIPHON); End to End Quality of Service in TIPHON Systems; Part 2: Definition of Quality of Service (QoS) Classes".

[2] ETSI TS 101 329-5: "TIPHON Release 3 Technology Compliance Specification; Part 5: Quality of Service (QoS) measurement methodologies".

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

dBm: power level with reference to 1 mW

dBm0: at the reference frequency (1 020 Hz), L dBm0 represents an absolute power level of L dBm measured at the transmission reference point (0 dBr point), and a level of L + x dBm measured at a point having a relative level of x dBr (See ITU-T Recommendation G.100, Annex A.4)

echo: unwanted signal delayed to such a degree that it is perceived as distinct from the wanted signal

talker echo: echo produced by reflection near the listener's end of a connection, and disturbing the talker

listener echo: echo produced by double reflected signals and disturbing the listener

loudness rating: as used in the G-Series Recommendations for planning; loudness rating is an **(LR)** objective measure of the loudness loss, i.e. a weighted, electro-acoustic loss between certain interfaces in the telephone network. If the circuit between the interfaces is subdivided into sections, the sum of the individual section LRs is equal to the total LR. In loudness rating contexts, the subscribers are represented from a measuring point of view by an artificial mouth and an artificial ear respectively, both being accurately specified.

overall loudness: loudness loss between the speaking subscriber's mouth and the **rating (OLR)** listening subscriber's ear via a connection

talker echo: loudness loss of the speaker's voice sound reaching his ear as a delayed **loudness rating** echo. See ITU-T Recommendation G.122, clause 4.2 and ITU-T Recommendation G.131, figure I.1 **(TELR)**

TCLw Terminal Coupling Loss weighted: weighted coupling loss between the receiving port and the sending port of a terminal due to acoustical coupling at the user interface, electrical coupling due to crosstalk in the handset cord or within the electrical circuits, seismic coupling through the mechanical parts of the terminal. For a digital handset it is commonly in the order of 40 dB to 46 dB.

TCLwst Weighted terminal coupling loss-single talk: weighted loss between Rin and Sout network interfaces when AEC is in normal operation, and when there is no signal coming from the user

TCLwdt Weighted terminal coupling loss-double talk: weighted loss between Rin and Sout network interfaces when AEC is in normal operation, and when the local user and the far-end user talk simultaneously

SLR Send Loudness Rating (from ITU-T Recommendation G.111): loudness loss between the speaking subscriber's mouth and an electric interface in the network. The loudness loss is defined here as the weighted (dB) average of driving sound pressure to measured voltage. The weighted mean value for ITU-T Recommendations G.111 and G.121 is 7 to 15 in the short term, 7 to 9 in the long term. The rating methodology is described in ITU-T Recommendations P.64, P.76, P.79.

RLR Receive Loudness Rating (from ITU-T Recommendation G.111): loudness loss between an electric interface in the network and the listening subscriber's ear. The loudness loss is here defined as the weighted (dB) average of driving e.m.f. to measured sound pressure. The weighted mean value for ITU-T Recommendations G.111 and G.121 is 1 to 6 in the short term, 1 to 3 in the long term. The rating methodology is described in ITU-T Recommendations P.64, P.76, P.79.

CLR Circuit loudness rating: loudness loss between two electrical interfaces in a connection or circuit, each interface terminated by its nominal impedance which may be complex

toll quality: In general "toll quality" is a term which is not well defined. Currently, there are two different views:

- ITU-T Recommendation G.109 provides the following guidance:

  "Finally, to relate the definitions provided by this Recommendation to concepts and terminology used in the past, a comment about "toll quality" is in order. "Toll quality" has been used by many different people to mean different things, but to network planners it really meant that technology being introduced into the network was robust to the effects of transmission impairments from other sources, and could thus be used in many configurations where inter-working with other systems would be necessary. In this context, the term "toll quality" does not have any absolute relation to speech transmission quality today, because, for example, the impairments of systems such as wireless access or packet-based transport will have the same impact regardless of whether on a local or a long-distance connection. Instead, the terminology provided here is recommended (i.e. "best" for R in [90, 100], "high" in [80, 90], "medium" in [70, 80])."

- Experts on low bit-rate coding (members of ITU-T Study Group 16 and SQEG) use the following explanation:

  "In summary, we define toll quality as equivalent to wire-line telephone quality. Basically the 32 kb/s ADPCM (ITU-T Recommendation G.726) is considered to be a toll quality coder, and when some low rate coders get standardized in ITU-T, 32 kb/s ADPCM is used as reference, and if a low rate coder produce equivalent performance to the 32 kb/s ADPCM, then this is considered to be toll quality."

Consequently, at this time the term "toll quality" should be considered as an internal term of speech coder experts only which is obsolete and which should be avoided in conjunction with the TIPHON QoS documentation. TIPHON equipment manufacturers and network designers should rather use the Quality Categories defined in ITU-T Recommendation G.109 or the QoS Classes specified in TIPHON (TS 101 329-2 [1]).

NOTE: Harmonization of the views regarding the term "toll quality" are envisaged to be discussed during the Study Period 2001-2004.

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ACR	Absolute Category Rating
ADSL	Asymmetric Digital Subscriber Line
ASL	Active Speech input Level
ATM	Asynchronous Transfer Mode
DTMF	Dual Tone Multi Frequency
GSM	Global System for Mobile communications
GSM HR	GSM Half Rate Speech Coder
GSM FR	GSM Full Rate Speech Coder
GSM EFR	GSM Enhanced Full Rate Speech Coder
ISDN	Integrated Services Digital Network
IP	Internet Protocol
ISP	Internet Service Provider
IWF	Inter Working Function
LAN	Local Area Network
MOS	Mean Opinion Score

PPP	Point to Point Protocol
NIC	Network Interface Card
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RSVP	Resource Reservation Set-Up Protocol
RTP	Real-Time Transport Protocol
SBM	Subnet Bandwidth Manager
SCN	Switched Communications Network
TCP	Transmission Control Protocol
TRM	Transmission Rating Model
UDP	User Datagram Protocol
VDSL	Very High Speed Digital Subscriber Line
VoIP	Voice over IP
VTOA	Voice and Telephony Over ATM
xDSL	ADSL, VDSL and other Digital Subscriber Line Techniques

4 General Considerations

The issues of end-to-end speech transmission quality need to be considered from various perspectives:

- Transmission planning
- User interaction
- Maintenance
- Verification

which are summarized in the following clauses.

4.1 Transmission planning

In order to deliver the intended end-to-end speech transmission quality in TIPHON scenarios transmission planning should be performed during the design phase of TIPHON related equipment. It is not sufficient to design equipment or networks just along the requirement limits of the respective TIPHON class.

An Advantage Factor A which is sometimes discussed for Internet-Telephony does not apply for business applications and subsequently does also not apply for TIPHON systems.

Any variation of transmission parameters should only be judged on the basis of E-model calculations for critical end-to-end connections. Any assumption whether or whether not a specific parameter variation will be perceived by the user should always be based on E-model calculations.

Special care should be taken with devices which dynamically vary one or more transmission parameters, e.g., Automatic Level Control (ALC) devices; experience with such devices have shown that they have the potential to impact end-to-end speech transmission quality severely.

4.2 User interaction

User interaction with regard to the change of certain transmission parameters may be provided by equipment which forms part of a TIPHON connection, e.g., a TIPHON terminal may include a PC client software which provides adjustment of Loudness Rating to the user (see clause 5.1.2 for further guidance).

It should be understood that this is one major difference between TIPHON networks and more traditionally planned networks where user interaction was of minor influence on end-to-end speech transmission quality.

4.3 Maintenance

After TIPHON equipment and networks have been designed, planned and rendered operative in compliance with one the TIPHON QoS classes it might - nevertheless - occur that users complain about too low speech quality.

In such cases, it is very important to be able to carry through a diagnosis of end-to-end speech transmission performance. For that it will be needed to keep track of all parameter changes (e.g., of Send and Receive Loudness Rating) carried out either automatically or by user interaction.

This should be considered already during the design phase of TIPHON equipment and networks, e.g., by providing tools to set parameter back to default values or by providing a log file function.

4.4 Verification

Even if a specific TIPHON system has been operated for some time at the desired level of customer satisfaction it may then be required to double-check the end-to-end speech transmission quality. This may become necessary, e.g., after a number of customer complaints have been received by the operator of the TIPHON system.

During such a verification phase it is of major importance to have easily access to the actual settings of all major transmission parameters - including those which were accessible to the user.

5 Guidance on main Transmission Parameters

5.1 Loudness Ratings

5.1.1 General Considerations

5.1.2 IP Terminals

This is for further study.

5.1.3 IP Gateways

This is for further study.

5.1.4 Network Elements

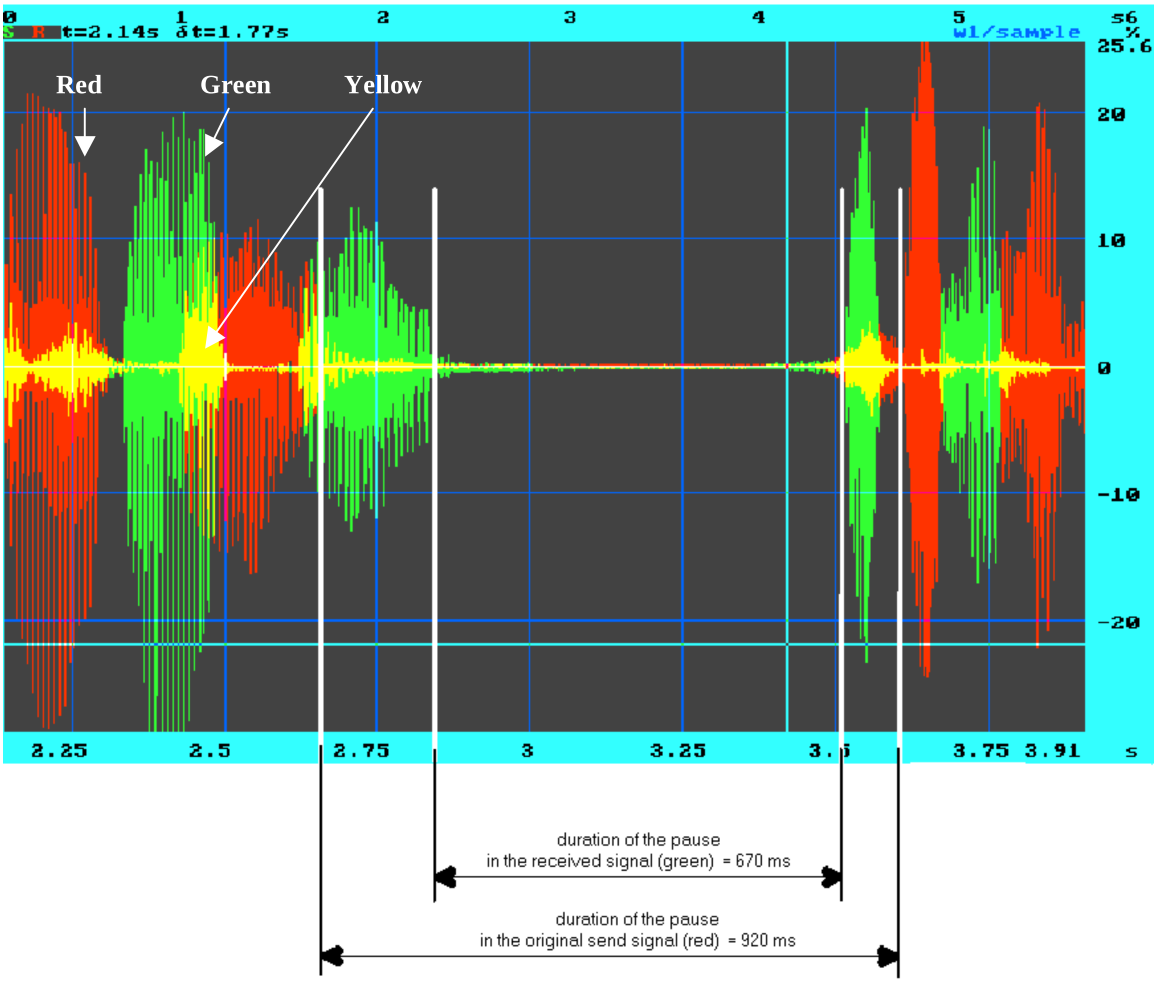
This is for further study.

5.2 Mean One-Way Delay

5.2.1 General Considerations

5.2.1.1 Delay Jitter

Figure 2 shows some measurement result regarding delay jitter:



Red = Original send signal; Green = receive signal (transmitted via the test set-up)

NOTE: due to the selected measurement mode the original send signal (red) is displayed 500 ms later and reduced by 6 dB in level than in reality. As a result the mean one-way delay is strongly variable between 450 ms and 700 ms.

Figure 2

Figure 2 shows the observed delay jitter. In the original speech sample (red signal) the pause sequence between the two words has approx. 920 ms duration; after having sent this speech sample over the test set-up the received speech (green signal) has a pause sequence of only 670 ms duration; i.e. the jitter amplitude is 250 ms.

5.2.2 IP Terminals

This is for further study.

5.2.3 IP Gateways

This is for further study.

5.2.4 Network Elements

This is for further study.

5.3 Echo Loss, Echo Cancellation

It is assumed that TIPHON end-to-end connections are equipped with proper echo control.

5.3.1 General Considerations

The determination of values for the Echo Loss within the SCN is based on reasonable assumptions.

With the understanding that the SCNs today are mostly fully digital with either analogue or digital terminals, two major cases with respect to Echo Loss appear:

- Analogue terminal with a Transhybrid Loss in the local exchange of 25 dB.
- Digital (wired) terminals in accordance with the long term objective given in I-ETS 300 245-2 and DECT terminals (base station digitally connected to the network) with a TCLw of 46 dB.

The acoustical properties of the terminals mentioned above are similar to those used in auditory evaluations according to ITU-T Recommendation P.800, i.e. the handset is assumed to be in accordance with ITU-T Recommendation P.310.

Echo canceller tail delay

For practical application the maximum tail delay of the echo canceller must be reduced by 6 ms to 8 ms; the remaining value must be divided by two (tail delay is a round-trip value) in order to get the correct value of the maximum mean one-way delay which an EC can handle.

Small and Medium sized Private Networks (e.g., Corporate Networks) which in a PSTN environment are operated without the deployment of separate echo cancellers, typically add 10 ... 20 ms to the mean one-way delay of a connection.

5.3.2 IP Terminals

This is for further study.

5.3.3 IP Gateways

This is for further study.

5.3.4 Network Elements

This is for further study.

5.4 Coding Distortion

5.4.1 General Considerations

The Impairment Factor method, used by the E-model of ITU-T Recommendation G.107, is now recommended for all systems except PCM according to ITU-T Recommendation G.711 the earlier method that used Quantization Distortion Units is no longer recommended.

For information values for the equipment impairment factor of various coding devices can be found in Appendix I to ITU-T Recommendation G.113; this Appendix I is intended to be updated on a frequent basis.

Table 1 gives an overview of the interdependency between end-to-end mean one-way delay and E-model Rating R for various types of codecs (including examples of packet loss conditions):

Table 1: R-values for indicated combinations of Ie and end-to-end mean one-way delay

Ie=	0	5	7	10	15	19	19	20	26
	G.711	GSM-EFR	G.726@32	G.729	G.723.1@6.3	G.729A+VAD w/ 2% loss	G.723.1@5.3	GSM-FR	G.729A+VAD w/ 4% loss
ms			G.728@16				G.723.1@6.3 +VAD w/ 1% loss	IS-54	
~0	94		87						
50	93		86	83		74			67
100	92	87	85	82	77	73	73	72	66
150	90	85	83	80	75	71	71	70	64
200	87	82	80	77	72	68	68	67	61
250	80	75	73	70	65	61	61	60	54
300	74	69	67	64	59	55	55	54	48
350	68	63	61	58	53	49	49	48	42
400	63	58	56	53	48	44	44	43	37
450	59	54	52	49	44	40	40	39	33

NOTE 1: R-values in this table have been calculated using the indicated values for Ie and T (T=Ta=Tr/2) along with the default values from Table 3 of ITU-T Recommendation G.107 for all other parameters.
NOTE 2: Unless indicated otherwise, examples do not include packet loss or Voice Activity Detection (VAD).
NOTE 3: Blackened cells indicate combinations of delay and codec that are impossible to realize.

5.4.2 IP Terminals

This is for further study.

5.4.3 IP Gateways

This is for further study.

5.4.4 Network Elements

This is for further study.

5.5 Speech Processing other than Coding

5.5.1 General Considerations

Recently, a number of speech processing devices have been defined, standardized and deployed in the network environment, the impact on speech transmission performance of which has not been considered in the E-model, yet. In the presence of long delay the interactions of specific modern network equipment, e.g.:

- Automatic Level Control (ALC);
- Digital Circuit Multiplication Equipment (DCME);
- Discontinuous Transmission (DTX);
- Echo Control Devices (ECD);
- Noise suppressors (NS);
- Voice Activity Detectors (VAD);

contribute with additional impairments - not only but mainly - during double talk situations. In order to include this impact in highly interactive communication situations into the E-Model calculation, it is desirable to estimate a preliminary Ie value for this effect.

The syllable cut-off and echo disturbances in double-talk situation which will frequently occur in highly interactive communication situations - if long delay is present - can be compared with the effect of handsfree telephony. For handsfree telephony it is well known that the impact perceived by the user lies in a range Ie = 10 .... 20.

Hence an equipment impairment value within this range may be chosen by the responsible transmission planner in order to consider this impact.

It should be noted that the impairments described here are independent from (and additional to) the impairments caused by long delay (pure delay, see clause 6.2.2).

NOTE: The consideration of this effect in the E-model is under study in ITU-T.

5.5.2 IP Terminals

This for further study.

5.5.3 IP Gateways

This is for further study.

5.5.4 Network Elements

This is for further study.

5.6 Transcoding in Network Elements

Transcoding in network elements should be avoided even in exceptional cases, because of the increase in coding distortion and coding delay (summing up of the values of both codecs).

6 Calculation Examples related to the Main Transmission Parameters

This clause provides guidance how transmission planning using the E-Model can be applied to TIPHON networks.

Examples for all TIPHON Scenarios and for all main transmission parameters are provided together with some background material. For further instructions see EG 201 050, ITU-T Recommendations G.108 and G.108.01.

In principle, the transmission planner has to work in different steps to fulfil his task. The first step is to draft a reference configuration of the network under consideration. Figure 3 shows - as an example - what such a reference configuration for Scenario 3 might look like:

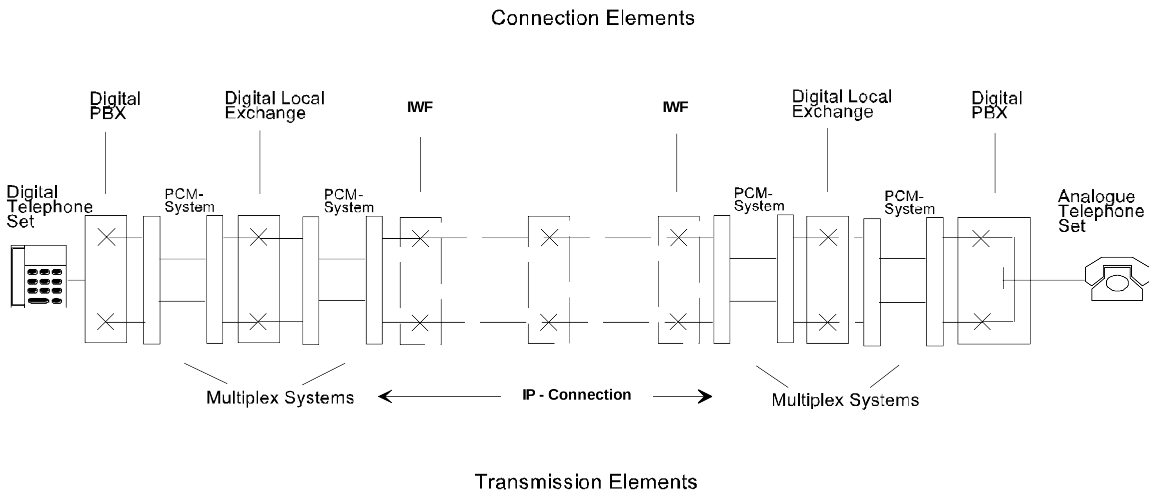


Figure 3: Reference configuration for a fully digital connection including an IP section and a terminating hybrid

The second step is to take into account the impairments with respect to speech quality; figure 4 gives some examples.

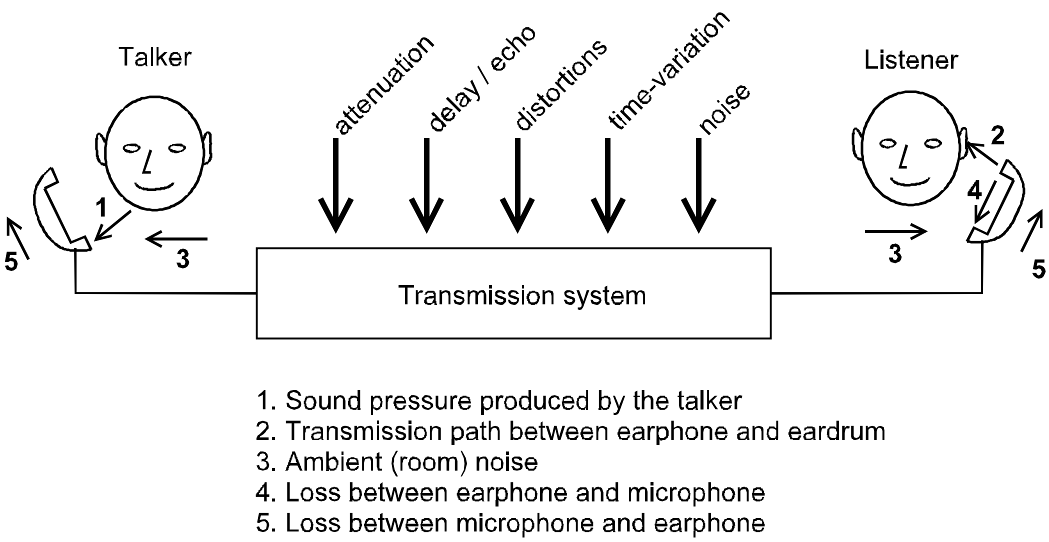


Figure 4: Transmission parameters influencing the quality of a handset telephone connection

The last step is to produce a working configuration - as shown in figure 5 - having all relevant parameters and required values at hand.

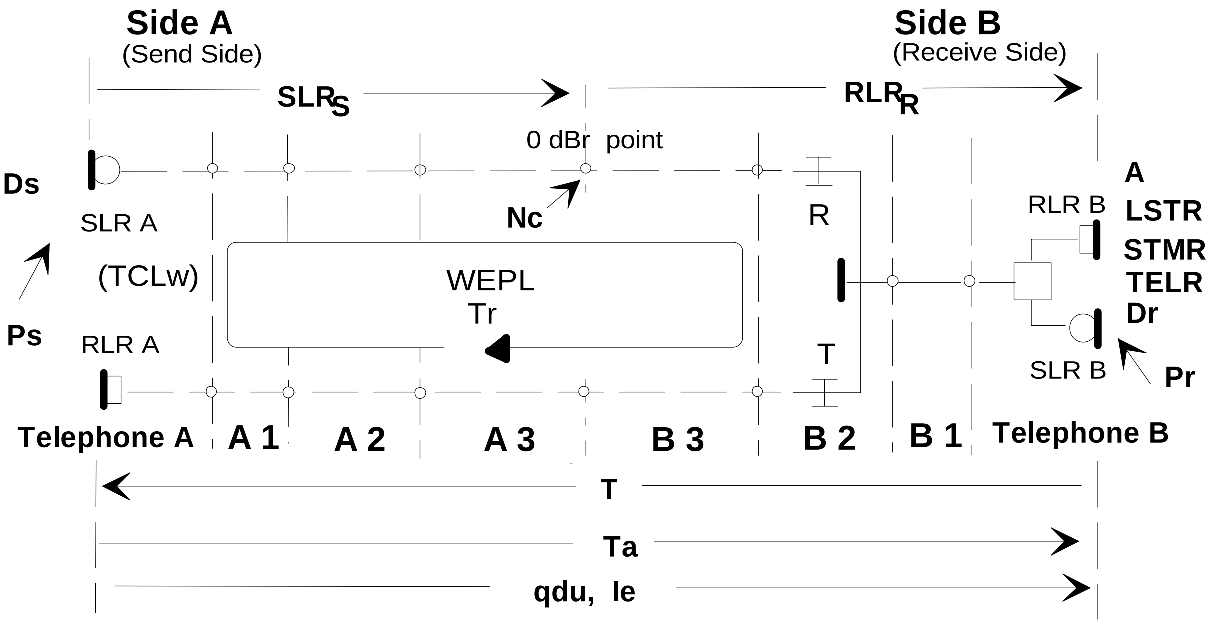


Figure 5: Working configuration for 4-wire/2-wire connections

All symbols and abbreviations are as defined in EG 201 050, whereas the IP section including the IWFs is represented by the segments A3 and B3.

It should be noted, that this clause provides examples only for the TIPHON QoS classes #1, #2 and #3. TIPHON QoS class #4 is explicitly omitted here, since it can - by definition - only be complied with by wideband speech transmission systems and since the E-model algorithm according to ITU-T Recommendation G.107 does not apply to wideband.

6.1 Examples with respect to Loudness Ratings

This is for further study.

6.2 Examples with respect to Mean One-way Delay

The term "mean one-way delay" is defined as half the sum of the transmission time in both transmission directions of a connection.

With a static de-jitter buffer at the receive side the end-to-end delay between the speaker and listener is assumed to be constant for the duration of a call and jitter will have been removed from the system.

Routing through the network (e.g. the number of hops) will increase transmission delay. Traffic congestion on the network will lead to packet loss and delay jitter.

Putting the available delay figures into context, it may be noted that network routing delays should in practice be quite small, of the order of a few ms, so that most of the delay is available for propagation. Taking this into account, and noting that, for planning purposes the delay in optical fibre systems is taken as 5 µs/km, the best combination of Terminal Mode and Network Class will result in a TIPHON end-to-end QoS of "High" for a connection up to about 8 000 km.

6.2.1 Delay due to speech processing and packetization

One of the mostly discussed items during the design phase of a piece of VoIP equipment is the "real" load of the network, e.g., the LAN, as a function of chosen speech codec.

However, this discussion considers only a part of a multi-dimensional problem, since the "parameters":

- codec type;
- size of the IP header;
- size of the IP packet (= payload);
- additional delay due the coding and packetizing process;
- robustness with regard to packet loss;

are all closely inter-dependent to each other.

The following figure is intended to provide some guidance with respect to this inter-dependency.

The y-axis of both diagrams is labelled "gross bit-rate" which is the total number of bits required to transport one second of speech (source) signal via a network.

The x-axis of the left diagram reports the total number of bytes payload per IP packet, whereas the figures denoted along the graphs present the number of coded speech frames per IP packet for each codec type.

For the right diagram, the only difference is, that the x-axis reports the additional minimum delay incurred due to speech processing and packetization.

Finally, the number of bytes which are to be assigned to the "IP header" is shown as an input value in between both diagrams.

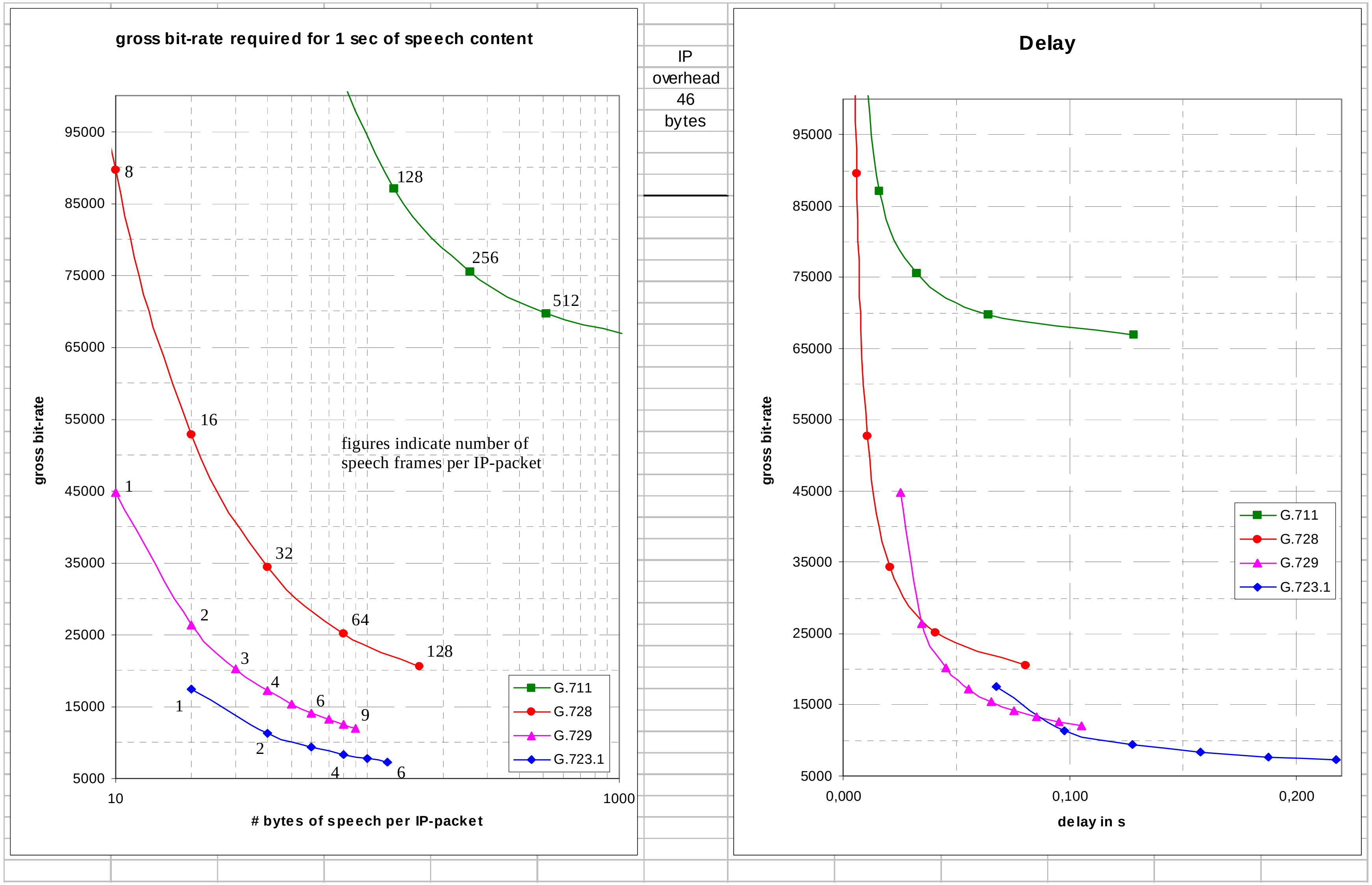

gross bit-rate required for 1 sec of speech content
gross bit-rate
bytes of speech per IP-packet
figures indicate number of speech frames per IP-packet
95000
85000
75000
65000
55000
45000
35000
25000
15000
5000
10
1000
128
256
512
8
16
32
64
128
1
2
3
4
6
9
1
2
4
6
G.711
G.728
G.729
G.723.1
IP overhead 46 bytes
Delay
gross bit-rate
delay in s
0,000
0,100
0,200
G.711
G.728
G.729
G.723.1

Figure 6

NOTE: Given suitable optimization of bandwidth, almost any link mechanism will suffice for audio communication (from high-performance modems upwards). The problems start to arise when the audio communication is concurrent with data collaboration. If the data bandwidth demands are too high, either the audio will suffer, or the data communications will break down (depending on how well optimized the communication is for real-time). Obviously, higher bandwidth links (like ISDN, Cable, ADSL) can mitigate this problem.

6.2.2 Planning examples regarding the occurrence of long delay

The following clauses are intended to provide detailed guidance on the effect of long delay, on the applicability and usage of the E-Model under such circumstances and explicitly on the proper use of the Advantage Factor A in such situations.

6.2.2.1 Introduction

The transmission impairments associated with long delay are best analysed by separating the echo-induced degradation and the subjective difficulty due to pure delay. Appropriate use of echo cancellers has been shown to indeed provide international or national satellite connections yielding listening-only quality (under echo-free conditions) and performance practically equivalent to the terrestrial connections for telephony. Note, that these results refer to electric echo only and additional studies are necessary to determine the effect of acoustic echo.

Thus, under echo-free conditions, the dominant impairments with respect to conversational quality are associated with the pure delay component.

Recently presented information suggests that:

- The effects of pure delay (no echo) on conversation dynamics can be detected well below 400 ms one-way delay if subjective experiments utilize highly interactive tasks and subjective measures related to specific conversational difficulties, such as ability to interrupt, are used.

- The effects of pure delay (no echo) on speech quality appear to moderately increase as the delay is increased.

However, as a standard set of conversational tests has not been agreed upon, experimental results may vary depending on the type of test carried out. Furthermore, obtained experimental results depend upon the type of conversational interactivity selected to evaluate the impact of delay.

Thus, designers and network planners must determine the type of services, and, hence, the communication interactivity needs that will be supported, if the performance of the system is to be evaluated appropriately.

6.2.2.1.1 Application of the Advantage Factor A with respect to the following examples

In conjunction with their task mentioned above, designers and network planners may wish to consider the "Advantage Factor A" which takes into account advantages of using a specific service, in this very case particularly an "Advantage of Access". This parameter has been introduced into transmission planning for the first time via the E-Model (ETR 250 and ITU-T Recommendation G.107). This factor enables the planner to take into account the fact that customers may accept some decrease in quality for access advantage: e.g. mobility or connections into hard-to-reach regions.

NOTE: In conjunction with the examples provided herein the term "region" is to be understood as referring to a geographical area, not further specifying related properties or standards.

Provisional values for A are given in ITU-T Recommendation G.107.

While some investigations (e.g., for GSM) tend to confirm the values for A given in ITU-T Recommendation G.107, these values have not been fully verified by auditory tests to date and are, therefore, provisional. In addition, it has been shown that the value for the Advantage Factor A for a specific service or technology will be strongly time variant (see Appendix II to ITU-T Recommendation G.113). Therefore, the Advantage Factor A should be used with care and with respect to the business interest of the respective network customer. The use of the Advantage Factor in transmission planning of networks and the selected values are subject to the planner's individual decision; however, the values for A given in ITU-T Recommendation G.107 should be considered as the maximum upper limit.

In cases, where connections to the very same destination under consideration are available with long delay as well as with significantly shorter delay (hereafter referred to as "Competition"), there will be - in general - no justification for the application of the Advantage Factor A (due to delay).

In cases, where connections to a specific destination under consideration are restricted to and available with long delay only (hereafter referred to as "Hard-to-reach"), there will be - in general - a certain justification for the application of the Advantage Factor A (due to delay).

For the purposes of the examples in clauses 6.2.2.3.2 and 6.2.2.3.3 the Advantage Factor A has been estimated by the responsible transmission planner to be A = 12. Note, that this is not a general planning rule, but a business and customer related decision for this single example case. In the course of another planning task the transmission planner may decide on a different value of the Advantage Factor.

Since the Advantage Factor compensates for a decrease in quality in comparison to a possible advantage of access, it should not be applied if no decrease in quality occurs. Hence, if there is no difference in quality between two services "Competition" and "Hard-to-reach" the application of an Advantage Factor should not be justified (see clause 6.2.2.3.1).

See ITU-T Recommendations G.107 and G.108 for further details.

6.2.2.1.2 Distinction between different communication situations for the following examples with regard to the grade of interactivity between the two parties

For the purpose of the following examples three different types of communication situations can be identified:

- Listening-only communication situation.

This kind of communication situation is considered as untypical and may occur in specific situations only (e.g., listening to a voice mail box or announcement machines). For E-model calculations regarding this listening-only communication situation the value for the pure delay (no echo) can be neglected.

- Typical communication situation.

This kind of communication situation is considered as typical for general conversations and may occur frequently (e.g., in normal conversation regarding matters of general interest). For E-model calculations regarding this typical communication situation all parameters have to be considered according to ITU-T Recommendation G.107.

- Highly interactive communication situation.

This kind of communication situation is considered as typical for active conversations and may occur frequently (e.g., in a conversation dedicated to the frequent exchange of technical or fiscal information). For E-model calculations regarding this highly interactive communication situation an additional value for impairment factor introduced due to double-talk has to be estimated (see the following clause).

6.2.2.1.3 Introduction of an additional Equipment Impairment Factor with respect to double-talk situations for the following examples

In the presence of long delay users performing highly interactive communication tasks will experience additional impacts (in comparison to the typical communication situation) affecting conversational speech quality.

Auditory test have shown that such highly interactive tasks in the presence of long delay will result in lower ratings scored by the subjects (see Annex B to ITU-T Recommendation G.114).

Since a standard set of conversational tests (for the evaluation of effects of pure delay) has not been agreed upon, the required transformation of the reduction in subjects' ratings into the Equipment Impairment Factor Ie has not been standardized, yet.

NOTE: Under study in Question 2D of ITU-T Study Group 12 during the 2001-2004 Study Period.

However, for the purposes of this example an equipment impairment value of Ie = 12 was chosen by the responsible transmission planner in order to consider this impact.

6.2.2.1.4 Purpose and general structure of the following examples

The following examples are intended to provide detailed guidance on the effect of long delay, on the applicability and usage of the E-Model under such circumstances and explicitly on the proper use of the Advantage Factor A in such situations.

The structure of the examples provided in the following clauses is summarized in table 2.

Table 2: Structure of the clauses describing the examples relating to long delay

	Listening-only communication situation	Typical communication situation	Highly interactive communication situation
"Competition"	clause 6.2.2.2.1	clause 6.2.2.2.2	clause 6.2.2.2.3
"Hard-to-reach"	clause 6.2.2.3.1	clause 6.2.2.3.2	clause 6.2.2.3.3

Figure 7 gives the general structure of the example end-to-end connection under consideration in the following clauses:

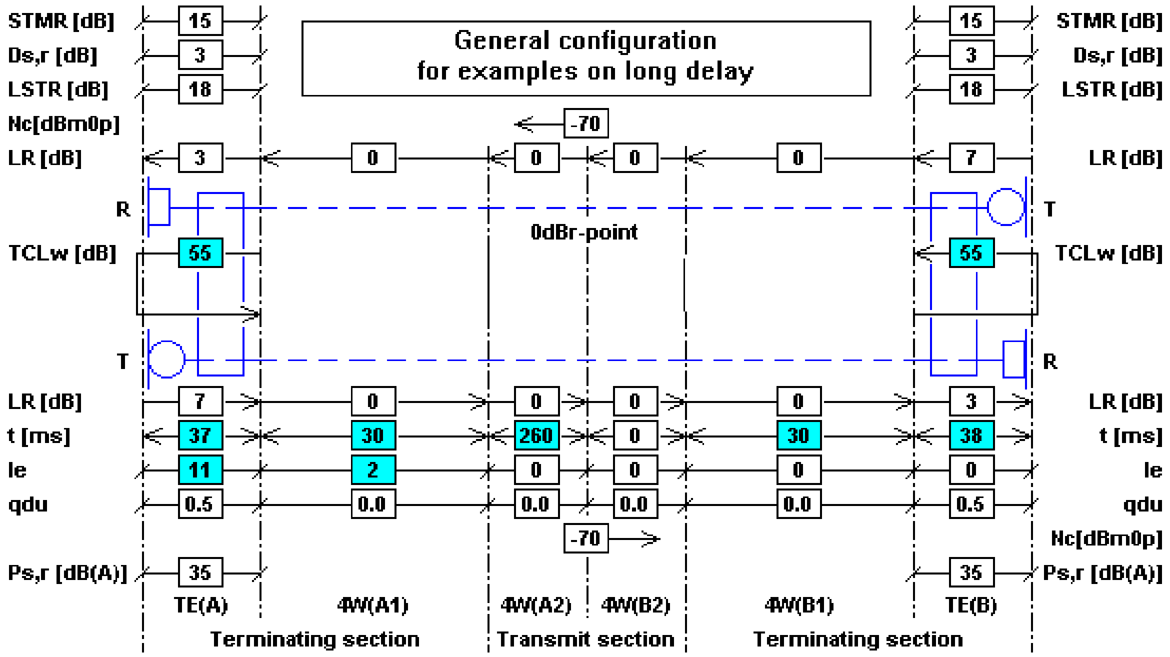


Figure 7: General configuration for examples regarding long delay

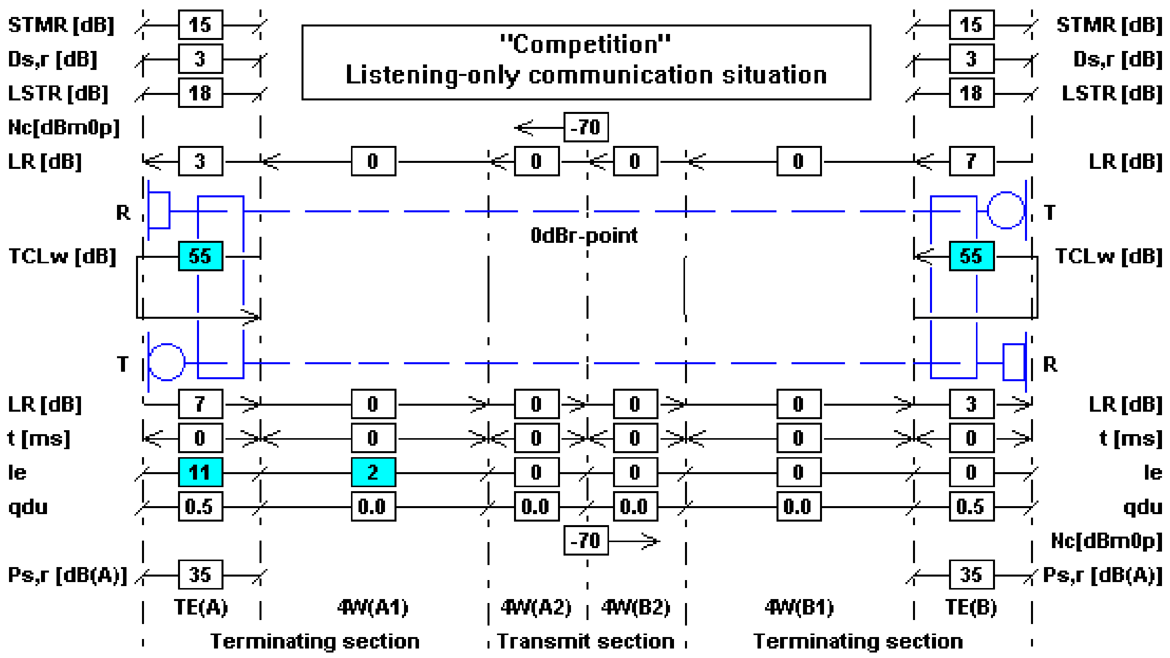
The calculations and considerations in the following clauses are based on the assumptions outlined below:

- Codec = ITU-T Recommendation G.729 A with VAD, Ie = 11 (see ITU-T Recommendation G.113);
- Terminal Mode B, delay = 75 ms (see TS 101 329-2 [1]) [split into 37 ms left and 38 ms right side];
- Network Class I, additional Ie = 2 (due to 0.5 % packet loss, see TS 101 329-2 [1] and ITU-T Recommendation G.113);
- Satellite (geo-stationary), delay = 260 ms (between earth stations, see ITU-T Recommendation G.114);
- Access networks, delay = 30 ms each (assumption), including 10 ms buffering for delay variation;
- Terminals, TELR = 65 dB each (assumption/perfect echo control integrated);
- all other parameters are assumed to be default (see ITU-T Recommendation G.107).

NOTE: that depending on application and/or actual network provider - in practice above stated values may vary which does, by no means, alter the validity of the examples given, since the purpose of the following calculations is the tutorial provision of guidance for designers of networks and equipment.

6.2.2.2 Connections to regions to which significantly shorter delay is available ("Competition")

6.2.2.2.1 Speech transmission performance as perceived in listening-only communication situations

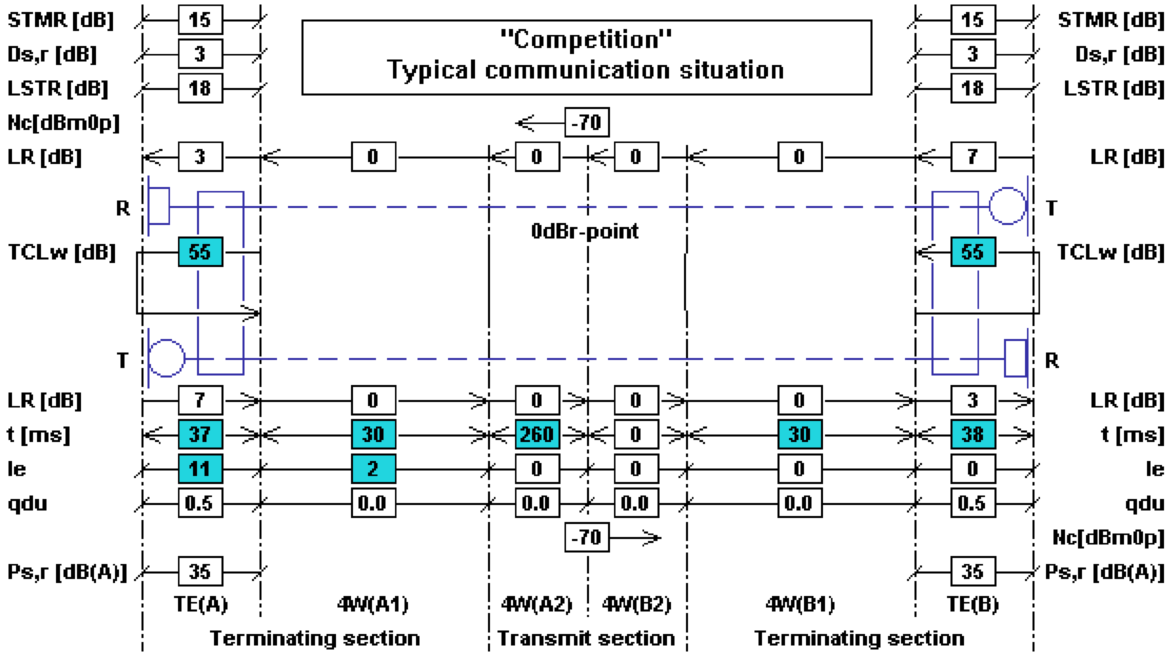


Impairment left side					Whole connection	Impairment right side				
R	Itot	Is	Id	Ie	Advantage Factor A	R	Itot	Is	Id	Ie
80.4	14.0	1.4	0.1	13	0	80.4	14.0	1.4	0.1	13

Figure 8

For this example it is assumed that no double talk communication situations will occur. Therefore, delay values are not considered in E-model calculations. No Advantage Factor can be applied, see clause 6.2.2.1.1.

6.2.2.2.2 Speech transmission performance as perceived in typical communication situations

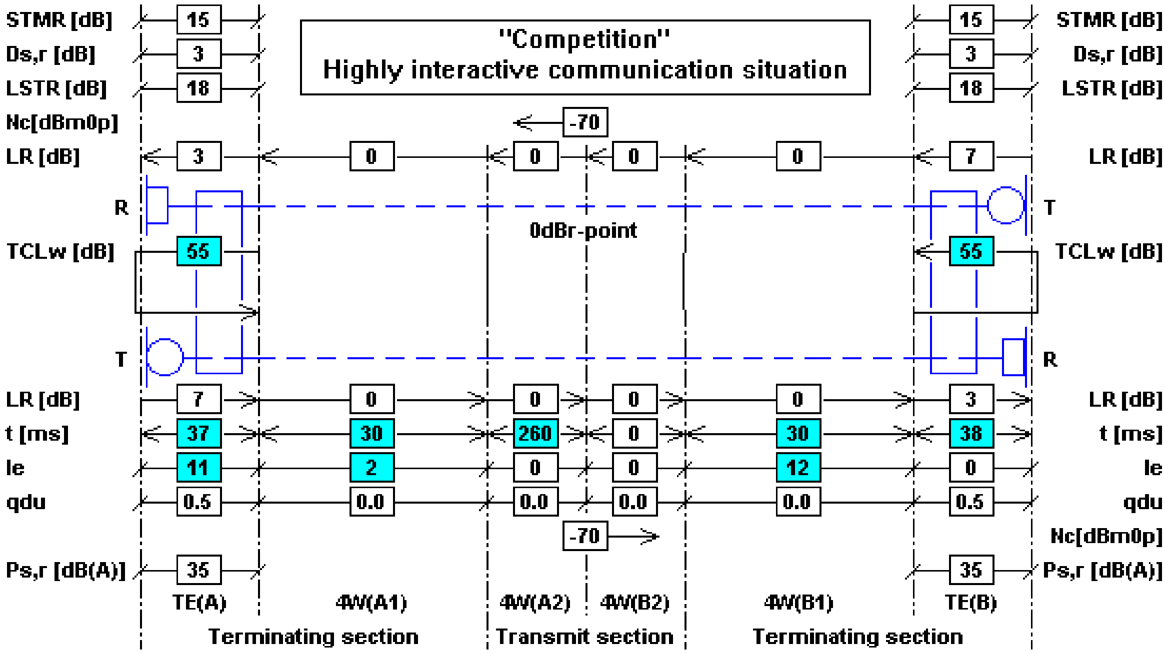


Impairment left side					Whole connection	Impairment right side				
R	Itot	Is	Id	Ie	Advantage Factor A	R	Itot	Is	Id	Ie
49.8	44.5	1.4	30.7	13	0	49.8	44.5	1.4	30.7	13

Figure 9

For this example it is assumed that double talk communication situations will occur from time to time. Therefore, delay values are considered in E-model calculations. No Advantage Factor can be applied, see clause 6.2.2.1.1.

6.2.2.2.3 Speech transmission performance as perceived in highly interactive communication situations



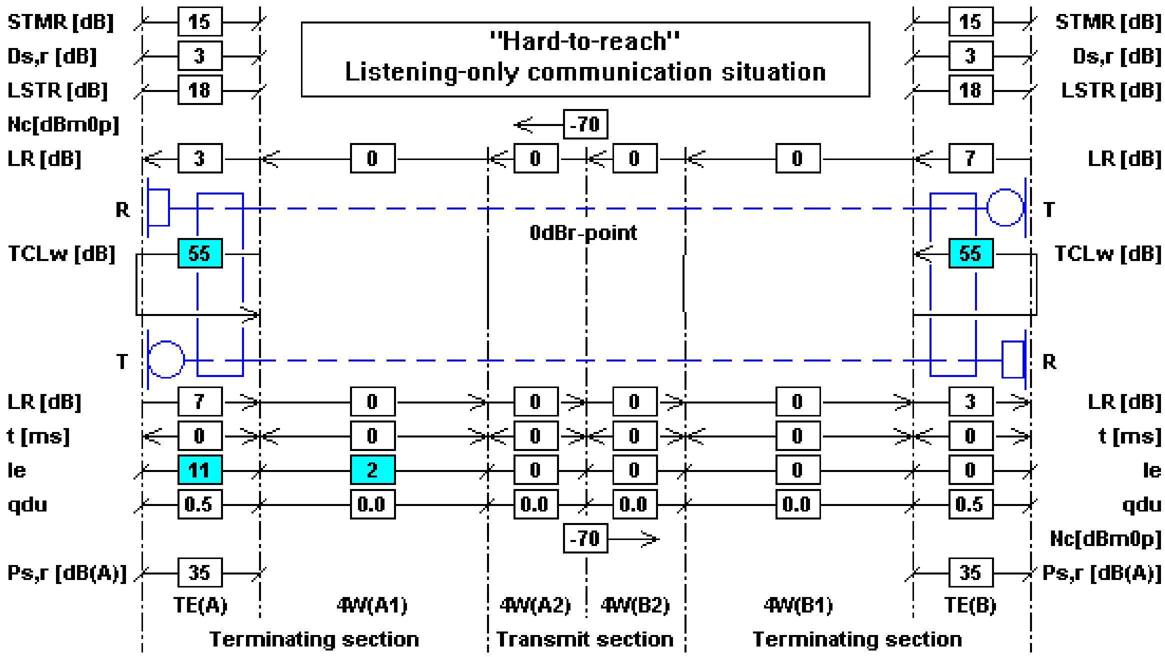
Impairment left side					Whole connection	Impairment right side				
R	Itot	Is	Id	Ie	Advantage Factor A	R	Itot	Is	Id	Ie
37.8	56.5	1.4	30.7	25	0	37.8	56.5	1.4	30.7	25

Figure 10

For this example it is assumed that double talk communication situations will occur frequently. Therefore, delay values are considered in E-model calculations. No Advantage Factor can be applied, see clause 6.2.2.1.1. An equipment impairment value of Ie = 12 was chosen by the responsible transmission planner in this example in order to consider this impact of long delay in highly interactive communication situations, see clause 6.2.2.1.3.

6.2.2.3 Connections to regions to which no shorter delay is available ("Hard-to-reach")

6.2.2.3.1 Speech transmission performance as perceived in listening-only communication situations

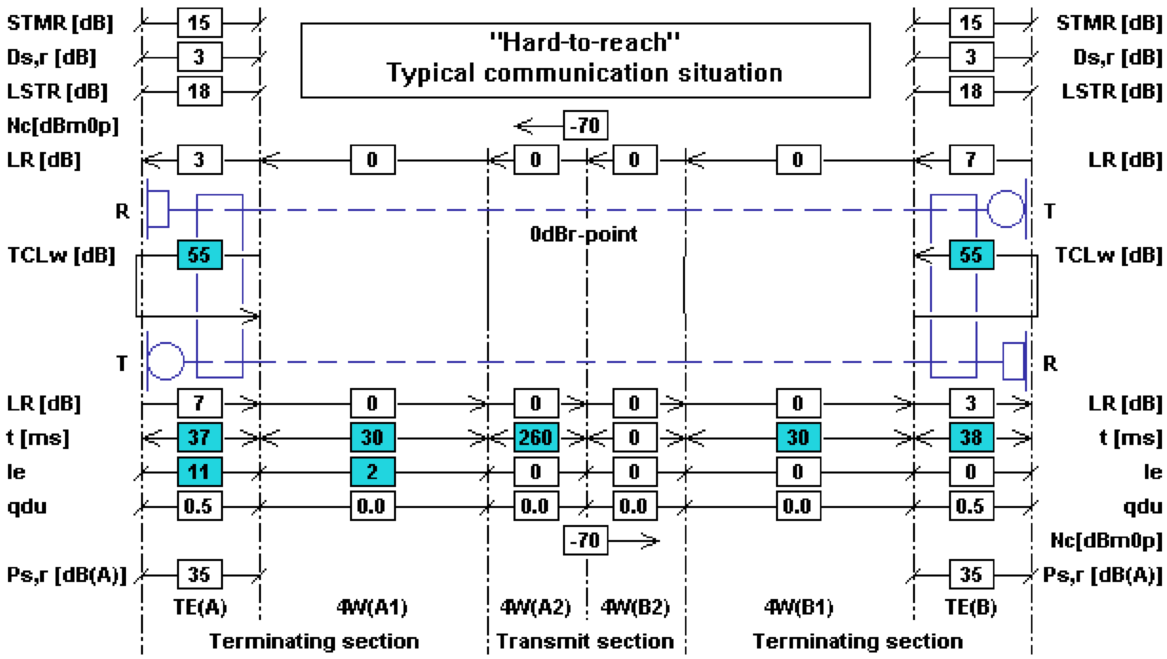


Impairment left side					Whole connection	Impairment right side				
R	Itot	Is	Id	Ie	Advantage Factor A	R	Itot	Is	Id	Ie
80.4	14.0	1.4	0.1	13	0	80.4	14.0	1.4	0.1	13

Figure 11

For this example it is assumed that no double talk communication situations will occur. Therefore, delay values are not considered in E-model calculations. No Advantage Factor can be applied since there is no decrease in R, see clauses 6.2.2.1.1 and 6.2.2.2.1.

6.2.2.3.2 Speech transmission performance as perceived in typical communication situations

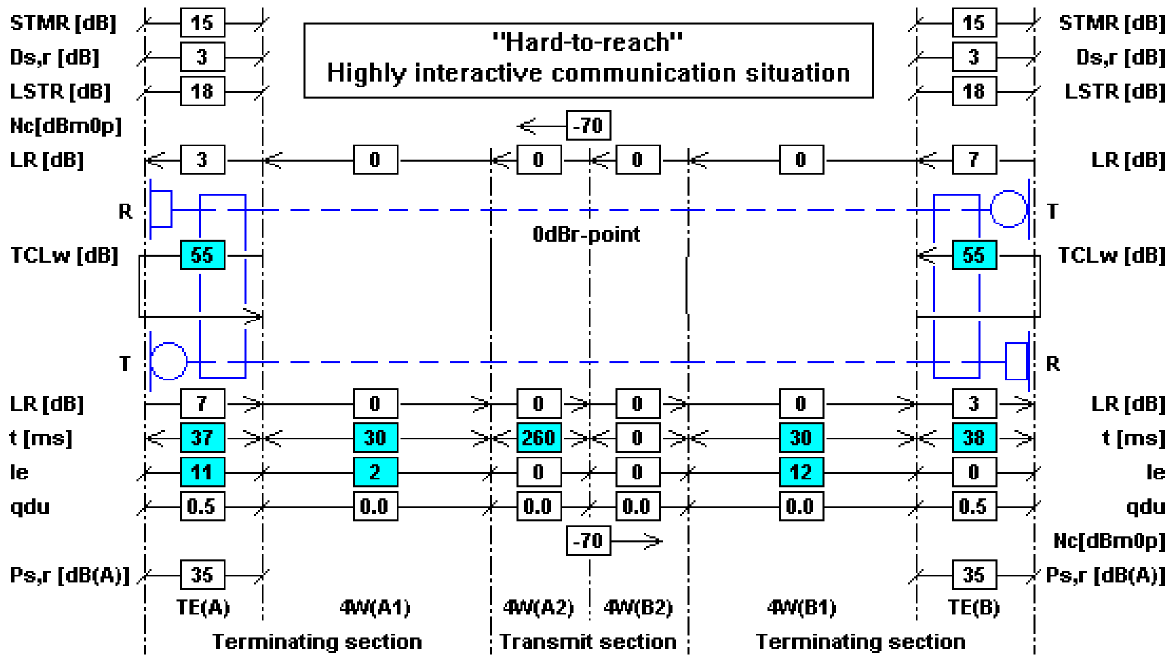


Impairment left side					Whole connection	Impairment right side				
R	Itot	Is	Id	Ie	Advantage Factor A	R	Itot	Is	Id	Ie
61.8	44.5	1.4	30.7	13	12	61.8	44.5	1.4	30.7	13

Figure 12

For this example it is assumed that double talk communication situations will occur from time to time. Therefore, delay values are considered in E-model calculations. A value of A = 12 has been assumed for the Advantage Factor.

6.2.2.3.3 Speech transmission performance as perceived in highly interactive communication situations



Impairment left side					Whole connection	Impairment right side				
R	Itot	Is	Id	Ie	Advantage Factor A	R	Itot	Is	Id	Ie
49.8	56.5	1.4	30.7	25	12	49.8	56.5	1.4	30.7	25

Figure 13

For this example it is assumed that double talk communication situations will occur frequently. Therefore, delay values are considered in E-model calculations. A value of A = 12 has been assumed for the Advantage Factor, see clause 6.2.2.1.1. An equipment impairment value of Ie = 12 was chosen by the responsible transmission planner in this example in order to consider this impact of long delay in highly interactive communication situations, see clause 6.2.2.1.3.

6.2.2.4 Summary on planning results for long delay

As can be seen from the example calculations above the user's perception the very same connection (in terms of the equipment used) will be perceived significantly different depending on the circumstances as outlined in clause 6.2.2.1 ("Competition" or "Hard-to-reach" and the kind of communication situation). Table 3 summarizes the values for the E-model Rating R resulting from the six example calculations.

Table 3: Summary of E-Model Rating R for examples of preceding clauses

	Listening-only communication situation (see Note 2)	**Typical communication situation**	**Highly interactive communication situation**
"Competition"	80 *(see 6.2.2.2.1)*	50 *(see 6.2.2.2.2)*	38 *(see 6.2.2.2.3)*
"Hard-to-reach"	80 *(see 6.2.2.3.1)*	62 *(see 6.2.2.3.2)*	50 *(see 6.2.2.3.3)*

NOTE 1: The application of the Advantage Factor A has been justified in two examples only; this is indicated in table 3 by circles. The remaining examples which did not qualify for the application of the Advantage Factor A are those without circles.

NOTE 2: The case of listening-only communications is included for completeness and is not generally applicable for network planning purposes, since most networks will carry voice traffic of all different types, including a wide variety of speech communication situations.

In cases, where connections to a specific destination under consideration are restricted to and available with long delay only ("Hard-to-reach"), there will be - in general - a certain justification for the application of the Advantage Factor A (due to delay). Although this will not include listening-only situations since no decrease in quality occurs.

For the purposes of the examples in the clauses 6.2.2.3.2 and 6.2.2.3.3 the Advantage Factor A has been estimated by the responsible transmission planner to be A = 12.
Note, that this is not a general planning rule, but a business and customer related decision for this single example case. In the course of another planning task the transmission planner may decide on a different value of the Advantage Factor.

In the presence of long delay users performing highly interactive communication tasks will experience additional impacts (in comparison to the typical communication situation) affecting conversational speech quality.

Since a standard set of conversational tests (for the evaluation of effects of pure delay) has not been agreed upon, the required transformation of the reduction in subjects' ratings into the Equipment Impairment Factor Ie has not been standardized, yet.

NOTE: Under study in Question 2D of ITU-T Study Group 12 during the 2001-2004 Study Period.

However, for the purposes of this example an equipment impairment value of Ie = 12 was chosen by the responsible transmission planner in order to consider this impact.

6.3 Examples with respect to the provision of proper echo control

In the following examples it is assumed that it was intention of the pre-installation planning to comply with the respective TIPHON QoS class. For illustrational purposes, the respective requirement limits from TS 101 512 have been taken as a starting point. In a first step, only TIPHON scenario #3, SCN to SCN over IP, has been subject to investigation.

In order to demonstrate various issues with a limited number of calculation examples, in each figure, the SCN on the left hand side is assumed fully digital, whereas the right hand SCN includes a typical analogue hybrid termination.

6.3.1 TIPHON QoS class "HIGH"

Figure 14 shows a connection with all values default according to ITU-T Recommendation G.107, except the mean one-way delay and the equipment impairment factor Ie for the coding distortion, which are set to the lower limits of TIPHON class #3. As no (or no proper) echo cancellation is provided, the overall transmission quality rating R as perceived on the left side is R_L = 40.6, while the rating R as perceived on the right side is R_R = 80.4.
This does not meet the requirements of TIPHON QoS class #3 with respect to R.

Figure 14 shows the same connection, but with proper echo cancellation provided, hence the overall transmission quality rating R as perceived on the left side is R_L = 87.3, while the rating R as perceived on the right side is R_R = 87.3.
The requirements of TIPHON QoS class #3 with respect to R are met for both sides.

Figure 15 shows - based on the connection of figure 14 - an example how the increase in overall quality reached by the provisioning of proper echo cancellation could be lost again. In practical applications, echo cancellation devices exist, which insert a digital loss pad of e.g. 6 dB, in the receive path - in order to reduce the maximum dynamic range in the echo path. Such additional loss is indicated in figure 15 which result in a decrease of the overall transmission quality rating R as perceived on the left side to R_L = 79.5, while the rating R as perceived on the right side decreases to R_R = 79.5.
This does not meet the requirements of TIPHON QoS class #3 with respect to R.

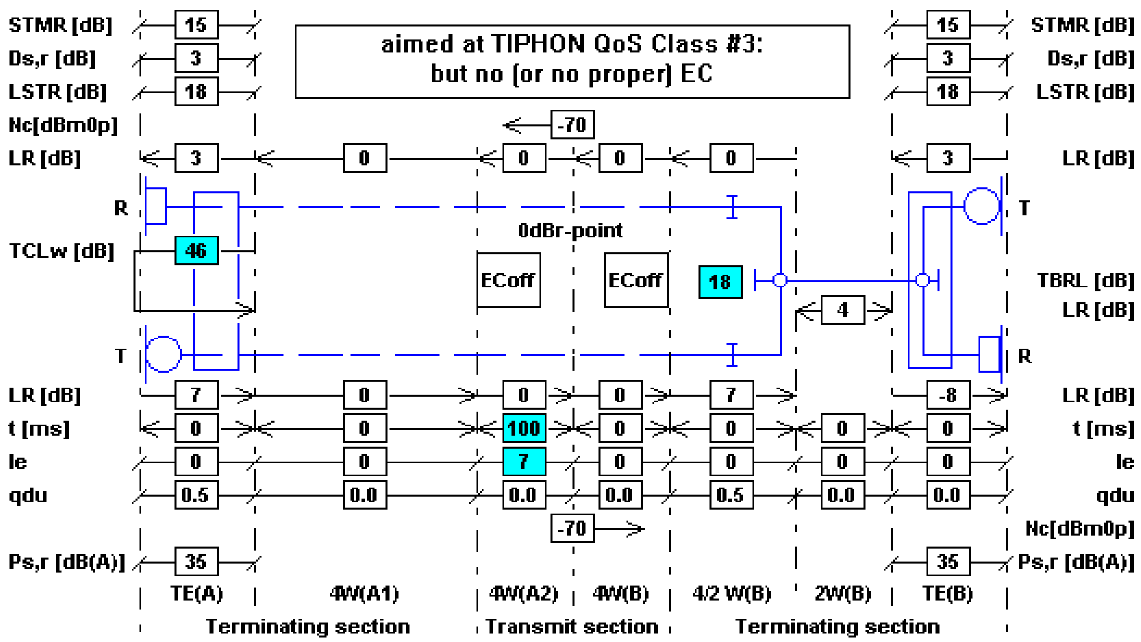


Figure 14: E-model Calculation: R as perceived on the left side is R_L = 40.6, R as perceived on the right side is R_R = 80.4

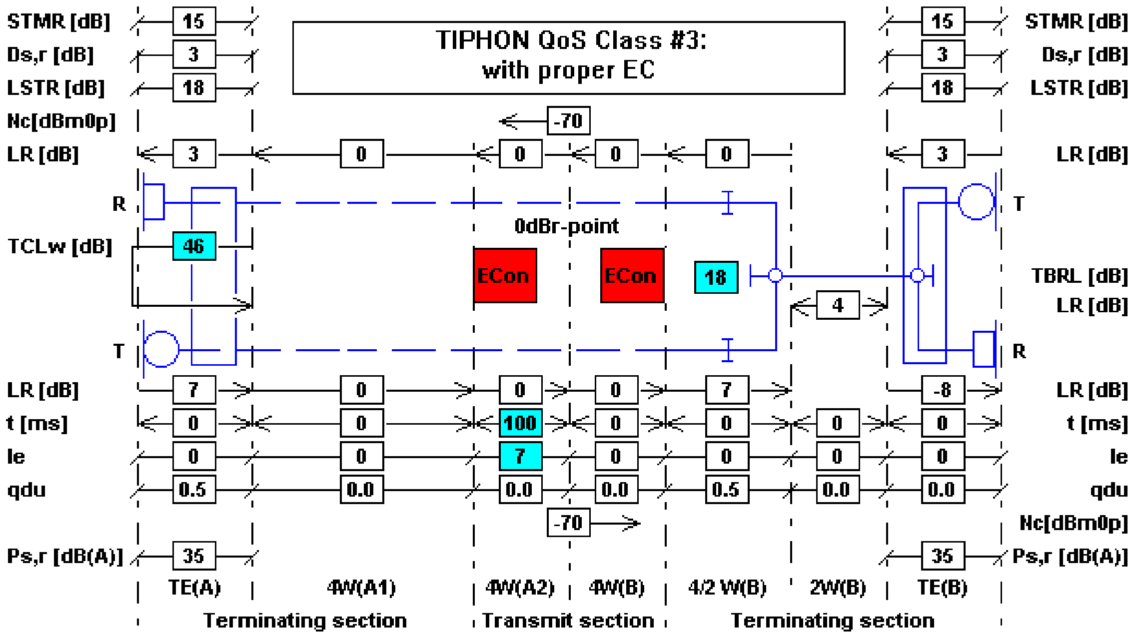


Figure 15: E-model Calculation: R as perceived on the left side is R_L = 87.3, R as perceived on the right side is R_R = 87.3

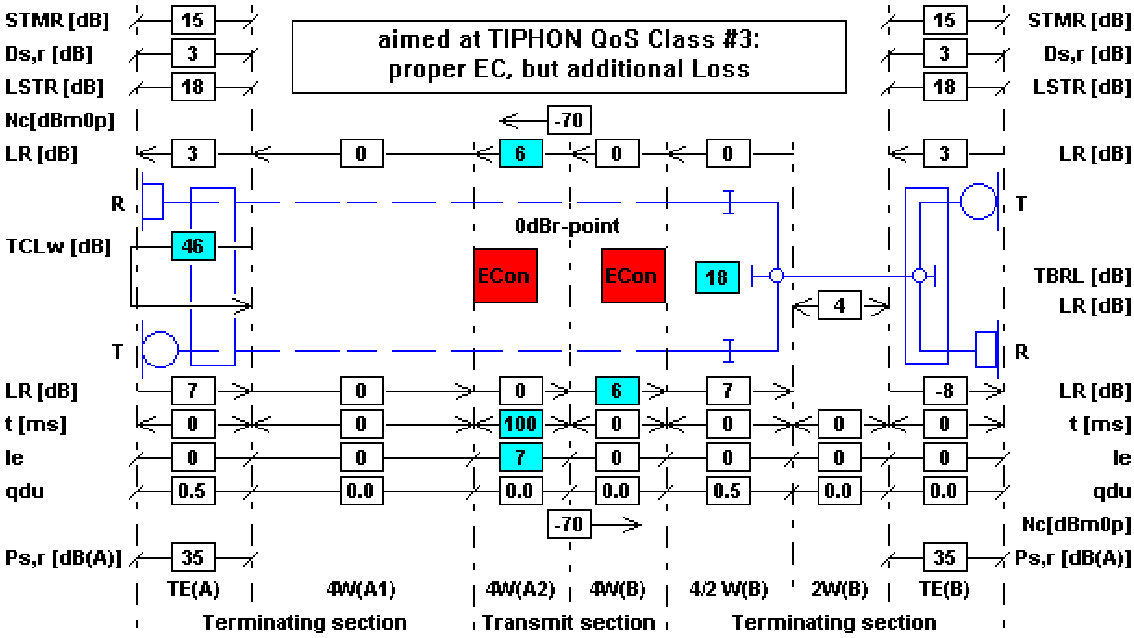


Figure 16: E-model Calculation: R as perceived on the left side is R_L = 79.5, R as perceived on the right side is R_R = 79.5

6.3.2 TIPHON QoS class "MEDIUM"

Figure 17 shows a connection with all values default according to ITU-T Recommendation G.107, except the mean one-way delay and the equipment impairment factor Ie for the coding distortion, which are set to the lower limits of TIPHON class #2. As no (or no proper) echo cancellation is provided, the overall transmission quality rating R as perceived on the left side is R_L = 19.1, while the rating R as perceived on the right side is R_R = 63.6. This does not meet the requirements of TIPHON QoS class #2 with respect to R.

Figure 17 shows the same connection, but with proper echo cancellation provided, hence the overall transmission quality rating R as perceived on the left side is R_L = 74.2, while the rating R as perceived on the right side is R_R = 74.2. The requirements of TIPHON QoS class #2 with respect to R are met for both sides.

Figure 18 shows - based on the connection of figure 17 - an example how the increase in overall quality reached by the provisioning of proper echo cancellation could be lost again. In practical applications, echo cancellation devices exist, which insert a digital loss pad of e.g. 6 dB, in the receive path - in order to reduce the maximum dynamic range in the echo path. Such additional loss is indicated in figure 18 which result in a decrease of the overall transmission quality rating R as perceived on the left side to R_L = 66.3, while the rating R as perceived on the right side decreases to R_R = 66.3. This does not meet the requirements of TIPHON QoS class #2 with respect to R.

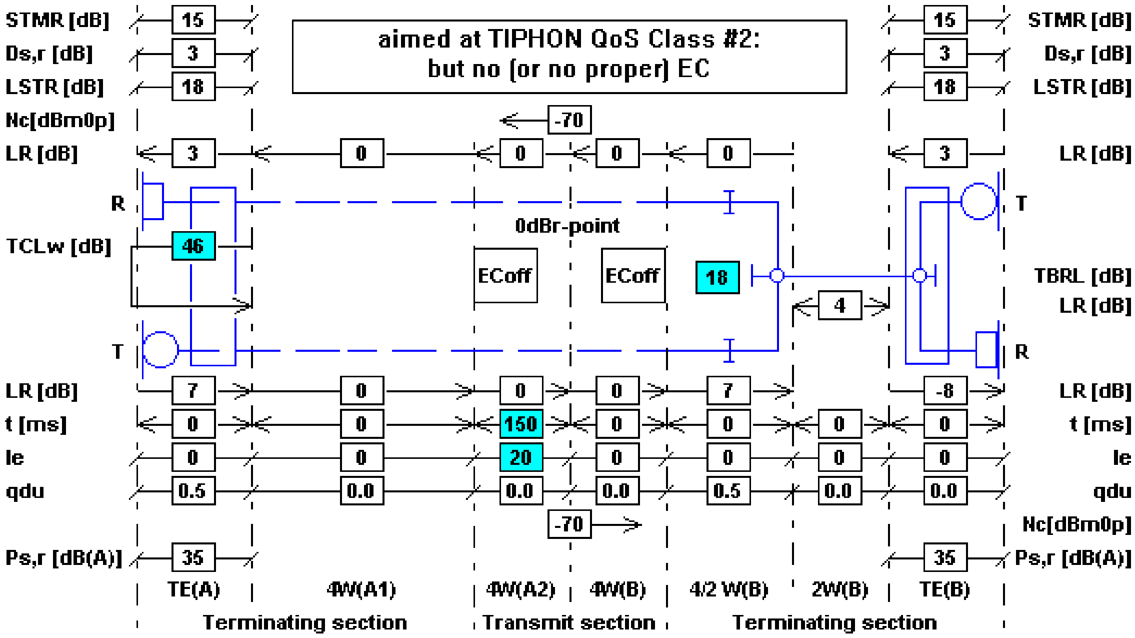


Figure 17: E-model Calculation: R as perceived on the left side is R_L = 19.1, R as perceived on the right side is R_R = 63.6

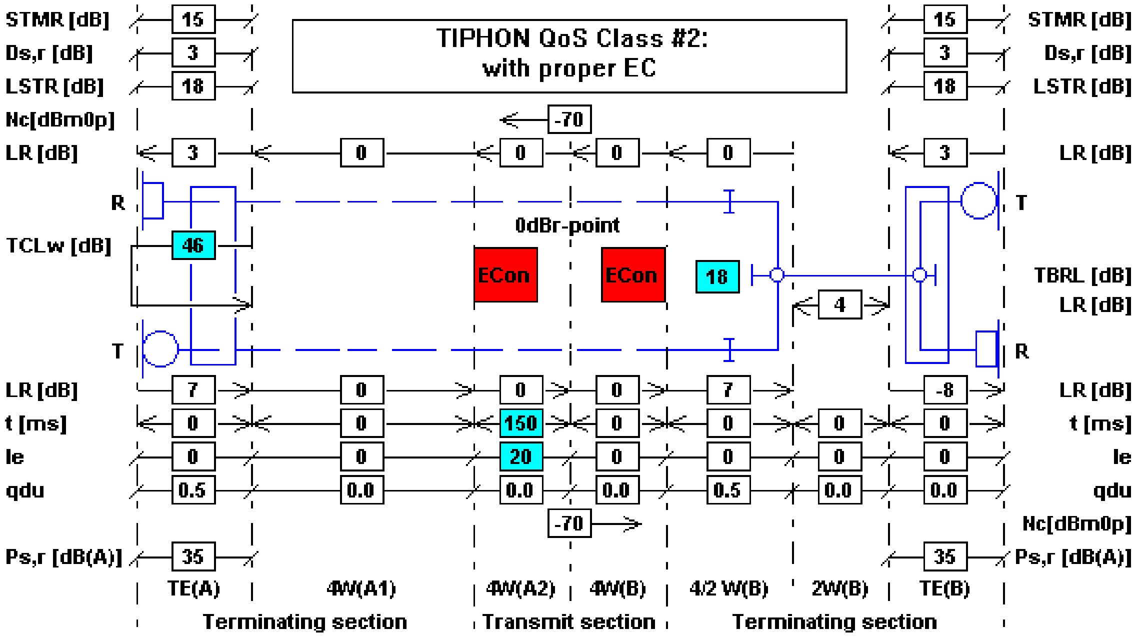


Figure 18: E-model Calculation: R as perceived on the left side is R_L = 74.2, R as perceived on the right side is R_R = 74.2

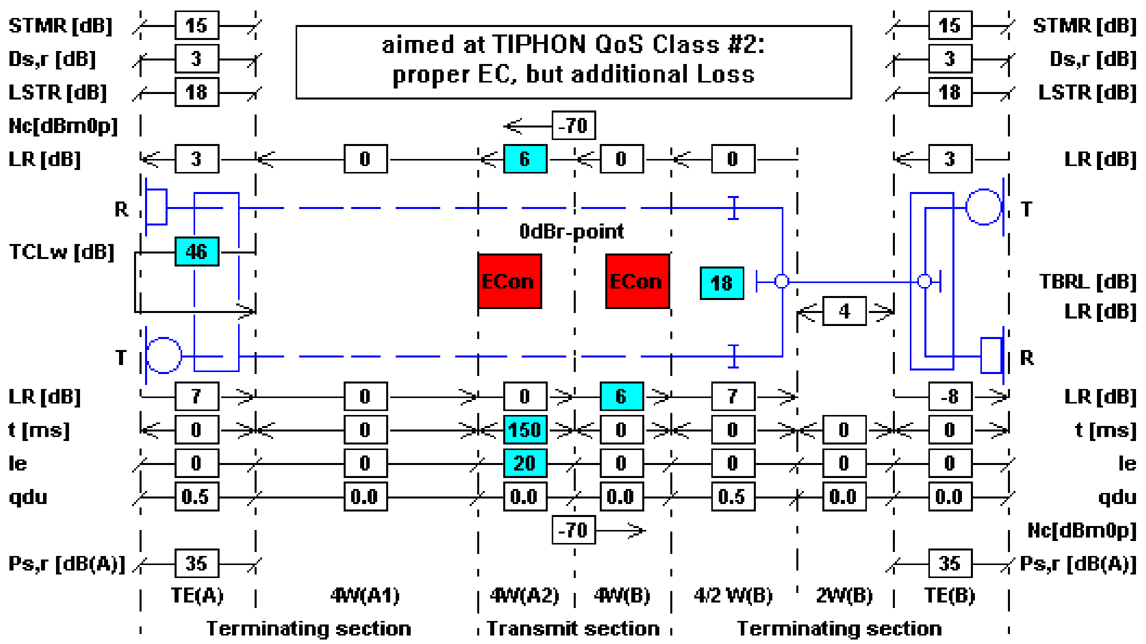


Figure 19: E-model Calculation: R as perceived on the left side is R_L = 66.3, R as perceived on the right side is R_R = 66.3

6.3.3 TIPHON QoS class "BEST EFFORT"

Figure 20 shows a connection with all values default according to ITU-T Recommendation G.107, except the mean one-way delay and the equipment impairment factor Ie for the coding distortion, which are set to the lower limits of TIPHON class #2. As no (or no proper) echo cancellation is provided, the overall transmission quality rating R as perceived on the left side is R_L = 0.0, while the rating R as perceived on the right side is R_R = 39.1. This does not meet the requirements of TIPHON QoS class #1 with respect to R.

Figure 21 shows the same connection, but with proper echo cancellation provided, hence the overall transmission quality rating R as perceived on the left side is R_L = 55.4, while the rating R as perceived on the right side is R_R = 55.4. The requirements of TIPHON QoS class #1 with respect to R are met for both sides.

Figure 22 shows - based on the connection of figure 21 - an example how the increase in overall quality reached by the provisioning of proper echo cancellation could be lost again. In practical applications, echo cancellation devices exist, which insert a digital loss pad of e.g. 6 dB, in the receive path - in order to reduce the maximum dynamic range in the echo path. Such additional loss is indicated in figure 22 which result in a decrease of the overall transmission quality rating R as perceived on the left side to R_L = 47.6, while the rating R as perceived on the right side decreases to R_R = 47.6. This does not meet the requirements of TIPHON QoS class #1 with respect to R.

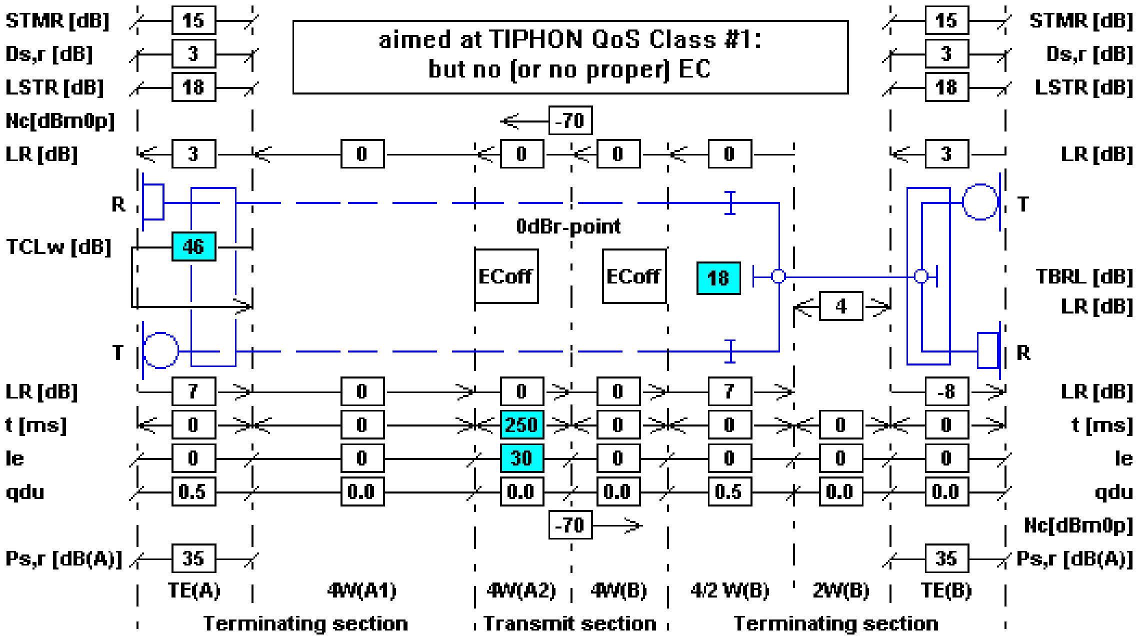


Figure 20: E-model Calculation: R as perceived on the left side is R_L = 0.0, R as perceived on the right side is R_R = 39.1

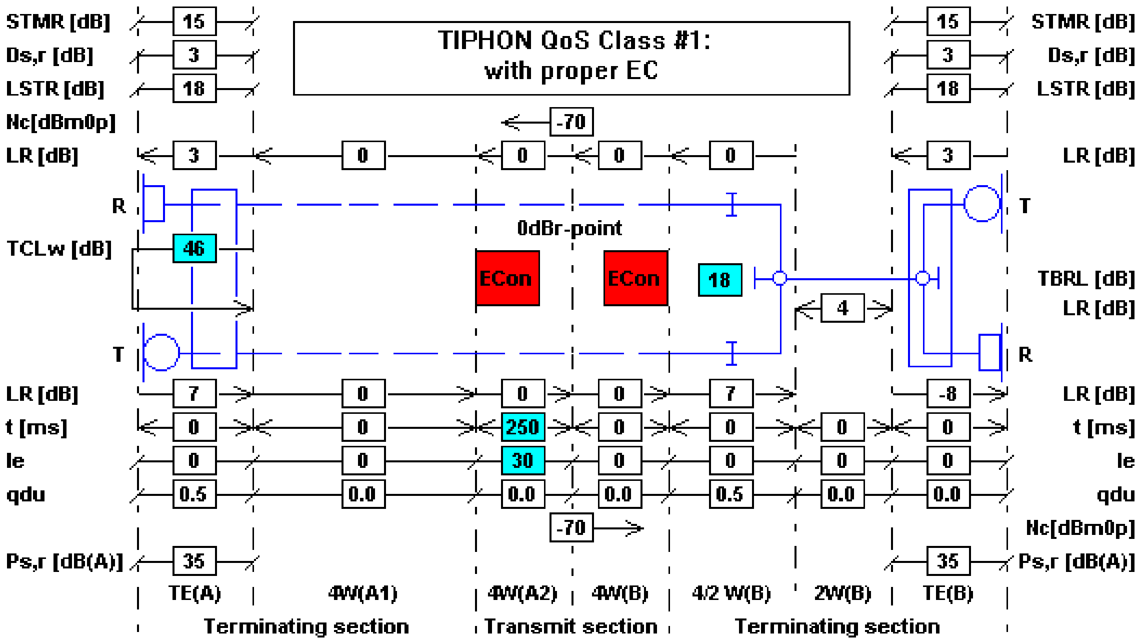


Figure 21: E-model Calculation: R as perceived on the left side is R_L = 55.4, R as perceived on the right side is R_R = 55.4

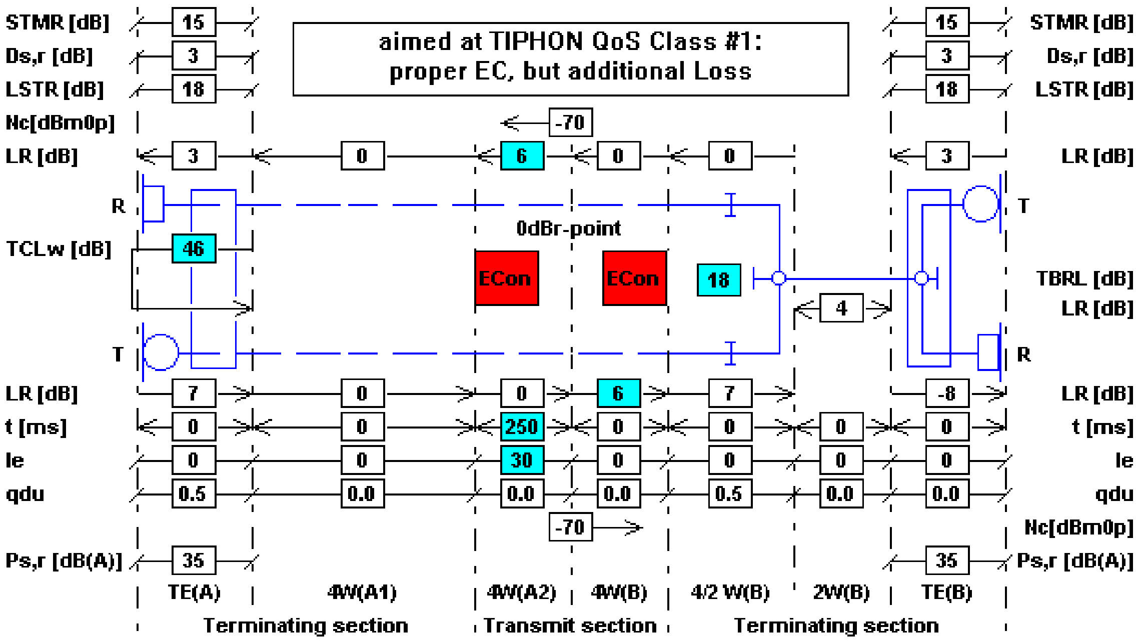


Figure 22: E-model Calculation: R as perceived on the left side is R_L = 47.6, R as perceived on the right side is R_R = 47.6

6.4 Examples with respect to Coding Distortion

This is for further study.

6.5 Examples with respect to Speech Processing other than Coding

This is for further study.

6.6 Interpretation of the results

All calculation results presented in the present document should be seen in conjunction with the 'Judgement of a connection on a linear quality scale' as given in figure 23:

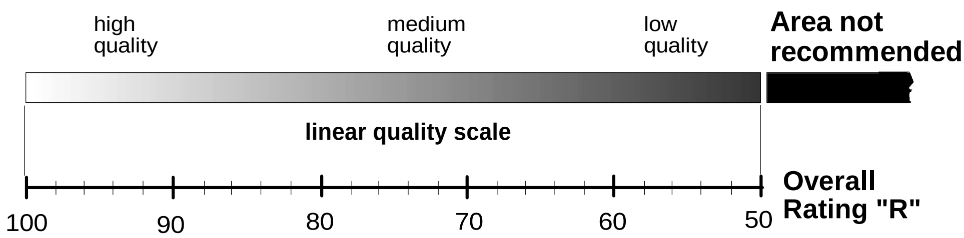


Figure 23: Judgement of a connection on a linear quality scale

Whereas Table shows the same relation in verbal form, guidance on the relation and interdependency between Auditory MOS, Objective MOS, and Predicted MOS is provided in the following clause.

Table 4: Relation between Rating Factor "R" and users satisfaction
(Table 1 of ITU-T Recommendation G.109: Definition of Categories of Speech Transmission Quality)

R-Value Range	Speech Transmission Quality Category	User satisfaction
90 ≤ R < 100	Best	Very satisfied
80 ≤ 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 R-values below 50 are not recommended. NOTE 2: Although the trend in transmission planning is to use R-values, equations to convert R-values into other metrics e.g. MOS, % GoB, % PoW can be found in Annex B of ITU-T Recommendation G.107.		

6.7 Guidance on the Relation and Interdependency between Auditory MOS, Objective MOS and Predicted MOS

For a better understanding of the contents of table 4, figure 24 is intended to show the relation and interdependency between Auditory MOS, Objective MOS, and Predicted MOS in detail.

The 'System' box contains all the equipment (acoustic or electric input/output) which is to be tested (either auditory or objectively).

The 'Auditory Test' is the subjective test with (auditory) MOS (Mean Opinion Score) as the result. This result can additionally be used to calibrate the objective test equipment (Comparison Rating Method) or (in the case of testing a pure codec device) to be transformed into the 'Equipment Impairment Factor' for use in the E-Model.

The 'Comparison Rating Method' is the objective measurement device (calibrated with the auditory test results) with 'Objective MOS' as the result. This result (in the case of testing a pure codec device) can additionally be used to be transformed into the 'Equipment Impairment Factor' for use in the E-Model.

The 'E-Model' is a parameter based method built up under use of the subjective test results of auditory tests done in the past (Auditory Test Library) with the 'System' parameters (and Ie-values) as inputs. The results of the E-Model calculations are 'Ratings' which can be transformed into 'Predicted MOS'.

Ideally, the Objective MOS as well as the Predicted MOS will be identical with the Auditory MOS.

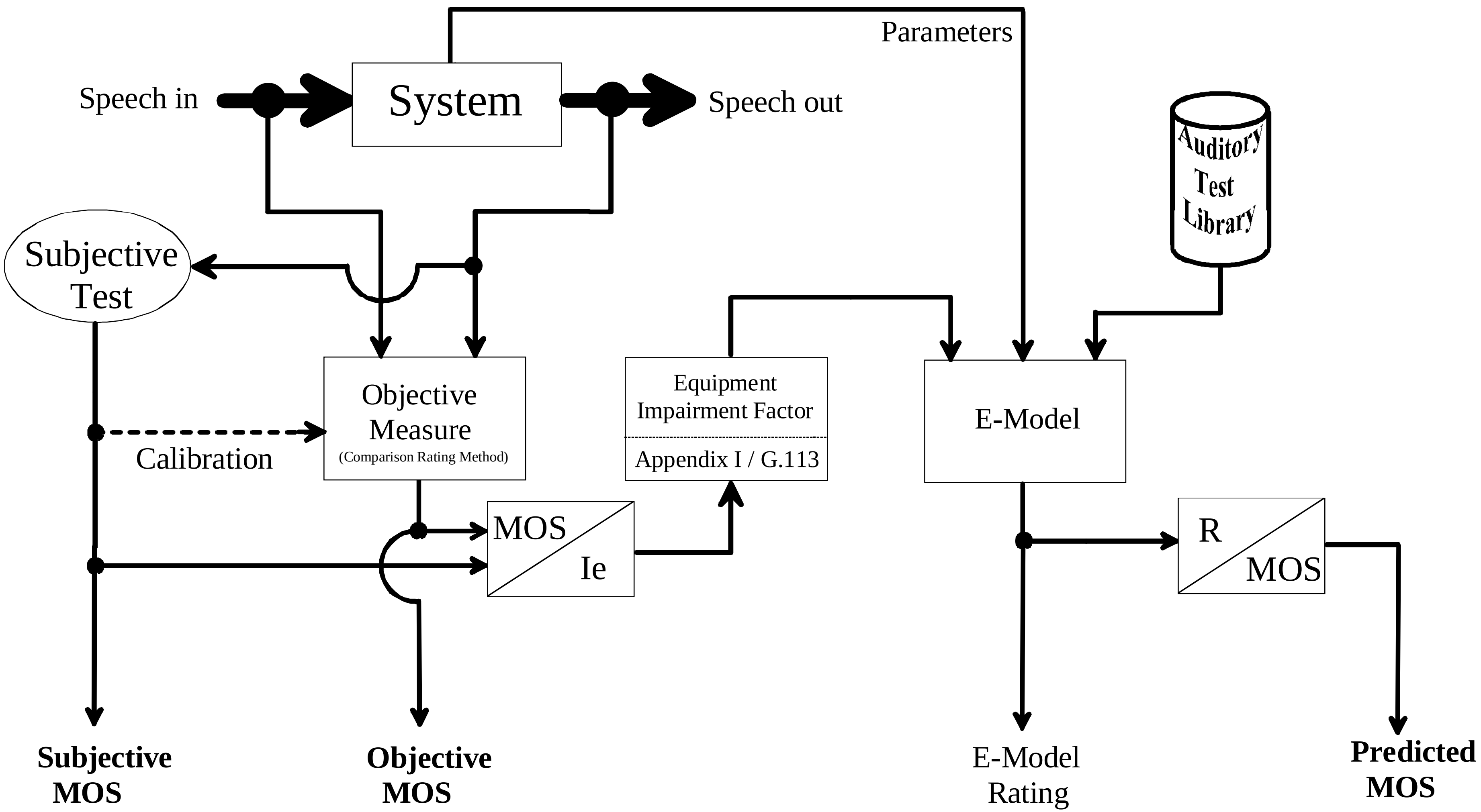


Figure 24: Relation and interdependency between Auditory MOS, Objective MOS, and Predicted MOS

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For the purposes of the present document the following standards should be considered together as a package:

ETSI EN 300 961 (V7.0.2): "Digital cellular telecommunications system (Phase 2+); Full rate speech; Transcoding (GSM 06.10 version 7.0.2 Release 1998)".

ETSI EN 300 962 (V7.0.1): "Digital cellular telecommunications system (Phase 2+); Full rate speech; Substitution and muting of lost frames for full rate speech channels (GSM 06.11 version 7.0.1 Release 1998)".

ETSI EN 300 963 (V6.0.1): "Digital cellular telecommunications system (Phase 2+); Full rate speech; Comfort noise aspect for full rate speech traffic channel (GSM 06.12 version 6.0.1 Release 1997)".

ETSI EN 300 964 (V6.0.1): "Digital cellular telecommunications system (Phase 2+); Full rate speech; Discontinuous Transmission (DTX) for full rate speech traffic channels (GSM 06.31 version 6.0.1 Release 1997)".

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For the purposes of the present document the following standards should be considered together as a package:

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ETSI EN 300 971 (V6.0.1): "Digital cellular telecommunications system (Phase 2+); Half rate speech; Comfort noise aspects for the half rate speech traffic channels (GSM 06.22 version 6.0.1 Release 1997)".

ETSI EN 300 972 (V6.0.1): "Digital cellular telecommunications system (Phase 2+); Half rate speech; Discontinuous Transmission (DTX) for half rate speech traffic channels (GSM 06.41 version 6.0.1 Release 1997)".

ETSI EN 300 973 (V6.0.1): "Digital cellular telecommunications system (Phase 2+); Half rate speech; Voice Activity Detector (VAD) for half rate speech traffic channels (GSM 06.42 version 6.0.1 Release 1997)".

For the purposes of the present document the following standards should be considered together as a package:

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ETSI EN 300 727 (V6.0.1): "Digital cellular telecommunications system (Phase 2+); Substitution and muting of lost frames for Enhanced Full Rate (EFR) speech traffic channels (GSM 06.61 version 6.0.1 Release 1997)".

ETSI EN 300 728 (V6.0.1): "Digital cellular telecommunications system (Phase 2+); Comfort noise aspects for Enhanced Full Rate (EFR) speech traffic channels (GSM 06.62 version 6.0.1 Release 1997)".

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History

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