COMMUNICATIONS
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INDUSTRY GUIDELINE

G634:2013

Quality of Service parameters for Voice over Internet Protocol (VoIP) services

G634:2013 Quality of Service parameters for Voice over Internet Protocol (VoIP) services Industry Guideline

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Communications Alliance Ltd was formed in 2006 to provide a unified voice for the Australian communications industry and to lead it into the next generation of converging networks, technologies and services.

EXPLANATORY STATEMENT

This is the Explanatory Statement for the G634:2013 ***Quality of Service parameters for Voice over Internet Protocol (VoIP) services*** Industry Guideline. This Explanatory Statement outlines the purpose of this Industry Guideline (the Guideline) and the factors that have been taken into account in its development.

Background

The Internet Protocol (IP) is used for a range of services, some of which are sensitive to delays in packet delivery and to packet loss e.g. voice, video. The performance of these services benefit from having a defined Quality of Service (QoS).

Objectives of the Guideline

This Guideline provides an indicator of quality for Voice over Internet Protocol (VoIP) services and information on factors that determine conversational voice quality on VoIP Services.

How the Objectives will be Achieved

The objectives will be achieved by the adoption of the QoS parameters suggested in this Guideline in a consistent manner by providers of VoIP Services.

Anticipated Benefits to Consumers

Consumers are likely to benefit from a consistent approach by service providers to the delivery of QoS for VoIP Services. Benefits include the ability to make an informed choice of VoIP Services as well as improved confidence that the VoIP Services will operate as expected and will operate between different networks.

Anticipated Benefits to Industry

A consistent approach to the definition of QoS for VoIP Service by service providers will reduce the complexity and cost of informing end-users. It will also increase the number of users that can be connected reliably.

Anticipated Cost to Industry

Anticipated costs include those associated with the use of an approach consistent with the information in this Guideline.

Acknowledgements

Nortel contributed content in multiple places in the document, including:

Section 4.2: Table 3 “Bandwidth per Voice Calls with Standard IP Header”; Section4.8: Echo Control, Figure 2 “Echo Level and one-way Delay”, Appendix C, Section C5 “Graphical representation of relationship between R and delay” - diagrams and Appendix D on Quality of Experience from *Essentials of Real-Time Networking: How Real-Time Disrupts the Best-Effort Paradigm*.

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Chairman
WC48 : VoIP QoS Revision Working Committee

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1 GENERAL

1.1 Introduction

1.1.1 The development of the Guideline has been facilitated by Communication Alliance through a Working Committee comprised of representatives from the telecommunications industry and Government regulatory agencies.

1.1.2 The Guideline should be read in the context of other relevant Codes, Guidelines and documents, including the G632:2012 ***Quality of Service parameters for networks using the Internet Protocol*** Guideline.

1.1.3 Statements in boxed text are a guide to interpretation.

1.2 Future Work

1.2.1 The Working Committee that developed this Guideline considered the application of "Static" Quality of Service (QoS) targets for networks (i.e. not requiring QoS negotiation between the Voice over Internet Protocol (VoIP) service and underlying transport on a call-by-call basis).

> *NOTE:*
>
> *1. 'Call-by-call' also refers to the different call scenarios for voice services on IP networks.*
>
> *2: The delivery of QoS for VoIP Services may depend on networks that meet different QoS targets however QoS for networks and services address different requirements. Refer to G632 for information on QoS targets for IP networks.*

1.2.2 Work is proceeding in international forums on "Dynamic" QoS Negotiation, which requires a higher level of coordination between providers of VoIP Services, on a service-by-service basis. This topic has been left for future work to allow time for international recommendations and standards to stabilize.

1.2.3 The assumption of growing IP bandwidth in access and core networks means that these dynamic methods will probably not be required for some services (e.g. voice), but may become more important for bandwidth-intensive applications (e.g. video-on-demand).

1.2.4 Extension of the Guideline to cover VoIP Services over wireless IP is considered future work. Techniques and standards for deploying VoIP Services over the radio access are currently immature – it is expected, however, that ITU Recommendations and 3GPP Specifications, along with associated technologies and voice quality tools will emerge over the next few years.

1.2.5 While the deployment and use of wideband codecs (e.g. AMR-WB, G.722.2) is increasing, the definition of parameters to guide wideband implementation (e.g. based on G107.1) is for future study.

1.3 Scope

1.3.1 The Guideline recommends Quality of Service (QoS) categories and identifies influencing impairments for Voice over Internet Protocol (VoIP) Services within Australia.

> *NOTES:*
>
> *1. QoS in this context refers to conversational voice quality on VoIP networks or, as described in Appendix D, Quality of Experience (QoE).*
>
> *2. The use of multiple network types (e.g. a VoIP call over a mix of packet and circuit switched networks) can degrade overall performance relative to the use of a single network type.*
>
> *3. Some networks can have high variability in performance and may not be suited to VoIP (e.g. the performance of some wireless networks varies with factors such as coverage, proximity to a base station/access point).*

1.3.2 The Guideline is based on ITU-T G.107 and provides information on QoS parameters for conversational voice quality for the end-user experience of VoIP service(s) over Managed Network(s).

> *NOTES:*
>
> *1. Refer to G632 for information on QoS performance in networks using the Internet Protocol (IP).*
>
> *2. The Guideline could be used for unmanaged (i.e. best effort) VoIP Services even where they do not meet the performance measures e.g. the information on codec selection, access links.*

1.3.3 The Guideline does not specify QoS parameters for services other than VoIP (e.g. video over IP, text over IP).

1.3.4 The Guideline does not specify QoS parameters for non-voice services carried over VoIP (e.g. Fax, dial-up modem, teletypewriter, burglar alarms, EFTPOS terminals).

1.3.5 The Guideline does not address processes for the measurement of VoIP QoS. Refer to G635 for information on the measurement of VoIP QoS.

1.4 Objective

The objective of this Guideline is to specify the categories of speech transmission quality in terms of limits of Transmission Rating Factor R and provide an overall indicator of the quality of Voice over Internet Protocol (VoIP) services. Providers of VoIP Services can use this Guideline for transmission planning purposes and to inform end-users. In addition, it provides information on the impairments that determine conversational voice quality for VoIP Services based on ITU-T Recommendations and Australian requirements.

1.5 Guideline Review

Review of the Guideline will be conducted within five years of publication.

2 ACRONYMS, DEFINITIONS AND INTERPRETATIONS

2.1 Acronyms

For the purposes of the Guideline, the following acronyms apply:

3GPP	3rd Generation Partnership Project
ACIF	Australian Communications Industry Forum
ADPCM	Adaptive Differential Pulse Code Modulation
AMR	Adaptive Multi-Rate
AMR-NB	Adaptive Multi-Rate Narrowband
AMR-WB	Adaptive Multi-Rate Wideband
CELP	Code Excited Linear Prediction
Codec	COder / DECoder
CSP	Carriage Service Provider
CE	Customer Equipment
DTMF	Dual Tone Multi Frequency
ECAN	Echo Canceller
ERLE	Echo Return Loss Enhancement
IP	Internet Protocol
IPDV	IP Packet Delay Variation
IPTD	IP Packet Transfer Delay
ITU-T	International Telecommunications Union – Telecommunications standardization sector
MOS	Mean Opinion Score
OLR	Overall Loudness Rating
POTS	Plain Old Telephone Service
QDUs	Quantising Distortion Units
QoS	Quality of Service
RFC	Request For Comment
RLR	Receive Loudness Rating
RTCP	Real Time Control Protocol
SLR	Send Loudness Rating
TCLw	Weighted Terminal Coupling Loss
TELR	Talker Echo Loudness Rating
UNI	User-to-Network Interface
VoIP	Voice over Internet Protocol
WEPL	Weighted Echo Path Loss

2.2 Definitions

For the purposes of the Guideline, the following definitions apply:

Carriage Service Provider

has the meaning given by section 87 of the Telecommunications Act 1997.

Carrier

has the meaning given by section 7 of the Telecommunications Act 1997.

Customer Equipment

has the meaning given by section 21 of the Telecommunications Act 1997.

E-model

means the computational model with the output of a scalar quality rating value, R, as defined in ITU-T Recommendation G.107.

> *NOTE: G.107 defines the E-model (used to generate R values) for narrowband codecs and G.107.1 extends the E model to cover wideband codecs.*

Internet Protocol

means the protocol defined in the Internet Engineering Task Force (IETF) Request For Comment (RFC) 791.

IP Packet Delay Variation (IPDV)

means the difference between the actual IP Packet Transfer Delay (IPTD) of a packet and a reference IPTD for a packet population of interest. The reference IPTD of a population of packets is the minimum IPTD for the packets within the population of interest.

IPDV is a statistical sample, measured over a packet population of interest. Unless otherwise stated, the default quantile is the 10^{-3} quantile i.e. 99.9% of packets should be received within the performance objective.

> *NOTE: IPDV is also referred to as "jitter", and is usually reported in milliseconds.*

IP Packet Loss Ratio (IPLR)

means the ratio of total lost IP packets to total transmitted packets in a population of interest. Total lost packets includes any delivered with errors or IPTD greater than 3 seconds.

> *NOTES:*
>
> *1. IPLR Ratio is also referred to as "Packet Loss" and is usually reported as a percentage.*
>
> *2. The upper limit value of 3 seconds for IPTD is based on the provisional value for the time limit for a successful packet outcome (refer to ITU T Rec. Y.1540 clause 5.5.4).*

IP Packet Transfer Delay (IPTD)

means the one-way time interval between the moment the first bit of an IP packet crosses an entry point of a network and the moment the last bit of the same packet crosses an exit point of the network.

> *NOTE: IP Packet Transfer Delay is also referred to as "delay" or "latency", and is usually reported in milliseconds.*

Loudness Rating

means a measure of the volume of speech based on ITU-T Recommendation G.121.

Managed Network

means an IP network with QoS-enablement e.g. a network that conforms with the parameters outlined in Guideline G632.

Network Boundary

has the meaning given by section 22 of the Telecommunications Act 1997.

Sidetone Path

means any path, acoustic, mechanical or electrical, by which a telephone user's speech and/or room noise is heard in their own ear(s).

> *NOTE: This is based on ITU-T P.10.*

Trombone Connection

means the use for a single call of two circuits in tandem between a remote switching stage and its controlling entity.

VoIP Service

means a voice communication service where the origination and/or the termination of the voice service is carried over an IP network.

> *NOTE: A VoIP Service is independent of the underlying transport method(s), e.g. DSL, Ethernet, HFC.*

3 VOIP SERVICE QUALITY INDICATORS

3.1 Measure of QoS for VoIP Services (Transmission Rating R)

3.1.1 Transmission Rating R is adopted as a predictive measure of voice quality, based on the computational model defined by ITU-T G.107 (the E-Model). The value of R may be derived by application of the planning guide defined by ITU-T G.108. R is expressed as a scalar (a single number) on a scale from 0 to 93.2 for narrowband voice services.

> *NOTES:*
>
> *1. The E model is only applicable where its parameter values can be determined on an end-to-end network basis, or as a complete "mouth-to-ear" experience. Assignment of those values into constituent network segments and operational boundaries is an area of further work*
>
> *2. As per the Telecommunications Act, Carriers and Carriage Service Providers can only manage and measure service quality to the defined Network Boundaries.*
>
> *3. Bundled offerings may cross the Network Boundary.*
>
> *4. Other measures following ITU-T recommendations such as ITU-T Rec. P.800 and ITU-T Rec. P.862 are not used as part of this Guideline.*
> *ITU-T Rec. P.800 uses subjective testing for the determination of a Mean Opinion Score (MOS). This approach of using the human ear is expensive, time consuming and inconvenient.*
> *ITU-T Rec. P.862 on Perceptual Evaluation of Speech Quality (PESQ) does not address determining factors for the evaluation of conversational voice quality such as delay, signal levels, echo impairment; this does not allow for the tuning of a network and cannot assist in identifying the source of a problem.*
>
> *5. The term wideband only refers to the choice of codec, as a voice service may still be using narrowband channel. For IP transport different bandwidth is required for different codecs.*
>
> *6. The R value is extendable, unlike MOS which needs different scoring for wideband codecs.*
>
> *7. G.107.1 extends the R value to a range of 0 to 129 for wideband codecs.*

3.1.2 R and its computation are defined in ITU Rec. G.107 and G.107.1.

> *NOTES:*
>
> *1. It is important to note that ITU-T G.108 is to be used as a network planning guide only. It does not imply specific performance that will be achieved by a particular connection or user device. As such, ITU-T G.108 refers to R as an indicator of QoS for planning purposes.*
>
> *2. The ITU-T uses QoS to refer to conversational voice quality on VoIP networks, or as described in Appendix D, Quality of*

Experience (QoE). Refer to ITU-T E.800 for the complete ITU-T definition of QoS.

3. It is advisable that the reader is fully aware of all the factors that determine R as described in ITU-T G.107/G.107.1 (the E-Model). See Appendix C of this document for a summary.

3.1.3 The parameters that contribute to the predictive measure of VoIP QoS include:

(a) Loudness Ratings and loss plan;

(b) Sidetone Path;

(c) D-value (related to handset design);

(d) echo loudness;

(e) codec distortion;

(f) immunity of the codec to packet loss;

(g) noise levels in room and circuit/codec; and

(h) advantage (gained from mobility or remote access).

NOTES:

1. The E-model does not model all network based impairments. Examples of the most severe impairments include mobile background noise and double talk echo. Further details on factors not covered by the E-model may be found in ITU-T G.108.1.

2. The E-model has particular limits when applied to services operated over connections with bandwidth limitations e.g. some wireless networks. For example, a wireless access network can have variable signal strength, which affects parameters such as delay and packet loss, which in turn affects voice call quality.

3. The E-model does not incorporate measures for enhancement techniques offered by mobile customer equipment, such as distortion masking or noise suppression. Other CE factors not easily accommodated include hands free kits and acoustical design.

4. The default value of 35dBA for room noise does not reflect the use of mobile devices in noisy environments e.g. it would increase to 55dBA for a quiet car, more for the use of a power tool in the next room.

5. Refer to AS/CAF S003 for the Australian loss plan. Customer Equipment Standards such as AS/CA S004, AS/CA S002 and AS/CA S003 should be used for the final loss plan analysis.

3.1.4 The key contributing parameters affecting conversational voice quality in a VoIP network are:

(a) delay;

(b) distortion;

(c) echo; and

(d) loss/level plan.

NOTES:

1. Also refer to Appendix D, Quality of Experience (QoE), ITU-T G.107/G.107.1, ITU-T G.108 and TIA TSB-116-A.

2. Delay Includes impairments due to propagation, processing and packetisation, queuing/jitter, and switching. Delay contributes to echo impairment, but is also an impairment on its own when the total delay becomes sufficiently high. End-to-end delay is the total of all delays in the voice path. The five main categories of delay are: processing delay, serialization delay (time taken to push the packet onto the wire), queuing delay (accumulates at network nodes), propagation delay and jitter buffer.

3. Distortion Includes impairments due to compression coding, end devices, lost/late voice packets, speech interruption, noise, quantizing distortion, and transcoding.

4. Echo Includes impairments due to hybrid inductive coupling (transhybrid loss) and acoustic coupling in the terminal handset/headset. Talker echo loudness rating (TELR) is the parameter defining the level of echo signal reflected back to the talker.

5. Loss/level plan Includes impairments due to non-optimal signal loudness — SLR (send loudness rating), RLR (receive loudness rating), CLR (circuit loudness rating) and TELR.

3.1.5 Proper control of the above four parameters ensures satisfactory end-user voice quality. It is relevant to note that the factors affecting voice quality on a VoIP network need to be considered as a whole; an isolated view of affecting parameters is incomplete.

3.1.6 It is also important to consider the resulting R for every call scenario, as this parameter provides a quantifiable figure for predicted end-user conversational voice quality.

NOTES:

1. The factors in 3.1.3 and 3.1.4 are combined to generate R. For the algorithm to combine the factors, refer to ITU-T Rec.s G.107/G.107.1 and G.108.

2. Quantising Distortion Units (QDUs) also contribute to R. However the audio signal for VoIP is typically digitally encoded and decoded in customer equipment so QDUs have less significance than when using an analogue (access) network.

3.2 Recommended Performance Values

3.2.1 Reference voice service categories A to H have been specified to provide a set of recommended VoIP Service performance values as set out in Table 1 below.

3.2.2 Refer to Appendix C for a definition of the categories of speech transmission quality based on ITU-T Rec. G.109. It highlights variations of R value based on parameters such as echo (refer to ITU-T G.131, ITU-T G.108, ITU-T G.108.1 and ITU-T G.108.2), equipment impairment I_e of codecs (refer to ITU-T G.113), and dependencies on packet loss and delay (refer to ITU-T G.114).

3.2.3 R values have strong dependency on the E-Model input parameters and impairments relative to the call scenario under consideration. It follows that specified performance values require reference to specific assumptions on parameters affecting the combined Transmission Rating Factor R.

3.2.4 To achieve the recommended performance values for Category D or better in Table 1 then one should have:

(a) a codec with impairment no worse than ITU-T G.711 (i.e. G.711 or a wideband codec);

(b) a managed access and core IP networks with IP Packet Loss Ratio less than 0.1%;

(c) IP Packet Delay Variation (IPDV) less than 50ms;

(d) Overall Loudness Rating (OLR) of 10dB;

(e) echo cancellation enabled; and

(f) one-way UNI to UNI mean delay of less than 100ms (i.e. achieving a mouth to ear delay of less than 150ms).

NOTES:

1. Refer to G632 for details on QoS parameters for networks using IP.

2. R-values less than 50 including the A parameter are not recommended when planning VoIP networks.

3."Best-efforts" services offer no performance target or performance guarantee.

4. Use of a wideband codec allows some relaxation on the packet network quality in determining the R value.

5. Appendix C.3.1 provides guidance to interpreting a calculated R-value with explicit user satisfaction expectations, as defined in ITU-T G.107 Annex B.

Category	R limit	% of Calls	Comment	Examples
'Best Efforts'	N/A	N/A	Best Efforts voice service; no guarantee on voice quality	Unmanaged voice service
A	≥ 50	95%	Nearly all users dissatisfied	
B	≥ 60	95%	Many users dissatisfied	
C	≥ 70	95%	Some users dissatisfied	G.729a codec on a wired network, achieving voice quality similar to that experienced on a cellular mobile service
D	≥ 80	95%	Satisfied	G.711 codec on a wired network, achieving voice quality similar to that experienced on a POTS voice service
E	≥ 90	95%	Very Satisfied	G.711 codec in an ideal network environment
F	≥ 100	95%		G.722.2 (wideband) codecs, QoS enabled network(s)
G	≥ 110	95%		
H	≥ 120	95%		

TABLE 1

Transmission Rating (R) limits for voice services

NOTES:

1. Current typical narrowband VoIP calls are unlikely to exceed an R value of 93. This would use G.711 codecs at each endpoint, broadband access links (i.e. greater than 800 kbps, and/or use of multiple virtual circuits and/or QoS enablement), well managed core networks (e.g. dimensioned for Class 0 of G632 performance and/or with appropriate QoS treatment), and no transcoding.

2. A voice call using wideband codecs, along with appropriate transport conditions, can achieve an R value greater than 100. The use of wideband codecs is becoming widespread in IP networks.

3. R values in this table are indicative of those expected for domestic terrestrial networks. By definition, this excludes calls that include an International call leg. In these cases call quality cannot be accurately modeled or predicted, if details of the terminating network will not be known.

4. Refer to Table 2 of ITU-T G.109 for examples of speech transmission quality with estimates of R values for a number of typical service/network scenarios.

5. Some access technologies are unable to operate on a Class 0 network and therefore are unlikely to achieve a performance level higher than Category B e.g. calls using geostationary satellite connections (see C4.1d in Appendix C), some wireless access technologies or low bandwidth access links.

6. This table typically represents calls made via an IP network and terminated either directly via IP, or via TDM with a single IP to TDM conversion.

7. The comments on the level of satisfaction/dissatisfaction originate from Table 1 of ITU-T Rec. G.109.

8. Where an end-point with a lower category service communicates with another end-point with a higher category service, the end-to-end voice quality will be representative of the lower category service.

9. One should measure and report the objective call quality without the A value (refer to Appendix C for more information).

Codec Choice and end-to-end Delay

3.2.5 The end-to-end delay is the total of all delays incurred in the voice path. The four main categories of delay are:

(a) **Processing delay**: time taken for speech to accumulate so that it can be put into a packet, loaded and transmitted. Where speech compression is used, the time needed for coding is added as well. The speed of any processors (DSP, CPU) involved also contributes to the final delay.

(b) **Serialization delay:** determined by the channel speed (bits/sec) and the number of bits in the packet. On high speed links serialization delay becomes negligible compared to other sources of delay.

(c) **Queuing delay:** accumulates at network nodes (routers and switches) across the network. Congestion can increase packet waiting times in buffers. Note that variation in queuing and buffering delays in the network account for most of the variation in packet transport times (i.e. jitter). The jitter buffer wait time is another instance of queuing delay.

(d) **Propagation delay:** time taken for the signal to travel through a transport medium (e.g. cable or fibre or wireless). In the conventional public network, propagation delay is the largest contributor to end-to-end delay. Note that propagation delay across a fixed distance is not a controllable parameter, since it is determined by the speed of the signal through the transmission medium (usually light through a fibre). However, it is possible to ensure that packets take the most direct route through the network to minimise queuing and propagation delay.

3.2.6 With VoIP, it is possible to trade-off different quality parameters - including delay - and still get acceptable overall voice quality.

3.2.7 A simplified example of the trade-off between codec type and delay that can still result in good speech quality as perceived by the user is shown in Table 2, with all other parameters of the E-Model at default values. Good speech quality (where users are 'very satisfied' or 'satisfied' defined as a Category E or D quality service) is indicated by an R value of 80 or more. Three popular codec types have been chosen for the comparison. These codecs have different qualities as indicated by their respective Impairment Factors, but the final voice quality result achieves an R value of not less than 80.

Codec	**Maximum Delay**
G.711	250 ms
G.729a	130 ms
G.723.1 (at 6.3 kbps)	Not possible

TABLE 2

Codec type vs. allowable delay with default E model values if R is to be not less than 80

NOTES:

1. The above table is based on the E model tool.

2. The table assumes ideal end-to-end speech conditions including ideal handsets, echo cancellation and IP network performance. If these are not present then the allowable delay is reduced.

3. The performance of wideband codecs (e.g. G.722.2/AMR-WB) is for future study.

4. While the improved quality of a wideband codec can partially compensate for long delays, it cannot eliminate the loss of interactivity which occurs with very long delays (e.g. satellite hops).

3.2.8 Delays can occur due to:

(a) distance (optical fibre - 5ms per 1000 km; satellite - ~250ms per hop);

(b) codec processing delay at both ends;

(c) routers (0.5-5ms per router - for very short voice packets);

(d) low bandwidth transmission links including access links; and

(e) LANs.

4 IMPLEMENTATION GUIDELINES

4.1 VoIP End-user Access Connection

4.1.1 To support services based on Class 0 network(s) (refer to G632), the end-user access connection should meet the minimum performance outlined in G632 Appendix B, (i.e. IPDV < 16 ms). This implies, for support of no more than one voice call:

(a) a minimum access speed of 800 kbps in each direction; or

(b) a means of reducing the impact of other traffic on VoIP traffic on the end-user access connection. This permits access speeds as low as 256Kbps or even 128Kbps in each direction.

4.1.2 Higher access speeds are required for

(a) higher data rate codecs; or

(b) multiple simultaneous calls.

4.2 VoIP Inter-Carrier Connection

4.2.1 To support Category D VoIP Services, connection(s) between packet networks should meet the minimum performance requirements of a Class 0 network as defined in G632 Appendix B.

4.2.2 In particular, for interconnection between packetised voice networks the effects of transcoding (successive encoding by different codecs) and tandeming (successive encoding by the same codec) needs to be considered. The total impairment factor (equipment impairment I_e in the E-Model) as a consequence of transcoding and/or tandeming for a particular call scenario is additive (note that $I_e = 0$ for the G.711 codec).

4.2.3 Table 3 is a guideline for Ethernet and ATM bandwidth per voice channel for G.711 and G.729a codecs, noting that 5% additional bandwidth should be allowed for Real Time Control Protocol (RTCP) packets.

Bandwidth per Voice Calls with Standard IP Header							
Codec		G.711			G.729a		
Codec Bit Rate		64 kbps			8 kbps		
Voice Sample (ms)		10	20	30	10	20	30
IP Payload size (bytes)		80	160	240	10	20	30
IPv4 Packet size (40 byte header)		120	200	280	50	60	70
IPv6 Packet size (60 byte header)		140	220	300	70	80	90
Ethernet							
Ethernet bytes (per packet)	IPv4	150	230	310	80	90	100
	IPv6	170	250	330	100	110	120
Ethernet bandwidth per voice flow (kbps)	IPv4	120	92	82.7	64	37	26.7
	IPv6	136	100	88	80	44	32
ATM Transport (ADSL/ADSL2+) (Includes 6 bytes for PPP)							
ATM bytes (PPPoAAL5oATM)	IPv4	159	265	371	106	106	106
	IPv6	212	265	371	106	106	159
ATM bandwidth per voice flow (kbps)	IPv4	127.2	106	98.93	84.8	42.4	28.27
	IPv6	169.6	106	98.93	84.8	42.4	42.4

TABLE 3

Bandwidth per voice calls with standard IP header

NOTES:

1. Layer 2 networks using VLAN tagging add 4 bytes per frame (single tagged) or 8 bytes per frame (dual-tagged).

2. Bandwidth per voice calls when using wideband codecs (e.g. AMR-WB or G.722.2) is for future study.

4.3 VoIP Packet Handling

4.3.1 CE should recognise VoIP packets with the recommended markings (as outlined in G632), and should handle them according to the priority defined by the implemented QoS scheme.

4.3.2 Network equipment of the voice and access service provider(s) should also recognise VoIP packets and give them priority consistent with the QoS provided by the contract with the end-user.

4.3.3 Where the network is aware that the QoS marking on a packet received from either an end-user or from an interconnecting network is inconsistent with the QoS contracted for by the owner of the destination address, the packet in each direction will be treated according to G632.

4.4 VoIP Packet Routing

4.4.1 Category C (or better) voice services should be carried on network paths meeting Class 0 as defined in G632. This will result in a UNI to UNI packet delay of less than 100ms, and should result in an end-to-end voice delay of less than 150ms.

4.4.2 Ideally, network equipment should route VoIP traffic by the path providing the shortest end-to-end delay. More broadly, selecting a path that meets the delay objective of IP traffic class is required.

4.4.3 For calls that involve interconnection to the TDM network, consideration should be given to ensuring a minimum end-to-end delay. To help achieve this providers of VoIP Services should minimise the use of Trombone Connections to distant points.

4.5 VoIP Packet Type and Priority

Protocols used for VoIP Services and therefore recommended to be given priority include the following:

(a) RTP Media;

(b) RTCP packets; and

(c) voice signaling.

4.6 VoIP Codec Choice and Codec Negotiation

4.6.1 The codec choice should be made by providers of VoIP Services in conjunction with overall network considerations that affect conversational voice quality in VoIP networks (refer to Appendix D, Quality of Experience (QoE) for a discussion on this topic).

4.6.2 Different codecs result in different bit rates (which affect the bandwidth required per call) and introduce different amounts of distortion (which affects voice quality) through their intrinsic compression process and their individual robustness to packet loss.

4.6.3 Where possible, RFC 3264 should be used for codec negotiation between end points.

4.6.4 To ensure interoperability it is recommended that G.711 (A-law) be included as an available codec should other preferred codecs not be available.

4.6.5 Packet loss concealment is recommended to be used in conjunction with waveform codecs (e.g. with G.711).

4.7 VoIP Echo Control

4.7.1 Equipment for a VoIP Service should support the impedance requirements in AS/CA S002 in order to achieve the objectives for echo cancellation and sidetone.

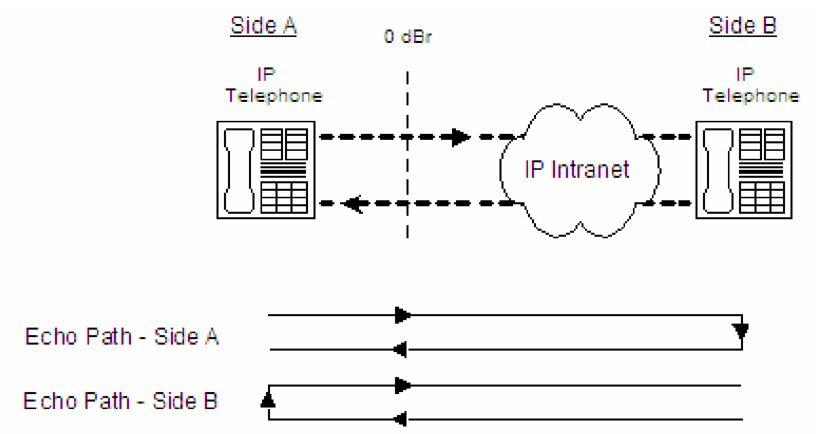
4.7.2 In general, VoIP networks have longer delays compared with traditional PSTN networks. With increasing delay any level of echo becomes increasingly audible.

4.7.3 For calls between an IP phone and a traditional PSTN phone, the echo control applied in the traditional network may not be sufficient. A crucial step in the engineering of the interface between a TDM and a VoIP Service is echo control, which must take account of the echo sources in the TDM side and the additional delay introduced by the VoIP side.

> *NOTES:*
>
> *1. The addition of a 2G mobile phone in place of a PSTN phone as part of the transmission path increases the delay (approximately 90 ms more in each direction).*
>
> *2. Refer to Section 7.2 of G.108 (09/99) for further detail on echo control.*

4.7.4 TELR is the sum of losses around the loop as shown in Figure 1 below. TELR represents the level of a talker's speech that comes back from an echo point in the network, often from the 2-wire to 4-wire hybrid in the far end line card.

4.7.5 The loss plan for an "all digital" connection is determined by the loudness ratings of the telephones — there are no additional losses in the network.



TELR (side B) =SLR (side B) + Loss in bottom path + ERL or TCLw (side A) + Loss in top path + RLR (side B)

FIGURE 1
Talker Echo Loudness Rating (TELR)

> *NOTES:*
>
> *1. Appendix C shows the effect of TELR variation.*
>
> *2. TCLw is the weighted terminal coupling loss.*

4.7.6 Figure 2 shows how echo impairment depends on the level of echo and delay.

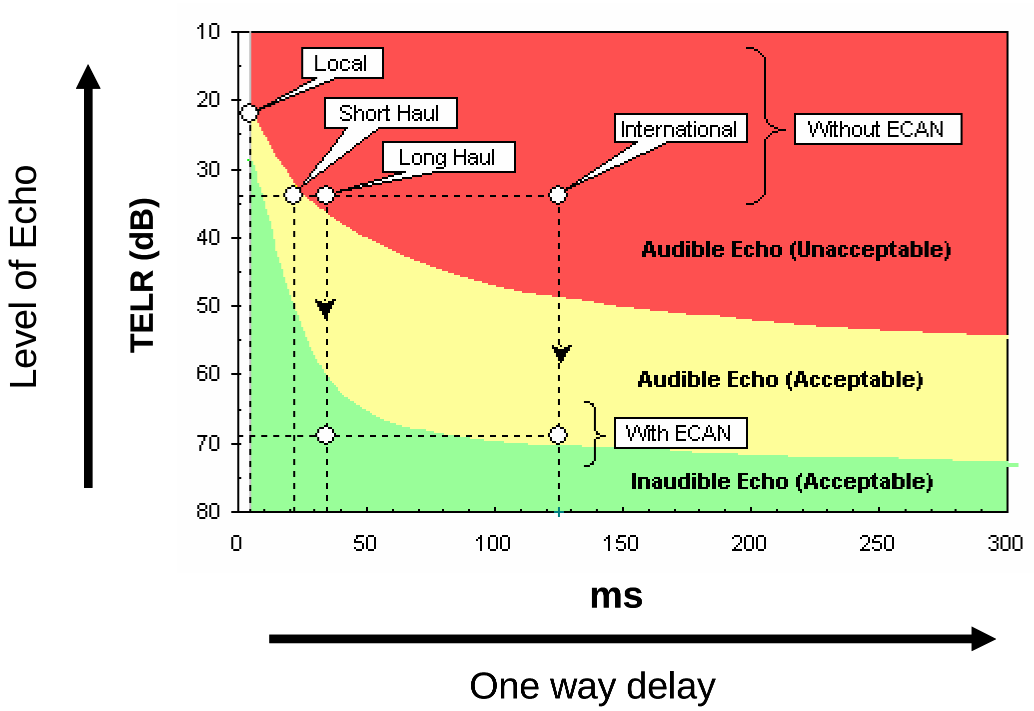


FIGURE 2
Echo Level and one-way Delay

NOTES: *1. Figure 2 is based on ITU-T G.131.* *2. Local = 100 Km, Short Haul = 2,800 Km, Long Haul = 5,000 Km, International = 14,000 Km.*

Proper location of an Echo Canceller (ECAN) in a VoIP network

4.7.7 An ECAN tracks the forward voice signal and the returning echo and builds a filter matching the echo characteristics. The filter is used to create a matching echo and is subtracted from the returning signal to remove the echo.

4.7.8 The audibility of echo depends on the level of the echo signal and on the delay imposed by the network; longer delay makes the echo more apparent. Location of the ECAN requires proximity to the far end to avoid the delay introduced by the packet network.

4.7.9 ECAN coverage (tail length) requires special consideration for coast to coast calls in Australia, including:

4.7.9.1 Optical signals travel at 5µs/km. Therefore, to cross 6000 km it would take (6000 x .005)ms, or 30ms. A round trip would take 60ms.

4.7.9.2 Australian loss plan considerations for TDM should follow Standards such as AS/CA S004, AS/CA S002 and AS/CA S003 for loss plan analysis.

4.7.9.3 The VoIP-TDM gateway needs to consider echoes that are not cancelled by the ECAN in the TDM network.

These will be calls that have delays up to the point where the TDM network puts cancellers on the trunks. As long as ECANs are at the correct point in the VoIP network, providers of VoIP Services will not need to worry about delay from the VoIP cross-country trunks getting into their tail circuit. Therefore, providers of VoIP Services do not need to have tail coverage equivalent to the delay across the country. They only need to have tail coverage sufficient to address the delay up to the point where the ECAN are added to TDM trunks. ACIF C519 states that ECANs must be employed when the round trip delay exceeds 34ms (Refer to ACIF C519 clause 6.2.12).

4.7.9.4 If for some reason, the TDM cancellers are not sufficient to remove echo to meet 65 dB TELR, then providers of VoIP Services would need to cover a longer tail, but this is not expected to be a common occurrence.

4.7.9.5 Optimum OLR is specified by ITU-T G.107 as shown in Figure 3.

NOTES:

1. SLR and RLR values depend on the telephone used. Australian Standards for CE characteristics such as AS/CA S004, AS/CA S002 and AS/CA S003 (with the loss plan specifically referenced in AS/CA S003) should be used for the final loss plan analysis – these standards are available from http://www.commsalliance.com.au/.

2. Where a digital network provides an analogue connection on the customer premises (e.g. with an Analogue Terminal Adapter or ATA), the ATA should emit 3dB lower levels than normally provided at an exchange, and expect 3dB higher levels, since there is no loss in the local loop.

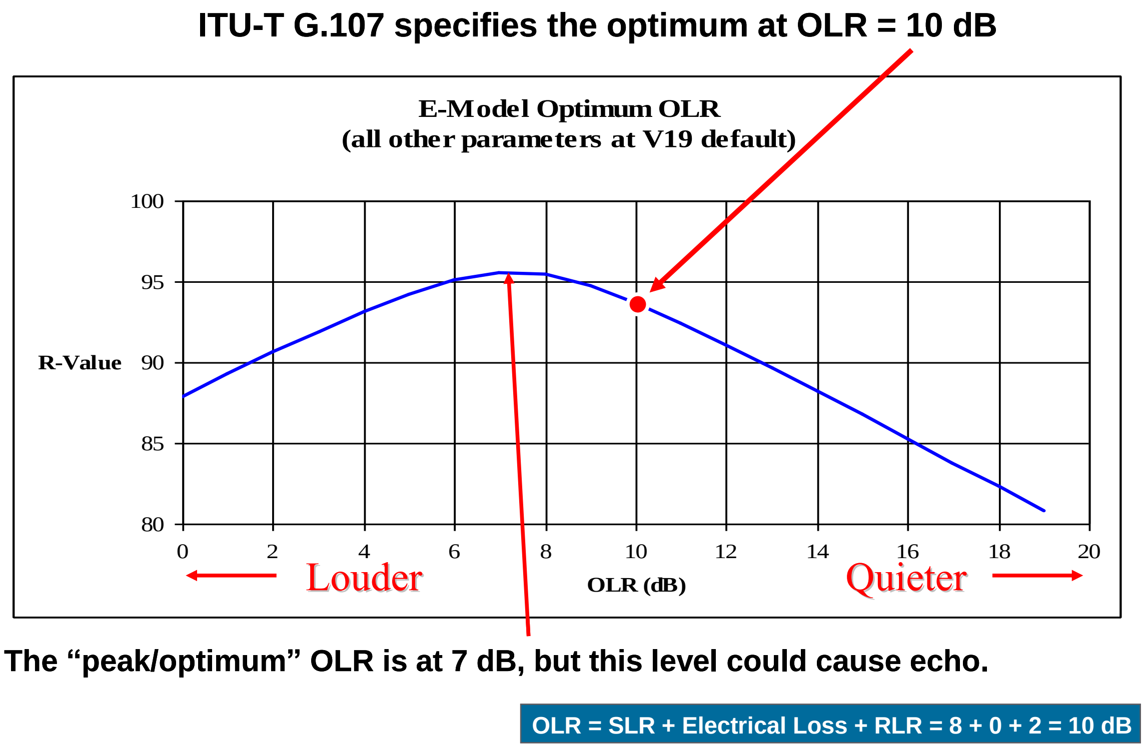


FIGURE 3
Optimum Overall Loudness Rating

4.7.10 Engineering considerations affecting echo control on a VoIP network need to be considered as a whole for the particular network (and call scenario) in question. Take care when using standards developed for PSTN networks (e.g. ITU-T G.168) as overall echo control considerations for a VoIP network are not totally addressed by these TDM standards.

4.8 VoIP Transcoding

4.8.1 Connecting to systems or endpoints that use a different codec requires transcoding.

> *NOTE: Transcoding implies successive encoding of a digital signal by different codecs. Each encoding degrades the quality, and degradation from the successive encodings is cumulative. The E-Model handles transcoding using an additive mode: i.e. the impairment factor for each codec is added to the total impairment for the call. The more codecs there are in succession, the lower the final R. (Note that there is also additional delay with transcoding, which is accounted for separately.)*
>
> *The following example shows this for multiple codecs in a voice path (packet loss is assumed to be < 10^{-3}): where successive encodings are made using Global System for Mobile - Enhanced Full Rate (GSM EFR) (Ie=5), G.711 (Ie=0), and G.729 (Ie=11), the Ie values add up to 16 (again, this does not take account of delay). Transcoding successively by the same codec (separated by G.711) is sometimes called Tandeming.*

4.8.2 For additional information on transcoding when interworking with mobile networks and using wideband codecs refer to G.722.2 Section 10.All end points should support the G.711 (A-law) codec as a fallback. This is to avoid transcoding (distortion) or the situation where endpoints are unable to negotiate a mutually agreeable codec (i.e. the call fails).

4.8.3 End-points should negotiate the codec to be used without enforced transcoding occurring on call gateway(s) at a point of interconnection.

4.8.4 Transcoding between G.711 and G.726 (32kbps) can occur multiple times provided that the signal remains digital, synchronous coding adjustment is used, with no data corruption (packet loss, etc.).

Note: G.726 (32 kbps) is used on DECT handsets.

4.8.5 One should avoid transcoding between CELP codecs (e.g. G.729) or between CELP and ADPCM (G.726/G.722) codecs.

4.8.6 One should count the number of compression codecs when assessing transcoding.

4.8.7 One should reduce the occurrences of transcoding and preferably eliminate them.

4.9 Other Components

4.9.1 Design considerations relating to the handling of fax tones from a codec selection viewpoint, handling of modem tones and the handling of DTMF tones are outside the scope of this guideline but should be considered.

Number of Simultaneous Calls (Call Admission Control)

4.9.2 Providers of VoIP Services should consider monitoring the number of simultaneous calls and take appropriate action according to the type or quality of service subscribed to, as exceeding available bandwidth will obviously result in a severely degraded voice experience.

Post dial delay – parameter definition and value

4.9.3 Post-Dial delay for:

(a) Category C should meet the performance targets for fixed lines in ACIF C519.

(b) Category B should meet the performance targets for mobile services in ACIF C519.

(c) The “Best Effort” category has no target value.

Voice Activity Detection (VAD)

4.9.4 The number of simultaneous calls sharing the available bandwidth when VAD is in use should be sufficient to ensure the probability of active speech on each call, to minimise dropped packets due to overload e.g. less than 1%.

NOTES:

1. VAD reduces bandwidth requirements for aggregate calls by 30–40% because only active speech is transmitted.

2. VAD is also known as silence suppression. When silence suppression is used, typical clipping of 5–8 ms can be noticed due to most gateway VAD implementations. It is often recommended to turn on comfort noise when silence suppression is turned on at the gateway.

3. Bandwidth required to support voice calls with silence suppression depends primarily on the voice activity level, i.e. the ratio of talk spurt/(talk spurt + silence), and the mix of voice calls and voice band data. There are methods developed for capacity engineering based on the central limit theorem (CLT) for voice traffic with silence suppression capability. The CLT model becomes progressively more accurate as the number of sources increase. With voice activity level greater than 30% and the number of voice sources exceeding about 700, it is suitable for capacity engineering.

4. The clipping of the initial sound of the first word in a talk spurt and the packet loss associated with simultaneous talking on the large majority of calls can cause VAD to degrade the perceived conversational voice quality.

Tradeoffs applicable to VoIP Services

4.9.5 Selection of an audio codec (waveform versus frame-based) is of major importance. Factors affected by codec choice include:

NOTE: G.711 and G.726 are waveform codecs which directly represent the analogue signal. Frame-based codecs (e.g., G.729 and AMR) parse the incoming signal into frames before encoding it. Frame-based codecs commonly use Code Excited Linear Prediction (CELP) compression.

(a) Delay: encoding of a waveform codecs is virtually instantaneous while encoding of a frame codec can introduce significant delay.

NOTE: A nominal estimate of the encoding delay of a codec is two times the processing sample size (duration) plus the look ahead, if any. The frame is the processing sample for a frame-based codec.

(b) Voice Bandwidth: Wideband codecs such as AMR-WB support voice frequencies from 50Hz-7kHz, which improves speech clarity compared to narrowband codecs like AMR-NB, G.711 and G.729, which support a bandwidth of 300Hz-3.3kHz.

(c) Data Bandwidth: frame codecs require less bandwidth.

NOTE: For instance, for 20 ms voice sample G.711 has an IP packet size of 200 bytes while G.729 has an IP packet size of 60 bytes (both including 40 byte header) which over Ethernet translates into 96.8 kbps versus 40.8 kbps respectively. Note that *even though G.729 has an 8-to-1 compression ratio for the speech data compared to G.711, for a 20-ms packet, the ratio is about 2-to-1 once the packet headers and other overheads are included.*

(d) Distortion: the distortion added by waveform codecs is more tolerable than that added by frame-based codecs.

(e) It is an advantage to use G.711 for conference and emergency calls. G.726 and all ADPCM codecs are more vulnerable to lost data than G.711.

(f) CELP codecs don't perform very well with non-speech signals such as DTMF tones, EFTPOS machines, intruder alert systems, health alert systems and music. It is recommended to switch to G.711 for transmission of fax(es) when a CELP codec (e.g. G.729 or AMR) is the main voice codec in use.

(g) Packet loss concealment and silence suppression: Some codecs have these two characteristics built-in, while G.711 requires them added externally.

4.9.6 The effect of echo is covered in section 4.7.

4.9.7 Access Jitter can be seriously affected on low speed networks (i.e. less than 10 Mbps). This is especially true on the end-user access network's upstream which often has less bandwidth. When data loading increases relative to voice, the probability of a data packet being put onto the wire increases. Even where voice packets are assigned priority over data packets, a voice packet must wait until the serialization of the current data packet is complete before it can be sent. The slower the link and the larger the data packets, the more this possibility increases the voice packet jitter.

NOTE: Jitter is a function of the loading of all the statistical multiplexers a packet passes through (access, routers, switches, gateways). When loading is unbounded (>90%) jitter is unbounded, delay rises asymptotically and Voice Quality is unpredictable/unstable.

4.9.8 The final goal when engineering a VoIP network is that such a network can provide acceptable levels of Conversational Voice Quality to end-users. When all E-Model factors are considered as a whole, the transmission rating R determines the level of Conversational Voice Quality that can be achieved. The level of acceptability of the new service is determined by how well it meets user expectations regarding perceived voice quality, given their experience with traditional PSTN and wireless technologies.

NOTE: Research has determined that users cannot detect a difference less than 3R and are likely to perceive a difference greater than 7R. Therefore, the end-user acceptability of a new VoIP Service can be quantified. Refer to Appendix D, Quality of Experience (QoE).

4.10 IP Network QoS Classes

4.10.1 G632 defines a number of IP Network QoS classes for network level QoS on networks using IP.

NOTE: The IP network QoS classes and parameter values in G632 are consistent with those in ITU-T Recommendation Y.1541.

4.10.2 Network performance that meets IP Network QoS class 0 in G632 will help meet the recommended QoS for VoIP Services.

5 REFERENCES

Publication	Title
3GPP Specification	
TS 06.60	Enhanced full rate speech transcoding http://www.3gpp.org/ftp/Specs/html-info/0660.htm
TS 26.190	Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Transcoding functions http://www.3gpp.org/ftp/Specs/html-info/26190.htm
TS 26.194	Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Voice Activity Detector (VAD) http://www.3gpp.org/ftp/Specs/html-info/26194.htm
Australian Standards	
AS/CA S002:2010	Analogue interworking and non-interference requirements for Customer Equipment for connection to the PSTN http://commsalliance.com.au/Documents/all/Standards/s002
AS/CA S003:2010	Customer Access Equipment for connection to a Telecommunications Network http://commsalliance.com.au/Documents/all/Standards
AS/CA S004:2013	Voice frequency performance requirements for Customer Equipment http://commsalliance.com.au/Documents/all/Standards/s004
IETF RFCs	
RFC 791	Internet Protocol http://tools.ietf.org/html/rfc791
RFC 3264	An Offer/Answer Model with the Session Description Protocol (SDP) http://tools.ietf.org/html/rfc3264
ITU-T Recommendations	
E.800 (09/2008)	Definitions of terms related to quality of service http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=9524

G.107 (12/2011)	The E-model, a computational model for use in transmission planning http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=11460
G.107.1 (12/2011)	Wideband E-model http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=11453
G.108 (09/1999)	Application of the E-model: A planning guide http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=4753
G.108.1 (05/2000)	Guidance for assessing conversational speech transmission quality effects not covered by the E-Model http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=5082
G.108.2 (03/2007)	Transmission planning aspects of echo cancellers http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=9062
G.109 (09/1999)	Definition of categories of speech transmission quality http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=4754
G.113 (11/2007)	Transmission impairments due to speech processing http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=9273
G.113 (2007) Amd.1 (03/2009)	Transmission impairments due to speech processing Amendment 1: Revised Appendix I – Provisional planning values for the wideband equipment impairment factor and the wideband packet-loss robustness factor http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=9273
G.114 (05/2003)	One-way transmission time http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=6254
G.121 (03/1993)	Loudness ratings (LRs) of national systems http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=764
G.131 (11/2003)	Talker echo and its control http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=7037

G.711 (11/1988)	Pulse code modulation (PCM) of voice frequencies http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=911
G.722 (09/2012)	7 kHz audio-coding within 64 kbit/s http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=11673
G.722.1 (05/2005)	Low-complexity coding at 24 and 32 kbit/s for hands-free operation in systems with low frame loss http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=8495
G.722.2 (07/2003)	Wideband coding of speech at around 16 kbit/s using Adaptive Multi-Rate Wideband (AMR-WB) http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=6506
G.723.1 (05/2006)	Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=8813
G.726 (12/1990)	40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM) http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=924
G.729 (06/2012)	Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP) http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=11675
P.10/G.100 (07/2006)	Vocabulary for performance and quality of service http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=8857
P.800 (08/1996)	Methods for subjective determination of transmission quality http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=3638
P.862 (02/2001)	Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=5374
Y.1541 (12/2011)	Network performance objectives for IP-based services http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=11462

Industry Guidelines	
G632:2012	Quality of Service parameters for networks using the Internet Protocol Guideline http://commsalliance.com.au/Documents/all/guidelines/g632
G635:2013	Testing Arrangements for Quality of Service parameters for Voice over Internet Protocol (VoIP) services http://commsalliance.com.au/Documents/all/guidelines/g635
Industry Code	
ACIF C519:2004	End-To-End Network Performance http://commsalliance.com.au/Documents/all/codes/c519
TIA Publications	
TSB-116-A	Telecommunications, IP Telephony Equipment, Voice Quality Recommendations for IP Telephony http://global.ihs.com/doc_detail.cfm?document_name=TIA%20TSB-116&item_s_key=00381377

APPENDIX A – VARIOUS SCENARIOS FOR VOICE SERVICES

A1 Single Carrier

A.1.1 IP Access and Core networks

This scenario represents calls within a managed VoIP Service, where the originating and terminating legs exist entirely within Carrier X's network domain (see Figure 4). In this case Carrier X manages the voice and IP network service. In this scenario the call signaling and media is carried as IP on an end-to-end basis. The call could be delivered via DSL, HFC, fibre or some other access network technology. Impacts on speech quality are primarily the end-to-end IP network characteristics, the CE characteristics and the choice of codec.

In this case end-to-end delay will usually be higher than the analogue case, particularly if the "last mile" is a lower-speed portion of the network (as can often be the case in the upstream direction). This can easily add 20-40ms of delay to the call (or more if using wireless CE).

To provide a high quality voice service:

(a) the CE must be able to identify voice packets, and give them an appropriate priority when sending them to Carrier X's network;

(b) Carrier X must identify the voice packets and treat them with the correct priority. To be considered "voice-grade", Carrier X must be able to meet criteria for delay and packet loss in the "last mile".

> *NOTE: This also applies to a Customer LAN segment of the network.*

Acoustic echo cancellation is handled within the VoIP CE and is not required within the network.

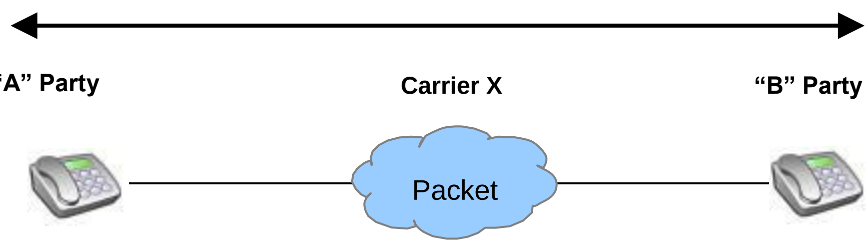


FIGURE 4
IP in Core and Access networks

A.1.2 TDM access and core, IP access network

This scenario covers the case where both TDM and IP segments are used, with an intervening TDM-IP voice gateway (see Figure 5). Speech quality impairments due to the packet CE, access and core networks are given the same considerations as in the example in A.1.1.

In this scenario additional end-to-end delay is incurred due to the addition of the TDM network component (typically small) and speech transcoding required at the TDM-IP gateway. For optimum performance the same codec should be used for interconnected networks (refer to Section 4.8).

For an indication of performance characteristics expected of PSTN services refer to ACIF C519.

Cancellation of network echo heard by the B party caused by reflections of B-Party speech at the A-Party PSTN 2-4 wire hybrid should take place at the packet-TDM gateway (refer to Section 4.7). Acoustic echo at the B party handset should be dealt with within that handset.

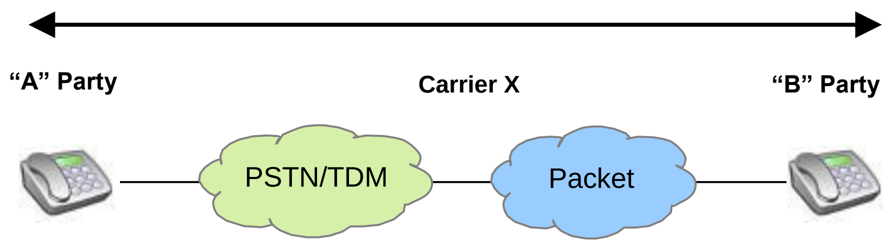


FIGURE 5
TDM Access and Core, IP in access network

A2 Two Carriers

A.2.1 TDM access and core, IP core and access

This scenario is similar to the example in A.1.2 from a functional perspective, with the exception that a TDM interconnect is used between Carrier Y and Carrier X for termination of calls to the PSTN (see Figure 6). The goals are to minimise the risk of transcoding, minimise delay and the potential for trombone trunking. For packet origination, it is recommended to support the carriage of voice as packet as far as possible. TDM originated calls are governed by regulation on where one can interconnect.

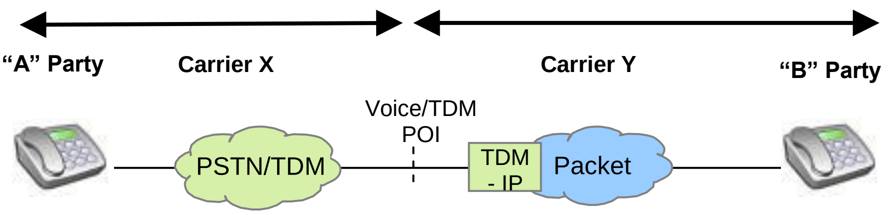


FIGURE 6
TDM access and core, IP core and access

A.2.2 TDM Access & IP Core, IP Access

This scenario is also functionally similar to the examples in A.1.2 and A.2.1, in that a single TDM-IP conversion is required. In this case, however, Carrier X offers a voice service to both the A party and B party (see Figure 7). The primary difference is that Carrier Y offers a packet layer access service to Carrier X, to enable delivery of service to the B party.

In this case, there is a packet interconnect (typically IP) required between Carrier X's network and the "B" party CE.

In order to predict impairments and end-to-end speech quality, the end-to-end packet layer characteristics must be known i.e. for the packet segment spanning Carrier X, Carrier Y and the B party's local network.

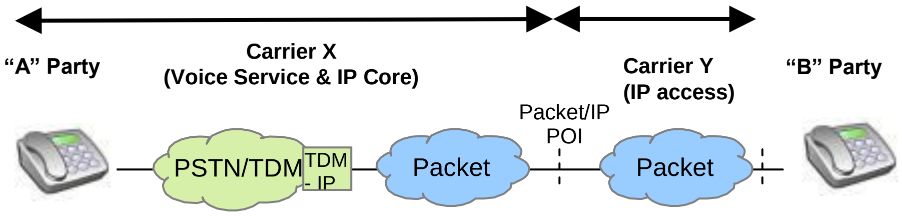


FIGURE 7
TDM Access & IP Core, IP Access

A.2.3 IP Access & IP Core, L2 Access

This scenario represents a competitive target architecture, where any carrier can provide a VoIP service to any party, with packet transmission from end to end.

One or more of the carriers may optionally use a L2 access network for the final connection to the party, such as the NBN.

In this case, there is a packet interconnect (typically IP) required between Carrier X's network and Carrier "Y", and a Layer 2 interconnect between Carrier "Y" and the "Z" Carrier Access network.

In order to predict impairments and end-to-end speech quality, the end-to-end packet layer characteristics must be known i.e. for the packet segment spanning Carrier X, Carrier Y and Carrier Z.

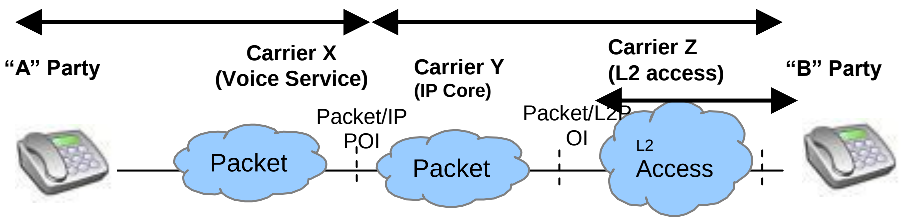


FIGURE 8
TDM Access, IP Core, L2 Access

A3 Three Carriers

A.3.1 IP Access, TDM Core, IP Access

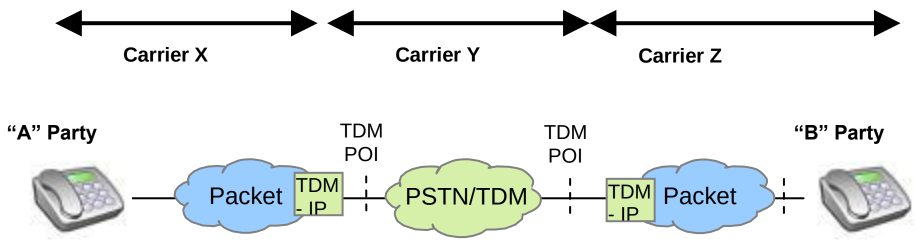


FIGURE 9
IP access, TDM Core, IP Access

In this scenario packet access is used by both the A party and B party, with TDM interconnection (see Figure 9). To ensure acceptable end-to-end speech quality careful consideration must be given to packet network delays, access bandwidth and voice codecs. In this case total network delay could severely impact speech quality. It is particularly important to ensure sufficient points of interconnect are used, thus minimising the effects of network tromboning. Given sufficient access bandwidth, optimal performance will also be achieved if both the A-party and B-party use G.711 codecs.

Where the A- and/or B-parties do not use G.711 codecs then providers of VoIP Service should minimise the use of this scenario with a TDM interconnect between service providers because of the impact of transcoding on voice quality.

This IP-TDM-IP scenario increases end-to-end delays, and has several points where various tones must be detected and acted upon asynchronously. A packet based interconnect is preferred ahead of an IP-TDM-IP scenario.

APPENDIX B – CODEC CHARACTERISTICS

B1 Codec characteristics and selection

B.1.1 Various codecs differ along the multiple characteristics including:

(a) access link speed required and traffic generated (as it affects call charges);

(b) baseline voice distortion;

(c) delay;

(d) immunity to packet loss; and

(e) immunity to transcoding.

Most of these factors are captured in the transmission rating factor R for the service.

B.1.2 Codecs should be selected to meet the target service category (refer to ITU-T G.114, Table I.4 for typical performance of some codecs).

APPENDIX C – PERFORMANCE VALUES BASED ON TRANSMISSION RATING FACTOR R

C1 Introduction

The sections in this Appendix are presented as relevant information when determining performance values based on the transmission rating factor R. Readers are encouraged to consult the complete referenced ITU-T Recommendations (see Section 5 for the list of References).

The earlier G.107 model supports narrowband codecs, while the more recent G.107.1 supports wideband codecs. They use a similar method, and the resulting value R can be compared on a common scale; Narrowband codecs score 0-93, while wideband codecs score 0-129.

The major difference for the end user is that the Ie parameter for narrowband codecs should be used on the narrowband model, and the Ie for wideband codecs should be used on the wideband model.

C2 ITU-T Recommendation G.107 - The E-Model, a computational model for use in transmission planning.

C.2.1 Section 3.1:

The transmission rating factor R is calculated as:

$$\boxed{R = Ro - Is - Id - Ie\text{-}eff + A}$$

Where:

(a) **Ro** represents the basic signal-to-noise ratio including noise sources such as circuit noise and room noise;

(b) **Is** is a combination of all impairments which occur more or less simultaneously with the voice signal;

(c) **Id** represents the impairments caused by delay;

(d) **Ie-eff** represents impairments caused by low bit rate codecs and includes impairment due to packet losses of random distribution; and

> *NOTE: Values of Ie-eff for different codecs are presented in Appendix B, Codec Characteristics. Also refer to G.113 Amendment 2 for provisional planning values for the equipment impairment factor Ie.*

(e) **A** represents the advantage factor, which allows for compensation of impairment factors when there are other advantages of access to the user. Provisional examples for A are given in Table 3 in section 3.6 of G.107.

> *NOTE: One should measure and report the objective call quality without the A value.*

C.2.2 Table 2 in section 3.7 of G.107 presents default values for input parameters of the E-Model.

C.2.3 Guidance for interpreting calculated R factors for planning purposes is given in Annex B of G.107.

C3 ITU-T Recommendation G.109 (09/99) – Definition of categories of speech transmission quality

C.3.1 Section 5 of ITU-T G.109, Definition of categories of speech transmission quality

This section provides the following table which gives definitions of the categories of speech transmission quality in terms of ranges of transmission rating factor R:

R-value range	Speech transmission quality category	User satisfaction
$90 \le R < 100$	Best	Very satisfied
$80 \le R < 90$	High	Satisfied
$70 \le R < 80$	Medium	Some users dissatisfied
$60 \le R < 70$	Low	Many users dissatisfied
$50 \le R < 60$	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/G.107.		

C.3.2 Section 6 of ITU-T G.109, "Examples of speech transmission quality provided in typical scenarios"

This section provides the following estimates of R values for a number of service/network scenarios in Table 2 of G.109:

Service/network scenario	R-value	Deviations from Table 3/G.107
ISDN subscriber to ISDN subscriber, local connection	94	Note 1
Analogue PSTN subscriber to analogue PSTN subscriber, 20 ms delay (average echo path losses; no active echo control)	82	Note 2
Mobile subscriber to analogue PSTN subscriber as perceived at mobile side	72	Note 3
Mobile subscriber to analogue PSTN subscriber as perceived at PSTN side	64	Note 4
Voice over IP connection using G.729a + VAD with 2% packet loss	55	Note 5
NOTE 1 – No deviations. NOTE 2 – TELR = 35 dB, WEPL = 50 dB, T = 20 ms, Tr = 40 ms, Ta = 20 ms. NOTE 3 – TELR = 68 dB, WEPL = 101 dB (ECAN with ERLE = 33 dB assumed), T = 110 ms, Tr = 220 ms, Ta = 110 ms, Ie = 20. NOTE 4 – TELR = 53 dB, WEPL = 101 dB (ECAN with ERLE = 33 dB assumed), T = 110 ms, Tr = 220 ms, Ta = 110 ms, Ie = 20. NOTE 5 – T = 300 ms, Tr = 600 ms, Ta = 300 ms, Ie = 19.		

C4 ITU-T Recommendation G.114 – One-way transmission time

C.4.1 Section 4 of ITU-T G.114 states:

(a) Regardless of the type of application, it is recommended to not exceed a one-way delay of 400 ms for general network planning.

(b) It is desirable to keep the delays seen by user applications as low as possible. The E model should be used to estimate the effect of one-way delay (including all delay sources, i.e., "mouth to ear") on speech transmission quality for conversational speech.

(c) Although a few applications may be slightly affected by end-to-end (i.e., "mouth to ear" in the case of speech) delays of less than 150 ms, if delays can be kept below this figure, most applications, both speech and non-speech, will experience essentially transparent interactivity.

(d) While delays above 400 ms are unacceptable for general network planning purposes, it is recognized that in some exceptional cases this limit will be exceeded. An example of such an exception is an unavoidable double satellite hop for a hard to reach location, the impact of which can be estimated by use of the advantage factor in the E-model.

(e) Regarding the use of the E-model for speech applications, the effect of delay can be seen in the following graph of Transmission Rating, R, versus delay (see Figure 10 below, or Figure 1 in G.114). Also shown are the speech quality categories of ITU-T Rec. G.109, which translate the R values to levels of user acceptance.

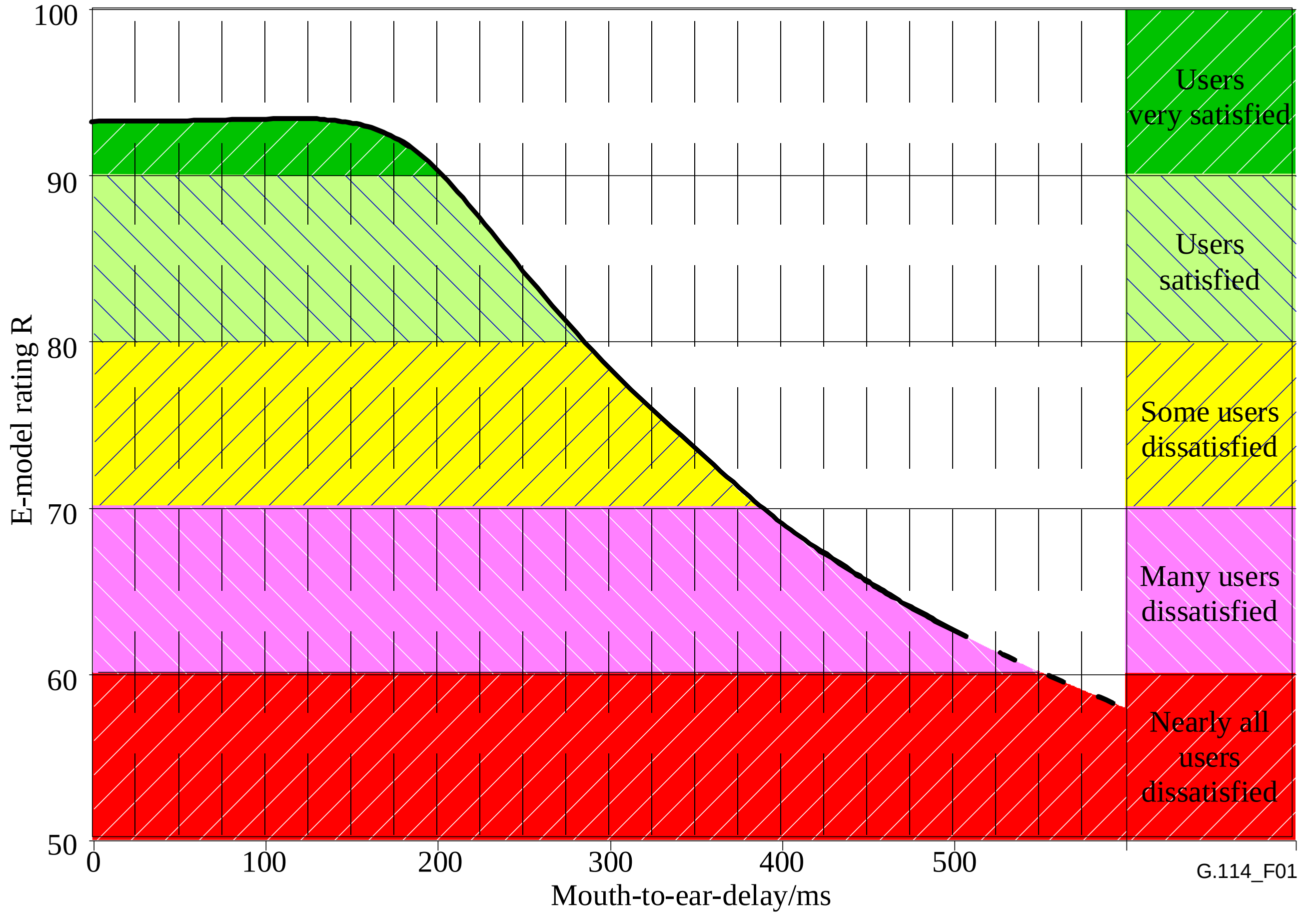


FIGURE 10

Impact of mouth to ear delay on R value

NOTES:

1. The curve in Figure 10 above is based on the effect of pure delay only, i.e., in the complete absence of any echo. This is calculated by setting the ITU-T G.107 E-model parameter Ta equal to the total value of one-way delay from mouth to ear, with all other E-model input parameter values set to their default values. The effect of echo, as would be incurred due to imperfect echo control, will result in lower speech quality for a given value of one-way delay.

2. The calculation also assumes a narrowband Equipment Impairment factor (Ie) of zero. Non-zero values, as would be incurred due to speech coding/processing, will result in lower speech quality for a given value of one-way delay, while a wideband codec will produce higher scores for the same delay.

3. For one-way delay values exceeding 500 ms, the graph is continued as a dashed line to indicate that these results are not fully validated, but is the best estimate of what should be expected and therefore provides useful guidance.

C5 Graphical representation of relationship between R and delay

The following diagrams are a useful way to illustrate variation of R with delay for several cases as shown below.

C.5.1 E-Model reference curve

Figure 10 shows the E-Model reference curve for G.711 codec and TELR = 65dB (which is effectively echo-free).

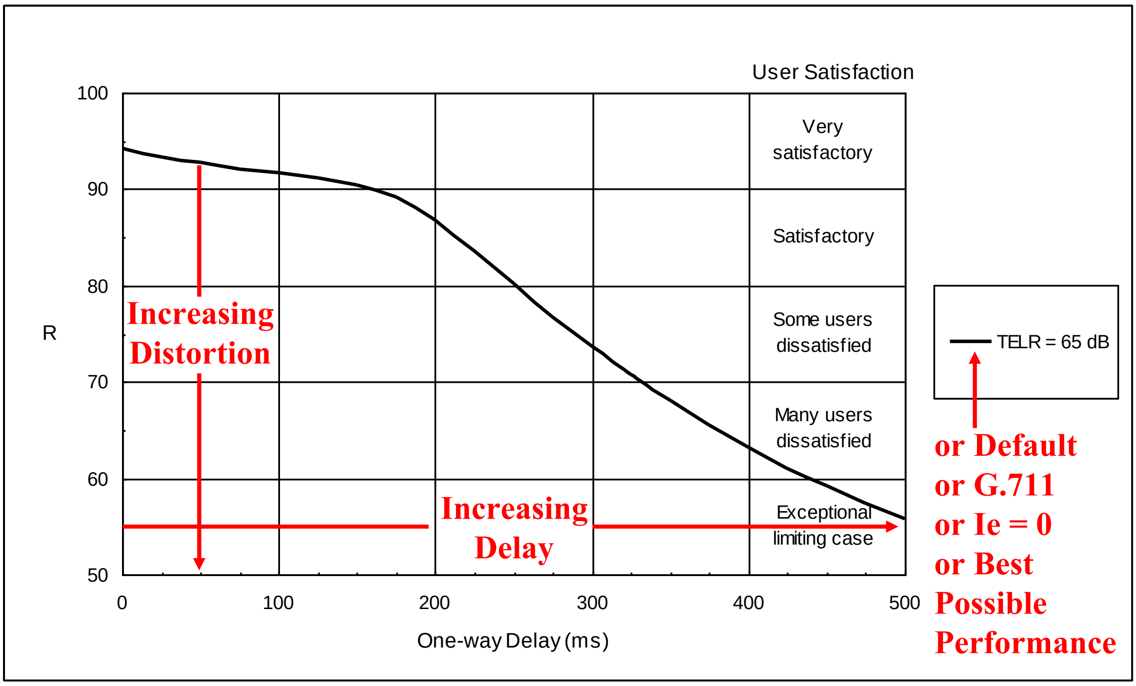


FIGURE 11
ITU-T G.107 Default Delay Impairment

C.5.2 Effect of Increasing Echo

Figure 12 shows the effect of Increasing Echo.

> *NOTE: The TELR value is also a function of the receivers used, and thus echo characteristics of the phones used also need to be considered for transmission planning purposes. For more information on TELR refer to ITU-T Recommendations G.131, G.108, G.108.1 and G.108.2.*

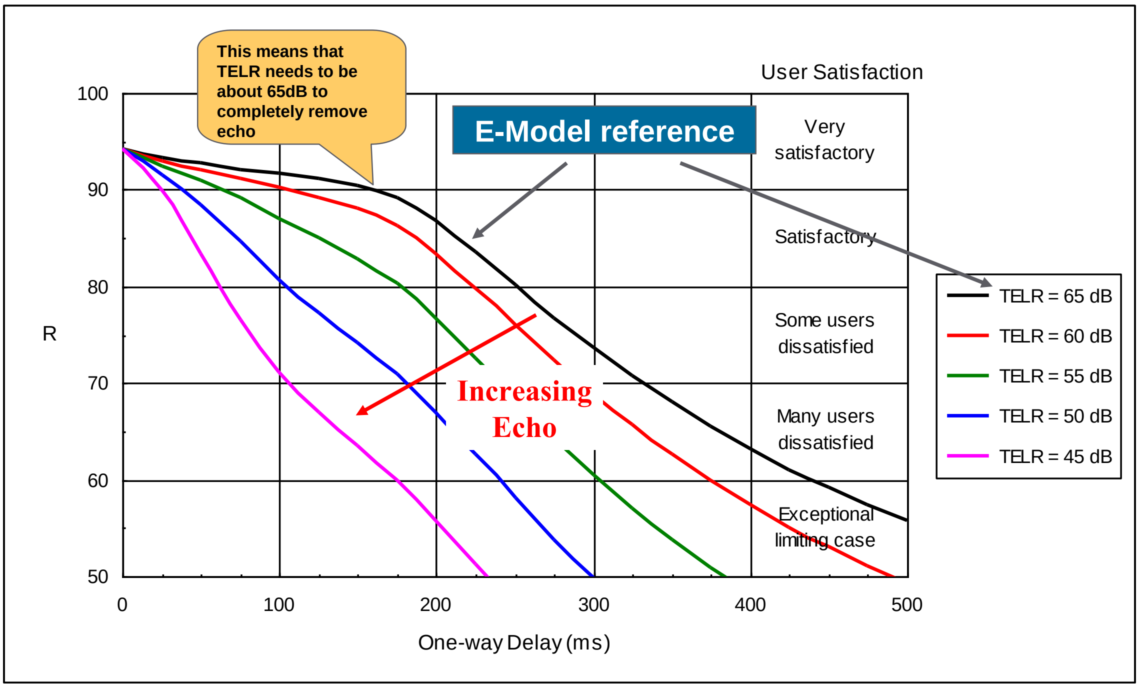


FIGURE 12
E-Model, Echo Impairment

C.5.3 Effect of codec change or transcoding

Figure 13 shows the effect of a change of codec or transcoding.

The change of codec or transcoding to a codec different to ITU-T Rec. G.711 will make the E-Model curve fall by the corresponding equipment impairment factor Ie, e.g. a G.729a codec with an Ie value of 11 will make the E-Model curve fall by 11R.

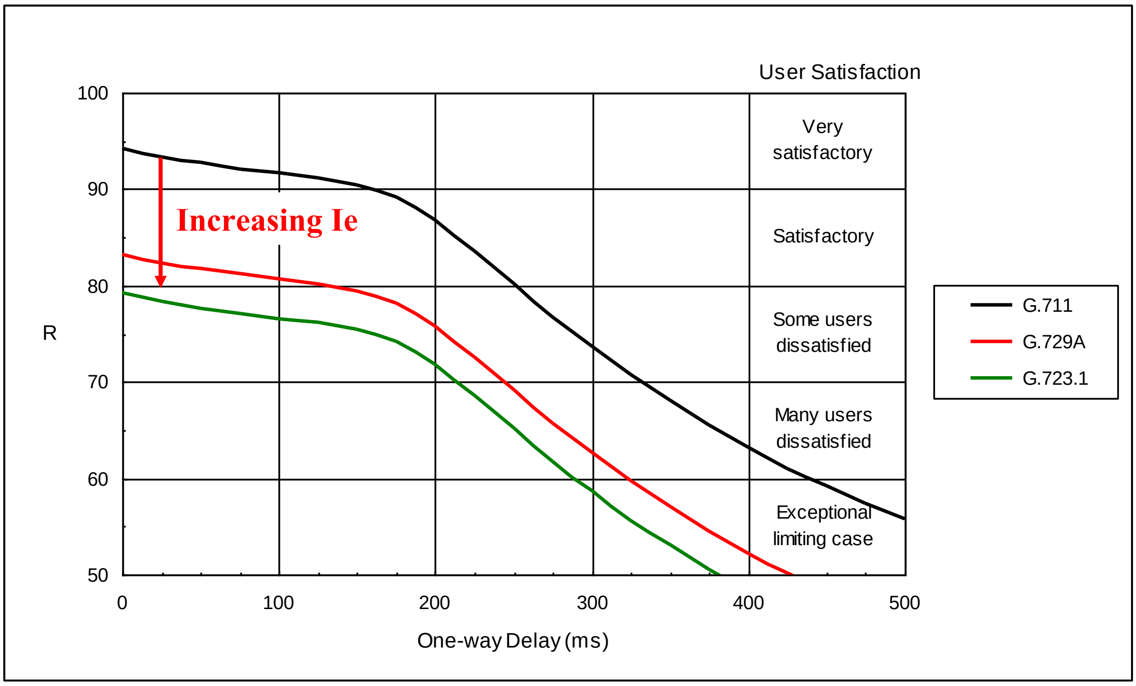


FIGURE 13
E-Model, Speech Compression Impairment

C.5.4 Packet Loss Impairment

The Ie value for a codec as specified in ITU-T G.113 increases with packet loss. Figure 14 shows an example for the G.729a codec.

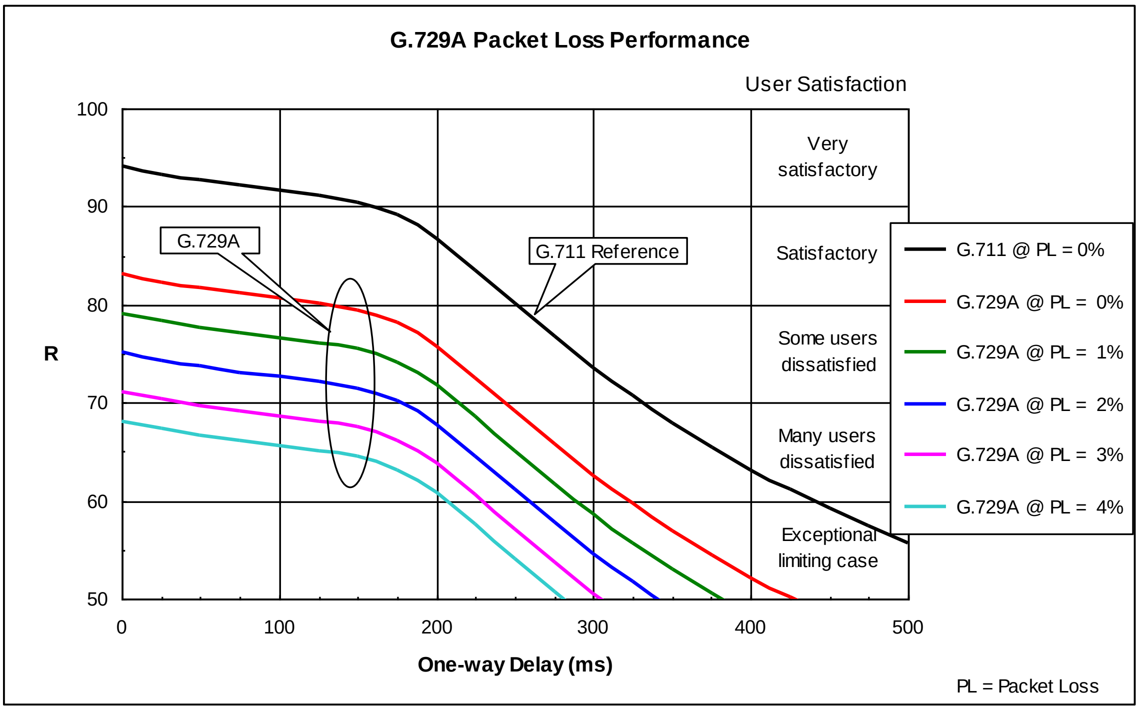


FIGURE 14
E-Model, G.729 Packet Loss Impairment

APPENDIX D - QUALITY OF EXPERIENCE (QoE)

D1 Introduction

D.1.1 QoE refers to the quality of a device, system, or service from the user's point of view. Other terms for similar and related concepts include user performance, human factors, user engineering, user interface design, Human-Computer Interface (HCI), and Man-Machine Interface (MMI).

D.1.2 QoE is associated with all technology used by humans to reduce work, solve problems, or reach goals. Voice telephony is a good example to explore QoE, since QoE of telephones has been well-studied and used to guide network and equipment design for decades. QoE shows up in telephony through:

(a) **Efficiency:** modern telecommunication services make it fast and easy to talk to someone.

(b) **Ease of Use:** the telephone dialpad is a simple user interface: a number sequence is pressed to set up the call, call progress tones tell the caller what is happening as call setup completes, the phone rings, the called party picks up the handset and talks.

(c) **Transparency:** How well does a telephone call approximate a face-to face conversation? The voice should have a good listening level without distortion or noise. Delay should be short enough, and there should be no echo or other annoying artifacts. Any impairments will annoy the user or will require that the user adapt to them. The better the "virtual reality" of the phone channel, the more the user can forget or ignore that the conversation is taking place on the phone.

D.1.3 The effectiveness of a device or system in addressing the user's needs and constraints determines its QoE.

D2 What is QoE?

D.2.1 Quality of Experience (QoE) is the user's perception of the performance of a device, a service, or an application. User perception of quality is a fundamental determinant of acceptability and performance for any service platform. The design and engineering of telecommunications networks must address the perceptual, physical, and cognitive abilities of the humans that will use them; otherwise, the performance of any service or application that runs on the network is likely to be unacceptable.

> *NOTE:*
>
> *1. Without proper understanding of user requirements, there is a risk of both under-engineering, where the network fails to meet the needs of the users, and over-engineering, where the specifications go beyond the user's needs, needlessly driving up the cost to provide the device or service.*
>
> *2. Figure 14 below illustrates some of the factors that influence the QoE of a service, application, or device.*

D.2.2 Successful design requires a thorough understanding of the needs and constraints of the eventual users of the system. QoE is best understood on the system level, since system characteristics and usage factors may interact, and this may be missed in subsystem-level analysis. For telecommunications networks, this means understanding the end-to-end performance.

D.2.3 QoE directly affects the bottom line. If service QoE is poor, the service provider may lose revenue or customers. When a conversation is impaired by excessive packet loss or delay, when an application is slow, or when an e-mail arrives late because the network was congested, communication effectiveness goes down. This affects the user's efficiency, and may push his costs up.

D.2.4 In telecommunications usage, the older term Quality of Service (QoS) has broadened in meaning and is now used to refer to the mechanisms intended to improve or streamline the movement of packets through the network (as in "Is QoS enabled on that network?"). In the past, the same term referred both to the intention (enabling mechanisms used to help ensure good service quality) and the outcome (the user's perception of the service quality), and described the user's perception of quality. We now use the term QoE for the user's perception of quality to eliminate any confusion.

D.2.5 Examples of user tasks or goals in the telecommunications realm include making an appointment (e.g. voice call), finding out when a movie is playing (e.g. internet browsing), or obtaining an item from an online retailer (e.g. e-commerce). When a user needs to spend attention and effort to manage the medium (e.g. accommodate complex setup, unstable session, signal distortion or artifacts, delay or other impairment), the task becomes more difficult to complete, and QoE is reduced. Each application will have its own combination of parameters to determine the QoE. Parameter values leading to acceptable or optimal performance may also be specific to the application.

D.2.6 Engineering for QoE is most effective when it is undertaken at the beginning of the design process. Overall requirements are determined from user needs for the target applications. Other factors such as the total number of users to be supported and the different applications that will run on the network are also taken into account. Requirements for individual network components are derived from the overall requirements. In some cases, it will be necessary to trade-off between factors. For example, the use of encryption may improve the user's feeling of security and privacy but can also increase delay and, therefore, reduce responsiveness. Guidelines for deployment options address the QoE implications of various choices. The user interface associated with the network management system and the effectiveness of quality monitoring features will also be improved by attention to QoE factors.

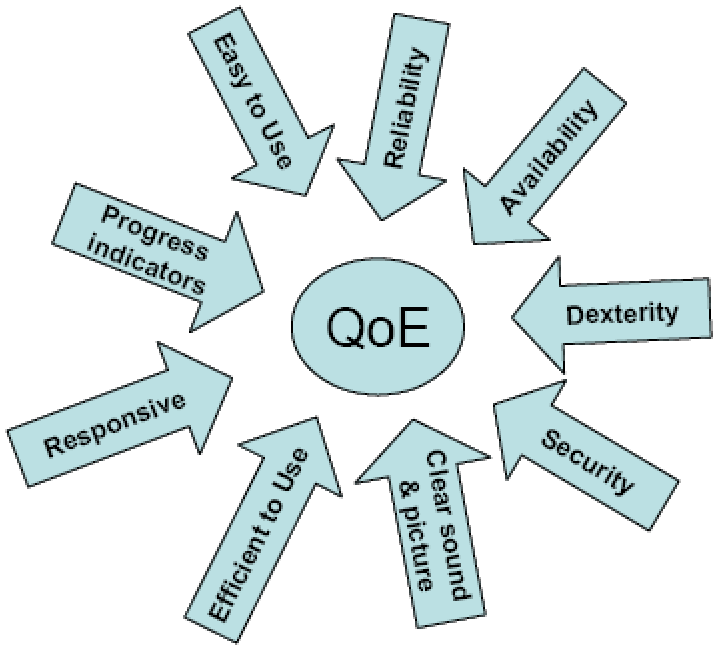


FIGURE 15

Some of the factors influencing the QoE of a service, application, or device

D.2.7 Efforts are more successful where QoE is an integral part of the design process. Retrofitting to improve low QoE is likely to be difficult, expensive, or inadequate, e.g. external ECANs are more expensive than integrated echo control. Tweaking the network to reduce delay may achieve some minor improvement, but many sources of delay will be hard-coded and therefore inaccessible to tuning. What does this mean for buyers of real-time converged networks? Vendors whose performance targets are derived from a comprehensive set of QoE parameters, and whose design intent begins with these targets are likely to achieve better overall QoE. Vendor selection criteria should include the vendor's attention to QoE, as well as system reliability and cost.

D3 Measuring QoE

D.3.1 Aside from the obvious grossly malfunctioning cases and user complaints, how can we determine the level of QoE our network or service provides? Quality of Experience is a subjective quantity and can be measured directly using behavioural science techniques. QoE can be measured in a laboratory setting or in the field, through user ratings, surveys, or observation of user behaviour. Specific techniques include user quality ratings of an application or service, performance measurement, such as the time taken to complete a task, or tabulation of service-related information, such as subscriber complaint rates or frequency of abandoned calls. A familiar QoE metric is subjective MOS.

D.3.2 In the previous section, we emphasize that the outcome is best where design and development proceeds using performance targets based on QoE. The performance targets, however, should not be expressed in terms of QoE metrics. This may seem counter-intuitive, given the previous discussion about QoE. Not only are subjective metrics more time-consuming and expensive to measure, they cannot always be translated into engineering characteristics. In concrete terms, if the specification was given as MOS, and verification testing showed that the performance was below target, how would we know what to fix?

D.3.3 Instead, we need to identify objectively measurable correlates of QoE, and determine the target for each. This approach facilitates design engineering, verification, and troubleshooting in the field, as well as providing customers with measurable performance targets the vendor will stand behind.

D.3.4 Objective parameters that contribute to QoE include:

(a) physical properties of the end device (such as size, weight, fit, button placement);

(b) timing and logic of system operation (such as feedback on progress of a hidden operation, how long the user must wait before going on to the next step, number of steps needed to complete a task);

(c) network characteristics (availability, call setup time, data loss [e.g. bit errors or missing packets], end-to-end delay/response time); and

(d) network / account administration (availability of user support, billing accuracy).

D.3.5 There are a few cases where two or more parameters interact, making it difficult to assess the QoE impact of one parameter individually. In most cases, the parameters can be separated into sensible domains. This allows the network characteristics to be considered separately from the physical properties of the end device.

D.3.6 Service pricing is not a component of QoE. A service that performs poorly remains poor even when it is free. Nevertheless, pricing remains a factor in a customer's decision whether to tolerate poor QoE or to complain about a problem.

D.3.7 The QoE results determine the range of allowable variation in each parameter that matches the perceptual and cognitive abilities of the user. The relationship between the range of variation and the acceptability of the performance allows us to define targets and tolerances for each parameter. When all parameters and their targets are properly identified, and a device is properly engineered to meet them, the resulting device will have high QoE.

D4 Quantifying QoE parameters

D.4.1 As noted previously, we need to relate subjectively measured QoE to a set of objective parameters, and determine the target for each parameter.

D.4.2 The particular values of QoE parameters determine or influence:

(a) the user's rating of service quality; or

(b) his/her performance on some relevant aspect of the service.

D.4.3 Subjective evaluation is done to quantify the relationship between the overall QoE and the objective parameters we believe determine the QoE. We vary the physical parameter (e.g. the resolution of a video image) and examine how the user's quality rating changes.

D.4.4 Figure 15 shows a hypothetical relationship of a generic parameter to some QoE measure. As our hypothetical parameter increases (x-axis), the subjective rating also increases (y-axis). The shape shown is common for QoE parameters, where the user rating bottoms out at the low end and tops out at the high end (so-called "floor" and "ceiling" effects). Other shapes are possible.

D.4.5 The positioning of the unacceptable, acceptable, and premium quality areas depends on another subjective measure, acceptability. Depending on human perceptual factors, user expectation, etc., the boundaries between the coloured regions can shift.

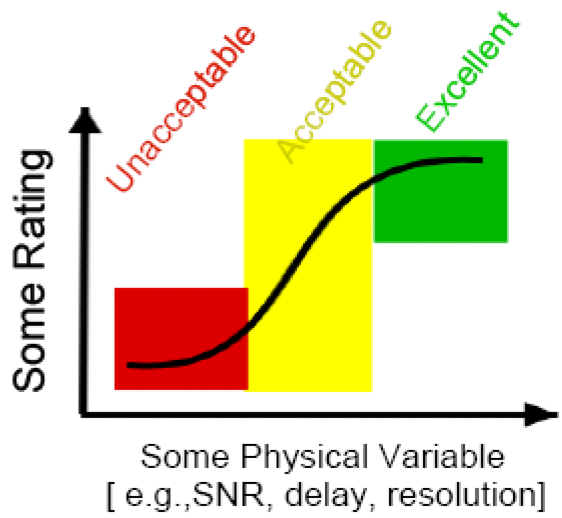


FIGURE 16
QoE variation as function of a variable

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This Working Committee was chaired by Gary Marshall.

James Duck of Communications Alliance provided project management support.

The Guideline was revised in 2012-2013.

Communications Alliance was formed in 2006 to provide a unified voice for the Australian communications industry and to lead it into the next generation of converging networks, technologies and services.

In pursuing its goals, Communications Alliance offers a forum for the industry to make coherent and constructive contributions to policy development and debate.

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