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Technical Report

Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; End-to-end Quality of Service in TIPHON systems; 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 multi-part deliverable covering End-to-end Quality of Service in TIPHON systems, as identified below:

TR 101 329-1: "General aspects of Quality of Service (QoS)";

TS 101 329-2: "Definition of speech Quality of Service (QoS) classes";

TS 101 329-3: "Signalling and control of end-to-end Quality of Service (QoS)";

TS 101 329-5: "Quality of Service (QoS) measurement methodologies";

TR 101 329-6: "Actual measurements of network and terminal characteristics and performance parameters in TIPHON networks and their influence on voice quality";

TR 101 329-7: "Design guide for elements of a TIPHON connection from an end-to-end speech transmission performance point of view".

Quality of Service aspects of TIPHON Release 4 and 5 Systems will be covered in TS 102 024 and TS 102 025 respectively, and more comprehensive versions of the Release 3 documents listed above will be published as part of Releases 4 and 5 as work progresses.

Introduction

The present document forms one of a series of technical specifications and technical reports produced by TIPHON Working Group 5 addressing Quality of Service (QoS) in TIPHON Systems. The structure of this work is illustrated in figure 1.

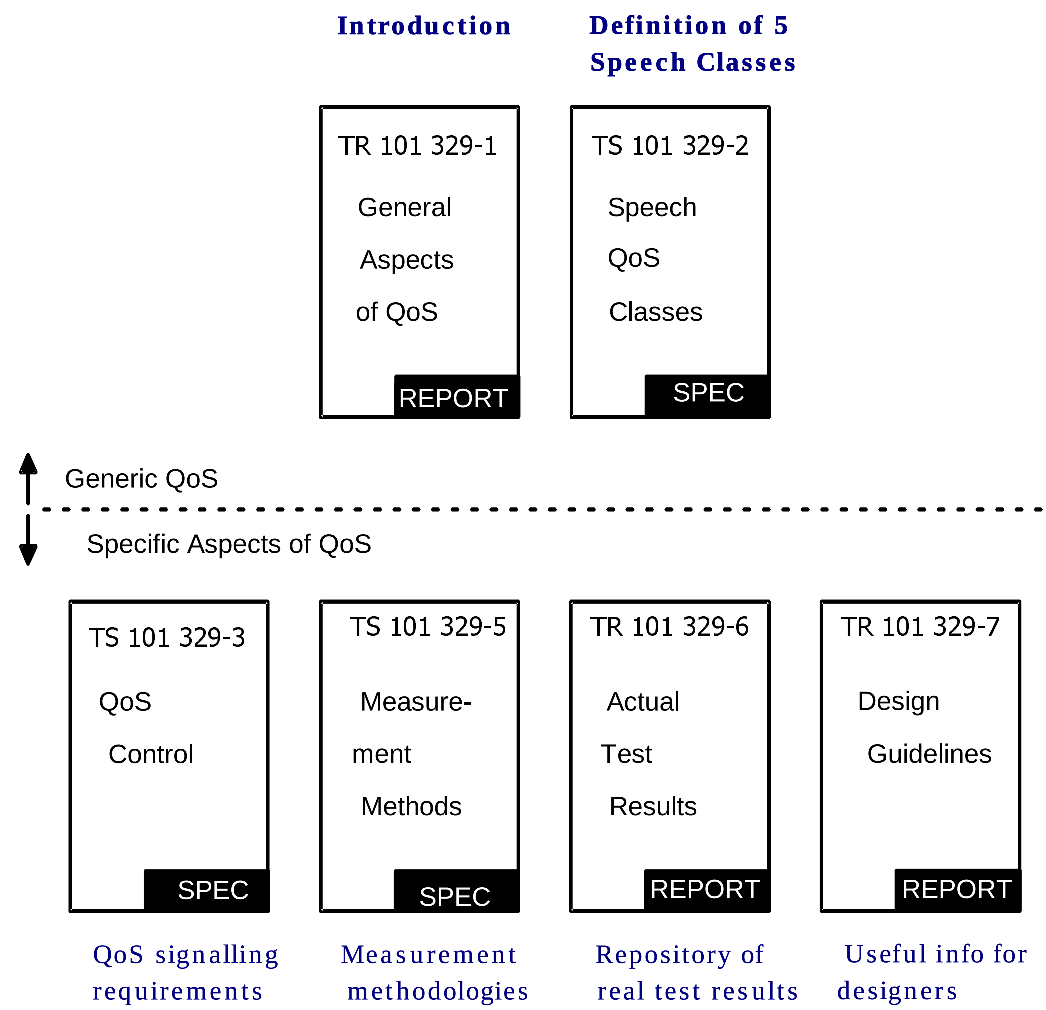


Figure 1: Structure of TIPHON QoS Documentation for Release 3

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 [3] which is a repository of real results.

Figure 2: Void

1 Scope

The present document provides a collection of informative background information and guidance to supplement parts 1 to part 6 of TS 101 329. The issues covered concern the practical design phases for both equipment and networks with respect to speech performance, and therefore is relevant to TIPHON equipment manufacturers, service providers and network designers.

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) Release 3; End-to-end Quality of Service in TIPHON Systems; Part 2: Definition of Speech Quality of Service (QoS) Classes".

[2] ETSI TS 101 329-5: "Telecommunications and Internet protocol Harmonization Over Networks (TIPHON) Release 3; End-to-end Quality of Service in TIPHON Systems; Part 5: Quality of Service (QoS) measurement methodologies".

[3] ETSI TS 101 329-6: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; End-to-end Quality of Service in TIPHON systems; Part 6: Actual measurements of network and terminal characteristics and performance parameters in TIPHON networks and their influence on voice quality".

[4] ITU-T Recommendation G.100: "Definitions used in Recommendations on general characteristics of international telephone connections and circuits".

[5] ITU-T Recommendation G.122: "Influence of national systems on stability and talker echo in international connections".

[6] ITU-T Recommendation G.131: "Control of talker echo".

[7] ITU-T Recommendation G.111: "Loudness ratings (LRs) in an international connection".

[8] ITU-T Recommendations P.64: "Determination of sensitivity/frequency characteristics of local telephone systems".

[9] ITU-T Recommendation G.109: "Definition of categories of speech transmission quality".

[10] ITU-T Recommendation G.726: "40, 32, 24, 16 kbit/s adaptive differential pulse code modulation (ADPCM)".

[11] ITU-T Recommendation G.114: "One-way transmission time".

[12] ETSI I-ETS 300 245-2: "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals; Part 2: PCM A-law handset telephony".

[13] ITU-T Recommendation P.800: "Methods for subjective determination of transmission quality".

[14] ITU-T Recommendation P.310: "Transmission characteristics for telephone-band (300-3400 Hz) digital telephones".

[15] ANSI/TIA/EIA-810-A-2000: "Telecommunications-Telephone Terminal Equipment-Transmission Requirements for Narrowband".

[16] ITU-T Recommendation G.168: "Digital network echo cancellers".

[17] ITU-T Recommendation G.107: "The E-Model, a computational model for use in transmission planning".

[18] ITU-T Recommendation G.113: "Transmission impairments due to speech processing".

[19] ETSI EG 201 050: "Speech Processing, Transmission and Quality Aspects (STQ); Overall Transmission Plan Aspects for Telephony in a Private Network".

[20] ITU-T Recommendation G.108: "Application of the E-model: A planning guide".

[21] ETSI ETR 250: "Transmission and Multiplexing (TM); Speech communication quality from mouth to ear for 3,1 kHz handset telephony across networks".

[22] ITU-T Recommendation P.833: "Methodology for derivation of equipment impairment factors from subjective listening?only tests".

3 Definitions 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

NOTE: See ITU-T Recommendation G.100 [4], 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 (LR): as used in the G-Series Recommendations for planning; loudness rating is an objective measure of the loudness loss, i.e. a weighted, electro-acoustic loss between certain interfaces in the telephone network

NOTE: 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 Rating (OLR): loudness loss between the speaking subscriber's mouth and the listening subscriber's ear via a connection

Talker Echo Loudness Rating (TELR): loudness loss of the speaker's voice sound reaching his ear as a delayed echo

NOTE: See ITU-T Recommendation G.122 [5], clause 4.2 and ITU-T Recommendation G.131 [6], figure I.1**.**

Terminal Coupling Loss weighted (TCLw): 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

NOTE: For a digital handset it is commonly in the order of 40 db to 46 dB.

weighted Terminal Coupling Loss-single talk (TCLwst): weighted loss between Rin and Sout network interfaces when AEC is in normal operation, and when there is no signal coming from the user

weighted Terminal Coupling Loss-double talk (TCLwdt): 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

Send Loudness Rating (SLR) (from ITU-T Recommendation G.111): loudness loss between the speaking subscriber's mouth and an electric interface in the network

NOTE: 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 [7] and G.121 (see Bibliography) 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 [8], P.76 (see Bibliography) and P.79 (see Bibliography).

Receive Loudness Rating (RLR) (from ITU-T Recommendation G.111): loudness loss between an electric interface in the network and the listening subscriber's ear

NOTE: 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 [7] and G.121 (see Bibliography) 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 [8], P.76 (see Bibliography), P.79 (see Bibliography).

Circuit Loudness Rating (CLR): 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 [9] 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 the range from 90 to 100, "high" in the range from 80 to 90 and "medium" in the range from 70 to 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 [10]) 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 [9] 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
AEC	Acoustic Echo Control
ALC	Automatic Level Control
ALC	Automatic Level Control
ASL	Active Speech input Level
ATM	Asynchronous Transfer Mode
DCME	Digital Circuit Multiplication Equipment
DTMF	Dual Tone Multi Frequency
DTX	Discontinuous Transmission
ECD	Echo Control Devices
GSM EFR	GSM Enhanced Full Rate Speech Coder
GSM FR	GSM Full Rate Speech Coder
GSM HR	GSM Half Rate Speech Coder
GSM	Global System for Mobile communications

IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
IWF	Inter Working Function
LAN	Local Area Network
MOS	Mean Opinion Score
MPLS	Multi Protocol Layer Switching
NIC	Network Interface Card
NS	Noise suppressors
PPP	Point to Point Protocol
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
VAD	Voice Activity Detection
VAD	Voice Activity Detectors
VDSL	Very High Speed Digital Subscriber Line
VED	Voice Enhancement Devices
VoIP	Voice over IP
VTOA	Voice and Telephony Over ATM
WAN	Wide Area Network
xDSL	ADSL, VDSL and other Digital Subscriber Line Techniques

4 General considerations

The realization of end to end speech quality in a TIPHON system is determined by a combination of user equipment design, service provider equipment performance and network transmission planning. To guarantee end to end speech quality:

- user equipment must meet specified performance requirements; and
- service provider equipment must meet specified performance requirements and be correctly configured by service providers;
- underlying transport networks, involved in the call end-to-end (IP as well as SCN), must be designed to deliver specific performance criteria at all times. It is implicit that guarantees can only be achieved over managed IP networks, engineered to deliver a given level of performance, and where traffic levels are controlled.

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

- user equipment and service provider equipment design by manufactures;
- system configuration by service providers;
- transmission planning by network operators.

The purpose of the present document is to provide design guidelines in each of these areas.

The following steps are likely to be involved in implementing a TIPHON system:

- Planning and configuration;
- Pre-qualification;
- User interaction;

- Maintenance;
- Monitoring and 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 systems, 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 (see ITU-T Recommendation G.107 and Appendix II to ITU-T Recommendation G.113) which is sometimes discussed for Internet-Telephony does not generally apply for business applications and consequently does also not apply for TIPHON systems. However, this is not a general planning rule, but a business and customer related decision for this single example case or for a specific service.

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; experiences 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).

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 parameters back to default values or by providing a log file function.

4.4 Monitoring & verification

Even if a specific TIPHON system has been operated for some time at the desired level of customer satisfaction it will be required to continuously monitor and check the end-to-end speech transmission quality.

Verification will require access to the actual settings of all major transmission parameters - including those which were accessible to the user.

4.5 Interconnection of TIPHON systems with other IP networks

Implementers of TIPHON networks are advised that IP networks other than those following the TIPHON regime, eventually may employ different QoS classification schemes than the one defined in [1].

As an example, in the following the TIPHON approach is compared to the TIA approach taken in [2].

Explanation of the arrangement of the figures

The figures are drawn with LSQ as the Y-axis and delay as the X-axis as these are the main design parameters over which the user has some control.

TIPHON sets three criteria for its QoS classes as shown in the following table which is an excerpt from [1].

	2H	2M	2A
OVR	80	70	50
Delay	100 ms	150 ms	400 ms
LSQ	86	73	50

Whilst the criteria for LSQ and delay are independent of each other, the criterion on OVR depends on LSQ, delay and other parameters. Some assumptions therefore have to be made for the other parameters as follows:

- Perfect echo cancellation is assumed for all TIPHON systems therefore, TELR = 65 dB and WEPL = 110 dB (default as per G.107). Whereas the need for proper echo control is recognized in general, other IP networks may consider the quality of echo cancellation as actually achieved in their QoS classification.
- The E-model default values are assumed for all terminal related parameters.
- The Overall E-Model Rating R is calculated according to E-model, with TELR, WEPL and terminal related parameters as above (G.107 default) and the values of delay and LSQ as on the graph.

Figure 3 illustrates the TIPHON QoS classes as described in TS 101 329-2 [1].

TIPHON QoS classes are defined by a combination of all three metrics, LSQ, delay and OVR. Therefore, the colored areas represent the TIPHON QoS classes as follows:

Green Narrowband High (2H)

Yellow Narrowband Medium (2M)

Red Narrowband Acceptable (2A)

Mauve Not Recommended (1)

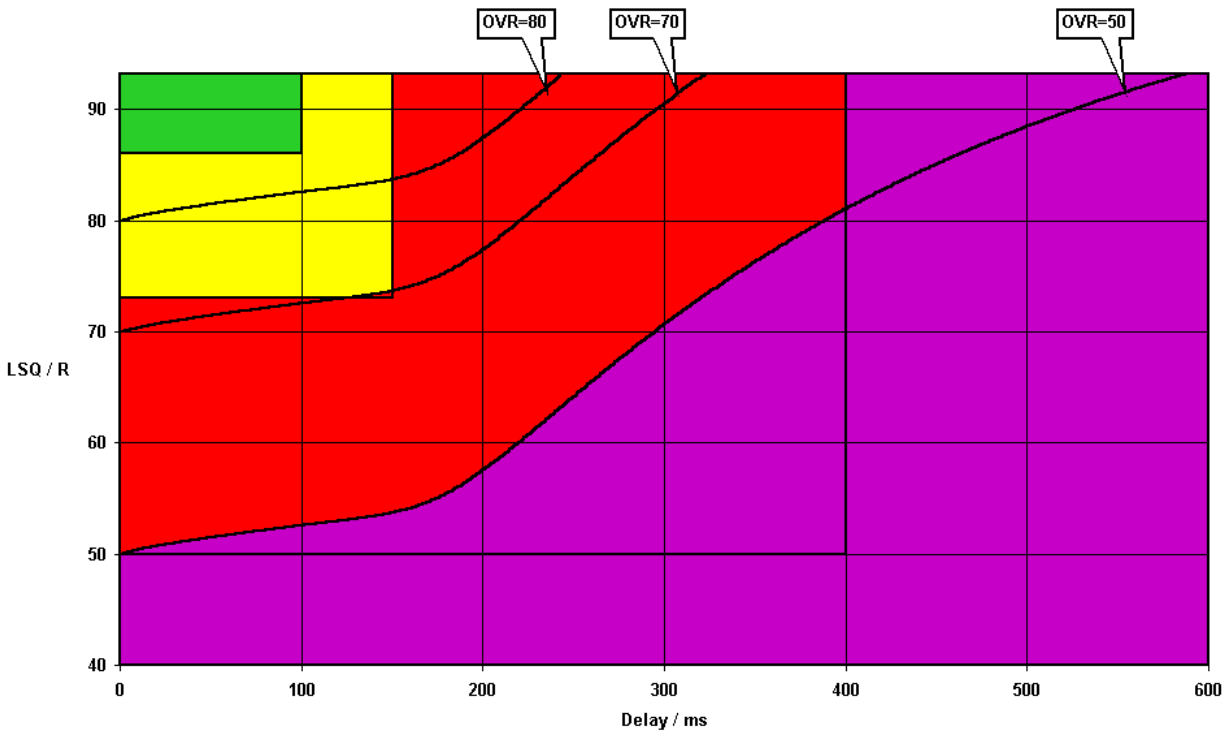


Figure 3: TIPHON QoS classes

Figure 3 shows how the criteria for LSQ and delay are more stringent for the high and medium quality classes than the criterion on OVR under the other conditions chosen. But for acceptable and not recommended quality, OVR is the most stringent criterion.

Figure 4 illustrates the TIA approach as contained in [2].

TIA QoS classes are defined by a pure E-Model approach, employing OVR, only. Therefore, the colored areas represent the TIA QoS classes as follows:

Green	High
Yellow	Medium
Red	Low
Mauve	3Not Recommended

Please, note that, the vertical and horizontal lines which indicate the TIPHON requirements on LSQ and delay have been added to the TIA diagram in figure 4 in order to accomplish a more convenient comparability with figure 3.

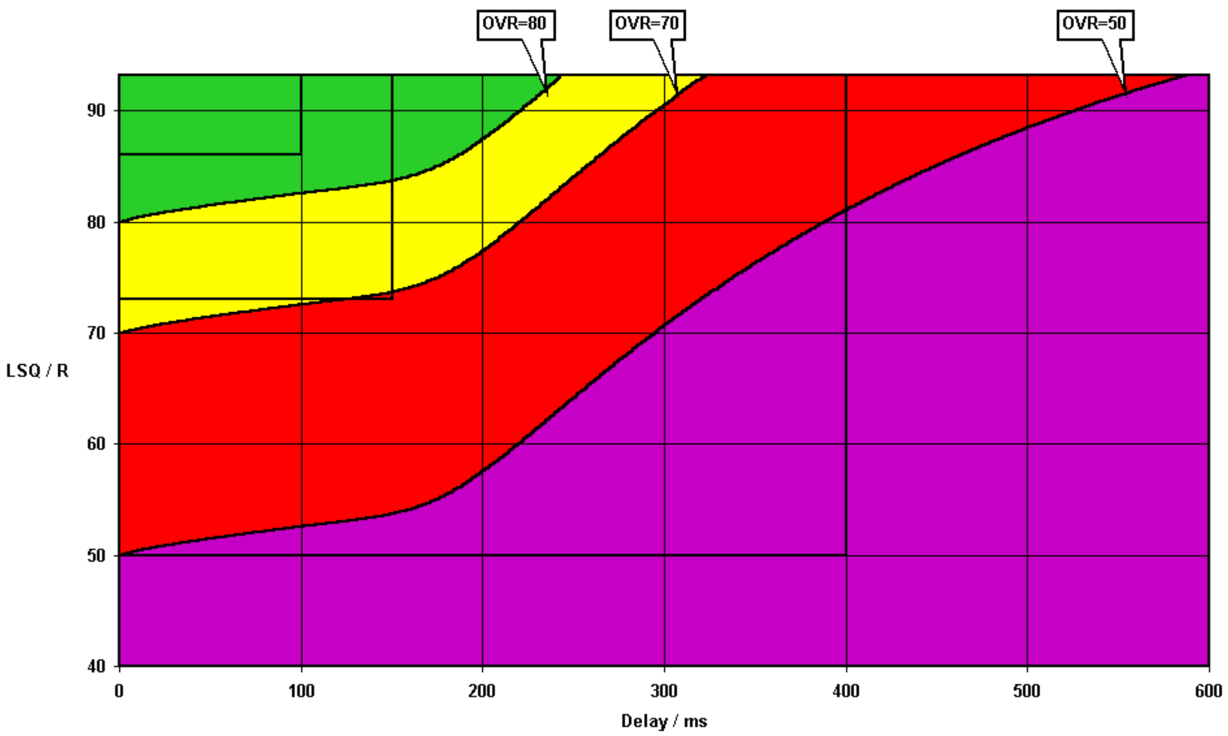


Figure 4: TIA QoS classes

Furthermore, it should be noted that in IP networks other than TIPHON one or all of the following may differ:

- name or description of QoS classes;
- guarantees provided or given for each individual class;
- availability and description of wideband classes.

In cases where TIPHON systems are interconnected to other IP networks, implementers of TIPHON systems, therefore, should thoroughly inspect the basic transmission parameters of the end-to-end connection.

Any conclusion of the resulting end-to-end speech transmission performance of such an interconnection scenario should be based on end-to-end E-model calculations. A mapping or transformation of TIPHON QoS classes with classes of other IP networks is, in general, not feasible.

5 Guidance on main transmission parameters

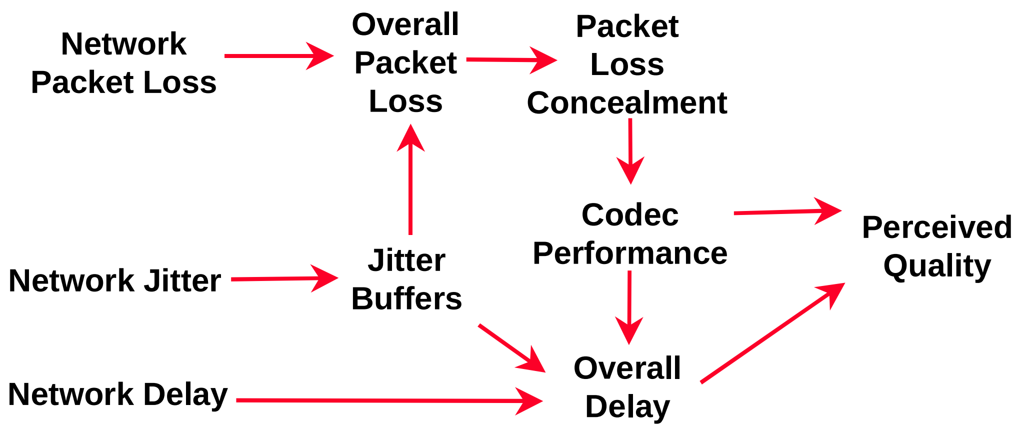


Figure 5: Interaction of transmission aspects, e.g. the influence of jitter buffers on packet loss and delay

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.

5.2 Mean one-way delay

5.2.1 Absolute delay

Absolute delay is the end-to-end delay from mouth to ear.

See ITU-T Recommendation G.114 [11] and clause 5.3 of TS 101 329-2 [1] for further information.

5.3 Delay jitter

5.3.1 Jitter buffer implementations

5.3.1.1 Static jitter buffers

A static dejittering mechanism delays the first arriving packet for a time equal to **the dejittering delay**. After this dejittering delay the dejittering buffer is read at constant rate. Packets that arrive before their play-out instant are temporary stored in **the dejittering buffer**. Packets that arrive too late (after they were supposed to be read out) are effectively lost. Hence, this dejittering delay should be chosen close to the maximum (e.g. a 99 % quantile) of the variable part of the end-to-end delay. Dejittering introduces additional delay. The choice of the dejittering delay involves a trade-off between delay and packet loss. When it is chosen too small a lot of packets will arrive too late for play out and will be considered to be lost (although they arrive at the receiver). In this case the dejittering buffer underflows (runs empty at the moment a packet is supposed to be read). The physical size of the dejittering buffer should be chosen large enough such that no (or rarely) buffer overflows will occur. It can be proven that a buffer size of twice the amount of packets that are generated during a time equal to **the dejittering delay** very rarely leads to buffer overflow.

Several approaches for filling the jitter buffer at the start of a connection have been found:

1) At the start of a connection, the first packet is delayed by the size of the jitter buffer. During this artificial initial delay, other packets are likely to arrive, thus creating a buffer of packets that can be used if, for some time, no new packets arrive. This is really the same as approach 3, except that the maximum size of the jitter buffer is not specified (but in reality it is usually limited and fixed).

2) The entire jitter buffer is filled with data, including silence packets, and kept full if possible. When a voice packet arrives too soon, silence packets may be removed from the buffer to make space for the new packet.

3) The buffer is filled to an initial value (the low water mark) but has a larger size (the high water mark). Packets are not played out until the low water mark is reached.

In general, packets that arrive out of order can be put back into the right order in the jitter buffer if they are still on time. Packets that arrive too late are discarded due to the limited size of the jitter buffer

Various ideas about the optimal size of a jitter buffer exist:

- An integral multiple of the expected packet inter-arrival time.
- Depends on network.
- Sufficient number.
- At least twice the anticipated jitter.
- Roughly twice the length of size of the expected packet arrival time variance.

Typical example jitter buffer sizes:

- 80 ms.
- 50 ms to 100 ms.
- 20 ms to 100 ms.
- Low water mark 30 ms, high water mark 150 ms.
- 150 ms.
- 50 ms.
- Up to 200 ms.

It is recommended that static jitter buffer sizes be configurable in a quantitative way, i.e., in terms of ms or bytes.

5.3.1.2 Dynamic Jitter Buffers

It is possible to use an algorithm that will dynamically adjust the dejittering delay, it will estimate the variable part of the delay of the first packet and adjust the dejittering delay accordingly. Such an algorithm has several advantages. The delay is minimized and buffer underflows are avoided. Also, when a static dejittering delay is used in a gateway or terminal it should be configured correctly at the establishment of each call to avoid buffer underflows and unnecessary delay. This configuration is not needed when a dynamic dejittering delay is used.

Finally, the codecs in the sender and receiver might not be (perfectly) synchronized. A dynamic dejittering algorithm will also compensate for this.

Several approaches to adapting the jitter buffer size have been found, for example:

- measure the variation of number of packets in the jitter buffer over a period of time and incrementally adapt the buffer size to match the calculated jitter. This approach works best with networks that provide a consistent jitter performance over time, such as ATM networks;
- count the number of packets that arrive late and create a ratio of these packets to the number of packets that are successfully processed. This ratio is then used to adjust the jitter buffer to target a predetermined, allowable late-packet ratio. This approach works best with the networks with highly variable packet-interarrival intervals—such as IP networks;
- a receive buffer with adaptive length controls its actual length via its filling level:
  - if the number of packets being at a time within the buffer increases, then the length of the buffer will be increased and at the same time at the output of the buffer short pause sequences from the original signal will be dropped in order to drain the buffer with a faster rate;
  - if the number of packets being at a time within the buffer decreases, then the length of the buffer will be decreased and at the same time at the output of the buffer short pause sequences will be additionally inserted into the original signal in order to drain the buffer with a slower rate.

Consequently, there will be a variable value of the end-to-end mean one-way delay between the talker's mouth and the listener's ear. It is important to clearly distinguish this effect from other delay variation discussions which refer to network internal processes, only. The impact of an end-to-end delay variation, as explained in this clause, is strongly dependant on the length of the dropped or inserted pieces of pauses; further emphasis rests on the correct implementation of the dynamic adaptation processes, e.g. the insertion of pieces of pauses into a syllable will have more serious impact than the insertion of pieces of pauses into a pause sequence.

The parameter of end-to-end delay variation is independent of the use of Voice Activity Detection (VAD) devices.

5.3.2 Jitter buffer monitoring capabilities

Often, the exact Jitter Buffer algorithm is partially unknown. This can cause many problems and uncertainties in implementing a Voice over IP service.

If a jitter buffer algorithm is partially unknown or cannot be disclosed, it remains necessary to be able to evaluate how well a particular jitter buffer performs under certain network conditions (e.g. specified loss, delay and jitter). This can be done by measuring key parameters of the jitter buffer in a test or live network.

The key parameters are illustrated in figure 6 and are:

1) Loss before the jitter buffer.

2) Loss after the jitter buffer, were inserted dummy packets are also seen as loss.

3) Delay (min, mean, max) added by the jitter buffer.

4) Filling (min, mean, max) of the jitter buffer.

5) Parameters of a jitter buffer that change over time, e.g. min and max size (= low and high water mark), how often these are changed, the min, mean and max values of these parameters.

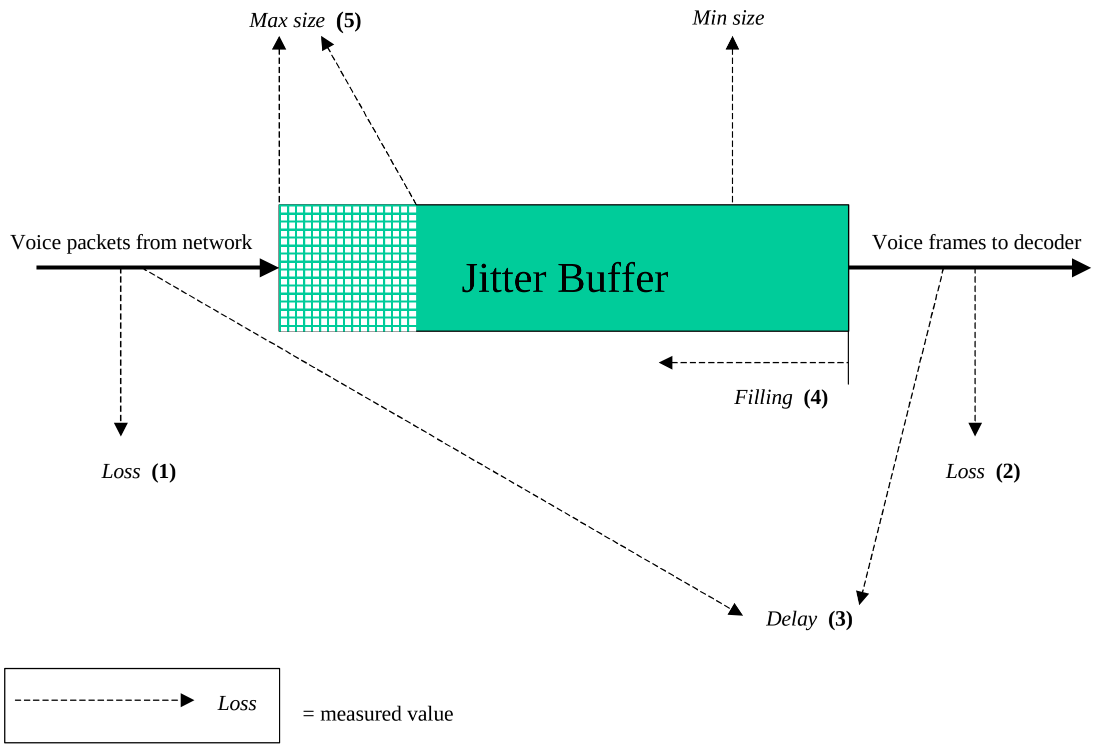


Figure 6: Key parameters for Jitter Buffer monitoring

It is recommended to make it possible to monitor these values:

- in debug mode of (test-)implementations of terminal equipment;
- in gateways: as part or regular element management,

in order for the service provider to be able to configure these devices.

5.3.3 Impact

As dynamic jitter buffers become more common, an estimation of their performance relative to that of static jitter buffers will be needed. Modelling dynamic jitter buffers is an order more difficult than modelling static jitter buffer due to the feedback involved in dynamic jitter buffers. A first estimation of the improvement that can be reached with dynamic jitter buffers may be found using simulation. Additionally, simulation can be used to "validate" the model of the static jitter buffer.

When the terminating device has a fixed-size static jitter buffer, i.e. one for which the size can't be changed, this device may or may not provide sufficient voice quality depending on the chosen jitter buffer size and the network delay. For example, some static jitter buffers are set to 200 ms. Therefore a service provider cannot guarantee end-to-end VoIP with sufficient speech quality in all cases, even though the network operator used by the service provider has adequately dimensioned their network. A service provider might have to test and approve/disapprove terminating devices as with ordinary phones and faxes. A limit for the allowable jitter buffer size must be determined for this.

When a terminating device has a configurable static jitter buffer size (either by the end user or the network operator, or both), a good value has to be chosen. As a proper value we recommend a value between 10 ms and 50 ms.

When a terminating device has a dynamic jitter buffer, also good parameter settings have to be chosen. This might be cumbersome as often the algorithm used for adjusting the buffer size (or low and high water marks) will not always be known, e.g. for competitive reasons.

Worst case, end users will blame the service provider for the unsatisfying speech transmission quality. In reality this is due to a mismatch between the network of the service provider and the jitter buffer implementation or configuration of the terminating device used.

5.4 Echo loss, echo cancellation

It is assumed that TIPHON end-to-end connections are equipped with proper echo control. However, to be sure about the actual perceived end-to-end speech quality, the service provider is advised to continously monitor (non-intrusive) and incidently check (intrusive test measurements) the talking quality with respect to echo.

5.4.1 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 [12] 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 [13], i.e. the handset is assumed to be in accordance with ITU-T Recommendation P.310 [14].

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 ms to 20 ms to the mean one-way delay of a connection.

5.4.2 IP terminals

ANSI/TIA/EIA-810-A-2000 [15] specifies the following requirements for weighted Terminal Coupling Loss (TCLw):

"The normalized value of TCLw loss shall be greater than 52 dB for IP sets and 45 dB for PCM sets when measured under free field conditions and with SLR normalized to 8 dB and RLR normalized to 2 dB. It is desirable that the normalized value of TCLw for IP sets to be greater than 55 dB and that the normalized value of TCLw for PCM sets be greater than 50 dB to meet ITU-T Recommendation G.131 [6] talker echo objective requirements".

5.4.3 IP gateways

IP Gateways should provide Echo Cancellers in accordance with ITU-T Recommendation G.168 [16].

5.5 Coding distortion

5.5.1 General considerations

The Equipment Impairment Factor of the E-model (ITU-T Recommendation G.107 [17]) is now recommended for all systems. 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 [18]; 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 [17] 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.

5.6 Speech processing other than coding

5.6.1 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 Enhancement Devices (VED);
- 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 to 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 Recommendations.

5.6.2 IP terminals

This for further study.

5.6.3 IP gateways

This is for further study.

5.7 Transcoding in network elements

Transcoding shall be avoided whenever possible. It will introduce additional delay and more importantly distortion. However, in certain cases (e.g. a GSM call traversing a TIPHON network and terminating in a PSTN) transcoding is inevitable. In table 2 the impact of transcoding on the speech quality is shown for different codec combinations. The color in the table shows how severe the impact on the speech quality is. When transcoding is done the signal is decoded to the G.711 format and then coded again. Hence, for all transcoding actions the G.711 is the intermediate format (see Appendix I of ITU-T Recommendation G.113 [18].

Table 2 : R factor for different transcoding combinations (perfect echo control and delay 0-150 ms.)

R-factor - perfect EC - delays between 0 and 142 (150) ms

	0	2	7	25	50	7	20	20	5	10	15	23	19
CODEC	G.711 (64 kb/s)	G.726 (40 kb/s)	G.726 (32 kb/s)	G.726 (24 kb/s)	G.726 (16 kb/s)	G.728 (16 kb/s)	GSM-FR (13 kb/s)	G.728 (12,8 kb/s)	GSM-EFR (12,2 kb/s)	G.729 (8 kb/s)	G.723.1 (6,3 kb/s)	GSM-HR (5,6 kb/s)	G.723.1 (5,3 kb/s)
G.711 (64 kb/s)	94,3	92,3	87,3	69,3	44,3	87,3	74,3	74,3	89,3	84,3	79,3	71,3	75,3
G.726 (40 kb/s)	92,3	90,3	85,3	67,3	42,3	85,3	72,3	72,3	87,3	82,3	77,3	69,3	71,3
G.726 (32 kb/s)	87,3	85,3	80,3	62,3	37,3	80,3	67,3	67,3	82,3	77,3	72,3	64,3	68,3
G.726 (24 kb/s)	69,3	67,3	62,3	44,3	19,3	62,3	49,3	49,3	64,3	59,3	54,3	46,3	50,3
G.726 (16 kb/s)	44,3	42,3	37,3	19,3	0	37,3	24,3	24,3	39,3	34,3	29,3	21,3	25,3
G.728 (16 kb/s)	87,3	85,3	80,3	62,3	37,3	80,3	67,3	67,3	82,3	77,3	72,3	64,3	68,3
GSM-FR (13 kb/s)	74,3	72,3	67,3	49,3	24,3	67,3	54,3	54,3	69,3	64,3	59,3	51,3	55,3
G.728 (12,8 kb/s)	74,3	72,3	67,3	49,3	24,3	67,3	54,3	54,3	69,3	64,3	59,3	51,3	55,3
GSM-EFR (12,2 kb/s)	89,3	87,3	82,3	64,3	39,3	82,3	69,3	69,3	84,3	79,3	74,3	66,3	70,3
G.729 (8 kb/s)	84,3	82,3	77,3	59,3	34,3	77,3	64,3	64,3	79,3	74,3	69,3	61,3	65,3
G.723.1 (6,3 kb/s)	79,3	77,3	72,3	54,3	29,3	72,3	59,3	59,3	74,3	69,3	64,3	56,3	60,3

R-value range	100 - 90	90 - 80	80 - 70	70 - 60	60 - 50	50 - 0
Speech transmission quality category according to G.109	best	high	medium	low	poor	not recommended

5.8 Packet loss

5.9 Example of TIPHON QoS parameter allocation

The intent of this clause is to provide an example of how the end-to-end performance parameters of the TIPHON QoS classes may be allocated between user equipment and service provider equipment and networkmay be achieved.

Table 3 shows the main parameters which impact end-to-end speech quality in VoIP systems, and their association with the terminal, the network, or both. Note that echo control is assumed to be taken care of.

Table 3: Main IP voice quality parameters and their association

Parameter	Associated with	
	user and service provider equipment	**Network**
Codec type	Yes	No
Packet Loss	No (see note 1)	Yes
Delay	Yes (see note 2)	Yes (see note 3)
Delay variation	No (see note 4)	Yes (see note 5)
NOTE 1: Assumes dejittering delay is chosen large enough such that no packets arrive too late for play out and that the dejittering buffer size is large enough such that no overflow will occur. NOTE 2 Due to speech coding/packetization and due to dejittering delay. NOTE 3: Due to network routing/propagation. NOTE 4: Assumes any delay variation from sending client is included in terminal delay. NOTE 5: Delay variation is introduced by the network, but compensated in the terminal.		

Inspection of table 3 shows that, with some simplifying assumptions, three of the four parameters - codec type, packet loss and delay variation - are associated with either the terminal or the network alone, but not both. This means that the end-to-end budget for these parameters can be totally allocated to the corresponding element. On the other hand, delay is associated with both the terminal and the network, and the overall end-to-end delay budget shall be allocated between them.

Recognizing that:

- overall end to end delay =
  (speech coding and packetization delay) + network routing delay + propagation delay + delay variation buffer size; and

- that jitter is introduced by the network, but compensated in the terminal, it is clear that delay in the terminal is essentially fixed, but delay in the network will be a function of distance and the number of router hops.

Therefore, three terminal "Modes" A, B, C may be used on the basis of delay to allow for different speech encoding and packetization schemes as shown in table 4. In this context, terminal delay includes encode/decode and packetization operations, but not jitter compensation (see later).

Table 4: Example of terminal modes

Terminal Mode	Delay	Application
A	< 50 ms	Allows for small speech frame duration and small number of frames per IP packet
B	< 75 ms	Allows for larger speech frame duration and/or small number of frames per IP packet
C	< 100 ms	Allows for larger speech frame duration and multiple frames per packet

Three network "Classes" I, II, III may also be used on the basis of packet loss and delay variation, but not propagation and routing delay as shown in table 5.

Table 5: Example of network classes

Network Class	Packet loss	Delay variation
I	< 0,5 %	< 10 ms
II	< 1 %	< 20 ms
III	< 2 %	< 40 ms
NOTE: Values of packet loss are based on subjective speech quality test results given in TR 101 329-6 [3]		

Taking the values of delay for the specific terminal mode and network class given in tables 4 and 5 and noting that (network routing delay + propagation delay) = overall delay for the TIPHON QoS class - (speech coding and packetization delay + jitter compensation delay) where the jitter compensation delay shall at least equal the network delay variation, table 6 shows the propagation and routing delay available to meet a given TIPHON QoS class.

Table 6: Available propagation and routing delay as a function of TIPHON QoS class, terminal mode and network class

Network Class	TIPHON QoS Class	Terminal Mode		
		A	B	C
I	High	40 ms	15 ms	X
	Medium	90 ms	65 ms	40 ms
	Best effort	340 ms	315 ms	290 ms
II	High	30 ms	5 ms	X
	Medium	80 ms	55 ms	30 ms
	Best Effort	330 ms	305 ms	280 ms
III	High	10 ms	X	X
	Medium	60 ms	35 ms	10 ms
	Best Effort	310 ms	285 ms	260 ms
NOTE: X = QoS Class can not be met (on the basis of delay) with this combination of Terminal Mode and Network Class. Figures in *italics* indicate where the QoS Class may not be met due to the combination of network packet loss and type of speech coder failing to meet the Listener Speech Quality requirements of table 3.				

6 Calculation and planning examples

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

It should be noted, that the examples provided in this clause are structured according to the respective transmission parameters and effects, i.e., loudness ratings, mean one-way delay, speech processing and packetization, the provision of proper echo control, coding distortion, speech processing other than coding, packet loss. However, each example will also provide useful guidance with respect to other transmission parameters or effects. For example, a planning example with its main focus on delay issues may to some extent also guide the reader who is concerned with packet loss issues.

Examples for all TIPHON Scenarios and for all main transmission parameters are provided together with some background material. For further instructions see for example EG 201 050 [19], ITU-T Recommendations G.108 [20] and G.108.01 (see Bibliography).

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 7 shows - as an example - what such a reference configuration for Scenario 3 might look like.

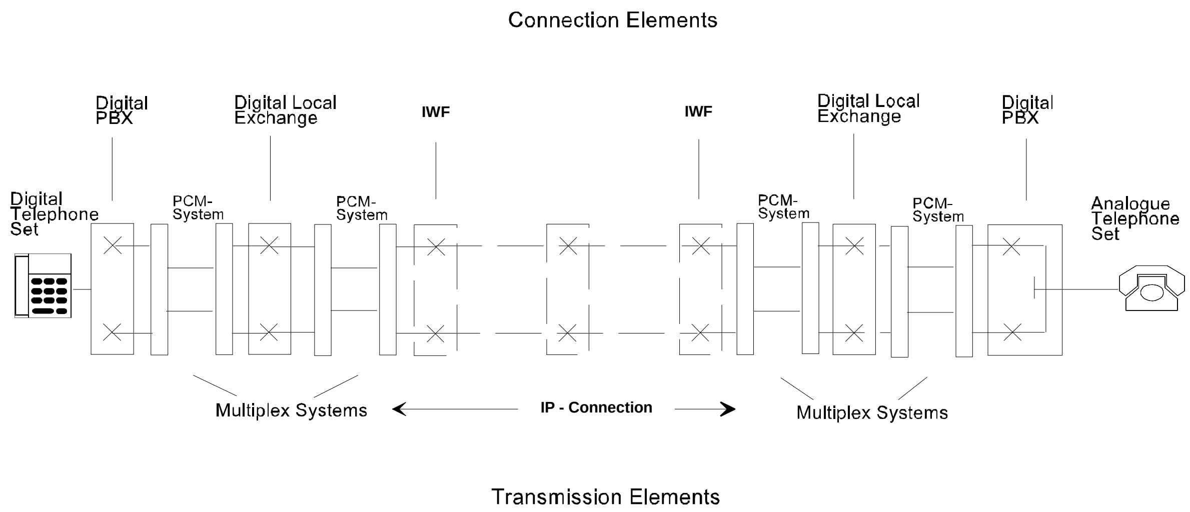


Figure 7: Reference configuration for a fully digital connection including an IP section and a terminating hybrid

The second step is to take into account the impairments with respect to speech quality; figure 8 gives some examples.

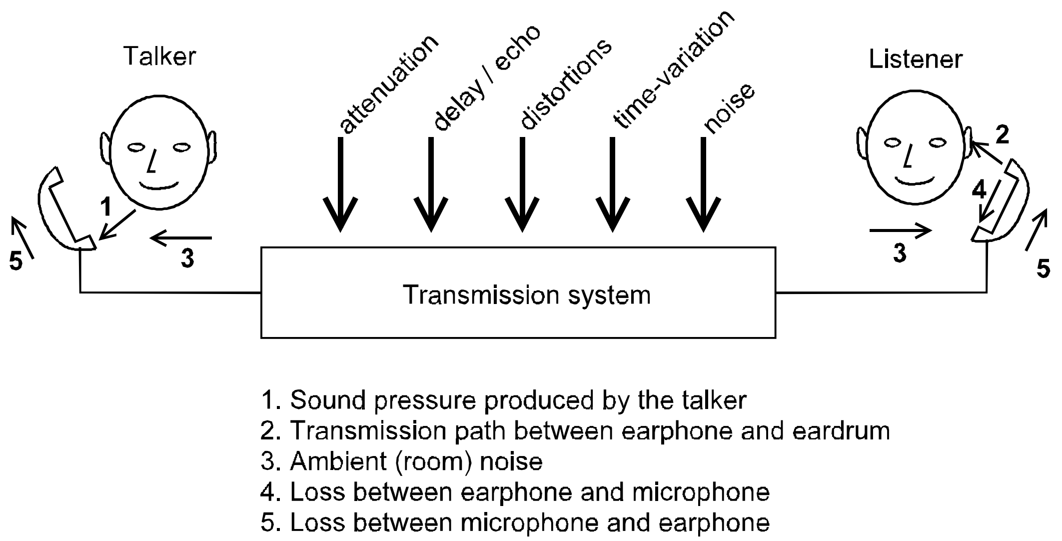


Figure 8: Transmission parameters influencing the quality of a handset telephone connection

The last step is to produce a working configuration - as shown in figure 9 - having all relevant parameters and required values at hand.

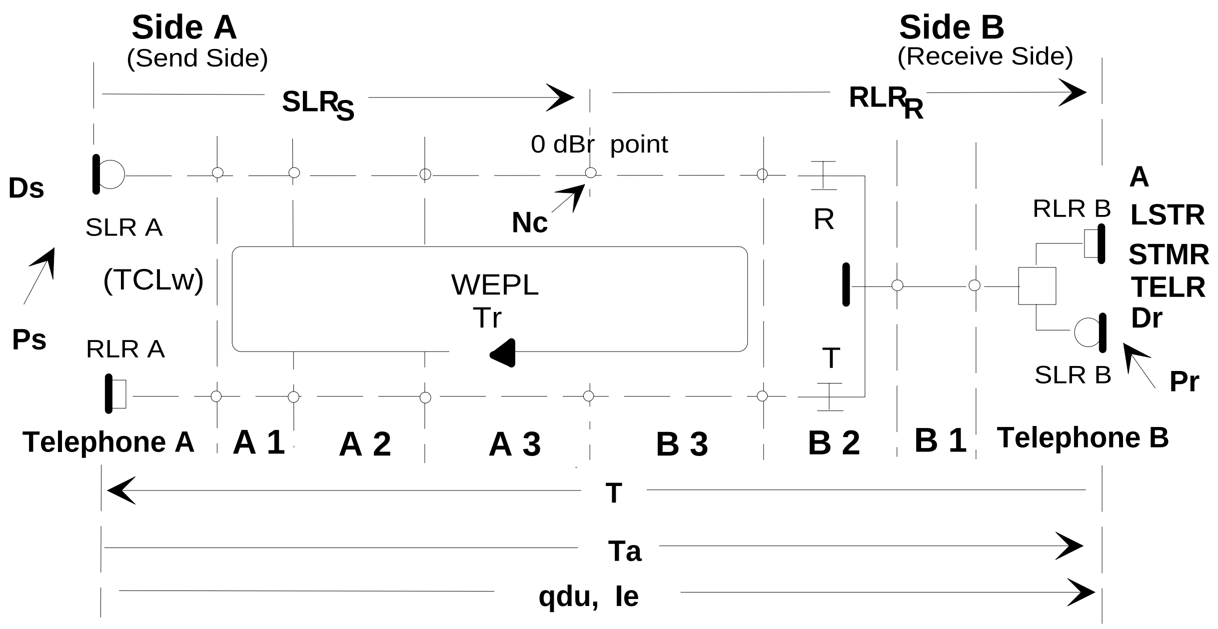


Figure 9: Working configuration for 4-wire/2-wire connections

All symbols and abbreviations are as defined in EG 201 050 [19], whereas the IP section including the IWFs is represented by the segments A3 and B3.

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.

Figure 10 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.

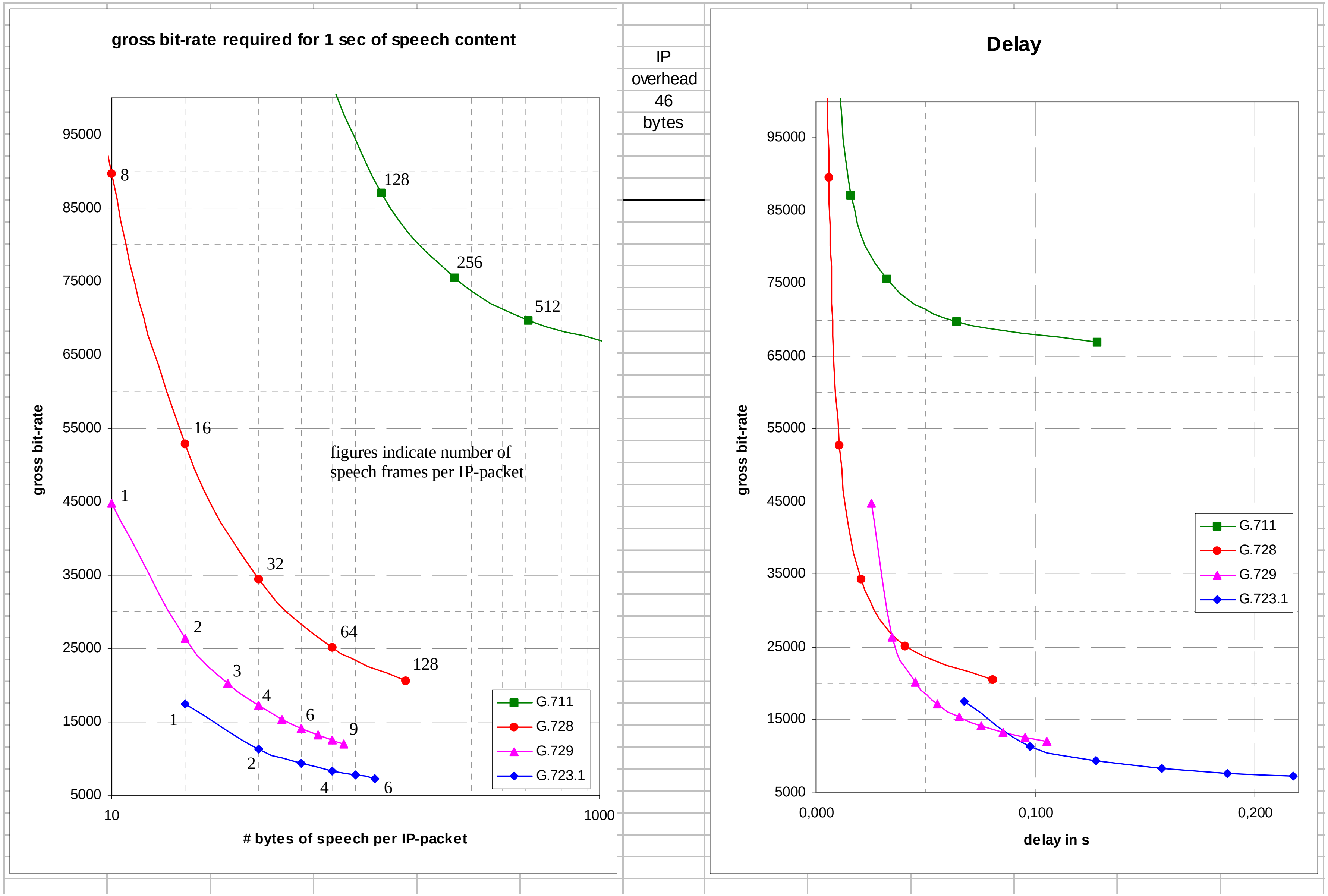


Figure 10: "Real" load of the network

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 analyzed 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

Whereas clause 4.1 provides the general statement that an Advantage Factor A does not apply for TIPHON systems, the following clauses are intended to provide guidance on how A can be applied in individual cases.

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 [21] and ITU-T Recommendation G.107 [17]). 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 [17].

While some investigations (e.g. for GSM) tend to confirm the values for A given in ITU-T Recommendation G.107 [17], 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 [18]). 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 [17] 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 [17] and G.108 [20] 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 [17].

- 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 [11]).

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 8 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 7.

Table 7: 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 11 gives the general structure of the example end-to-end connection under consideration in the following clauses.

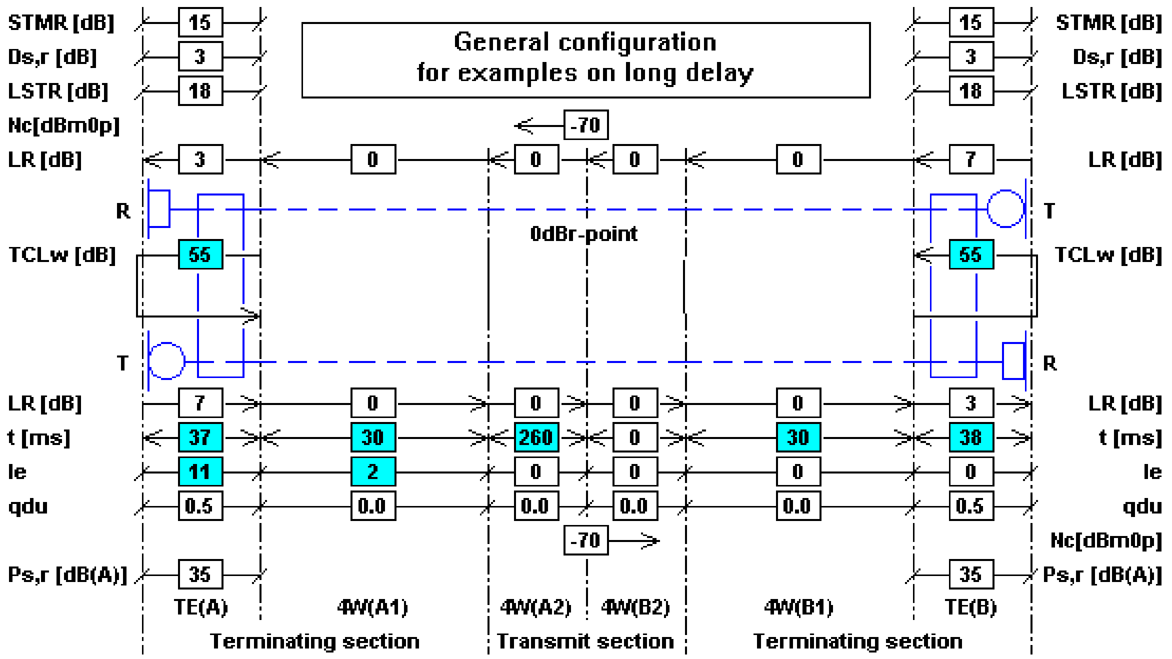


Figure 11: General configuration for examples regarding long delay

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

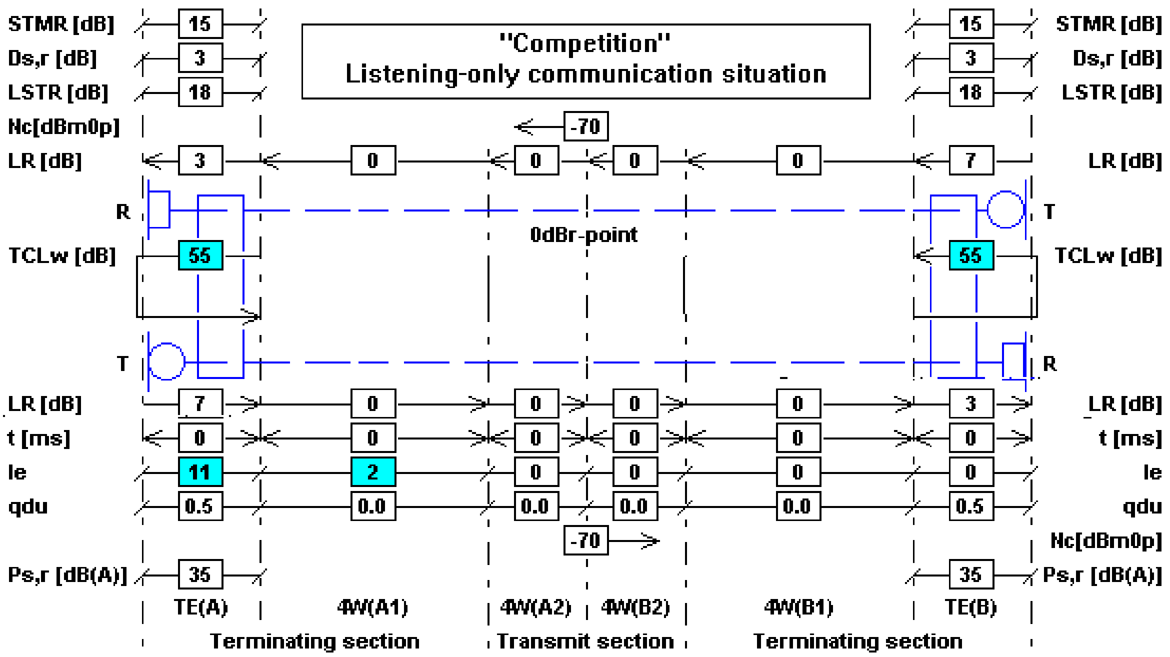
- Codec = G.729 A with VAD, Ie = 11 (see ITU-T Recommendation G.113 [18]);
- Terminal Mode B, delay = 75 ms (see TS 101 329-2 [1]) [split into 37 ms left and 38 ms right side];
- Network Class I, additional Ie = 2 (due to 0,5 % packet loss, see TS 101 329-2 [1] and ITU-T Recommendation G.113 [18]);
- Satellite (geo-stationary), delay = 260 ms (between earth stations, see ITU-T Recommendation G.114 [11]);

- 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 [17]).

NOTE: 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

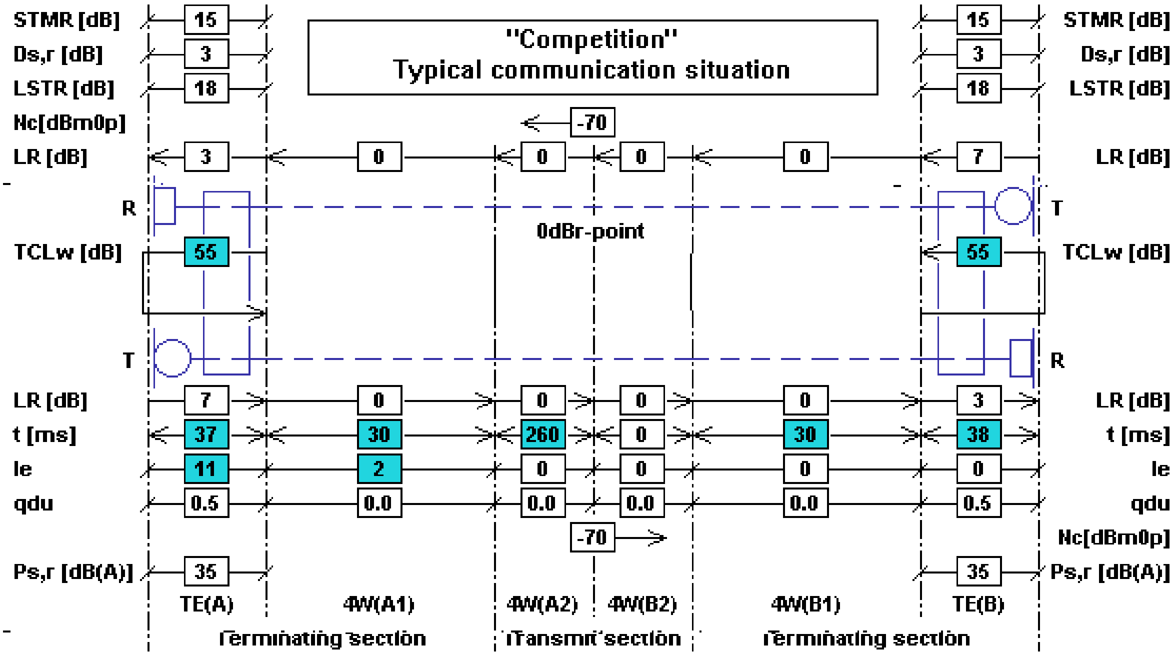


Impairment left side					Whole connection	Impairment right side				
R	Itot	Is	Id	Ie	Advantage Factor A	R	Itot	Is	Id	Ie
80,4	14,0	1,4	0,1	13	0	80,4	14,0	1,4	0,1	13

Figure 12: Competition & listening-only communication situations

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

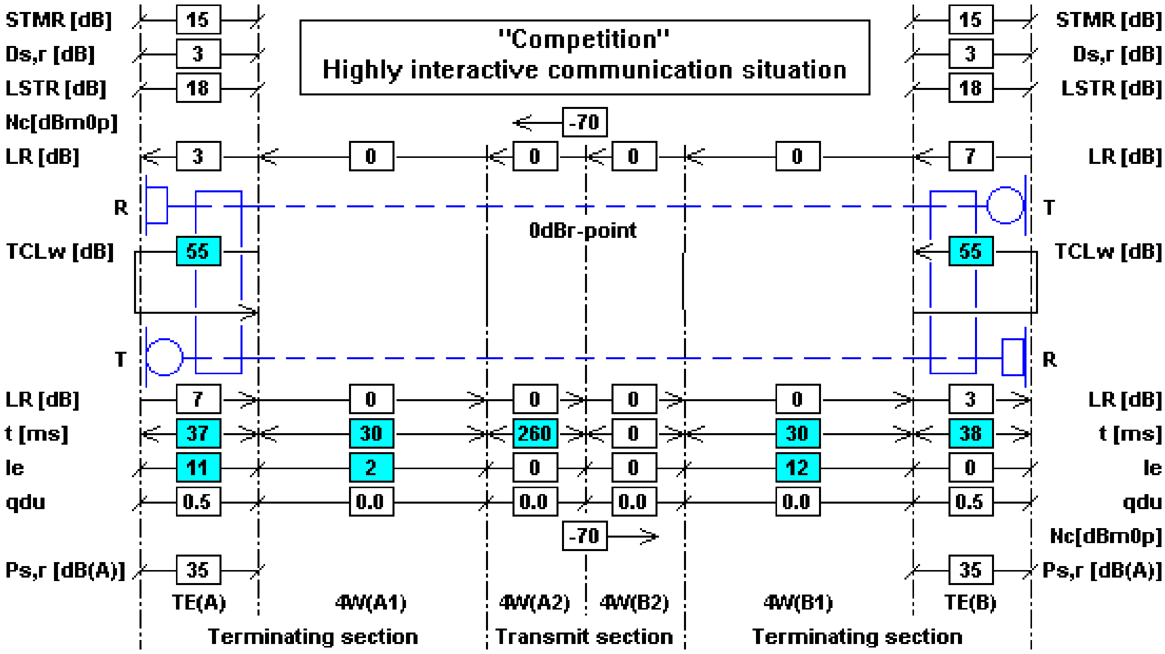


Impairment left side					Whole connection	Impairment right side				
R	Itot	Is	Id	Ie	Advantage Factor A	R	Itot	Is	Id	Ie
49,8	44,5	1,4	30,7	13	0	49,8	44,5	1,4	30,7	13

Figure 13: Competition & typical communication situations

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



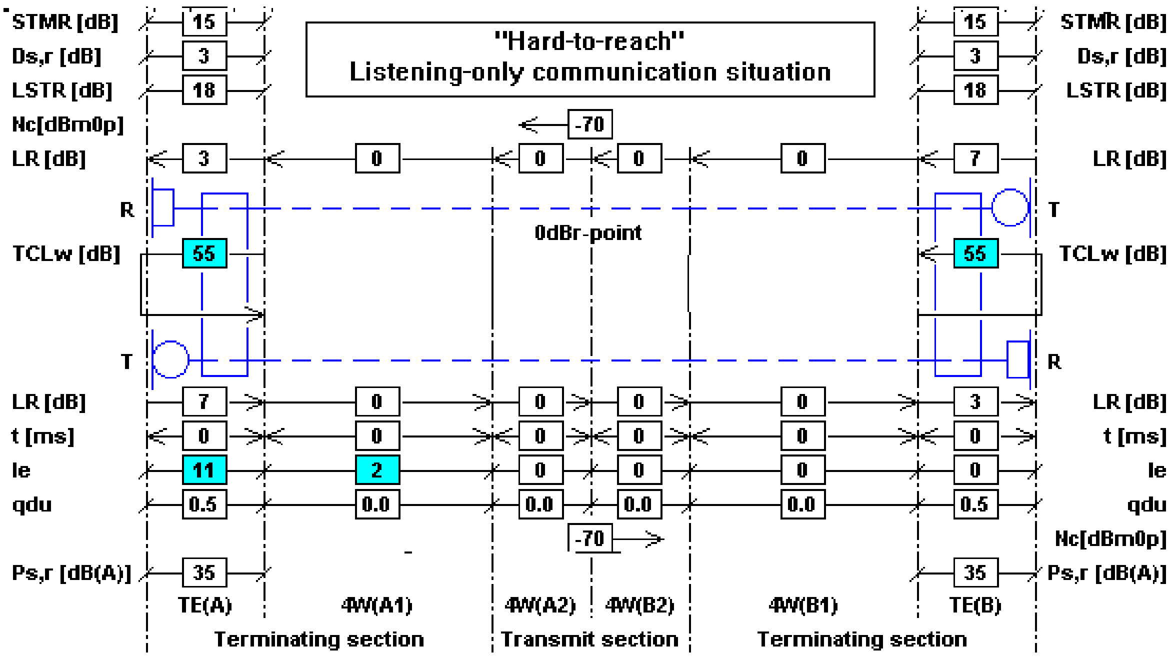
Impairment left side					Whole connection	Impairment right side				
R	Itot	Is	Id	Ie	Advantage Factor A	R	Itot	Is	Id	Ie
37,8	56,5	1,4	30,7	25	0	37,8	56,5	1,4	30,7	25

Figure 14: Competition & highly interactive communication situations

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

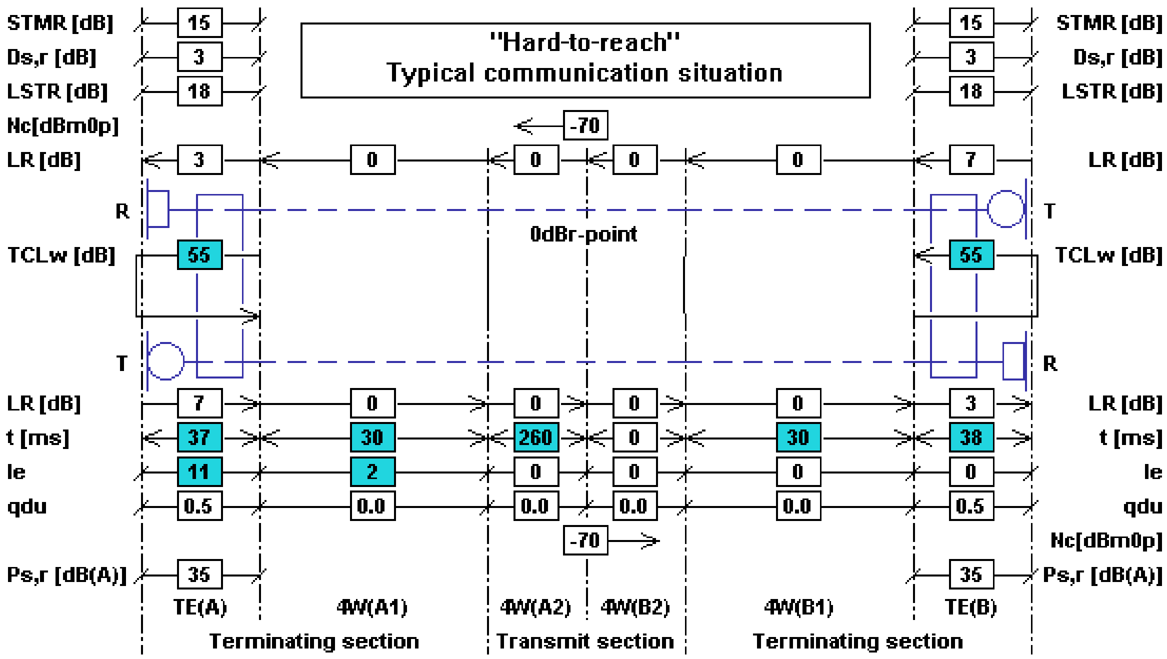


Impairment left side					Whole connection	Impairment right side				
R	Itot	Is	Id	Ie	Advantage Factor A	R	Itot	Is	Id	Ie
80,4	14,0	1,4	0,1	13	0	80,4	14,0	1,4	0,1	13

Figure 15: "Hard-to-reach" & listening-only communication situations

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

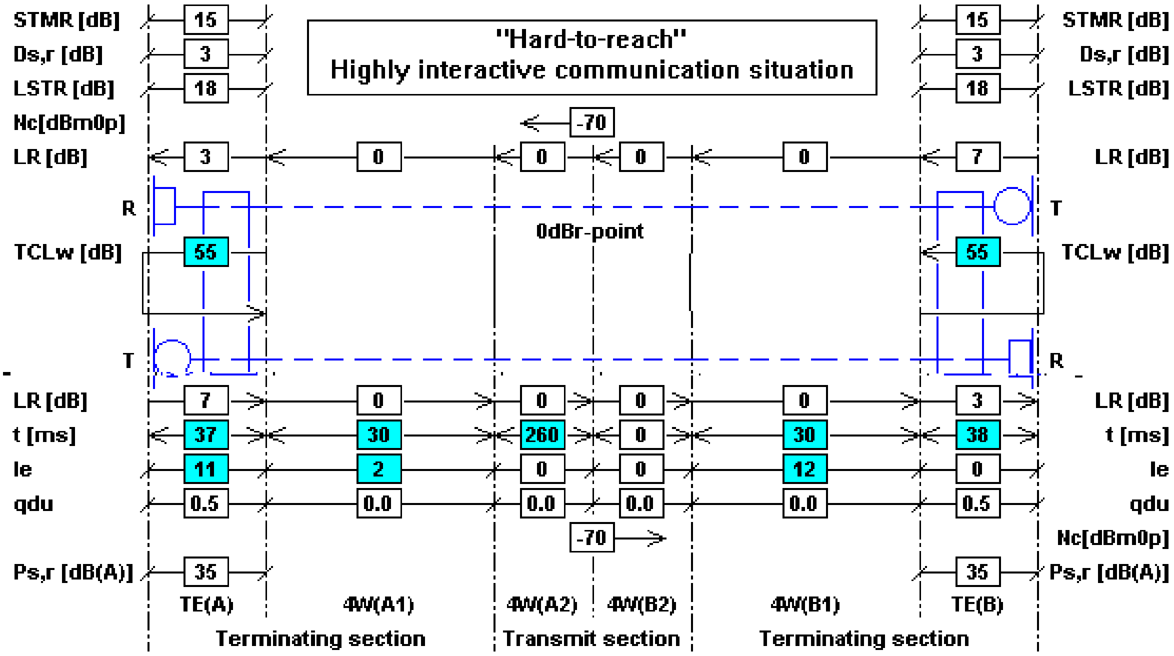


Impairment left side					Whole connection	Impairment right side				
R	Itot	Is	Id	Ie	Advantage Factor A	R	Itot	Is	Id	Ie
61,8	44,5	1,4	30,7	13	12	61,8	44,5	1,4	30,7	13

Figure 16: "Hard-to-reach" & typical communication situations

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 17: "Hard-to-reach" & highly interactive communication situations

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 8 summarizes the values for the E-model Rating R resulting from the six example calculations.

Table 8: 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 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 8 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.

6.2.3 VoIP end-to-end delay budget planning for private networks

This clause provides a more detailed view of VoIP one-way end-to-end delay sources in a private IP network or intranet. End-to-end delay will be used synonymously with one-way delay in this clause. Clause 6.3.2.2 covers delay sources in an example worst-case end-to-end private network. Clauses 6.3.2.3 and 6.3.2.4 show detailed end-to-end delay budget planning in a VoIP network for a G.729A vocoder and shows how the end-to-end delay is affected by the voice packet size, link speed and maximum data packet size. Although this clause covers the delay budget planning for the G.729A vocoder only, the same planning rules can be applied to any other vocoder.

6.2.3.1 VoIP end-to-end delay sources overview

Figure 18 shows a VoIP end-to-end private network connection and lists the main delay sources for each section of the network. There are basically two types of delay source, fixed or variable and each delay source in figure 18 is listed in one of the two categories.

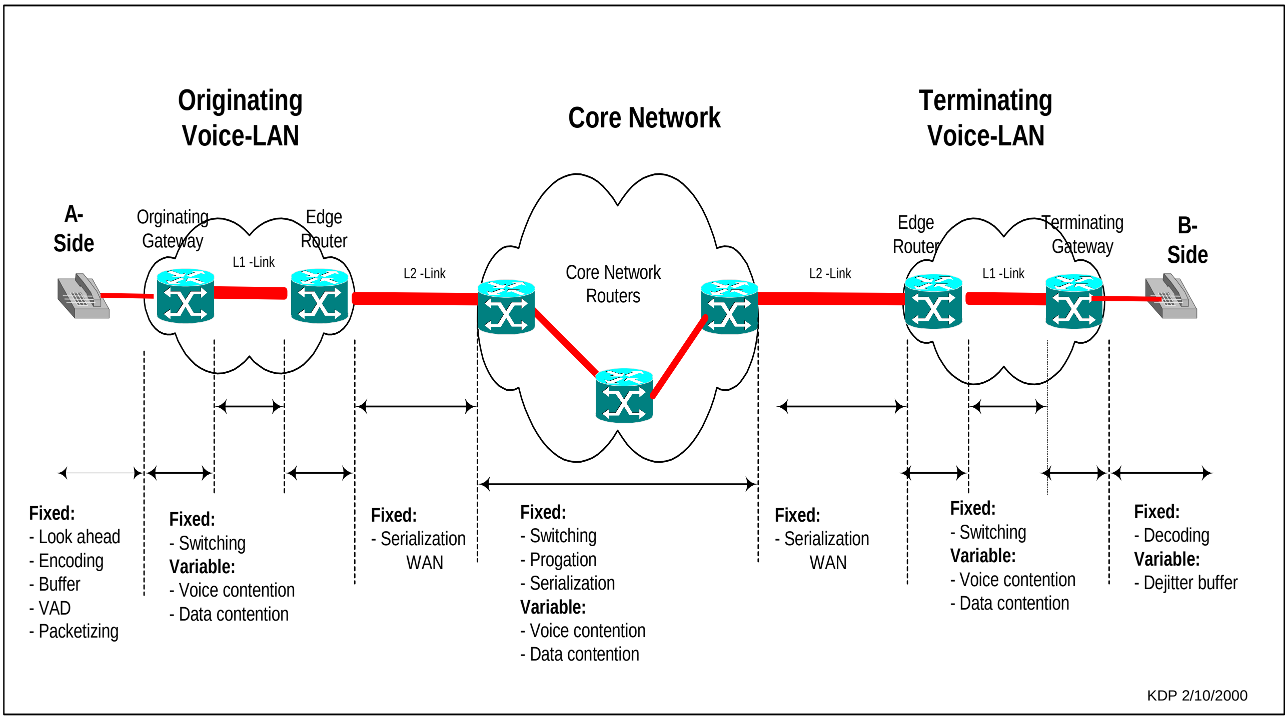


Figure 18: VoIP End-to-End delay sources for Private network case

6.2.3.2 VoIP end-to-end delay sources definitions

Vocoder Encoding

Details on the vocoder delays are from ITU-T Recommendation G.114 [11] and also see section 5.2.1 of TIA/EIA/TSB116. This consists of fixed delays, look ahead, the encoding process and packetization. There is also the additional serialization delay to transmit the packets over the 10/100 Base T link, but this is negligible (much less than 1ms) so it is ignored.

Originating Voice-LAN

Fixed switching delay:

- through the edge switch can be significant since forwarding engines in the edge switch are not very fast.

Variable Voice contention delay:

- is delay due to contention between voice packets for the link bandwidth. Average queue delays caused by contention between voice packets sharing the same queuing priority can be modeled using the queuing theory formula for fairly constant bit rate traffic sharing a single queue is:
  - Average voice queuing time is: $t_{Q\text{-}av} = t_{dls} \times \sigma/2 \times (\sigma\text{-}1)$
  - Worst case queuing time is (95 % of distribution): $t_{Q\text{-}wo} = 2 \times t_{Q\text{-}av}$
  - Where t_{dls} is Voice packet link serialization delay
  - σ is the link utilization of voice packets

Variable Data/voice contention delay:

- is delay due to contention between voice packets and data packet, where data packet has already started transmission. When forwarding node uses priority scheduling algorithms for differentiated QoS between voice and data classes, than the maximum time voice packet is delayed by the data packet is:
  - $t_{D\text{-}max}$ = (Maximum # Data MTU bytes + 48 overhead)/(link speed kbps/8).

Important planning rule: need to use priority scheduling for voice class traffic, as well as RTP header compression and data packet fragmentation on slow speed links to minimize the contribution of this variable delay source.

Fixed Serialization WAN delay:

- is delay due to voice packets transmission on the WAN L2- link. The link rate can vary from 56kb/s to OC3 and up. The formula for serialization delay is:
  - $t_{V\text{-}max}$ = (Voice packet bytes + 48 overhead)/(link speed kbps/8).

Important planning rule: in order to minimize the effect of this delay source, avoid using slow serial links in any of the end-to-end network connections.

Core Network

Fixed switching delay:

- includes packet switching engine delay (see originating Voice-LAN section for details) and any other network multiplexing equipment delays. An estimate of 1 ms of delay for each hop is used in the calculation table in the next section.

Fixed propagation delay:

- is the cumulative delay due to the physical "speed of light" limitations of propagation through the network. Details for this are contained in ITU-T Recommendation G.114 [11]. For the purpose of this exercise a figure of 5 μs/km is used in the calculation of the table in clause 6.2.3.3.

Fixed Serialization delay network:

- is the same as defined earlier, but since the link rate in the core network is usually in the broadband range, the total effect of this delay source is small enough (< 1,5 ms) that it is ignored in the calculation table in the next section.

Variable Voice contention delay:

- is same as defined earlier:

Average voice queuing time is: $t_{Q\text{-}av} = t_{dls} \times \sigma/2 \times (\sigma\text{-}1)$

Worst case queuing time is (95 % of distribution): $t_{Q\text{-}wo} = 2 \times t_{Q\text{-}av}$

Total core network worst case queuing time is (95 % of distribution): = $t_{Q\text{-}wo}$ × (number of hops -1)

Since the link rate in the core network is usually in the broadband range, the tdls delay source is small in addition σ, link utilization ratio for voice packet is small, that the total effect of this delay source can be ignored in the calculation tables in clause 6.2.3.3.

Variable Data/voice contention delay:

- is same as defined earlier:
  - $t_{D\text{-}max}$ = (Maximum # Data MTU bytes + 48 overhead)/(link speed kbps/8).

Total core network maximum data MTU queuing time is: = $t_{Q\text{-}wo}$ × (number of hops -1).

Important planning rule: need to use priority scheduling for voice class traffic, as well as RTP header compression and data packet fragmentation on slow speed links to minimize the contribution of this variable delay source.

Terminating Voice-LAN

Fixed Serialization WAN:

- is delay due to voice packet transmission on the WAN L2- link. The link rate can vary from 56kb/s to OC3 and up. The formula for serialization delay is:
  - $t_{V\text{-}max}$ = (voice packet bytes + 48 overhead)/(link speed kbps/8)

Important planning rule: in order to minimize the effect of this delay source, avoid using slow serial links in any of the end-to-end network connections.

Vocoder decoder

Variable dejitter buffer delay:

- Is the delay required to buffer all the variable delays in the network so that the voice packets can be played at constant bit-rate to the decoder. The size of dejitter buffer is, vocoder encoding compression amount plus the total variable delay in the end-to-end connection.

Fixed decoder:

- Details on the vocoder decoding delay is detailed in ITU-T Recommendation G.114 [11].

Variable voice contention delay:

Queuing delay is due to competition between voice packets for the link capacity. If enough CBR voice streams are multiplexed the maximum queuing delay caused by contention between voice flows of the same priority can be calculated using the M/G/1 model. If all flows use the same packet size the M/G/1 model boils down to the M/D/1 model. When several queues are traversed (and queuing delays incurred in successive nodes are assumed to be statistical independent) the probability density function of the total queuing delay is a convolution of the individual probability density functions. The formulas below are only valid for identical nodes (i.e. same load and link rate), see also [1]. When the nodes are not identical one can calculate the maximum queuing delay per node and accumulate them.

Note that this approach provides only a very worst case estimation of the maximum end-to-end queuing delay. In the case of N identical M/D/1 nodes loaded to a load ρ.

The average voice queuing time is: $$t_{Q-av} = t_{dls} \cdot N \cdot \frac{\rho}{2(1-\rho)}$$

And the variance of the voice queuing time: $$\sigma_N = t_{dls} \cdot \sqrt{N} \sqrt{\left(\frac{\rho}{2(1-\rho)}\right)^2 + \frac{\rho}{3(1-\rho)}}$$

According to [1] the maximum (1-P)-quantile) end-to-end queuing delay is: $w_N = t_{Q-av} + \alpha(P) \cdot \sigma$

t_{dls} Voice packet link serialization delay

ρ is the link utilization of voice packets

***N* number of nodes**
***P* is the fraction of packets that arrive later than w_N.**
W_N is the maximum queuing time over *N* nodes

With $\alpha(P) = \frac{E_N^{-1}(P) - N}{\sqrt{N}}$ where $E_N^{-1}(P)$ is the inverse of the Erlang tail distribution with *N* stages.

Some values for $\alpha_N(P)$ are given in table 9.

Table 9: Values for $\alpha_N(P)$ for different number of nodes and different quantiles

	95%	96%	97%	98%	99%
N	0,05	0,04	0,03	0,02	0,01
1	1,9957	2,2189	2,5066	2,9120	3,6052
2	1,9402	2,1303	2,3730	2,7110	3,2798
3	1,9028	2,0778	2,3001	2,6077	3,1211
4	1,8768	2,0427	2,2526	2,5421	3,0225
5	1,8575	2,0171	2,2186	2,4956	2,9537
6	1,8424	1,9974	2,1927	2,4605	2,9020
7	1,8303	1,9816	2,1720	2,4327	2,8614
8	1,8201	1,9685	2,1550	2,4100	2,8284

6.2.3.3 VoIP End-to-end Delay Budget Case 1

Table 10: Case 1a - VoIP End-to-end delay budget

Case 1a:	L1 = 10 Mb/s; L2 = 128 kb/s; Data MTU max = 128					
Delay type	Codec type:	Units	G.729 10,00 Fixed (ms)	G.729 10,00 Variable (ms)	G.729 20,00 Fixed (ms)	G.729 20,00 Variable (ms)
A-side phone						
Encoding process delay	Codec Look ahead	ms	5,0		5,0	
	Encoding compression	ms	10,0		10,0	
	1xbuffer	ms	10,0	~	10,0	~
Packetization delay		ms	0,0	~	10,0	~
# of Voice bytes/pkt		Bytes	*10,0*		*20,0*	
Orginating Voice-LAN						
Switching	1 hops, @ > 100 pps	ms	10,0		10,0	
Voice contention queuing	voice packets queuing @ 128 kb/s (Max 2 × SD)	ms		1,5		2,9
Data Queuing	Max. data unit 128 bytes +48 O/H @ 128 kb/s	ms		11,0		11,0
Serialization WAN delay	Voice pkt + 48 O/H @ 128kb/s	ms	3,6	~	4,3	~
Core Network						
Switching	5 hops, @ > 1k pps	ms	5,0		5,0	
Voice contention queuing	voice packets queuing @ 1 544 kb/s (Max 2 × SD)	ms		0,1		0,3
Data Queuing	5 hops, Max data 128+48 O/H @ 1 544 kb/s avg	ms		3,6		3,6
Serialization core	Voice pkt + 48 O/H @ 1 544 kb/s	ms	1,2	~	1,4	~
Propogation delay	5 000 km @ 5us/km	ms	25,0	~	25,0	~
Terminating Voice-LAN						
Serialization WAN delay	Voice pkt + 48 O/H @ 128 kb/s	ms	3,6	~	4,3	~
Switching	1 hops, @ > 100 pps	ms	10,0		10,0	
B-side phone						
Dejitter buffer delay	1 comp. delay + network variable delay	ms	10,0	16,2	10,0	17,8
Decoding delay		ms	10,0	~	10,0	~
	Min/Max		103,5 103,5	32,5 135,9	114,9 114,9	35,7 150,6

Table 11: Case 1b - VoIP End-to-end delay budget

Case 1b:	L1 = 10Mb/s; L2 = 128kb/s; Data MTU max= 512					
Delay type	Codec type:	Units	G.729 10,00 Fixed (ms)	G.729 10,00 Variable (ms)	G.729 20,00 Fixed (ms)	G.729 20,00 Variable (ms)
A-side phone						
Encoding process delay	Codec Look ahead	ms	5,0		5,0	
	Encoding compression	ms	10,0		10,0	
	1xbuffer	ms	10,0	~	10,0	~
Packetization delay		ms	0,0	~	10,0	~
# of Voice bytes/pkt		Bytes	*10,0*		*20,0*	
Orginating Voice-LAN						
Switching	1 hops, @ > 100 pps	ms	10,0		10,0	
Voice contention queuing	voice packets queuing @ 128 kb/s (Max 2 × SD)	ms		1,5		2,9
Data Queuing	Max. data unit 512 bytes + 48 O/H @ 128 kb/s	ms		35,0		35,0
Serialization WAN delay	Voice pkt + 48 O/H @ 128 kb/s	ms	3,6	~	4,3	~
Core Network						
Switching	5 hops, @ > 1k pps	ms	5,0		5,0	
Voice contention queuing	voice packets queuing @ 1 544 kb/s (Max 2 × SD)	ms		0,1		0,3
Data Queuing	5 hops, Max data 512+48 O/H @ 1 544 kb/s avg	ms		11,6		11,6
Serialization core	Voice pkt + 48 O/H @ 1 544 kb/s	ms	1,2	~	1,4	~
Propogation delay	5 000 km @ 5 us/km	ms	25,0	~	25,0	~
Terminating Voice-LAN						
Serialization WAN delay	Voice pkt + 48 O/H @ 128kb/s	ms	3,6	~	4,3	~
Switching	1 hops, @ > 100 pps	ms	10,0		10,0	
B-side phone						
Dejitter buffer delay	1 comp. delay + network variable delay	ms	10,0	48,2	10,0	49,8
Decoding delay		ms	10,0	~	10,0	~
			103,5	96,4	114,9	99,6
	Min/Max		103,5	199,9	114,9	214,5

6.2.3.4 VoIP End-to-end Delay Budget Case 2

Table 12: Case 2a - VoIP End-to-end delay budget

Case 2a: L1 = 10Mb/s; L2 = 1544kb/s; Data MTU max= 128

Delay type	Codec type:	Units	G.729 10,00 Fixed (ms)	G.729 10,00 Variable (ms)	G.729 20,00 Fixed (ms)	G.729 20,00 Variable (ms)
A-side phone						
Encoding process delay	Codec Look ahead	ms	5,0		5,0	
	Encoding compression	ms	10,0		10,0	
	1xbuffer	ms	10,0	~	10,0	~
Packetization delay		ms	0,0	~	10,0	~
# of Voice bytes/pkt		Bytes	*10,0*		*20,0*	
Orginating Voice-LAN						
Switching	1 hops, @ > 100 pps	ms	10,0		10,0	
Voice contention queuing	voice packets queuing @ 128 kb/s (Max 2 × SD)	ms		0,1		0,2
Data Queuing	Max. data unit 128 bytes +48 O/H @ 1 544 kb/s	ms		0,9		0,9
Serialization WAN delay	Voice pkt + 48 O/H @ 1 544 kb/s	ms	0,3	~	0,4	~
Core Network						
Switching	5 hops, @ > 1k pps	ms	5,0		5,0	
Voice contention queuing	voice packets queuing @ 1 544 kb/s (Max 2 × SD)	ms		0,1		0,3
Data Queuing	5 hops, Max data 128+48 O/H @ 1 544 kb/s avg	ms		3,6		3,6
Serialization core	Voice pkt + 48 O/H @ 1 544 kb/s	ms	1,2	~	1,4	~
Propogation delay	5 000 km @ 5 us/km	ms	25,0	~	25,0	~
Terminating Voice-LAN						
Serialization WAN delay	Voice pkt + 48 O/H @ 128 kb/s	ms	0,3	~	0,4	~
Switching	1 hops, @ > 100 pps	ms	10,0		10,0	
B-side phone						
Dejitter buffer delay	1 comp. delay + network variable delay	ms	10,0	4,8	10,0	5,1
Decoding delay		ms	10,0	~	10,0	~
			96,8	9,6	107,1	10,2
	Min/Max		96,8	106,4	107,1	117,3

Table 13: Case 2b - VoIP End-to-end delay budget

Case 2b: L1 = 10Mb/s; L2 = 1544kb/s; Data MTU max = 512

Delay type	Codec type:	Units	G.729 10,00 Fixed (ms)	G.729 10,00 Variable (ms)	G.729 20,00 Fixed (ms)	G.729 20,00 Variable (ms)
A-side phone						
Encoding process delay	Codec Look ahead	ms	5,0		5,0	
	Encoding compression	ms	10,0		10,0	
	1xbuffer	ms	10,0	~	10,0	~
Packetization delay		ms	0,0	~	10,0	~
# of Voice bytes/pkt		Bytes	*10,0*		*20,0*	
Orginating Voice-LAN						
Switching	1 hops, @ > 100 pps	ms	10,0		10,0	
Voice contention queuing	voice packets queuing @ 1 544 kb/s (Max 2 × SD)	ms		0,1		0,2
Data Queuing	Max. data unit 512 bytes +48 O/H @ 1 544 kb/s	ms		2,9		2,9
Serialization WAN delay	Voice pkt + 48 O/H @ 1 544 kb/s	ms	0,3	~	0,4	~
Core Network						
Switching	5 hops, @ > 1k pps	ms	5,0		5,0	
Voice contention queuing	voice packets queuing @ 1 544 kb/s (Max 2 × SD)	ms		0,1		0,3
Data Queuing	5 hops, Max data 512+48 O/H @ 1 544 kb/s avg	ms		11,6		11,6
Serialization core	Voice pkt + 48 O/H @ 1 544 kb/s	ms	1,2	~	1,4	~
Propogation delay	5 000 km @ 5 us/km	ms	25,0	~	25,0	~
Terminating Voice-LAN						
Serialization WAN delay	Voice pkt + 48 O/H @ 1 544 kb/s	ms	0,3	~	0,4	~
Switching	1 hops, @ > 100 pps	ms	10,0		10,0	
B-side phone						
Dejitter buffer delay	1 comp. delay + network variable delay	ms	10,0	14,8	10,0	15,0
Decoding delay		ms	10,0	~	10,0	~
	Min/Max		96,8	29,5	107,1	30,1
			96,8	126,3	107,1	137,2

6.2.4 E-Model analysis of the VoIP over MPLS reference model

This clause analyses the requirements for Voice QoS on MSF Networks. MSF Networks are intended to support a full range of services including existing PSTN and Cellular services, these applications are analysed in the context of a Global Reference Connection. A number of scenarios are addressed which are typical of current PSTN and Cellular practice taking into account possible MSF Network deployments. It is concluded that ATM meets the requirements and that any Voice over IP or Voice over MPLS alternatives would need to match ATM CBR network performance.

6.2.4.1 Introduction

The ITU-T standards for voice network QoS are defined in relation to a Global Reference Connection, which is intended to represent the worst case international situation. This paper takes a PSTN call from Japan to East-coast USA and a GSM call from Australia to East-coast USA as being representative of Global Reference Connections having clear commercial significance.

The analysis is split into three distinct parts. In the first part we analyse scenarios where the VoMPLS deployment is constrained to the core of the network; in the second part, the MPLS deployment is extended into the access network; and in the third part, the impact of deploying different voice encoding schemes is analysed.

Three scenarios are presented:

- Calls traversing multiple MSF networks in each country deployed by different operators but inter-working through the current PSTN.
- Calls traversing a single MSF network in each country for the Japan to USA PSTN case, demonstrating the effect of current DCME practice.

- Calls traversing a single MSF network in each country for the Australia to USA GSM case, demonstrating the effect of current DCME practice.

The scenarios are analysed using the ITU-T E-Model transport modelling method G.107 [17]. The analysis covers the case of G.711 voice coding within the MSF domains, other coding schemes are for further study. The E-Model allows multiple sources of impairment to be quantified and the overall impact assessed. The result is expressed as an R-Value which is a rating of the assessment that real users would express if subjected to the voice impairments. Equations to convert E-model ratings into other metrics e.g. MOS, %GoB, %PoW can be found in annex B of G.107 [17]

The conclusion is drawn that, for a voice over packet technology to be freely deployable within MPLS networks then it should meet the following requirements:

- Intrinsic delay < 10 msec (Intrinsic delay = packetization + average read-out buffer delay).
- Negligible packet loss.

Intrinsic Delay is a characteristic of the voice over packet technology, which is incurred independently of the transmission delay. It is incurred once for each MSF network at the adaptation layer. Transmission delay is cumulative end-to-end and is evaluated based on SONET/SDH transmission.

6.2.4.2 MPLS in core networks only

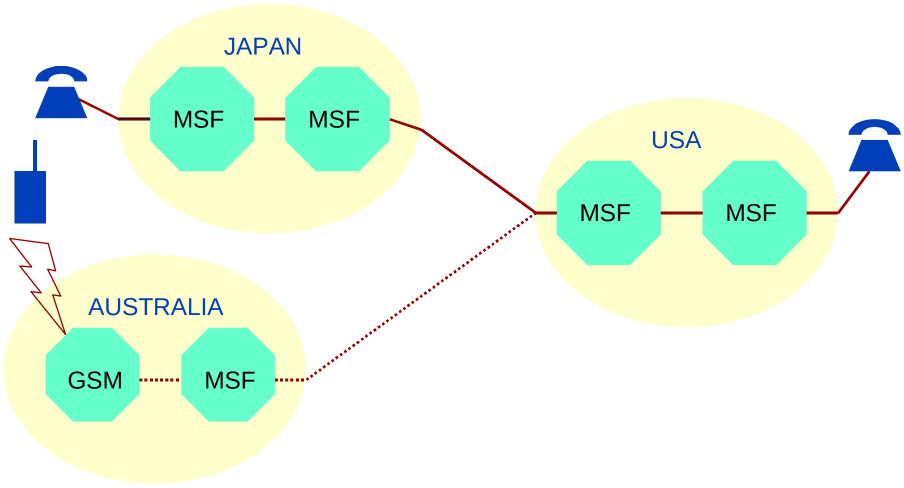
6.2.4.2.1 Scenario 1 - Multiple MPLS Networks

In this scenario two MPLS networks are traversed in Japan and the USA for the PSTN case and one MPLS network in Australia and two in the USA for the GSM case. The analysis covers a range of intrinsic delay (packetization + average read-out buffer) from 10 msec to 100 msec and the packet loss for each MPLS domain is set at 0 %, 0,5 % or 1,0 %. Each MPLS domain is assumed to have the same performance. It is assumed that the transmission delay corresponds to 1,5 times the greater circle distance between the two users.

A number of further assumptions are made on the basis of best possible practice in order to separate the contribution of multiple networks from other sources of impairment, in particular:

- DCME on the Japan to USA link is at full rate e.g. 32 kb/s G.726 [10] and Voice Activity Detection is not included.
- The Australia to USA link is G.711 i.e. there is no DCME.
- MSF domains use G.711 with packet loss concealment algorithm employed.
- GSM domain uses full rate codec and no Voice Activity Detection.
- Wired PSTN phones are analogue with echo-cancellers employed.

The results of the Multiple MSF (MPLS) scenario are illustrated in figure 19.



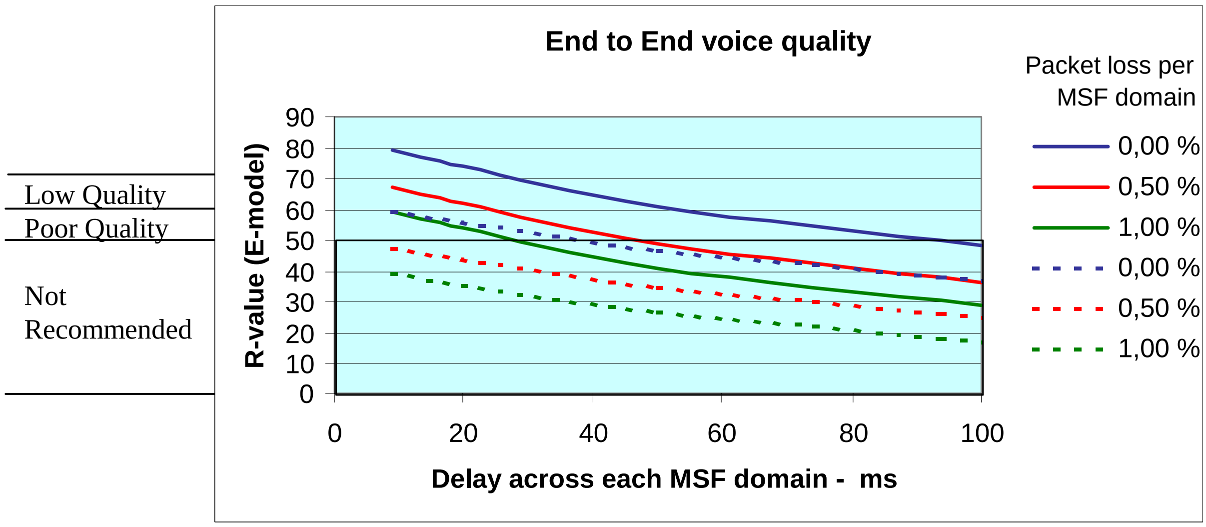


Figure 19: End-to-end voice quality with multiple msf networks

The results are presented as a graph showing the E-model rating against the Intrinsic Delay for a single MPLS network. Separate lines are shown for the three packet loss cases, the PSTN example uses solid lines and the GSM example dotted lines. It can be seen that with an Intrinsic Delay of 10 msec and 0 % packet loss then the PSTN case achieves a rating of 80 which is the normal target for PSTN. The GSM case achieves only a rating of only 60 which is rated as poor quality in the E-Model. It can be seen by inspection that any significant relaxation of the delay or packet loss objectives leads to operations with a rating of less than 50 which is outside recommended planning limits.

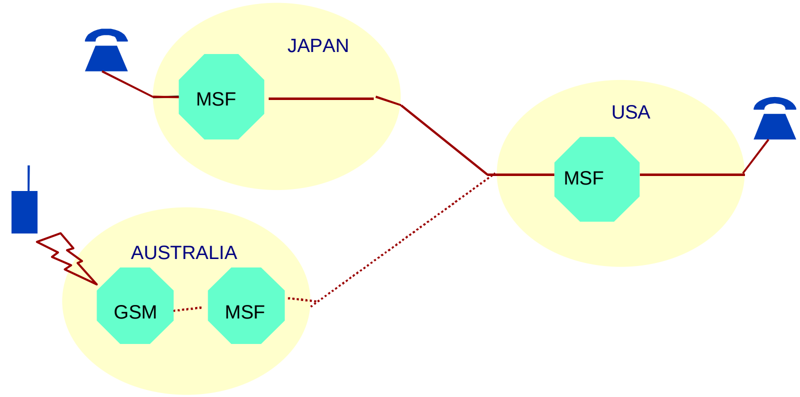
6.2.4.2.2 Scenario 2 - PSTN and typical DCME practice

In this case the scenario is simplified to a single MPLS network in Japan and the USA but the DCME scenario is changed to show the impact of Voice Activity Detection and downspeeding. The Intrinsic Delay of a single MPLS network is analysed from 10 msec to 100 msec and the packet loss scenarios cover the same 0 %, 0,5 % and 1,0 % possibilities.

The voice processing assumptions are as follows:

- DCME on the Japan to USA link uses Voice Activity Detection and includes the downspeeding of the G.728 coding to 12,8 kb/s.
- MSF domains use G.711 with packet loss concealment.
- Wired phones are analogue with echo-cancellers deployed.

The results from this scenario are shown in figure 20.



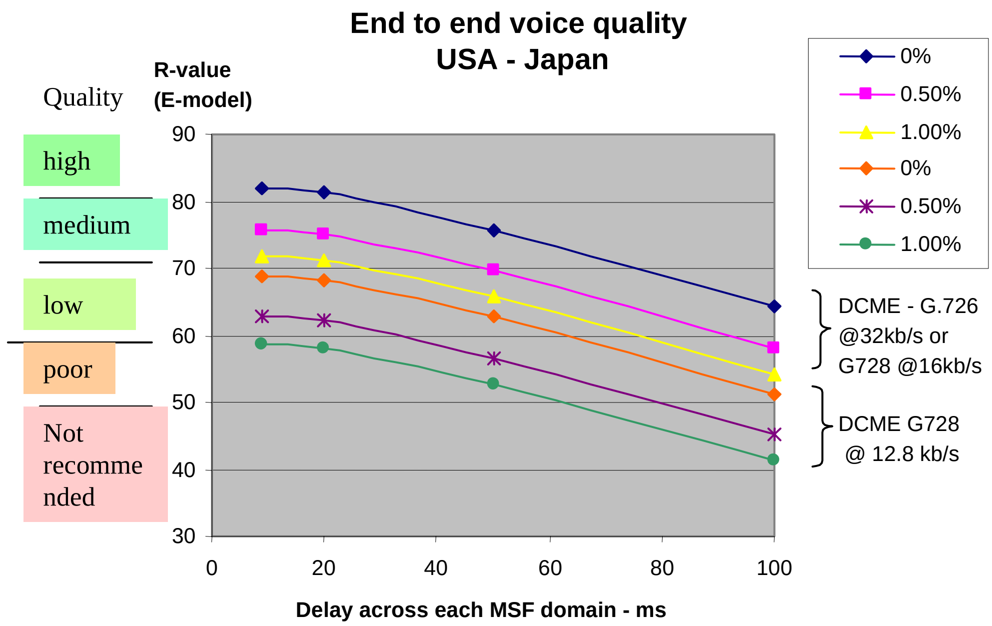


Figure 20: PSTN QoS with DCME options

The results are presented as a graph showing the E-model rating against Intrinsic Delay. Three lines are shown for full rate DCME i.e. G.726 at 32 kb/s and G.728 at 16 kb/s, one for each of the packet loss cases. A further three lines are shown for the case of G.728 DCME downspeeded to 12,8 kb/s.

It can be seen that with DCME downspeeding an intrinsic delay of 10 msec and 0 % packet loss is in the low quality range. Any significant relaxation would lead to poor quality or operation outside of planning limits.

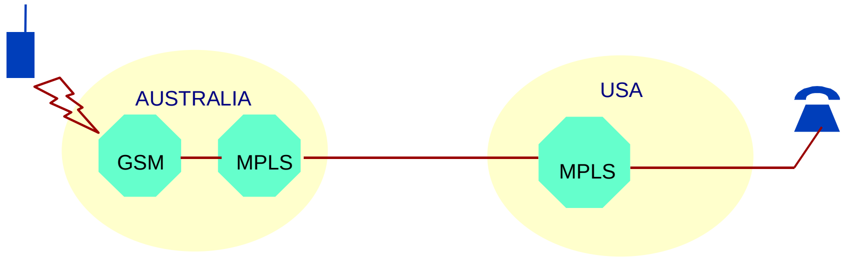
6.2.4.2.3 Scenario 3 - GSM and Typical DCME practice

In this scenario the network is simplified to a single MSF domain in Australia and another in the USA and the analysis covers the impact of typical DCME practice. In this case only 0 % packet loss is considered. Three DCME cases are considered, G.711 i.e. no DCME, G.726 at 32 kb/s and G.728 with downspeeding to 12,8 kb/s. The DCME equipment also includes voice activity detection. The GSM listener receives better QoS than the PSTN listener as a result of the asymmetrical operation of echo handling. Echo generated at the 2-4 wire conversion in the PSTN side is removed by an echo canceller whereas the GSM side, being 4-wire throughout, relies on the terminal coupling loss achieved by the handset itself to control any acoustic echo. For this calculation a weighted terminal coupling loss of 46 dB is assumed for the terminal.

The voice processing assumptions are as follows:

- MSF domains use G.711 with packet loss concealment.
- Wired phones are analogue with echo-cancellers deployed.

The results from this scenario are shown in figure 21:



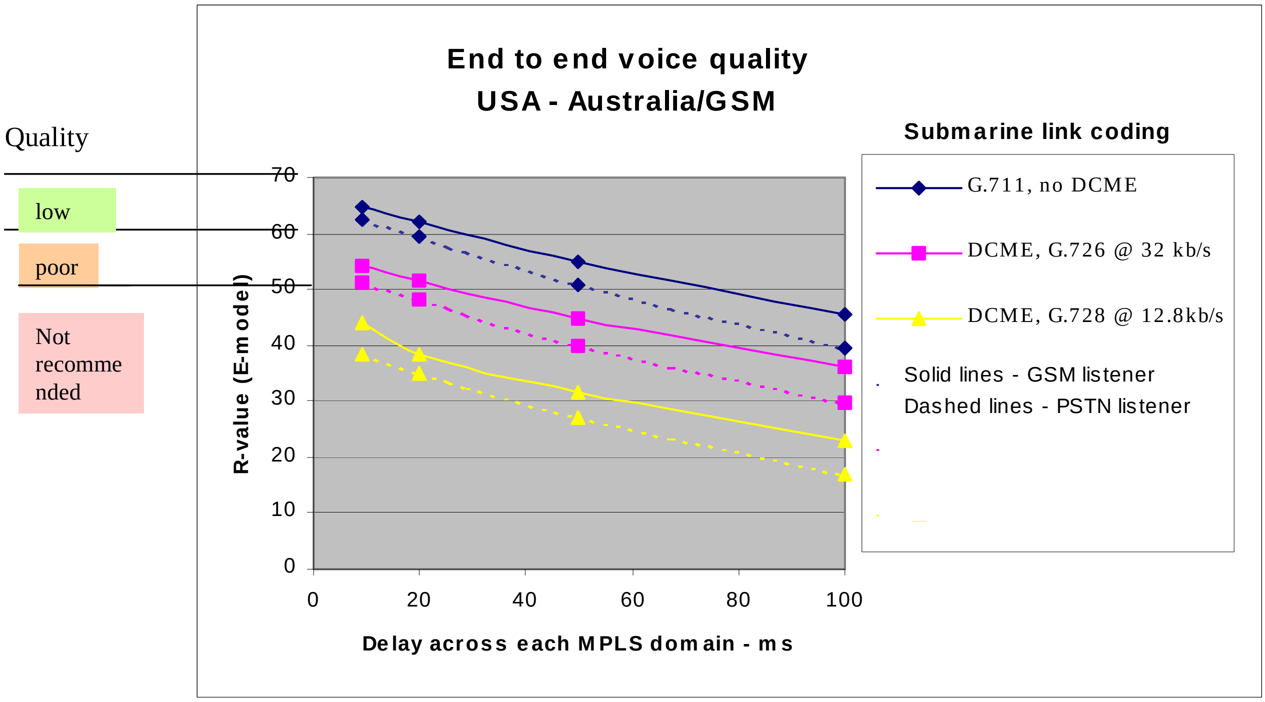


Figure 21: GSM QoS with DCME options

It can be seen by inspection that it is difficult to provide acceptable QoS for GSM calls on Global Reference Connections. DCME is typical practice in this case.

6.2.4.2.4 VoMPLS core network summary

Multiple MPLS islands are a natural consequence of switch deployment practice. A carrier wishing to deploy MPLS as a PSTN solution would wish to continue normal investment to cope with growth and retiring obsolete equipment. This will lead to multiple MPLS islands within a single carriers' network as well as islands which arise due to calls which are routed through multiple operators. It is possible to deploy equipment intelligently and to plan routing to avoid excessive numbers of islands, but if deployment is driven by growth and obsolescence then the transition to a full MPLS solution will take 15 to 20 years, during which time multiple islands will be the normal situation. Solutions, which lead to retrofit requirements in order to solve QoS problems, are very unlikely to be cost effective.

In international networks DCME is deployed at the transport layer, if calls are switched through multiple transit nodes then multiple stages of DCME are invoked. A typical rule for international network planning is to allow a worst case of three DCME equipments deployed in tandem. This will lead to QoS results considerably worse than the scenarios analysed above.

The scenarios represent simple routing cases, it is possible with call forwarding to incur multiple echo cancellers which can lead to further deterioration in each of the scenarios.

For a technology to be suitable for deployment as the core of a MSF network, then it should meet the following requirements:

- The intrinsic delay (packetization + average readout buffer) should be < 10 msec.
- Packet loss should be negligible.

6.2.4.3 Extending VoMPLS into the access network

The following scenarios analyse the impact of extending VoMPLS into the access network.

6.2.4.3.1 Scenario 4 - VoMPLS access on USA to Japan

In this scenario the core network comprises 2 MPLS networks in USA plus 2 MPLS networks in Japan linked by sub cable which may have DCME employed. The intrinsic delay within each core MPLS network is set to 10 ms delay and zero packet loss is assumed. The encoding scheme used is G.711 throughout. Figure 22 illustrates the deployments analysed. Four cases are considered:

- (A) MPLS access network each end, full echo control, no DCME
- (B) MPLS access network each end, no echo control, no DCME
- (C) MPLS access network one end; analogue PSTN other end, full echo control, DCME @32kb/s
- (D) MPLS access network one end; Analogue PSTN other end, full echo control, no DCME

Case A & B:

```
    TE --|MPLS|---|MPLS|--|MPLS|---------|MPLS|--|MPLS|--|MPLS|--TE

   Dig.  Access    Core    Core SUB-cable Core    Core   Access  Dig
```

Case C & D:

```
   TE --|CO|---|MPLS|--|MPLS|-----------|MPLS|--|MPLS|--|MPLS|--TE

   An   PSTN   Core    Core   SUB-cable  Core    Core   Access  Dig
```

Figure 22: Scenario 4 - Impact of VoMPLS access systems

The results for the analysis are shown in table 14 and figure 19 which provide results for various access delays (per access domain). For cases A and B the performance is symmetrical (digital terminals have identical performance) whereas for cases C and D the performance is slightly different at each end due to the different nominal loudness ratings of the analogue and digital terminals. The figures in the table refer to the listener at the analogue PSTN terminal - the performance at the digital terminal is slightly worse by about 5 points.

Table 14: R-Values for Scenario 4

Delay - ms	10	20	50	100	150
Case (A)	92,8	91,9	83,9	73,4	65,9
Case (B)	80,8	77,9	67,9	54,0	44,3
Case (C)	84,1	83,0	79,4	73,4	68,3
Case (D)	93,6	93,0	90,2	84,2	75,8

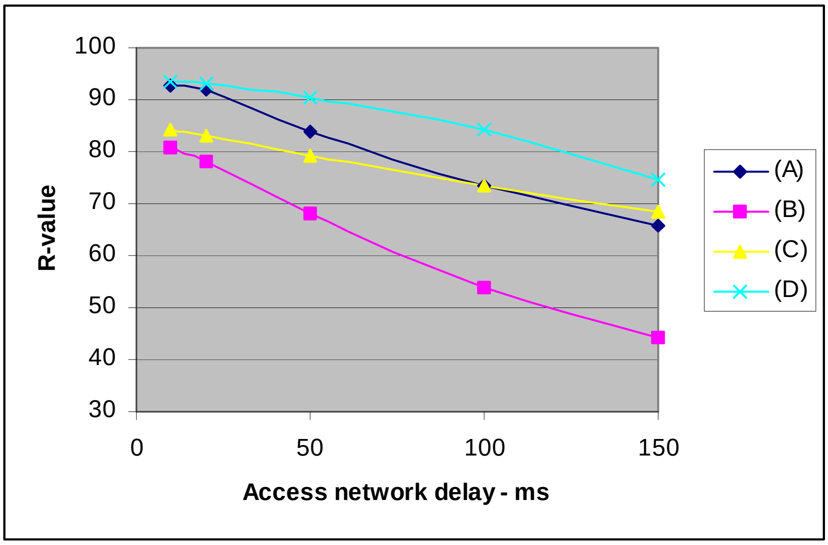


Figure 23: R-Values for scenario 4

The results show that if the MPLS access delay is restricted to 50 ms or below generally satisfactory results can be achieved for most scenarios.

6.2.4.3.2 Scenario 5 deployment of GSM and VoMPLS access

In this scenario the core network comprises 2 MPLS networks in USA plus 1 MPLS network and a mobile network in Australia linked by sub cable which does not have DCME employed. Each core MPLS network has 10 ms intrinsic delay and zero packet loss. Encoding G.711 throughout MPLS domains. Figure 24 illustrates the deployments analysed. Four cases are considered:

- E - Mobile = GSM FR codec, full echo control, no DCME.
- F - Mobile = GSM FR codec, no echo control, no DCME.
- G - Mobile = GSM EFR codec, full echo control, no DCME.
- H - Mobile = GSM EFR codec, no echo control, no DCME.

```
TE --|MPLS|---|MPLS|--|MPLS|-----------|MPLS|--|MPLS|--|GSM|---TE

Dig   Access   Core    Core  SUB-cable   Core   Core   Access  Dig
```

Figure 24: Scenario 5 - VoMPLS access with GSM

The results from the E-model analysis are given in table 15 and figure 25.

Table 15: R-Values for Scenario 5

Delay - ms	10	20	50	100	150
Case (E)	73,3	72,7	69,8	63,7	58,1
Case (F)	61,7	60,5	55,9	47,6	40,1
Case (G)	88,3	87,7	84,8	78,7	73,1
Case (H)	76,7	75,5	70,9	62,6	55,1

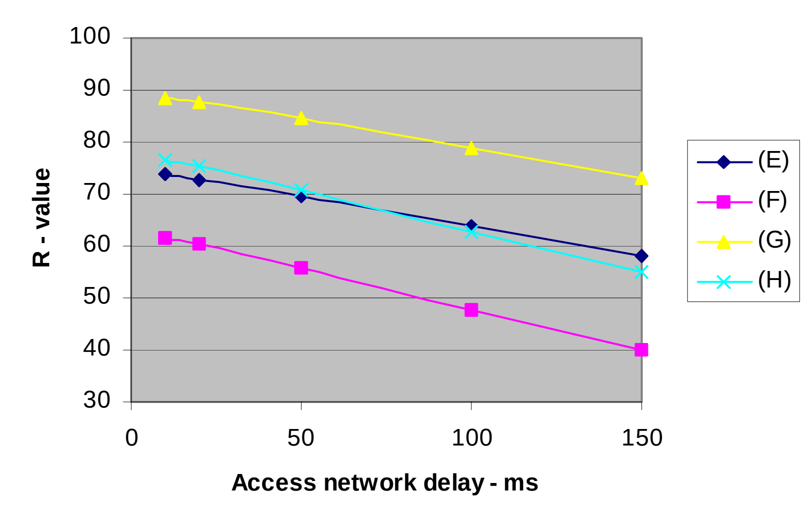


Figure 25: R-Values for scenario 5

Again the results show that MPLS access delays should be restricted to the order of 50 ms or below to achieve acceptable voice quality for the majority of connections.

The results also highlight the advantage of using the GSM EFR codec over the GSM FR codec and that even when working fully digital full echo control provides a measurable benefit.

6.2.4.4 Effects of Voice Codecs in the access network

In the final scenarios the impact of deploying voice codecs within the access network is considered.

6.2.4.4.1 Scenario 6 - Deployment of Codecs in one Access Leg (USA – Japan)

Again the core network comprises 2 MPLS networks in USA plus 2 MPLS networks in Japan linked by sub cable which has no DCME employed. Each core MPLS network has 10 ms intrinsic delay and zero packet loss. Encoding is G.711 throughout the core network. A fixed delay of 50 ms and zero packet loss is assumed in the access MPLS network. The configuration is illustrated in figure 26.

```
TE --|CO|---|MPLS|--|MPLS|-----------|MPLS|--|MPLS|--|MPLS|--TE

 (2ms)  (10ms)  (10ms)          (10ms)  (10ms)  (50ms) (var)
```

Figure 26: Scenario 6 - Effects of Codecs in one Access Leg

The results for various voice codec deployments are presented in table 16 and figure 27 which provide the R-values as experienced by the user of the PSTN and the MPLS access system.

Table 16: R-values for Scenario 6

Connection	PSTN	MPLS
G.711 to G.711	88,9	84,6
G.711 to G.729A + VAD (8kb/s)	73,7	69,9
G.711 to G.723A + VAD (6.3kb/s)	62,4	58,0
G.711 to G.723A + VAD (5.3kb/s)	58,4	54,0
G.711 to GSM-FR	61,7	57,3
G.711 to GSM-EFR	76,7	72,3

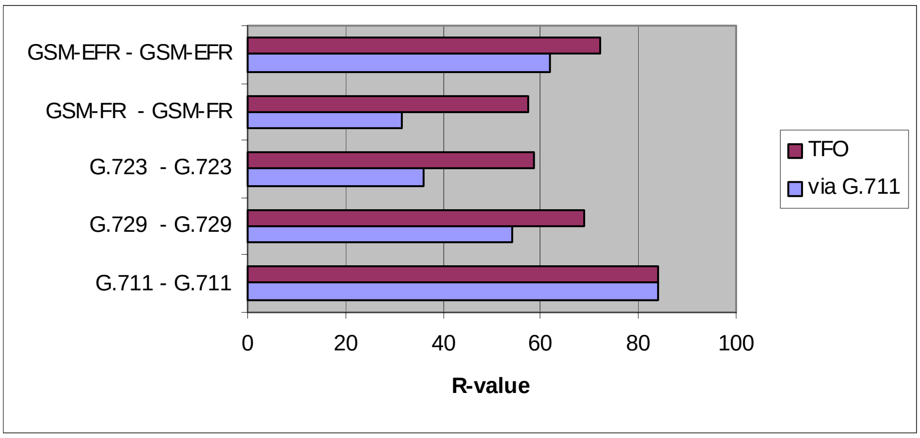


Figure 27: R-values for scenario 6

The results show asymmetrical performance due to the different nominal loudness ratings of the analogue and digital terminals. Generally acceptable performance is attained although the performance for the low bit rate G.723 coding scheme is marginal. In these examples since VoMPLS access is used for one leg of the connection only transcoding is performed once.

6.2.4.4.2 Scenario 7 - Codec Deployment in both Access Legs (USA - Japan)

The deployment configuration for this scenario is as scenario 6 with the exception that MPLS access systems are used at both ends. The configuration is illustrated in figure 28. and the resultant R-values provided in table 17 and figure 29.

```
TE --|MPLS|---|MPLS|--|MPLS|---------|MPLS|--|MPLS|--|MPLS|--TE


Dig    Access    Core    Core SUB-cable Core    Core    Access  Dig

(var)  (50ms)   (10ms)  (10ms)         (10ms)  (10ms)  (50ms) (var)
```

Figure 28: Scenario 7 - Codec Deployment in Both Access Legs

Table 17: R-value Results for Scenario 7

Connection	R-value
G.711 to G.711	83,9
G.729A+VAD to G.711 to G.729A+VAD (8.0kb/s)	54,2
G.729A+VAD (8.0kb/s) tandem free operation	68,9
G.723A+VAD to G.711 to G.723A+VAD (6.3kb/s)	36,2
G.723A+VAD (6.3kb/s) tandem free operation	58,6
GSM-FR to G.711 to GSM-FR	31,7
GSM-FR tandem free operation	57,2
GSM-EFR to G.711 to GSM-EFR	61,7
GSM-EFR tandem free operation	72,2

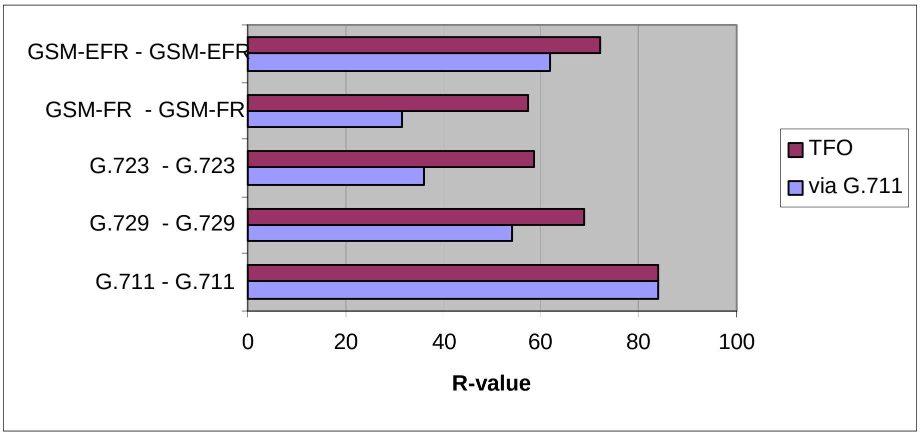


Figure 29: R-Value results for Scenario 7

The benefits of eliminating transcoding - tandem free operation (TFO) - can be clearly seen from these results. Further it can be seen that the performance attained by low bit rate G.723 is extremely poor when transcoding is performed at both access gateways.

6.2.4.4.3 Scenario 8 codec deployment and mobile access (USA - Australia)

The core network comprises 2 MPLS networks in the USA plus 1 MPLS network and a mobile network in Australia linked by sub cable which does not have DCME employed. Each core MPLS core network has 10 ms intrinsic delay and zero packet loss. The access network has 50 ms delay and zero packet loss. Full echo control is employed. For the UMTS mobile network, a delay of 60 ms and a codec impairment factor (Ie) of 5 is assumed based on the predicted performance of the GSM AMR codec. The results are provided in table 18 and figure 31.

```
    TE --|MPLS|--|MPLS|--|MPLS|---------|MPLS|--|MPLS|--|UMTS|---TE


   Dig   Access   Core    Core SUB-cable Core    Core   Mobile  Dig

(var)  (50 ms)  (10 ms)  (10 ms)         (10 ms)  (10 ms)  (60 ms) (var)
```

Figure 30: Scenario 8 - Codec Deployment and Mobile Access

Table 18: Results for Scenario 8

Connection	R-value
UMTS to G.711	78,7
UMTS to G.729A via G.711	63,9
UMTS to G.723A via G.711	53,6
UMTS to GSM-EFR	69,3
UMTS to UMTS - TFO	76,6

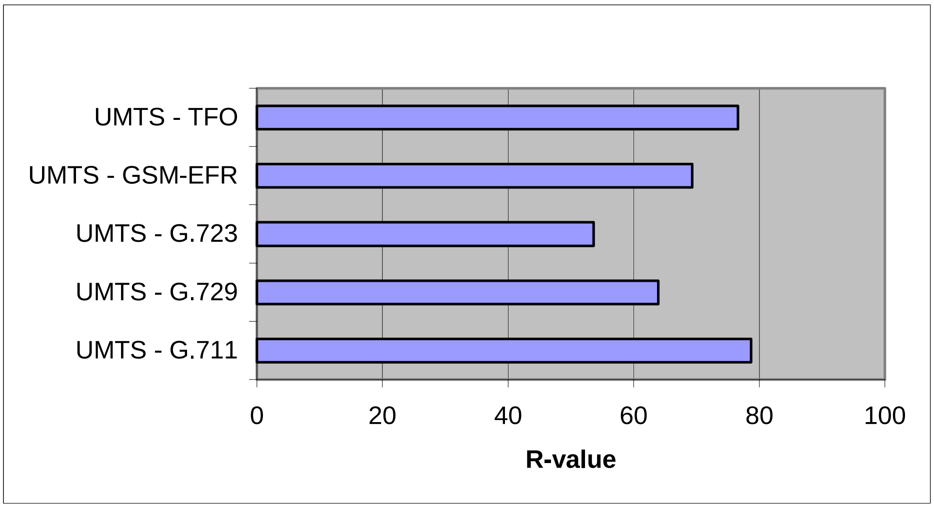


Figure 31: R-Value results for Scenario 8

Again these results highlight the significant benefit arising from the use of tandem free operation.

6.2.4.4.4 Voice codec summary

The scenarios in this clause highlight the critical impact that the voice coding scheme deployed in the access network will have on the overall voice quality. For international reference connections acceptable voice quality may not be attained with some of the very low bit rate codecs. The benefits of avoiding transcoding wherever possible can also clearly be seen.

6.2.4.5 Overall conclusions

The following key conclusions may be drawn from the study:

1) For VoMPLS core networks, per domain the intrinsic delay should not exceed 10 ms and the packet loss should be negligible.

2) When MPLS is extended to the access domain (in conjunction with the use of digital terminals) an additional 50 ms per access domain may be tolerated.

3) Wherever possible codec compatibility between the end-terminals should be negotiated to avoid the requirement for transcoding.

4) Where terminal compatibility cannot be achieved transcoding should be limited to one function per connection.

5) Low bit rate G.723 coding should be avoided unless transcoderless operation can be attained.

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 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 "narrowband high (2H)"

Figure 32 shows a connection with all values default according to ITU-T Recommendation G.107 [17], 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 #2H. 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 #2H with respect to R.

Figure 33 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 #2H with respect to R are met for both sides.

Figure 34 shows - based on the connection of figure 33 - 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 34 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 #2H with respect to R.

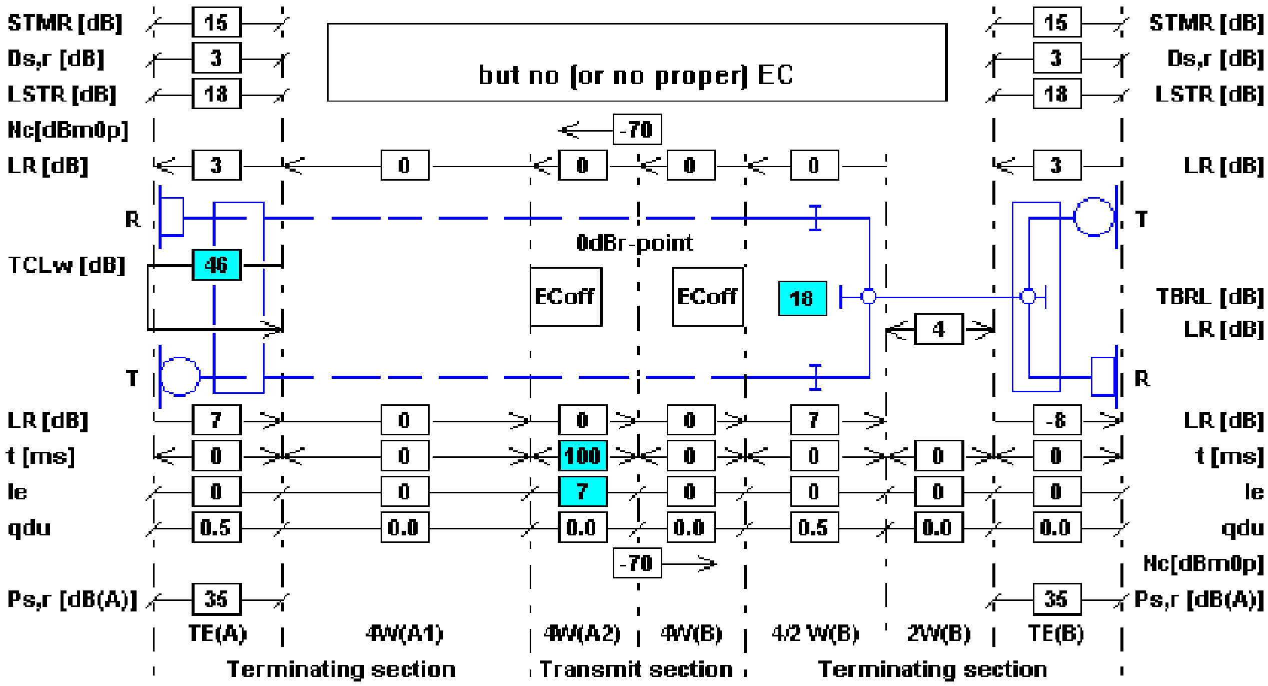


Figure 32: E-model Calculation: R as perceived on the left side is R_L = 40,6, R as perceived on the right side is R_R = 80,4

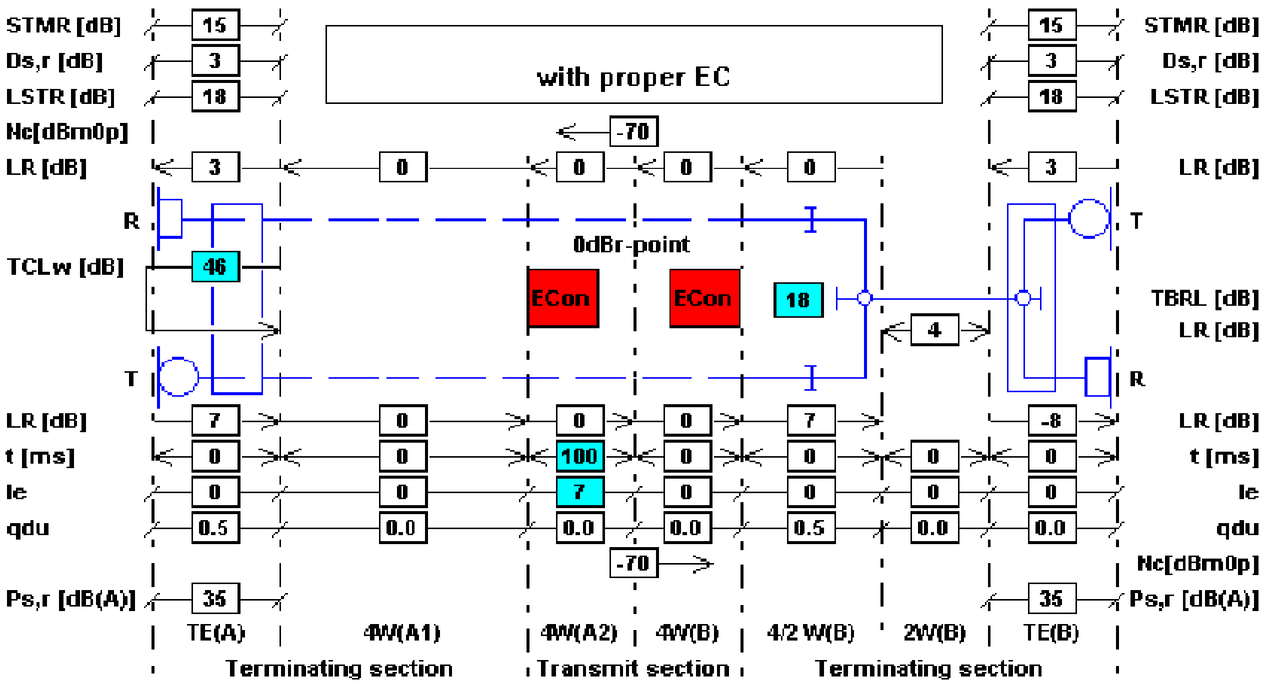


Figure 33: E-model Calculation: R as perceived on the left side is R_L = 87,3, R as perceived on the right side is R_R = 87,3

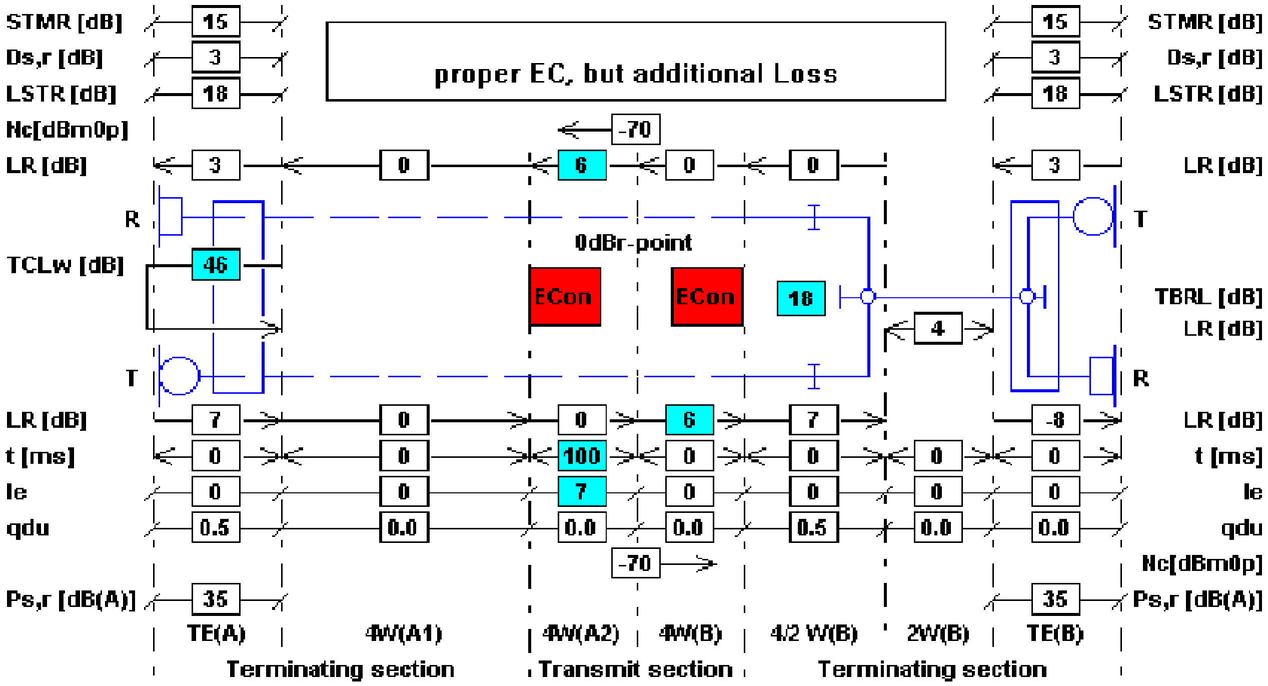


Figure 34: E-model Calculation: R as perceived on the left side is R_L = 79,5, R as perceived on the right side is R_R = 79,5

6.3.2 TIPHON QoS class "narrowband medium (2M)"

Figure 35 shows a connection with all values default according to ITU-T Recommendation G.107 [17], 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 #2M. 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 #2M with respect to R.

Figure 36 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 #2M with respect to R are met for both sides.

Figure 37 shows - based on the connection of figure 36 - 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 37 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 #2M with respect to R.

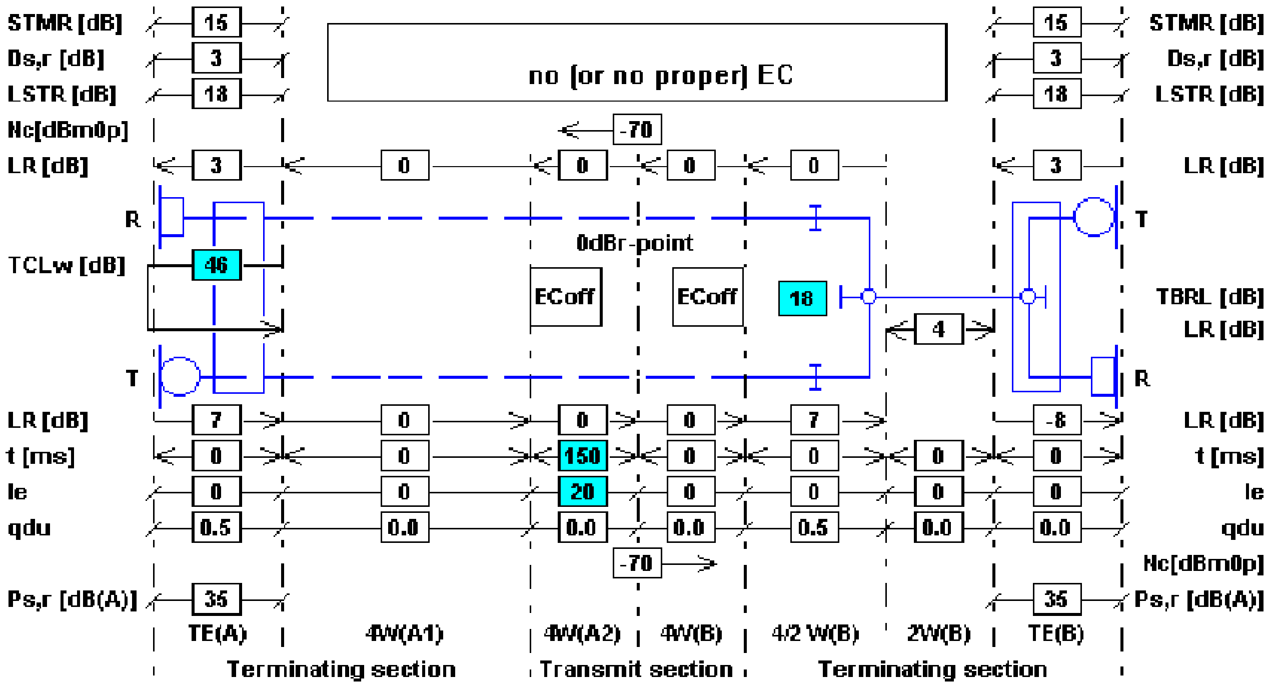


Figure 35: E-model calculation: R as perceived on the left side is R_L = 19,1, R as perceived on the right side is R_R = 63,6

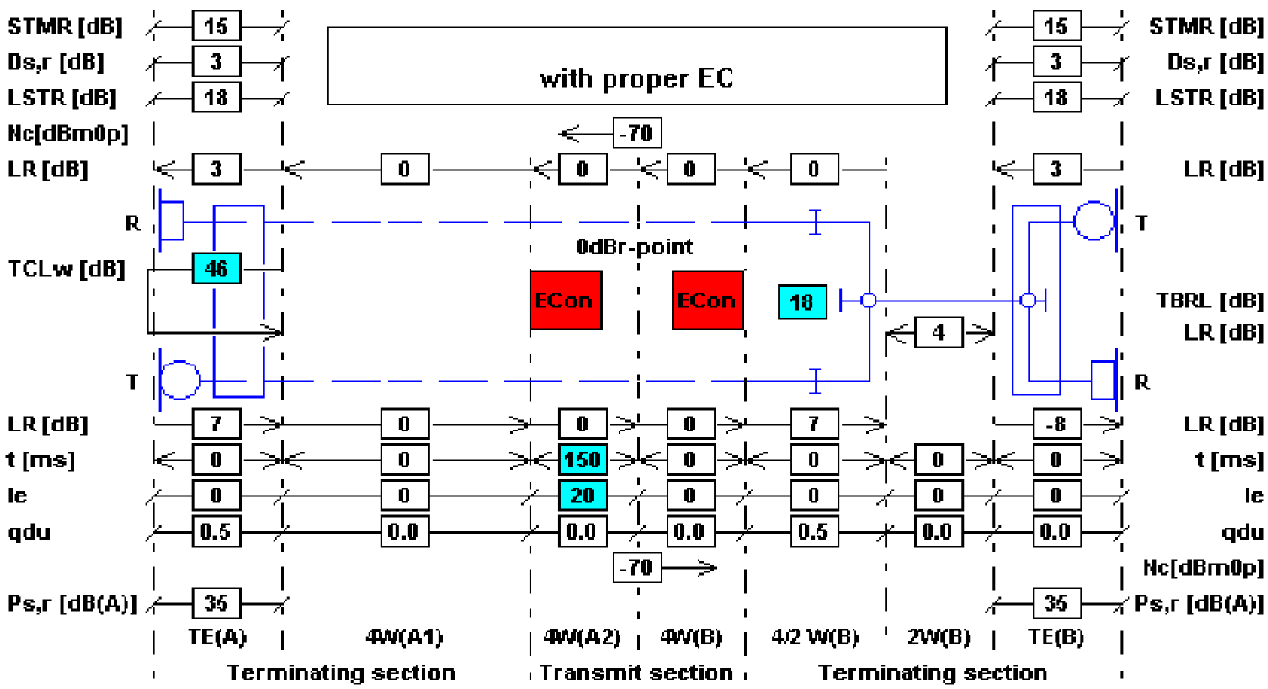


Figure 36: E-model calculation: R as perceived on the left side is R_L = 74,2, R as perceived on the right side is R_R = 74,2

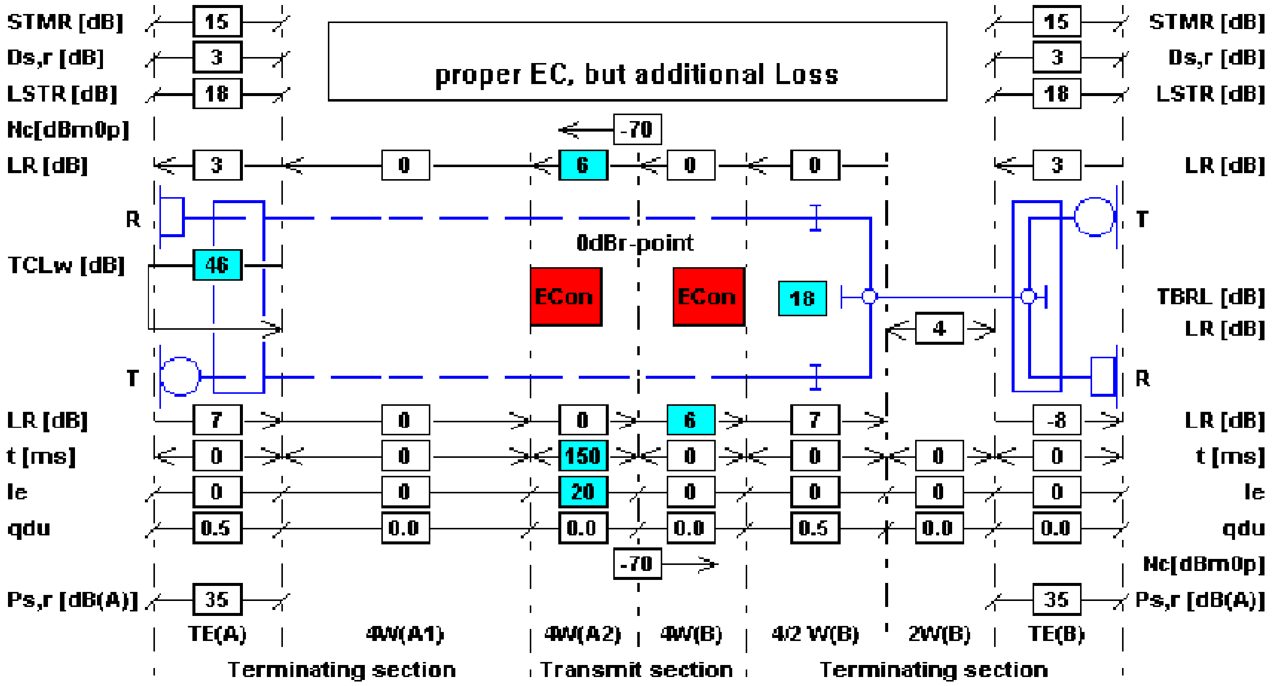


Figure 37: E-model calculation: R as perceived on the left side is R_L = 66,3, R as perceived on the right side is R_R = 66,3

6.3.3 TIPHON QoS class "narrowband acceptable (2A)"

Figure 38 shows a connection with all values default according to ITU-T Recommendation G.107 [17], 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 #2A. 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 #2A with respect to R.

Figure 39 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 #2A with respect to R are met for both sides.

Figure 40 shows - based on the connection of figure 39 - 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 40 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 #2A with respect to R.

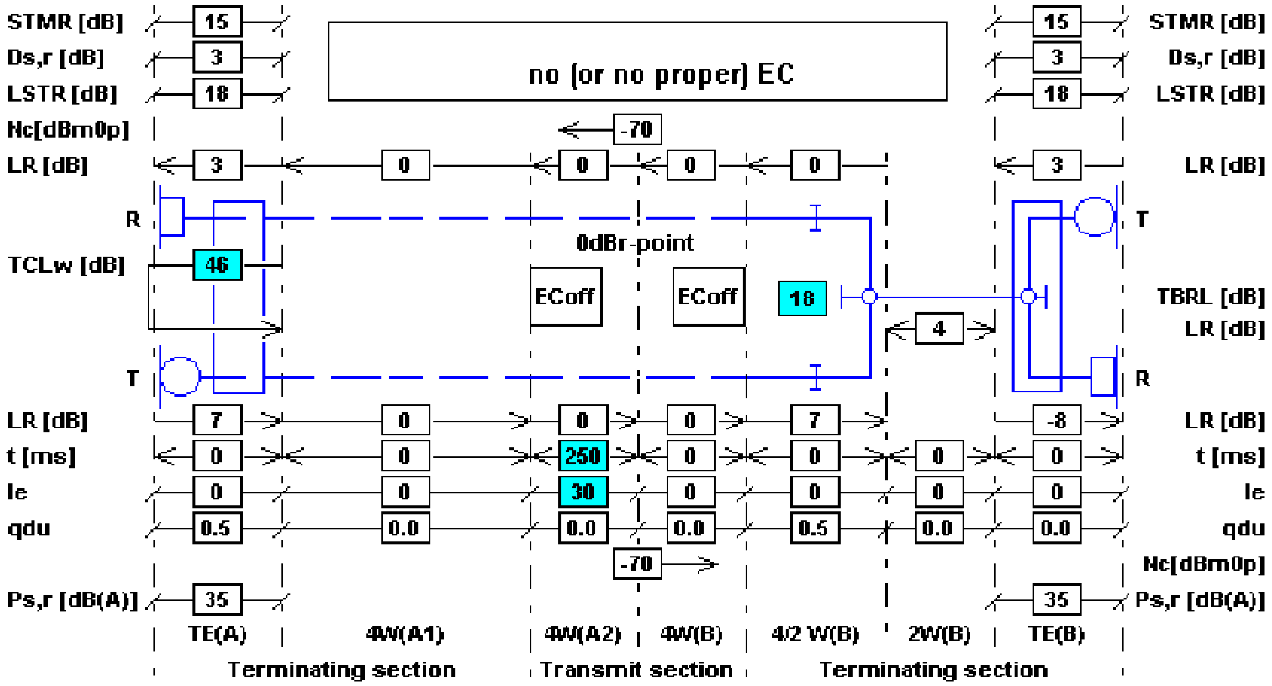


Figure 38: E-model Calculation: R as perceived on the left side is R_L = 0,0, R as perceived on the right side is R_R = 39,1

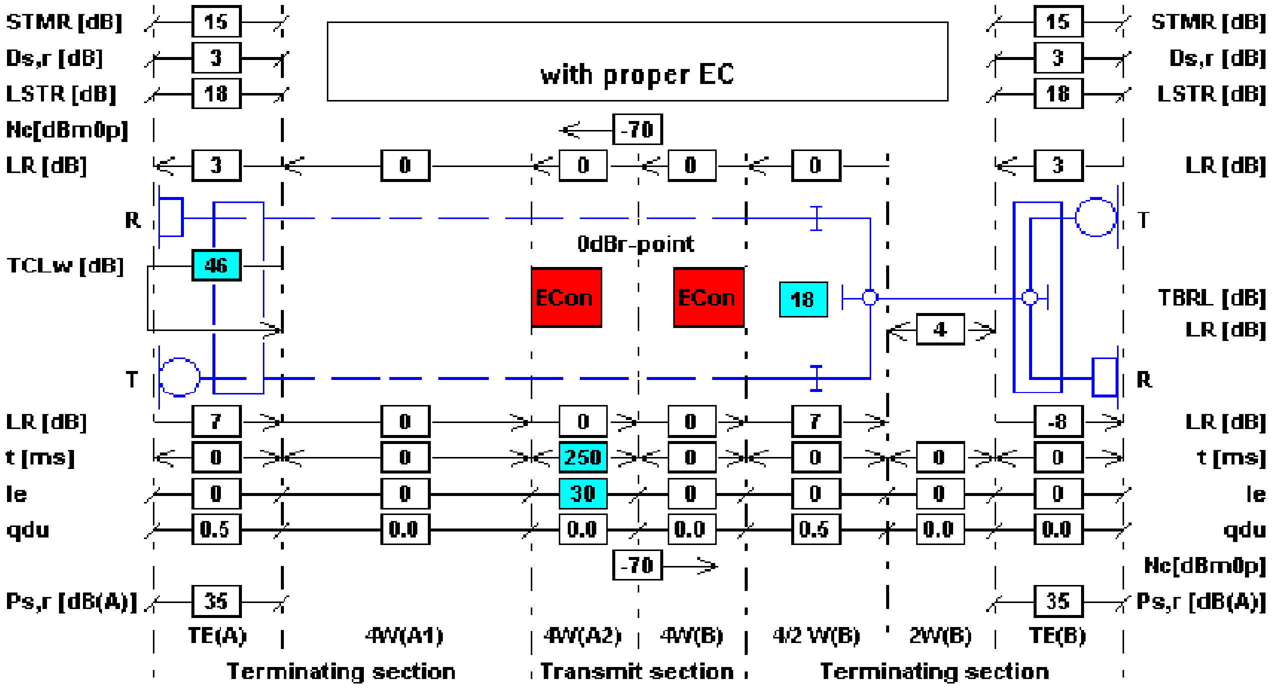


Figure 39: E-model calculation: R as perceived on the left side is R_L = 55,4, R as perceived on the right side is R_R = 55,4

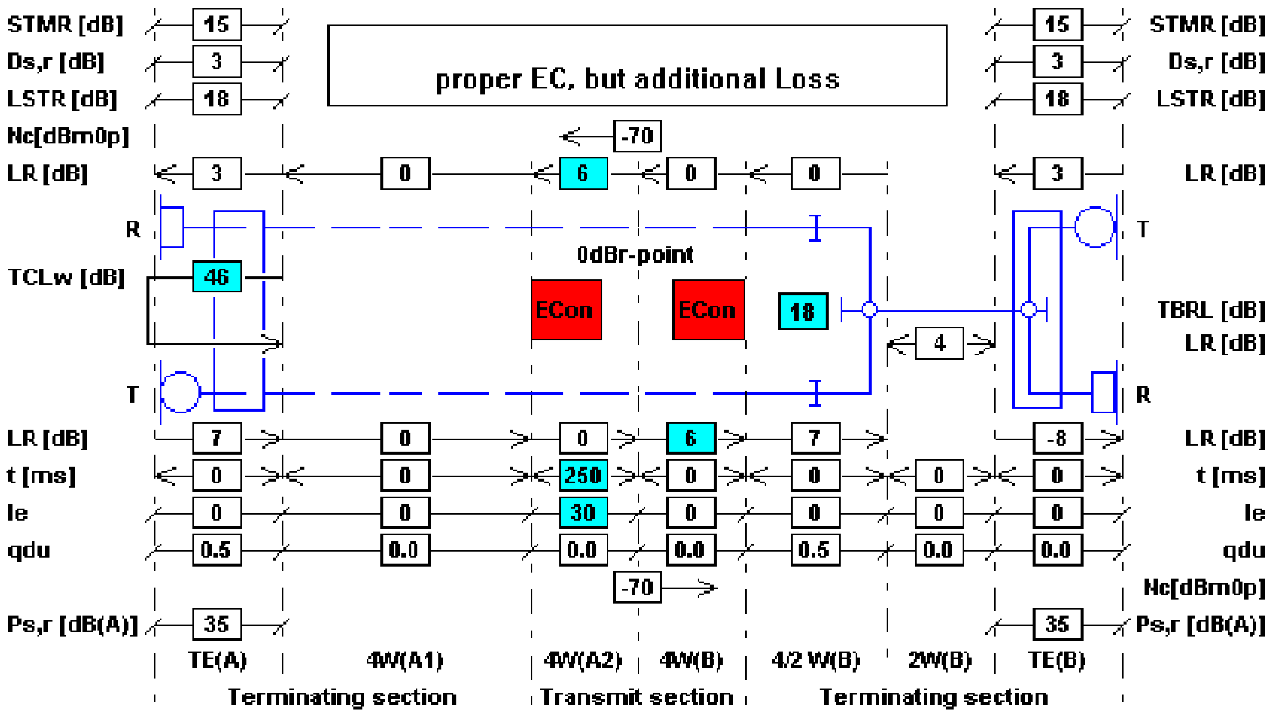


Figure 40: E-model Calculation: R as perceived on the left side is R_L = 47,6, R as perceived on the right side is R_R = 47,6

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 Examples with respect to packet loss

This is for further study.

6.7 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 41:

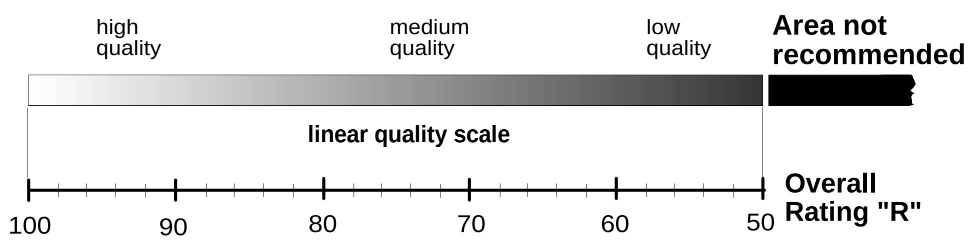


Figure 41: Judgement of a connection on a linear quality scale

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 clause 6.8.

Table 19: Relation between Rating Factor "R" and users satisfaction (Table 1 of ITU-T Recommendation G.109 [9]: 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 [17].		

6.8 Guidance on the relation and Interdependency between Auditory MOS, Objective MOS and Predicted MOS

For a better understanding of the contents of table 18, figure 42 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.

ITU-T Recommendation P.833 [22] provides the methodology for the derivation of Equipment Impairment Factors Ie from auditory test results. If new equipment impairment factor values for different codecs (or existing codec under new conditions, e.g. packet loss) have to be derived, then the overall consistency with the established framework is of primary importance.

Draft ITU-T Recommendation P.DIEIM describes a methodology for the derivation of Equipment Impairment Factors Ie from instrumental (or objective) measurement results.

It has not yet been proven that the quantitative degradation in the listening-only and in the conversational modes are similar, but for reasons of simplicity a strong interrelation is assumed. Users of transmission rating models, however, should be aware that differences may exist, and that most of the data derived reflects the listening-only situation only.

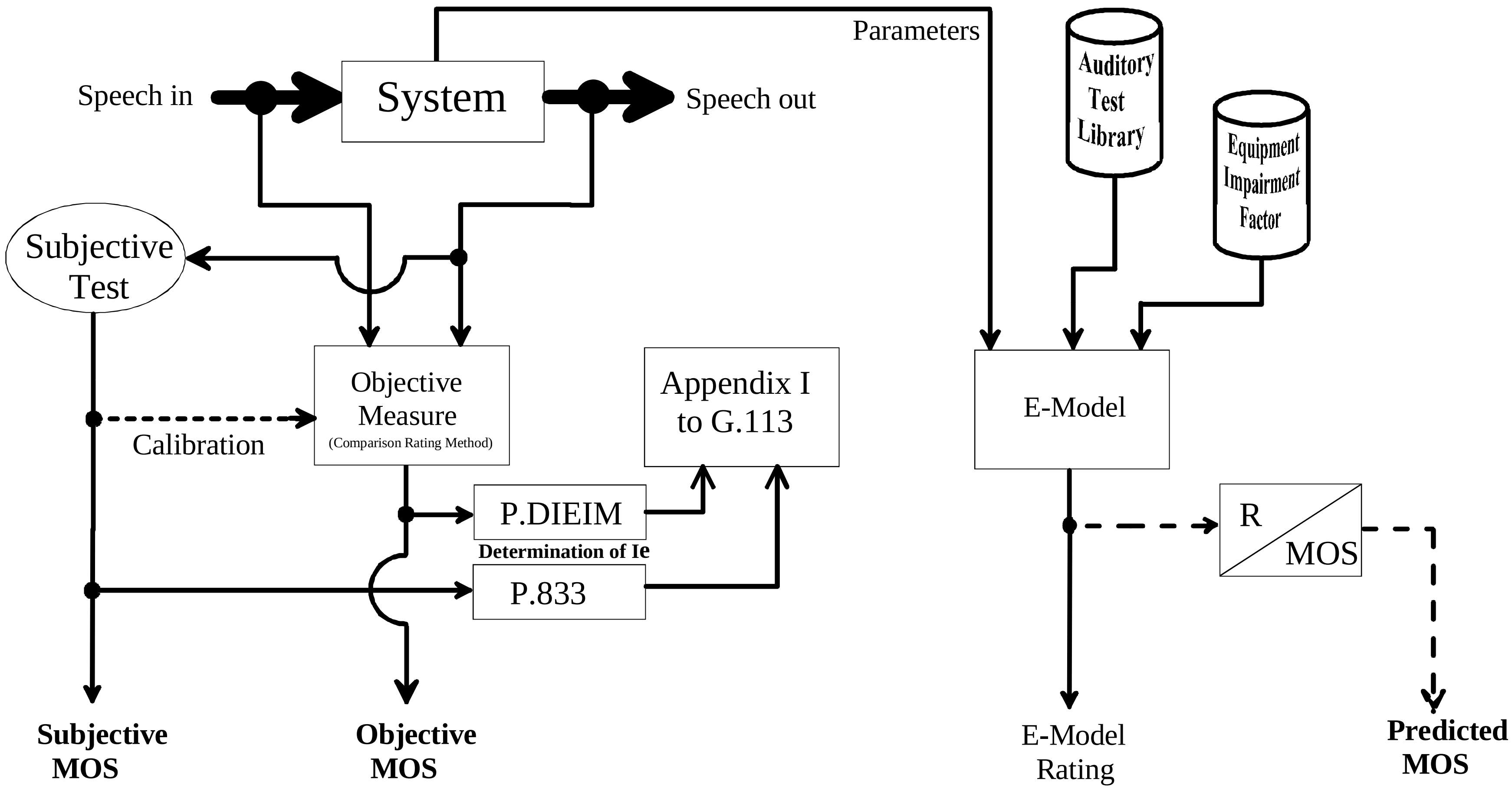


Figure 42: Relation and interdependency between auditory MOS, objective MOS, and predicted MOS

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

ETSI EN 300 962: "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 1997)".

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

ETSI EN 300 964: "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)".

ETSI EN 300 965: "Digital cellular telecommunications system (Phase 2+); Full rate speech; Voice Activity Detector (VAD) for full rate speech traffic channels (GSM 06.32 version 6.0.1 Release 1997)".

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

ETSI EN 300 969: "Digital cellular telecommunications system (Phase 2+); Half rate speech; Half rate speech transcoding (GSM 06.20 version 6.0.1 Release 1997)".

ETSI EN 300 970: "Digital cellular telecommunications system (Phase 2+); Half rate speech; Substitution and muting of lost frames for half rate speech traffic channels (GSM 06.21 version 6.0.1 Release 1997)".

ETSI EN 300 971: "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)".

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ETSI EN 300 973: "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)".

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History

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