Statement of [c-i-c start] [c-i-c end]

On 28 November 2011, I [c-i-c start] [c-i-c end], make the following statements:

1 I provide this statement on behalf of AAPT Limited (**AAPT**) as part of the Australian Competition and Consumer Commission (**ACCC**) inquiry into varying the exemption provisions in the final access determinations for the Wholesale Line Rental (**WLR**), Local Carriage Service (**LCS**) and Public Switched Telephone Network Originating Access service (**Inquiry**).

2 The purpose of this statement is to respond to a witness statement made by an undisclosed employee of Telstra (**Telstra Witness Statement**) which assesses the technical equivalence of VoIP and traditional fixed line PSTN services generally referred to in this statement as the plain old telephone service (**POTS**).

3 The Telstra Witness Statement was provided by Telstra to the ACCC as Attachment J to Telstra's submission made on or around 14 October 2011. A non-confidential version of the Telstra Witness Statement is available on the ACCC website at www.accc.gov.au.

Confidentiality

4 Certain information contained in this statement is confidential to AAPT and has been provided for the purpose of the Inquiry only. The relevant information has been appropriately identified as "[c-i-c]" and shaded yellow within this statement and must be treated as confidential.

Background

5 I am the [c-i-c start] [c-i-c end] at AAPT and have been employed by AAPT for [c-i-c start] [c-i-c end]. I have been the [c-i-c start] [c-i-c end]. I also previously managed the [c-i-c start] [c-i-c end] department at AAPT where I was responsible for all the technology streams including Time Division Multiplexing and VoIP voice technologies.

6 In my role as [c-i-c start] [c-i-c end] I am responsible for all of AAPT's carrier technology including all aspects of engineering, operations and customer service at AAPT.

7 I have been provided with a non-confidential copy of the Telstra Witness Statement. A copy of the non-confidential Telstra Witness Statement is provided at **Attachment A** to this statement.

8 Unless otherwise indicated, the meaning of any technical terms used in this statement has the same meaning as used in the Telstra Witness Statement.

VoIP is not substitutable for POTS

9 I do not agree with the overall conclusion encapsulated at paragraph 18 of the Telstra Witness Statement:

> "*Insofar as it relates to voice, from the customer's perspective, Carrier Grade VoIP provides a service that is substitutable for POTS in respect of service quality, features and emergency calls*…"

10 Based on my professional experience, it is my view that Carrier Grade VoIP cannot at this time be considered to be substitutable for POTS due to the operational limitations of the ULLS in respect of service restoration, features and other technical aspects, from the customer's perspective or otherwise. In addition, I also consider that Carrier Grade VoIP cannot be considered an economic substitute for a single line POTS service.

11 I set out below my views in relation to the technical differences between a POTS and VoIP service to support the conclusion that Carrier Grade VoIP cannot be considered a substitute for a POTS service which serves as a customer's primary telephone line.

Service quality

12 At paragraph 16 of the Telstra Witness Statement, "Carrier Grade VoIP" is defined to be the category of VoIP "where all aspects of the communication, end to end, are managed by the carriage service provider who supplies the service". For a carrier other than Telstra, the ability to manage "all aspects of the communication" on an "end to end" basis to ensure service quality will depend in large part on the service quality provided by Telstra in relation to the supply of the unconditioned local loop service (**ULLS**).

13 For example, there are a number of characteristics of VoIP provided via the unconditioned local loop (**ULL**) and certain ULLS processes imposed by Telstra which may limit AAPT's ability to provide a Carrier Grade VoIP service which is substitutable for a POTS. These include:

a) *Service provisioning and fault restoration* - While service level assurances for provisioning and restoring a POTS and a standard ULLS are similar, the time taken for AAPT to provide or restore a VoIP service itself will invariably take longer than providing or restoring a POTS.

(i) This is because VoIP provided via the ULL can only be provided by leveraging voice over broadband. Voice interference faults for VoIP are not included in Telstra's ULL service level agreement. Instead, Telstra is only subject to a 'best endeavours' obligations to restore service quality. Ultimately, this means that service levels for VoIP are less deterministic and are subject to more variation in service quality and service restoration times than a POTS.

(ii) Putting aside the delay AAPT may face in gaining access to Telstra's facilities to install a DSLAM, access to the ULL is only one part of AAPT's VoIP provisioning or fault restoration processes. For example, for AAPT to provision VoIP, AAPT not only requires supply of the ULLS from Telstra, AAPT must also conduct further works such as equipment isolation and testing in order to provide the end to end VoIP service or to restore a VoIP service. This could result in doubling of the time taken for an access seeker to provide an active VoIP service to a customer or to restore a faulty VoIP service compared to the supply or restoration of a POTS.

(iii) In addition, Telstra often has processes which can delay AAPT's fault restoration of VoIP. For example, where a faulty port is discovered immediately following jumpering for ULL access or an existing port becomes faulty, Telstra does not record this as a fault event. Instead Telstra requires AAPT to enter in a new ULL provisioning order before Telstra will move the jumpered wires to a working port. This will invariably mean delay in providing the VoIP service to the customer or in rectification of the faulty service.

b) *Number porting* - When a customer churns to AAPT and the customer wants to keep the same number, it is a complex business process that is difficult to co-ordinate with Telstra and will generally take anytime between 5 days and one month to complete the port to a VoIP service whereas for a POTS, churning will occur within 3 days because number porting will not be required since the customer's number remains on Telstra's network and there is no impact to the customer's voice service.

c) *Ubiquity* – Telstra is the only provider with a ubiquitous network, which is attractive to those customers who have a national presence. As there are currently no alternative wholesalers of a standalone voice only service, AAPT can only service a customer with various offices throughout Australia by relying on supply by Telstra in exchange service areas where AAPT has no infrastructure. Even where AAPT has a DSLAM installed, ULLS (and therefore VoIP) can only be supplied if there is an unconditioned wire line between an exchange and an end-user's premises. Where a large pair gain system has been installed, the wire line is considered to be "conditioned" and ULLS (and therefore VoIP) cannot be supplied. In addition, the further out a customer is from an exchange, the less bandwidth there is available for AAPT to provision VoIP over broadband via the ULL.

14 As set out at paragraph 25 of the Telstra Witness Statement, service quality when deploying VoIP may be affected in respect of dial-up customer equipment such as fax, modem, EFTPOS and security alarms. While there may be a number of techniques which can be employed to allow such equipment to function on a VoIP network, the need to implement a "work-around" would not be positively received by a customer who requires a simple solution to provide, for instance, EFTPOS in all their retail outlets.

15 Each of the technical differences between the POTS and VoIP (provided over ULL) described at paragraphs 13 to 14 above can, on their own, manifest into serious service quality issues which can substantially affect the customer's perception regarding the substitutability of the two services. Where those differences occur together, the divergence in service quality is further compounded. In such circumstances, it would be difficult for anyone to conclude that POTS and VoIP services are technically equivalent.

Features

16 In addition to the PSTN features described in the Telstra Witness Statement such as a different sounding dial tone, there are other features which are not provided by VoIP which the customer may value above any additional features that could be provided by VoIP, such as the Telstra directory service "Call-connect 1234".

Reliability and power

17 VoIP services rely on mains power, while a traditional voice service is powered via the phone line. As such, an end-user relying upon a VoIP service as their primary telephone line would not be able to make phone calls during a power failure. In the case of an emergency, this could be life threatening. While this may be the same case for a cordless phone, this does not remove the inherent power limitation of VoIP.

Although many customers do install cordless phones, they will also often retain the use of a phone which is powered via the phone line. For those customers that specifically forgo a phone-line powered voice service, they will often require the service provider to install power backup which would add another layer of cost and complexity in the provisioning, supply and maintenance of a VoIP solution when compared to traditional POTS.

Overall costs and usage costs

18 VoIP services may not be charged in the same way as a POTS service is charged and this may be an important consideration for some customers. While VoIP calls can have a lower per call rate than a POTS service, there could be higher installation and set-up costs, including the cost of any extra equipment that may be required. In addition, a VoIP service may also count towards a customer's internet download quota, which may require extra usage management by the customer to ensure they do not exceed their data limit.

19 [c-i-c start] [redacted] [c-i-c end]

NBN Roll out

20 While it is acknowledged that in the future, voice services will be delivered via VoIP on the NBN, this will not be a complete reality until the NBN roll out is completed, which is predicted to take at least ten years. By that time, all retail service providers will be accessing the same infrastructure from a wholesale-only provider within a non-discriminative and level-playing field environment.

21 Moreover, VoIP provided over the NBN, which is a fibre network, is not the same as VoIP provided over Telstra's existing copper network. VoIP provided over the NBN is unlikely to suffer the same limitations of VoIP provided over copper which can often be constrained by distance (length of the copper wire), poor copper quality and large pair

gain systems. Accordingly, VoIP provided over the NBN is more likely to be a technical substitute for a POTS.

Conclusion

22 Given the many Telstra imposed operational challenges and the technical differences that I have set out above between VoIP provided via the ULL and a traditional POTS, which can adversely affect the service quality of a voice service delivered as VoIP over copper, it is my view that, currently, VoIP delivered over the ULL is not sufficiently equivalent or comparable to the POTS to be considered an effective substitute.

Dated: 28th November 2011

[c-i-c start] [c-i-c end]

[c-i-c start] [c-i-c end]

STATEMENT OF [C-I-COMMENCES] [C-I-C] [C-I-C ENDS]

On 23 September 2011, I, **[c-i-commences]** [c-i-c] **[c-i-c ends]** state as follows:

1 I provide this statement on behalf of Telstra Corporation Limited ("**Telstra**") strictly for the purpose of the Australian Competition & Consumer Commission's inquiry into varying the exemption provisions in the final access determinations for the wholesale line rental service, local carriage service and the public switched telephone network originating access service.

A. Confidentiality

2 The information in this statement is confidential to Telstra. I have prepared this statement on the basis that the information in it will be treated as confidential.

B. Background

3 I am the **[c-i-c comments]** [c-i-c] **[c-i-c ends]** at Telstra Corporation Limited ("**Telstra**"). I have been the **[c-i-c commences]** [c-i-c] **[c-i-c ends]** of Telstra since July 2000. I also previously managed the **[c-i-c commences]** [c-i-c] **[c-i-c ends]** from August 1996 to their closure in December 2005.

4 In my role as **[c-i-c commences]** [c-i-c] **[c-i-c ends]** I am responsible for investigating future technologies which will impact on Telstra's business. I am also exposed to and have familiarity with the technologies that Telstra requires to run its current and future businesses.

5 **[c-i-c commences]** [c-i-c]. **[c-i-c ends]**

6 Further information on my employment history, academic expertise, memberships, appointments, conferences, research accomplishments and publications are set out in my Curriculum Vitae which forms **Attachment A** to this Statement.

7 As a result of the above postings and expertise, I am familiar with most new technologies and developments which are available or becoming available in the communications industry.

8 I have been asked to comment on the equivalence of Voice over Internet Protocol ("**VoIP**") and the traditional voice services provided over the Public Switched

Telephone Network ("**PSTN**") and in particular, voice services provided by access seekers using unconditioned local loop services ("**ULLS**") and Digital Subscriber Line Multiplex ("**DSLAM**") equipment.

C. Provision of voice services

9 There are a number of ways to provide voice services using ULLS and DSLAM equipment (the function of a DSLAM is discussed in paragraph 12 below). These are, in broad terms:

(a) POTS (or "plain old telephone service") which provides voice services using standard switching technology over the PSTN (though I note that POTS can also be provided without the aid of a DSLAM or MSAN by transmitting signals through the copper network to a POTS exchange);

(b) POTS emulation, involving POTS being carried as analogue voice from the customer to the DSLAM where it is converted into VoIP with VoIP then being carried to the core network; and

(c) VoIP, involving VoIP being carried from the customer to the DSLAM and then into the core network. VoIP services include over-the-top (or best efforts) VoIP and Carrier-Grade VoIP: see section D of this statement for further details.

10 I have been provided with a copy of a statement previously provided by **[c-i-c commences]** [redacted] **[c-i-c ends]** dated **[c-i-c commences]** [redacted]. **[c-i-c ends]** is Telstra's Principal Domain Expert - Access Wholesale and Regulatory. A copy of **[c-i-c commences]** [redacted] **[c-i-c ends]** statement is at **Attachment B**.

11 **[c-i-c commences]** [redacted] **[c-i-c ends]** statement provides a useful summary of the above modes of provision of voice services at paragraphs [61] to [72]. I consider **[c-i-c commences]** [redacted] **[c-i-c ends]** summary to be accurate. I set out some additional observations, clarifications and updates below.

12 At paragraphs [27] to [28] and [52] to [59], **[c-i-c commences]** [redacted] **[c-i-c ends]** provides a summary of the function of the DSLAM. A DSLAM is a piece of equipment which is responsible for mulitplexing, which is a process where broadband customer data is combined with analogue voice service over the same copper wires between the customer and the exchange. DSLAM's are also used to

provide broadband data over the copper lines to the customer, even if there is no voice service (so called "naked DSL"). Strictly speaking, a DSLAM only supports Digital Subscriber Line Technology ("**DSL**"), the most common example of which is Asymmetric Digital Subscriber Line ("**ADSL**").

13 A Multi-Service Access Node ("**MSAN**") is a variation of the DSLAM technology. An MSAN typically supports an additional function such as a voice card for analogue (low frequency) voice termination. It is important to note that although an MSAN can be used to provide carrier grade VoIP, it is not necessary. For example, carrier grade VoIP may be provided over a broadband access (which traverses DSLAM equipment) by placing the voice termination function at the customer's premises.

14 By reference to the three methods of providing voice referred to in paragraph 9 above, the DSLAM/MSAN is able to provide voice services with the aid of the following equipment:

(a) for POTS, through the use of a splitter and a PSTN (see paragraphs [61] to [64] of **[c-i-c commences]** [c-i-c] **[c-i-c ends]** statement);

(b) for POTS emulation, through the use of a splitter (or more specifically a voice card), a Softswitch and a PSTN gateway (see paragraphs [65] to [69] of **[c-i-c commences]** [c-i-c] **[c-i-c ends]** statement);

(c) for VoIP, through an analogue telephone adaptor ("**ATA**") or VOIP phone, a Softswitch and a PSTN gateway (see paragraphs [70] to [72] of **[c-i-c commences]** [c-i-c] **[c-i-c ends]** statement).

15 If VoIP is carried directly from the customer (via a ULLS) to an internet service provider's Softswitch (as illustrated in the diagram on page 21 of **[c-i-c commences]** [c-i-c] **[c-i-c ends]** statement), then either an MSAN or DSLAM can be used for that purpose.

D. VoIP

16 VoIP can essentially be broken down into two categories of service:

(a) 'Carrier Grade VoIP' where all aspects of the communication, end to end, are managed by the carriage service provider who supplies the service; and

(b) 'Other VoIP' where the voice service is provided with no management, control or intervention by the carrier (eg. Skype or Engin).

17 For the purposes of this statement, my focus is upon 'Carrier Grade VoIP'. An example of Carrier Grade VoIP is **[c-i-c commences]** [c-i-c] **[c-i-c ends]**, which is discussed in paragraph 23 below.

18 Insofar as it relates to voice, from the customer's perspective, Carrier Grade VoIP provides a service that is substitutable for POTS in respect of service quality, features and emergency calls. Each of these matters are discussed below. The key distinction between VoIP and POTS is in respect of the need for power, which is also addressed below.

Service quality of VoIP

19 There are a number of factors that determine perceived voice quality in a telephone network – these include total end to end delay, sound distortion caused by conversion of the human voice to digital format (encoding), echo, signal impairments, loudness and design factors relating to the phone itself.

20 By appropriate selection of network and voice service design parameters, a Carrier Grade VoIP provider can, from the customer's perspective, achieve a similar voice quality experience to that of the PSTN today in terms of delay, distortion, echo and loudness.

21 In this regard, I attach at **Attachment C** a copy of Industry Guideline G634:2007 *Quality of Service parameters for Voice over Internet Protocol (VoIP) services* ("**the Guideline**") published by the Communications Alliance Ltd ("**CAL**"). These guidelines outline the relationship between these voice and IP network design parameters and the voice quality likely to be experienced by the end users. The CAL is the primary telecommunications industry body which, amongst other things, provides technical specifications and guidelines on the provision of telecommunications services in Australia.

22 Using VoIP technology, the voice signal is captured in data information "packets" which are then transmitted over an IP data network. For Carrier Grade VoIP services, the underlying IP network is designed and managed by the carrier in such a way that IP voice packets are given priority over other data packets. This is

important as it ensures voice packets continue to be transmitted when the network is busy or congested. Further, the underlying network capacity must be designed to ensure that sufficient bandwidth will always exist for the expected number of calls within the carrier's network. These techniques are referred to in paragraph [72] of **[c-i-c commences]** [c-i-c] **[c-i-c ends]** statement. **[c-i-c commences]** [c-i-c].

23 [c-i-c]. **[c-i-c ends]**

24 Voice quality, as perceived by end users of a telephony system, is typically measured using a 1 to 5 scale known as the "Mean Opinion Score" ("**MOS**"). MOS represents an end user's subjective rating of the audio quality within a voice communication. Within a VoIP network, the user perceived MOS can be predicted or estimated through measurement of a number of critical network parameters, namely, packet loss, packet delay and jitter (or variation in delay). The MOS score is then predicted by incorporating these values with known values for other network parameters (such as the digital voice encoding used) into an industry model called the E-model. **[c-i-c commences]** [c-i-c] **[c-i-c ends]**.

25 One area where service quality may be affected when deploying VoIP is in respect of dial-up customer equipment such as, fax, modem, EFTPOS or security alarms, all of which use data over a voice channel. These technologies, which were originally designed to work with PSTN technology, must be managed carefully due to the different nature of VoIP communication channels (for example, variable network delays, the potential for packet loss and different forms of digital encoding). It should also be noted that while interim support for these devices will be needed, in the longer term it is anticipated that customers will migrate to IP broadband based data terminals (eg. IP based eftpos or home alarm equipment).

26 In the meantime, there are a number of techniques available for the support of existing fax, EFTPOS and alarm services to enable them to function on VoIP. In short, this involves configuring settings in the ATA and the PSTN gateway so that fax, EFTPOS and/or alarm can function on the VoIP network.

Emergency services on VoIP

27 Emergency services (eg 000 calls) are reachable by a VoIP system. Critically, the emergency services provider can see the location of the caller, provided the VoIP service provider has met its industry obligations to provide all necessary

information to the Integrated Public Number Database ("**IPND**"). The IPND is an industry-wide database containing all listed and unlisted public telephone numbers and associated information. It is a critical source of information for emergency and law enforcement purposes. In this regard, attached to this statement at **Attachment D** is a copy of a paper published by CAL entitled *Access to emergency services for users of VoIP and Internet Telephony* which sets out a service provider's obligations in this respect.

28 It should be noted that, for emergency calling, a limitation that can apply is that a VoIP device could be moved to a different location on the same IP network, potentially rendering address information inaccurate. This risk is minimised in two ways today. First, Australian VoIP service providers are obligated to signal to the emergency organisation that a call is originating from a VoIP device – enabling appropriate measures to be taken to validate the correct address when the call is answered (as with emergency calls from mobile phones). Secondly, Carrier Grade VoIP providers can restrict support of their VoIP service to phones connected only over their access networks with the result that a VoIP phone moved elsewhere, such as to a general Internet socket, will not receive service.

Features of VoIP

29 Insofar as voice is concerned, the core features/characteristics of typical PSTN services are available on VoIP, but may be accessed in slightly different ways. Similar to a PSTN phone, VoIP has the following core characteristics:

(a) ability to dial a telephone number and be connected to that number; and

(b) ability to be alerted to an incoming call through a ring tone.

30 The differences in core characteristics are relatively minor and do not affect the service quality from the customer's perspective. One example is in respect of dial tone. The dial tone for VoIP may sound different to that on the PSTN. This is because the dial tone for PSTN is an audio signal from the network whereas for VoIP it is generated by the device itself or by the ATA. Another example is in respect of call forwarding services on VoIP where the star code used to activate this service may be different from PSTN (depending upon the design and configuration of the softswitch used by the VoIP provider).

31 VoIP Softswitch technology typically supports additional features that are not available on the PSTN service today. Examples of these include:

(a) Do Not Disturb function;

(b) Call Screening or Selective Call Rejection (eg. based on calling line identity, time of day, day of week);

(c) High Definition Voice – providing significantly higher audio quality than available on today's POTS services;

(d) IP Video phones, video calls and video conferencing; or

(e) Multiple lines or numbers over a single broadband access (**[c-i-c commences]** c-i-c **[c-i-c ends]**).

Power

32 A traditional wired POTS phone is provided with continuous power from a local exchange. In the case of a voice service provided by POTS emulation, with analogue voice delivered from the MSAN to the customer's premises, continuous power is supplied through the customer equipment.

33 VoIP based end user devices must be powered via either mains power or battery backup. Battery backup (kits for which are readily available on an off-the-shelf basis), is seldom provided due to a number of potential issues (including, amongst other things, the cost and inconvenience of battery back up). Customers who use cordless phones with the PSTN service are also faced with the same issue, namely that their voice service is dependent upon the availability of power, unless they also use a wired phone. To my knowledge, a vast number of consumers use cordless phones.

E. National Broadband Network (NBN) development

34 The NBN is an Australian Government initiative to deliver a high speed broadband access network that can be accessed by all Australians. To achieve this, 93% of premises will be connected via optical fibre, while the remaining 7% will be connected via fixed wireless and satellite technologies. The NBN Co, a government owned organisation, will build, own and operate the network.

35 The NBN has, to date, been in an initial trial phase with a progressive roll out planned over a 10 year period. Early launch sites have recently been announced with full roll out expected by late 2012.

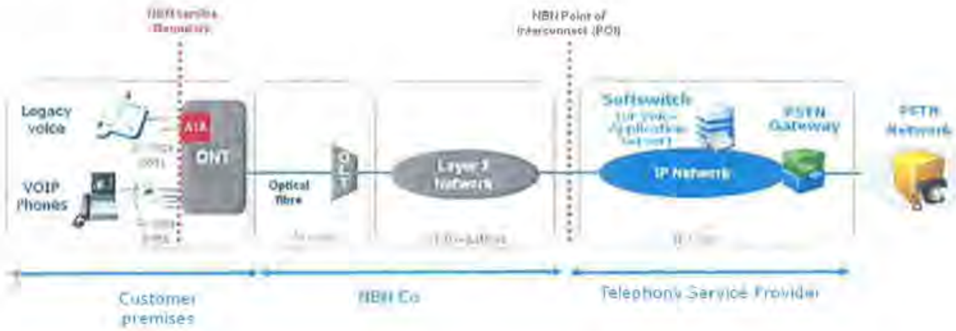
36 **[c-i-c commences]** [c-i-c] **[c-i-c ends]**.

37 The NBN network will be implemented using packet based technology, known as Layer 2 Ethernet aggregation, for switching and transmission. The network will support managed Quality of Service levels for voice or data services as discussed in paragraph 43 below. Retail or wholesale service providers wishing to offer voice or data services will interconnect to the NBN at a Point of Interconnect ("**POI**") provided by NBN Co.

38 Equipment known as an Optical Network Terminal ("**ONT**") will terminate the optical fibre used in the NBN access network within the customer premises. The ONT will also provide physical interfaces (or ports) for the connection of home telephone wiring and customer equipment to the NBN.

39 Two physical interface types will be offered on the ONT; (i) "UNI-V" or ATA ports for the connection of legacy "POTS" handsets and (ii) UNI-D Ethernet ports for connection of data equipment such as Internet routers or personal computers or VoIP phones.

40 Voice Services delivered over the NBN fibre access can be delivered in two ways. First, as described, existing POTS handsets can be connected directly to the NBN ATA port. VoIP technology will be used to connect the ATA to the service provider's softswitch as shown in the Design Diagram below. Secondly, a VOIP phone or third party ATA can be connected to the service provider's softswitch via the Ethernet data port.



Design Diagram

41 Given that voice services will be delivered via VoIP on the NBN, it appears that the Government considers that VoIP services provide a substitute for existing POTS services.

42 The NBN will offer a battery backup capability for the voice (UNI-V) port on the ONT. Thus, in the event a power outage, 50V DV power will be supplied for use by a legacy POTS phone, enabling the service to continue operation for the life of the battery. It should be noted, however, that in such an event, mains powered telephones such as cordless phones will not continue to operate, as they will not on today's PSTN, unless connected to a separate battery backup provided by the customer.

43 In order to support voice quality, the voice services on the NBN will make use of Quality of Service techniques implemented on the underlying network. Services that use the NBN ATA (UNI-V) port will use a high priority traffic class suited to voice. Characteristics of this class of service include low packet delay, low packet loss and a committed bandwidth. For service providers wishing to support VOIP phones or devices on the data (UNI-D) port, this high priority traffic class will also be available. These techniques are also referred to in paragraph [72] of **[c-i-c commences]** [c-i-c] **[c-i-c ends]** Witness Statement.

F. Conclusion

44 In summary I make the following observations and conclusions:

(a) Carrier grade voice services can be provided using today's voice and IP network technology and can be delivered using either DSLAM or MSAN equipment.

(b) Through appropriate design and configuration, carrier grade VoIP networks can provide a voice quality comparable to today's POTS services.

(c) VoIP technology can provide enhanced service features not available on POTS today.

(d) An industry-wide agreed solution is in place today for calling emergency services on VoIP.

(e) The voice technology to be used in the NBN is exclusively VoIP.

(f) If required, the alternative solution to POTS "Lifeline" (being the continuous power provided for PSTN phones from a local exchange) for VOIP is battery backup (this will be the case for NBN services). This same issue occurs for POTS customers using cordless phones today.

(g) **[c-i-c commences]** [c-i-c].

DATED: [c-i-c]..

[c-i-c] **[c-i-c ends]**

“Attachment A

This attachment is entirely confidential”

STATEMENT OF

1 I am for Telstra Corporation Limited ("**Telstra**") and am authorised to make this statement on behalf of Telstra.

2 This statement is structured as follows:

(a) confidentiality;

(b) position and experience;

(c) the public switched telephone network;

(d) delivering voice services using Telstra's PSTN;

(e) Telstra's supply of ADSL broadband services;

(f) the unconditioned local loop service;

(g) the line sharing service;

(h) connecting ULLS and LSS;

(i) ULLS and LSS networks; and

(j) delivering voice services using ULLS and LSS.

(A) Confidentiality

3 The information in this statement is confidential to Telstra. I have prepared this statement on the basis that the information in it will be treated as confidential.

(B) Position and experience

4 I have obtained the following qualifications relevant to the role I perform at Telstra (which I describe below) and the evidence I give in this statement:

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(C) The Public Switched Telephone Network

15 Telstra's PSTN is a nation wide fixed line telecommunications network. The PSTN is used to provide voice telephony and data (for example, facsimiles and dial-up and broadband internet access) services. It is connected to, though separate from, Telstra's wireless networks which provide, for example, mobile telephony.

16 The PSTN consists of the Customer Access Network (known as the "**CAN**") and the Inter-Exchange Network (known as the "**IEN**"). The CAN is that part of the PSTN that connects a "customer" (also referred to as an "end-user") to an "exchange". The IEN is that part of the PSTN that connects exchanges together so that a call can be routed from a calling-party to a called-party where those parties are connected to different local exchanges. Telstra has approximately 5,116 exchanges located throughout Australia.

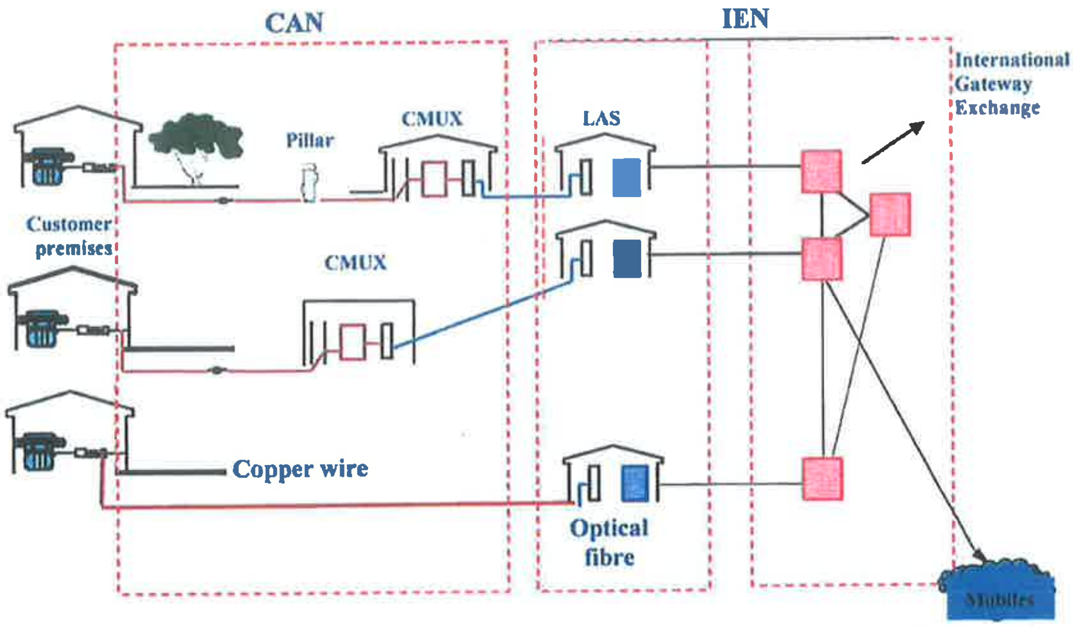
9

10

11

12

17 A simplified representation of the basic architecture of Telstra's PSTN is set out below.



18 In the above diagram, the acronyms used, which have not already been described in this statement, have the following meanings:

(a) Pillar - is a cross connection point which connects cables directly to the end-user with those to the exchange;

(b) CMUX - Customer Multiplexer, which enables a number of customers to be connected via an optical fibre transmission system to a LAS.

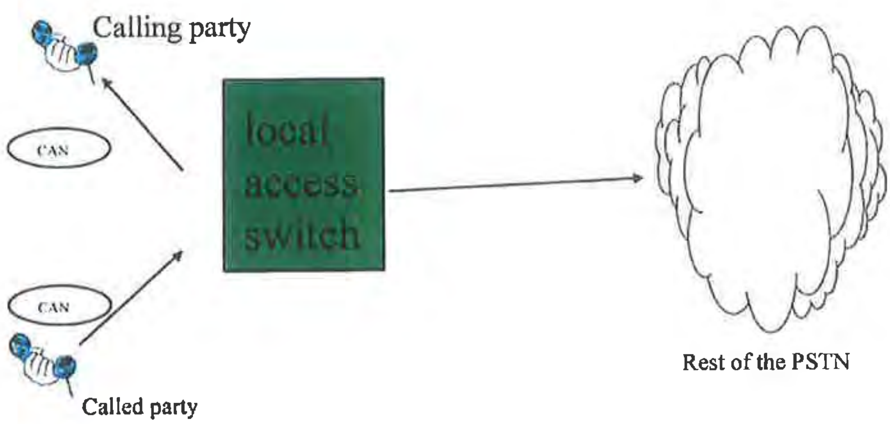
(c) LAS - Local Access Switch, which is a switch in the IEN which connects to end-users.

19 A PSTN is made up of switches connected by transmission systems. Telephone switches allow a call to be routed from one end-user's device to another. They do this by establishing a temporary connection between the end-users. Without telephone switches, an end-user would need a dedicated telephone line connecting to each person with whom he or she wanted to communicate.

(D) Delivering voice services using Telstra's PSTN

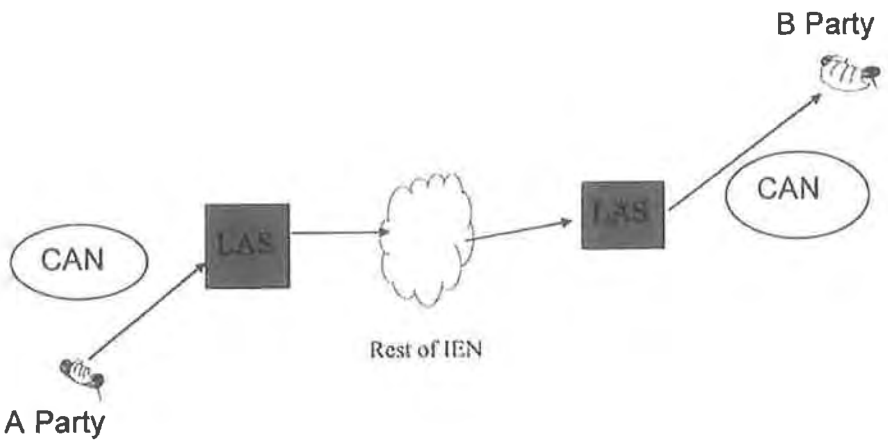
20 To make a telephone call from one person's handset (calling party) to another (called-party), the handsets must be linked by a network capable of transmitting the call.

21 The following is a description of a call between two customers connected to the same LAS. This call would be a local call. To make a call the calling party lifts the handset. The LAS responds by sending a dial tone through the CAN to the calling party. The calling party then dials the called party's number. If the line for the called party number is free, the LAS connects the two lines. This sends a ring tone to the calling party and a ring to the called party. This may be illustrated as follows:



22 The following is a description of a call between two customers connected to different LASs. In this case, the calling party picks up the handset and then dials the called party number. The LAS passes the called party number to the control equipment which then signals to the LAS to which the called party is connected. The control equipment of the called party switch checks to see if the called party is free. If the called party is free then the control equipment signals each of the switches to connect the two lines and sends a ring tone to the calling party and a ringing signal to the called party.

This may be illustrated as follows:



(E) Telstra's supply of ADSL broadband services

23 A broadband service enables an end user to send and receive digital information at high speed to another device, usually a computer.

24 Broadband services are delivered using both dedicated infrastructure and infrastructure that is also used to deliver PSTN voice services.

25 The technology used to transmit the information at high speed from the customer's home to the carrier's data network over the copper wires is called Digital Subscriber Line Technology ("**DSL**"), the most common example of which is Asymmetric Digital Subscriber Line ("**ADSL**").

26 The messages in the form of packets are sent from the customer's computer to the ADSL modem. This modem enables the high speed transmission of data over the customer's copper line, the same copper line that is also used for the telephony service. It does this by using higher frequencies than those used for the voice service. This is analogous to different radio stations transmitting simultaneously, one in the AM band (medium frequency) and one in the FM band (very high frequency).

27 The copper line is connected to a **CMUX**. The CMUX incorporates equipment that separate the analogue voice from the digital ADSL transmissions and multiplexes the packets of information from all customers and converts them into light pulses which are transmitted over optional fibre. The ADSL electronics in the CMUX are known in the industry as the Digital Subscriber Line Multiplexer (**"DSLAM").** Multiplexing is a

process where multiple streams of data are combined into a single higher speed stream in order to increase efficiency.

28 The DSLAM is then connected using optical transmission technology to the carrier's data network (separate from the PSTN) which controls how the packets are sent to the customers' Internet service provider ("**ISP**"). Routers in the ISP's data centre then determine where the packets should be sent next. The ISP will have connections to other carriers' networks that form part of the public Internet.

(F) The Unconditioned Local Loop Service

29 In August 1999, the Australian Competition and Consumer Commission ("**Commission**") declared the ULLS for the purposes of Part XIC of the TPA. The ULLS gives access seekers control over a line, between the local access switch and the end-users premises, allowing the access seeker to supply both voice services and data services, including ADSL, to end-users.

30 The Commission's declaration describes the ULLS as follows:

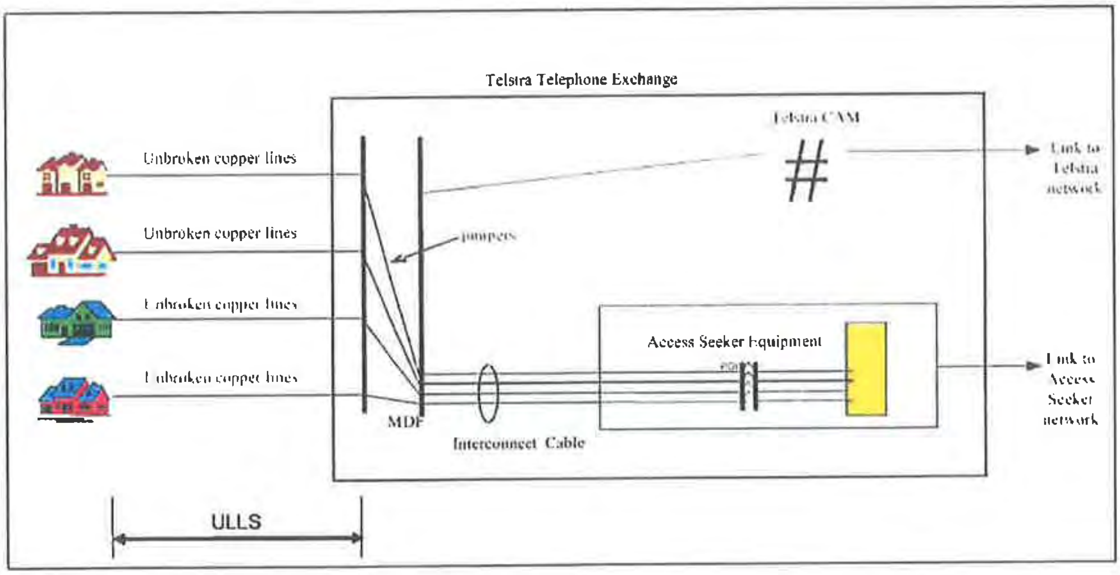
> *"The Unconditioned Local Loop Service is the use of unconditioned communications wire between the boundary of a telecommunications network at an end-user's premises and a point on a telecommunications network that is a potential point of interconnection located at or associated with a customer access module and located on the end-user side of the customer access module."*

31 In May 2000, the Commission made some minor variations to the original service description for the ULLS. Those variations concerned the definitions of "communications wire" (varied so as to include aluminium based wire) and "customer access module" (varied so as to refer to a "Local Switch" rather than a "Local Access Switch").

32 In July 2006, the Commission again declared the ULLS. The service description for the ULLS contained in that further declaration is in like terms to that in the original declaration and set out in paragraph 31 above. When I refer to the "ULLS" in this statement, I refer without distinction to the ULLS as originally declared, as later varied and as further declared.

33 The "unconditioned communications wire", which forms the subject of the ULLS (and the LSS, which I discuss further below), is part of the CAN. That wire is a continuous copper (or, under the variation, aluminium) based pair of wires (a "**Local Loop**") running from a customer's premises to a "customer access module" ("**CAM**"), which is generally located in an exchange. The service is described as being "unconditioned" because it concerns access to the raw copper (or, under the variation, aluminium) wires that form a Local Loop. By way of contrast, a pair of wires is classified as being "conditioned" if there is equipment at some point along an individual loop that changes its electrical characteristics to enable it to provide a telecommunications service. This may be necessary, for example, if a given loop travels a long distance from an exchange to a customer's premises. In this circumstance, without the addition of the equipment to enhance the loop's characteristics, it may not be able to properly carry telecommunications services delivered from the exchange. The majority of loops do not require conditioning in order to supply telecommunications services from the exchange.

34 The diagram below depicts the ULLS. In the diagram each of the "unbroken copper lines" running from a customer's premises represents an individual ULLS line (or a Local Loop). Under the variations to the ULLS service description discussed at paragraph 31 above, the lines could have been aluminium rather than copper.



35 The nature of the ULLS was described by the Commission in its report titled *Declaration of Local Telecommunications Services* (July 1999) which accompanied the original declaration of the ULLS as follows (at pages 14-15):

"The service description is intended to cover the situation in which an end-user chooses to churn from one service provider (e.g. Telstra) to another service provider for services provided over the line. In such a situation the access seeker would acquire use of the line. It is also intended to cover the situation where a line has been deployed but is not currently being used to supply services to end-users.

With this service there is no prescribed bandwidth. This is because the access seeker is receiving the use of the twisted copper pair without conditioning or specific carriage technology. This enables the access seeker to add its own carriage technology in order to supply, for example, high speed data carriage services to end-users or alternatively multiple telephony services to medium and large corporates (supplying up to 30 voice channels on a single copper pair) or a combination of voice and data services."

36 I consider that the above passage accurately describes the nature of the ULLS.

(G) The Line Sharing Service

37 In October 2002, the Commission declared the LSS for the purposes of Part XIC of the TPA. The LSS gives access seekers control over the high frequency part of a line between the local access switch and the end-users premises, allowing the access seeker to supply various services to end-users, including data services such as ADSL, and voice services, using Voice Over Internet Protocol technology (described in further detail below).

38 The LSS declaration sets out the service description for the LSS as follows:

"The High Frequency Unconditioned Local Loop Service is the use of the non-voiceband frequency spectrum of unconditioned communications wire (over which wire an underlying voiceband PSTN service is operating) between the boundary of a telecommunications network at an end-user's premises and a point on a telecommunications network that is a potential point of interconnection located at, or associated with, a customer access module and located on the end-user side of the customer access module."

39 The LSS is very similar to the ULLS except that with the LSS, a Local Loop is used by two carriers or service providers. As I explain below, with the ULLS the access seeker obtains the exclusive use of the given Local Loop. However, with the LSS the use of a Local Loop is shared between the access seeker who obtains the exclusive use of the high frequency (or "non-voiceband") portion of the given Local Loop and another carrier (typically Telstra) who uses the low frequency (or "voiceband") portion of that loop. In other words, when an access seeker obtains access to the LSS, while it gains the direct connection to, and use of, the given Local Loop, the use of that loop is shared so that one carrier or service provider can provide voice services over the low frequency portion of the loop while another carrier (the access seeker) can provide high-speed data services over the high frequency portion of the line at the same time.

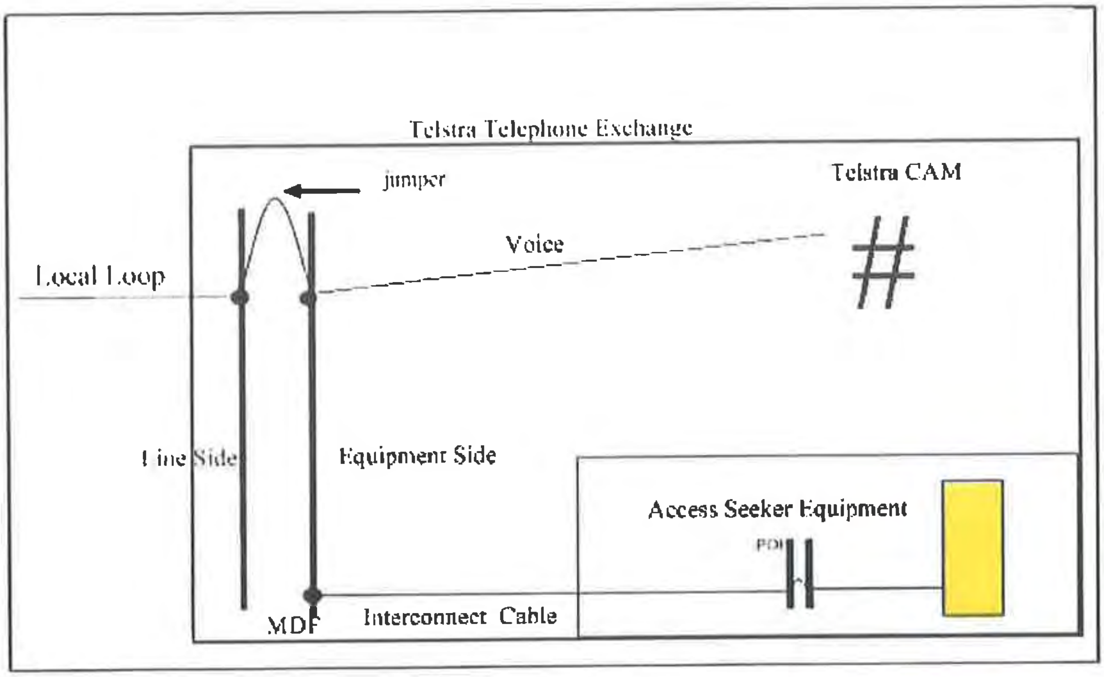
40 The high frequency portion of a Local Loop can be used to deliver high-speed telecommunications services such as broadband Internet access and also certain digital voice telephony services (such voice services are sometimes referred to as "voice over internet protocol" (or "**VOIP**") or "voice over digital subscriber line" (or "**VODSL**")). The low frequency portion of a Local Loop can be used to deliver regular analogue voice telephony services.

(H) Connecting ULLS and LSS

41 The connection process for supplying ULLS to access seekers is commonly referred to as a "cutover". By this process, the Local Loop is physically cutover to the access seeker's network. It is physically disconnected from Telstra's PSTN and connected to the access seeker's network.

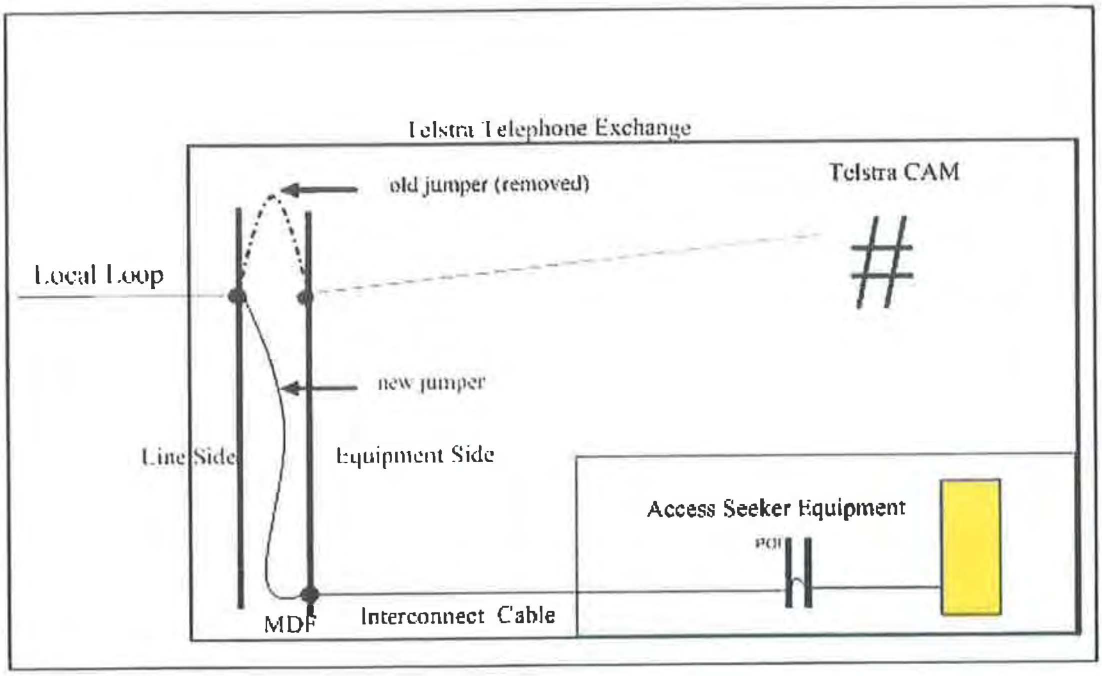
42 At the exchange, a Local Loop physically terminates at the "Main Distribution Frame" ("**MDF**"). The MDF consists of "line side" and "equipment side" termination blocks. Local Loops running into the exchange from "the street" (ultimately from customers' premises) terminate on the line side termination block and network equipment is connected to (or terminates on) the equipment side termination block. Linking the termination blocks on either side of the MDF is a pair of wires for each Local Loop referred to as a "jumper". In other words, for each Local Loop a jumper runs between the two sides of the MDF so that a Local Loop is connected to network equipment and, as a result, to the broader telecommunications network.

43 This is illustrated in the diagram below.



44 In order to supply the ULLS, the existing jumper connecting the Local Loop to Telstra's network (as illustrated above) must be disconnected. A new jumper is then installed connecting the Local Loop at the line side of the MDF to the access seeker's "interconnect cable" on the equipment side of the MDF. That interconnect cable runs to the access seeker's equipment which is typically located in space used by the access seeker in Telstra's exchange. This is what is referred to as "jumpering".

45 The jumpering process is illustrated in the diagram below. In that diagram the dotted line represents the old jumper that connected the Local Loop to Telstra's network prior to it being removed in order to give the access seeker access to the Local Loop.



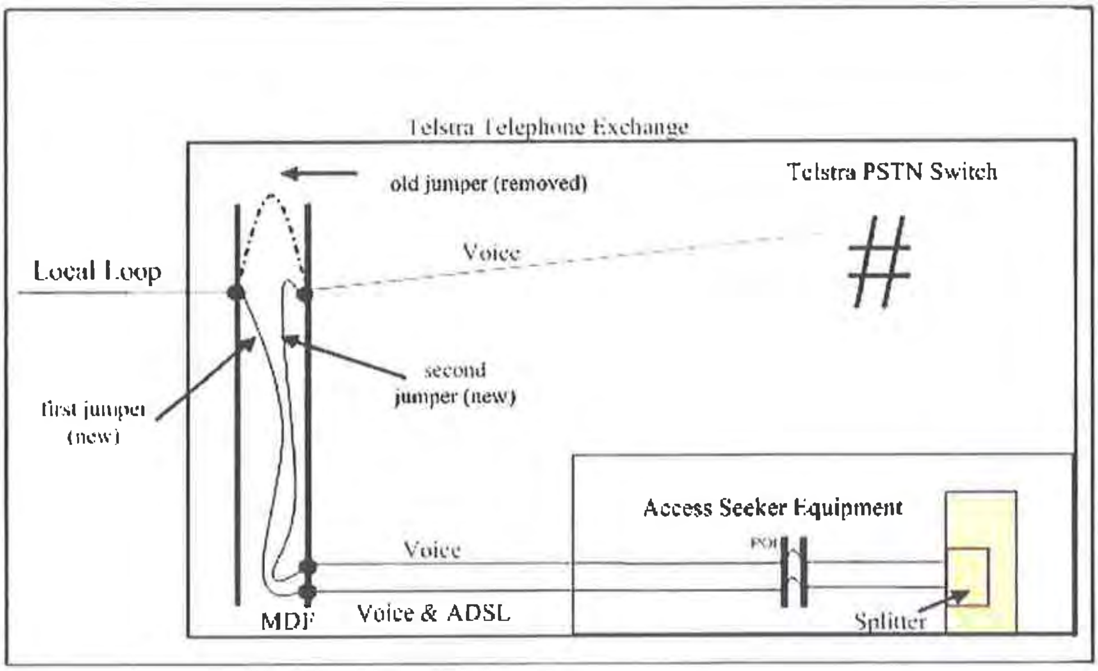
46 Once the Local Loop has been "jumpered" to the access seeker's network, Telstra loses complete "visibility" and control of that loop. Physically, the Local Loop is no longer connected to Telstra's broader PSTN as it is severed from the IEN.

47 The connection process for the LSS involves a similar "jumpering" process to that in respect of the ULLS, save that two new jumpers are required for the LSS rather than one. Also, unlike with the ULLS, in order to supply the LSS it is necessary to use a device known as a "splitter". This is a device that splits a Local Loop into two independent channels; one channel for the voiceband frequency and another channel for the non-voiceband frequency. I describe the way in which a splitter is used in the connection process in the next paragraph of my affidavit. A splitter can also be used by an access seeker that acquires ULLS, but the splitter is not necessary unless the access seeker provides POTS voice services in addition to data services. POTS is an acronym for the "plain old telephone service" (which is, in turn, equivalent to the standard telephone service supplied by Telstra).

48 Just as with the ULLS, the existing jumper that physically connects the Local Loop to Telstra's network (as illustrated in the diagram accompanying paragraph 43 above) must be disconnected. A new jumper (**"first jumper"**) is then installed connecting the Local Loop to the access seeker's interconnect cable on the equipment side of the MDF (as with the ULLS). That interconnect cable runs to the access seeker's equipment (which is

typically located in Telstra's exchange) which incorporates a splitter which separates the voice and non-voiceband channels of the loop. The non-voiceband channel continues through the access seeker's equipment and then to the access seeker's network. The voiceband channel is sent back from the access seeker's equipment to the equipment side of the MDF across a separate interconnect cable. A further jumper ("**second jumper**") is connected to this interconnect cable which leads to Telstra's equipment and back into the PSTN.

49 The jumpering process in respect of the LSS is illustrated in the diagram below.



50 Regular voiceband telecommunications services are carried, typically, from Telstra's equipment, through the second jumper and the access seeker's equipment and back through the first jumper ultimately to the end user's premises. The access seeker supplies high bandwidth telecommunications services directly from its own equipment, through the first jumper, and ultimately to the end user's premises.

51 Once the Local Loop has been "jumpered" to the access seeker's network, Telstra is no longer able to supply any services over that loop using the non-voiceband channel.

(I) ULLS and LSS access seeker's equipment and networks

ULLS and LSS Networks

52 In the above description of ULLS and LSS, I refer to the access seeker equipment. That access seeker equipment is generally located in Telstra's telephone exchanges in an area referred to as the Telstra Exchange Building Access ("**TEBA**") space. The TEBA space is leased by the access seeker from Telstra. In that space, the access seeker installs a cabinet which contains a DSLAM. The DSLAM is connected to transmission cables which forms part of the access seeker's network and which is in turn connected to a number of other pieces of equipment making up that network. The DSLAM is also connected to mains power and battery power.

53 Set out below is a photograph of a TEBA space in which access seekers have installed cabinets housing DSLAMs. The large cables protruding from the top of the cabinets are the interconnect cables.

54 A DSLAM may be configured in different ways to provide different functionality. For example, in addition to network termination cards:

(a) a DSLAM connected to ULLS which is configured to provide ADSL services would contain shelves with ADSL cards;

(b) a DSLAM connected to ULLS which is configured to provide voice and data services would contain shelves with ADSL cards, a splitter and voice cards;

(c) a DSLAM connected to LSS would contain a shelf with ADSL cards and a splitter;

(d) a DSLAM configured to supply either LSS or ULLS would contain shelves with a splitter and combination cards (capable of being used for a variety of services) or a splitter and some voice cards and some ADSL cards.

55 The number of cards required in a DSLAM is determined by the number of services being supplied using that DSLAM. Each card contains a specific number of ports. Each port in turn services one copper pair or Local Loop. The number of ports serviced by a card will vary and is generally increasing as card technology improves.

Use of DSLAMs

56 If an access seeker decides to move its customers from a LSS to a ULLS, it may need to change in the configuration of the DSLAM from one of the configurations listed in paragraph 55 of my statement to another configurement listed in that paragraph. Whether changes need to be made will depend on whether the DSLAM has already been configured to provide voice services. For example a DSLAM containing a combination card or both ADSL and voice cards may be used to provide both ULLS and LSS.

57 DSLAMs may also be relocated and/or resold. The DSLAM shelf, voice and ADSL cards can be reinstalled in another exchange. The cables connecting the DSLAM to the Telstra equipment would have to be purchased afresh as they are pre-cut to the appropriate length. However, the cost of these cables is a negligible component of the overall DSLAM cost).

58 DSLAMs have a relatively short lifespan by reason that the technology used in them is evolving rapidly. For example, as I referred to in paragraph 56 above, the number of ports per card has increased over time. Additionally, the nature of the service supplied using a DSLAM has changed significantly. Each change requires an upgrade to all or part of a DSLAM. For example, ADSL technology has changed over time and is enabling faster speeds of data transmission (from ADSL1 to ADSL2+). Each change in technology

requires a change in equipment in the DSLAM. Therefore, the lifespan of a DSLAM is determined mostly by changes in technology and capacity rather than wear and tear.

Determining the scope of ULLS or LSS networks

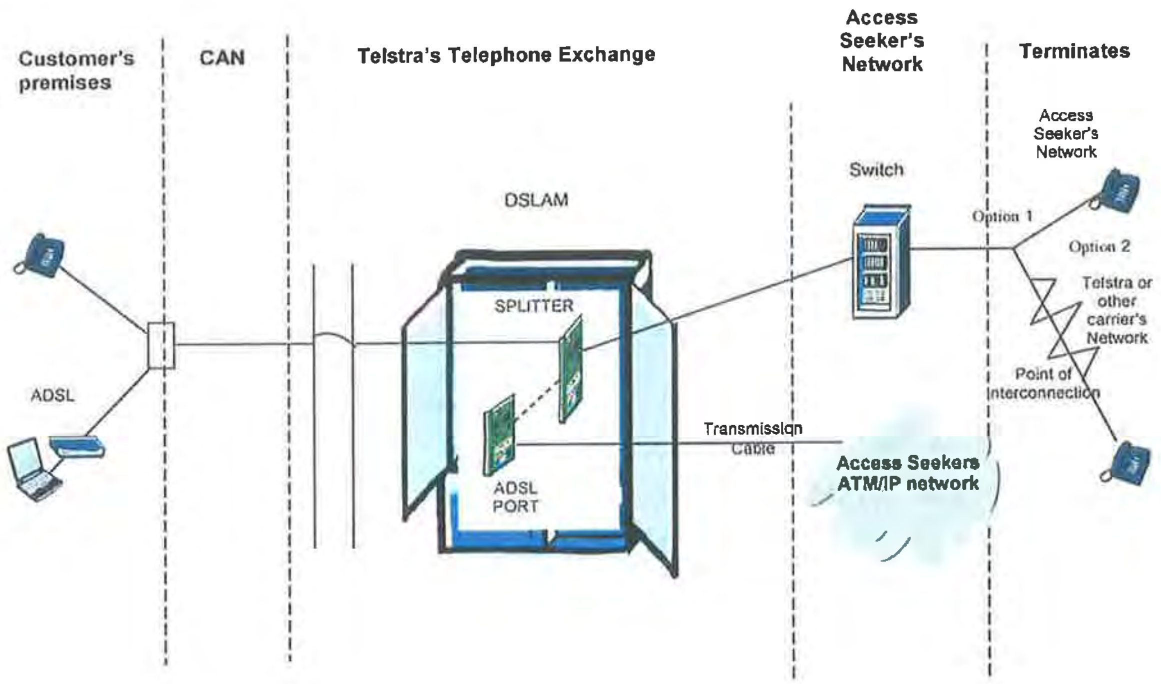
59 One way of ascertaining whether a particular exchange service area ("**ESA**") contains a DSLAM which is owned or operated by a carriage service provider other than Telstra is by determining, based on publicly available information such as websites of alternative providers, whether ADSL2+ services are supplied in that ESA. If ADSL2+ services are supplied in a particular ESA by a carriage service provider other than Telstra, this indicates that a non-Telstra DSLAM is operating in that ESA as Telstra does not currently resupply its ADSL2+ services to wholesale customers anywhere in Australia.

(J) Delivering voice services using LSS and ULLS

60 At present, a telecommunications service provider wishing to provide a standard telephone service ("**STS**") quality voice service using a ULLS or LSS network can adopt one of three technology choices. An acquirer of ULLS or LSS may supply voice services on the line using standard switching technology (ULLS only), POTS emulation (ULLS only) or VOIP (ULLS or LSS). I consider each of these technologies in turn below.

Standard switching technology

61 A ULLS access seeker can deliver voice calls from a customer connected to its ULLS network using standard switching technology. To do this, the access seeker would need to have switching equipment. The diagram below sets out the path of the voice call using standard switching technology.



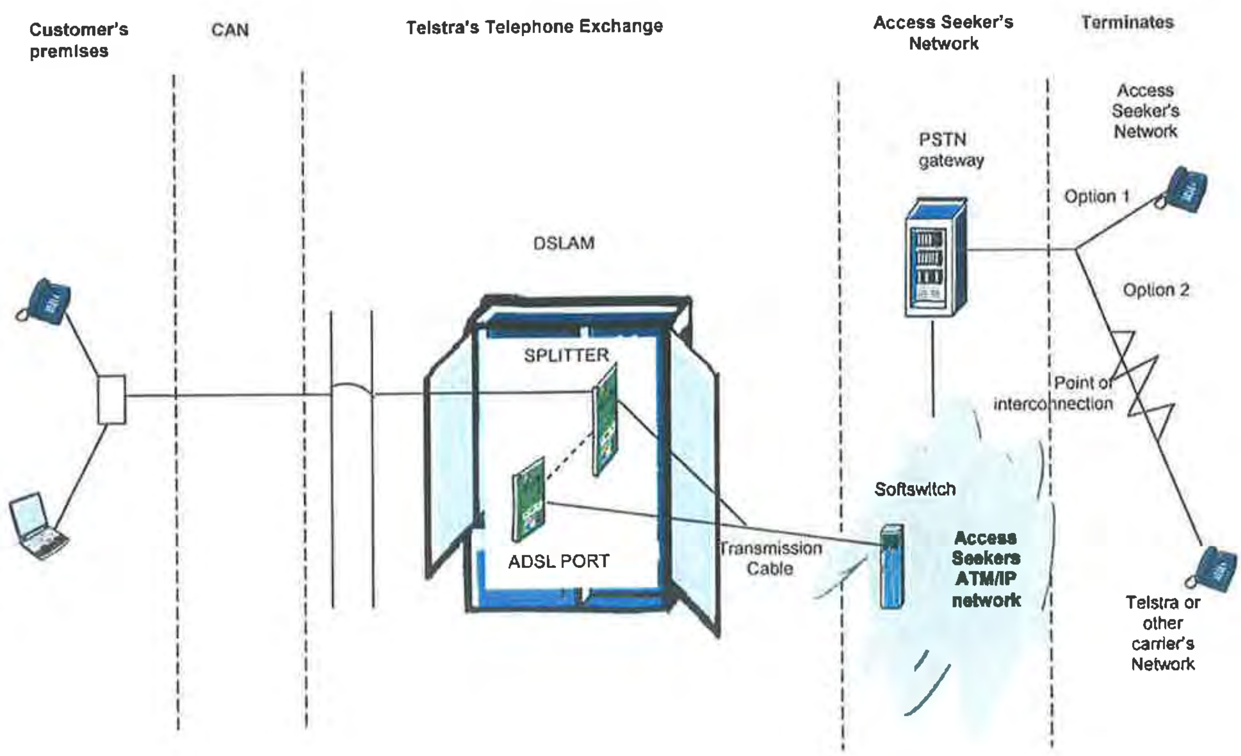
62 The call starts at the calling party's telephony device and passes through the CAN, travelling down the copper (or aluminium) pair connected to the access seeker's DSLAM. That copper (or aluminium) pair is the ULLS acquired from Telstra. The voice call travels on the low frequency part of the pair. Once the line enters the DSLAM it is sent through a splitter which "splits" the low frequency (voice) and high frequency (data) section of the line.

63 The call then passes through the access seeker's transmission cable and is sent to the access seeker's switch which may be located in one of Telstra's telephone exchanges or in other premises. The switch then directs the call to its destination (the called party) which may be to an end-user connected to the access seeker's network or an end-user connected to Telstra or another carrier's network. If the call is to be terminated on Telstra or another carrier's network it passes a point of interconnection between the access seeker's network and the other carrier's network before terminating at the called party's telephony device.

64 The voice service supplied by the access seeker using its ULLS network and standard switching technology is similar to the standard telephone service supplied by Telstra to its customers.

POTS emulation

65 A voice call may also be delivered by a ULLS access seeker using technology which emulates standard switching using "soft switches". I refer to this technology in this statement as "POTS emulation". POTS emulation is a method providing a telephony service which is very similar to the one I refer to in paragraphs 63 to 64 above (even to the extent of using a traditional telephone handset), but one which uses an Internet Protocol network to a greater extent. The diagram below sets out the path of the voice call using POTS emulation.



66 The call starts at the calling party's telephony device and passes through the CAN, travelling down the copper pair connected to the access seeker's DSLAM sited in the same way described above in connection with standard switching. The line used to transport the call from the customer's premises to the DSLAM is controlled by the access seeker using the ULLS acquired from Telstra. The voice call travels on the low frequency part of the line. Similarly, once the line enters the DSLAM it is sent through a splitter which "splits" the low frequency (voice) and high frequency (data) section of the line.

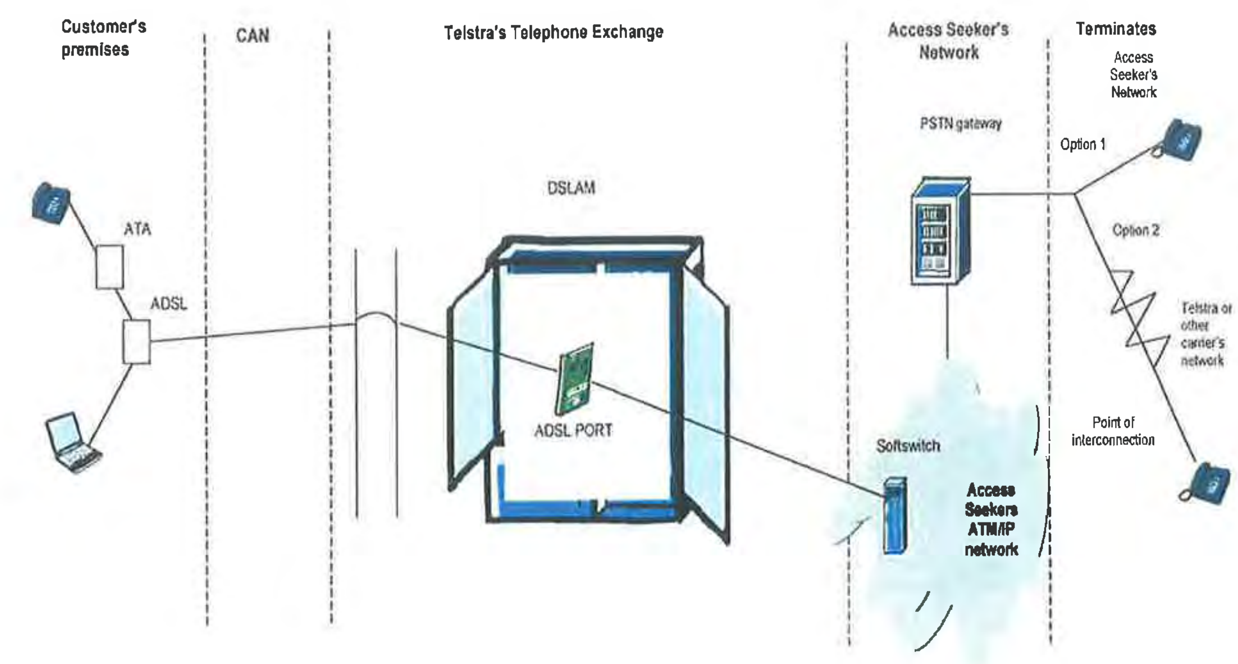
67 The voice then passes through the access seeker's transmission cable and is sent to a device known as a "soft switch". A soft switch routes the call to its destination. The soft switch is essentially a computer which reads information about the routing of a call and then uses that information to direct the call to its destination. From the soft switch the call information is transmitted using packet switching and is directed to a point on the access seeker's network closer to its destination. Once the packets of information reach a point on the access seeker's network which is the closest point on that network to its destination, the packets of information pass through a PSTN gateway and are converted to a standard voice signal.

68 The voice signal is then directed to the called party which may be an end-user connected to the access seeker's network or an end-user connected to Telstra or another carrier's network. If the call is to be terminated on Telstra or another carrier's network it passes a point of interconnection between the access seeker's network and the other carrier's network before terminating at the called parties' telephony device.

69 The voice service supplied by the access seeker using its ULLS network and POTS emulation is the same, from an end-user's perspective, as a voice service supplied using standard switching. The quality of the voice service is equivalent to that provided using standard switching.

Voice Over Internet Protocol

70 A voice call may also be delivered by either a ULLS or LSS access seeker using VOIP. The diagram below sets out the path of the voice call using VOIP.



71 To provide this service the analogue voice signal is converted into data at the end-user's premises and then is carried through most, if not all, of the network/s it passes as packets of data rather than as an analogue voice signal. To be able to use VOIP, the calling party must have a device which converts the signal from a voice signal to packets of data. This device is referred to as an analogue telephone adapter ("**ATA**"). Once the voice signal has been converted into packets of data it travels along the CAN and into the DSLAM in the same manner as data in the supply of an ADSL service. The packets of data enter the DSLAM through an ADSL port. They are then conveyed to a soft switch. From the soft switch to the call party's telephone, the call is carried in the same manner as I have described above (at paragraph 67 - 68) for POTS emulation.

72 The carriage of a call by POTS emulation or VOIP does not necessarily result in an inferior quality service to an end-user as compared with a call which is carried using standard switching. In circumstances where an Internet Protocol path is congested, the packets of information carrying the voice call may be afforded priority over the packets of information carrying other data, with the result that the quality of the call will not be deteriorated by any congestion on the network and will therefore be equivalent to that of a call carried by a traditional switching technology.

DATED:

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ATTACHMENT C

COMMUNICATIONS
ALLIANCE LTD

INDUSTRY GUIDELINE
G634:2007
Quality of Service parameters for Voice over Internet Protocol (VoIP) services

G634:2007 Quality of Service parameters for Voice over Internet Protocol (VoIP) services
Industry Guideline
First published as G634:2007

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EXPLANATORY STATEMENT

This is the Explanatory Statement for the G634:2007 ***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*, available from http://support.nortel.com/go/main.jsp?cscat=DOCDETAIL&id=292677&poid=12483

Gary Marshall
Chairman
Voice over IP Quality of Service Working Committee

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1 GENERAL

1.1 Introduction

1.1.1 The development of the Guideline has been facilitated by the 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 G632:2007 ***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, along with associated technologies and voice quality tools will emerge over the next few years.

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).

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 (Note)
ADPCM	Adaptive Differential Pulse Code Modulation
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

NOTE: ACIF and SPAN merged in September 2006 to form Communications Alliance.

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.

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 outined 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. A wideband voice service is capable of providing R-values greater than 100 – refer to ITU-T G.107 Amdt 1 and ITU-T G.109 section 9.*

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

> *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 Experiece (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 (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/ACIF S003 for the Australian loss plan. Customer Equipment Standards such as AS/ACIF S004, AS/ACIF S002 and AS/ACIF 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, 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 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;

(b) a managed core IP network 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. 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%	Not generally available	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 high quality 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 a higher R value than the conditions in Note 1. At present the use of wideband codecs is not widespread.

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, 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 minimize 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.

3.2.8 Delays can occur due to:

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

(b) codec processing delay at both ends;

(c) routers (0.5-5 ms 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

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 minimize the use of Trombone Connections to distant points.

4.5 VoIP Packet Type and Priority

4.5.1 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/ACIF S002 in order to achieve the objectives for echo cancellation and sidetone.

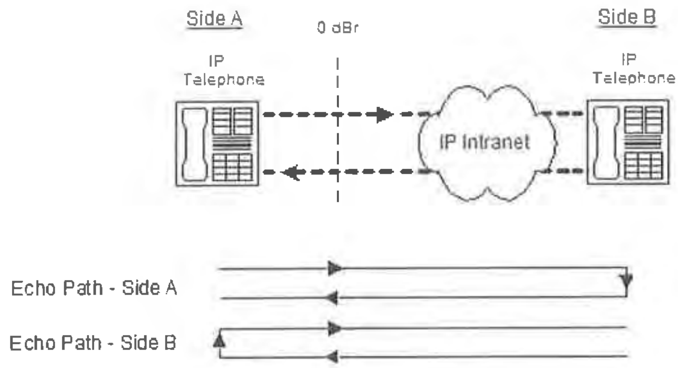
4.7.2 In general, VoIP networks have longer delays compared with traditional PSTN networks. With increasing delay any level of echo becomes increasingly audible.

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.

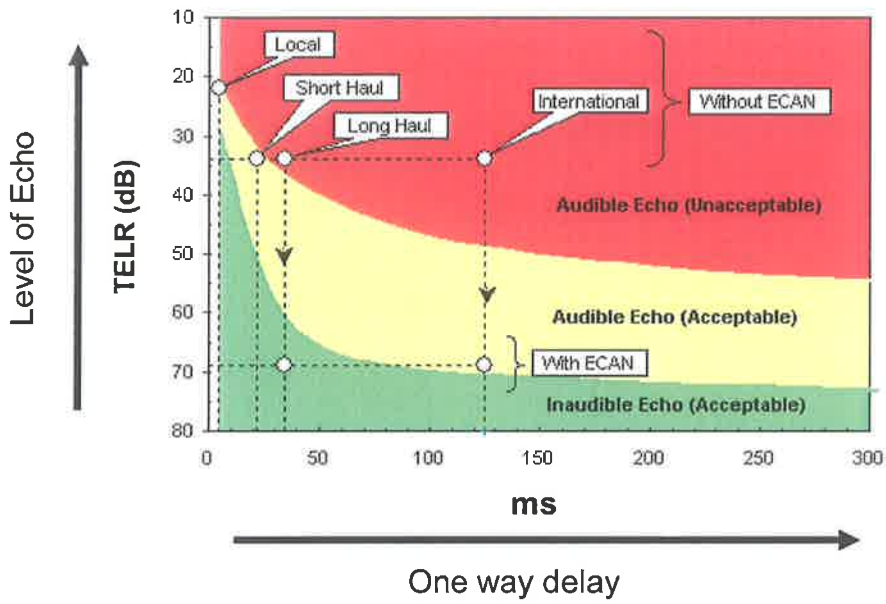


FIGURE 2

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

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

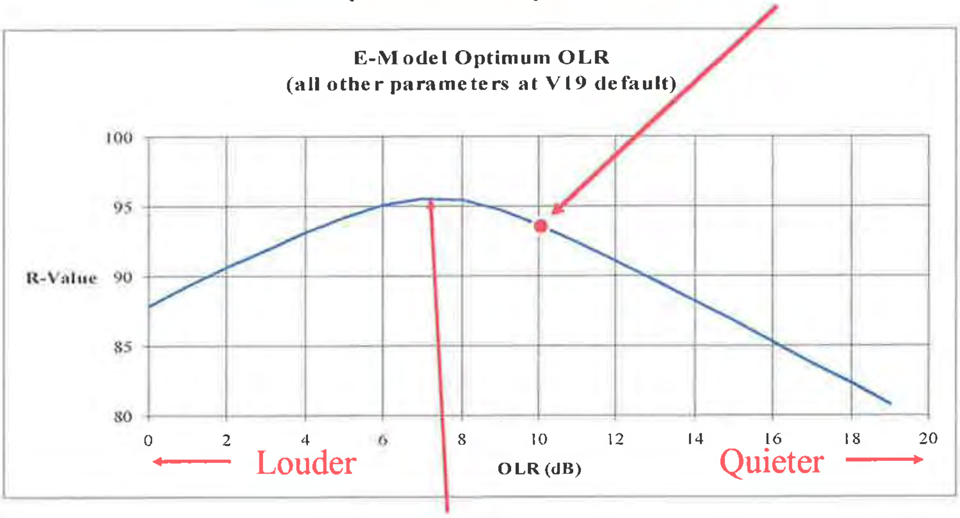
4.7.9.3 The VoIP-TDM gateway needs to consider echoes that are not cancelled by the ECAN in the TDM network. These will be calls that have delays up to the point where the TDM network puts cancellers on the trunks. As long as ECANs are at the correct point in the VoIP network, providers of VoIP Services 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.

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

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

ITU-T G.107 specifies the optimum at OLR = 10 dB



The "peak/optimum" OLR is at 7 dB, but this level could cause echo.

OLR = SLR + Electrical Loss + RLR = 8 + 0 + 2 = 10 dB

FIGURE 3
Optimum Overall Loudness Rating

4.7.11 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 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) parse the incoming signal into frames before encoding it. Frame-based codecs commonly use Code Excited Linear Prediction (CELP) compression.*

(d) 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.*

(e) 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.*

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

(g) 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.

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

(i) 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
Australian/ACIF Standards	
AS/ACIF S002:2005	Analogue interworking and non-interference requirements for Customer Equipment for connection to the PSTN
AS/ACIF S003:2006	Customer Access Equipment for connection to a Telecommunications Network
AS/ACIF S004:2006	Voice frequency performance requirements for Customer Equipment
IETF RFCs	
RFC 791	Internet Protocol
RFC 3264	An Offer/Answer Model with the Session Description Protocol (SDP)
ITU-T Recommendations	
E.800 (08/94)	Terms and definitions related to quality of service and network performance including dependability
G.107 (03/05)	The E-model, a computational model for use in transmission planning
G.107 Amdt 1 (06/06)	
G.108 (09/99)	Application of the E-model: A planning guide
G.108.1 (05/00)	Guidance for assessing conversational speech transmission quality effects not covered by the E-Model
G.108.2 (03/07)	Transmission planning aspects of echo cancellers
G.109 (09/99)	Definition of categories of speech transmission quality
G.113 (02/01)	Transmission impairments due to speech processing
G.113 Amdt 2 (01/07)	Transmission impairments due to speech processing Amendment 2: Revised Appendix I – Provisional planning values for the equipment impairment factor I_e and packet-loss robustness factor B_{pl}
G.114 (05/03)	One-way transmission time
G.121 (03/93)	Loudness ratings (LRs) of national systems
G.131 (11/03)	Talker echo and its control
G.711 (11/88)	Pulse code modulation (PCM) of voice frequencies

G.722 (11/88)	7 kHz audio-coding within 64 kbit/s
G.722.1 (05/05)	Low-complexity coding at 24 and 32 kbit/s for hands-free operation in systems with low frame loss
G.722.2 (07/03)	Wideband coding of speech at around 16 kbit/s using Adaptive Multi-Rate Wideband (AMR-WB)
G.723.1 (05/06)	Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s
G.726 (12/90)	40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)
G.729 (01/07)	Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)
P.10 (12/98)	Vocabulary of terms on telephone transmission quality and telephone sets[superseded by G.100/P.10 07/06 (prepublished)
P.800 (08/96)	Methods for subjective determination of transmission quality
P.862 (02/01)	Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs
Y.1541 (05/02)	Network performance objectives for IP-based services
Industry Guidelines	
G632:2007	Quality of Service parameters for networks using the Internet Protocol Guideline
G635:2007	Testing Arrangements for Quality of Service parameters for Voice over Internet Protocol (VoIP) services
Industry Code	
ACIF C519:2004	End-To-End Network Performance
TIA Publications	
TSB-116-A	Voice Quality Recommendations for IP Telephony, TIA Telecommunications Industry Association Standards and Engineering Publications

URLs for some of the above references are:

http://www.3gpp.org/ftp/Specs/html-info/0660.htm

http://www.ietf.org/rfc.html

http://www.itu.int/ITU-T/publications/recs.html

http://commsalliance.com.au/documents

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 can be worse 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.

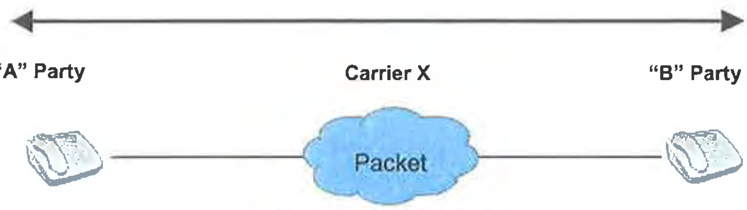


FIGURE 4
IP in Core and Access networks

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.

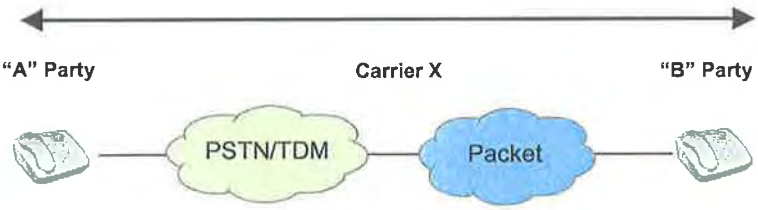


FIGURE 5
TDM Access and Core, IP in access network

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.

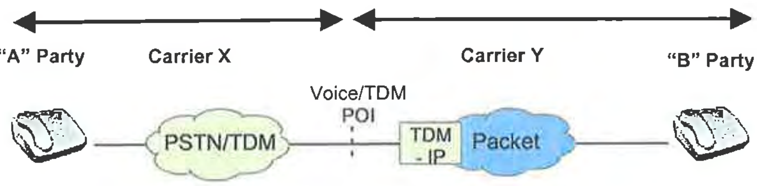


FIGURE 6
TDM access and core, IP core and access

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.

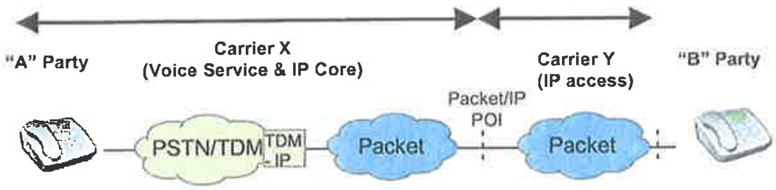


FIGURE 7
TDM Access & IP Core, IP Access

A3 Three Carriers

A.3.1 IP Access, TDM Core, IP Access

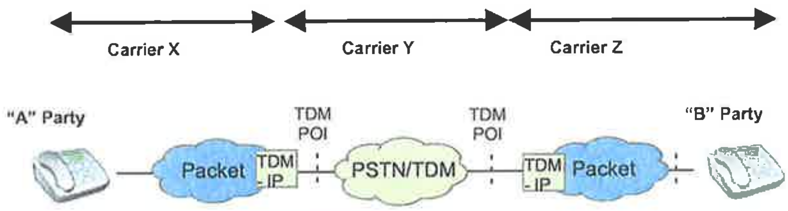


FIGURE 8
IP access, TDM Core, IP Access

In this scenario packet access is used by both the A party and B party, with TDM interconnection (see Figure 8). 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.

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).

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
$90 \leq R < 100$	Best	Very satisfied
$80 \leq R < 90$	High	Satisfied
$70 \leq R < 80$	Medium	Some users dissatisfied
$60 \leq R < 70$	Low	Many users dissatisfied
$50 \leq R < 60$	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 9 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.

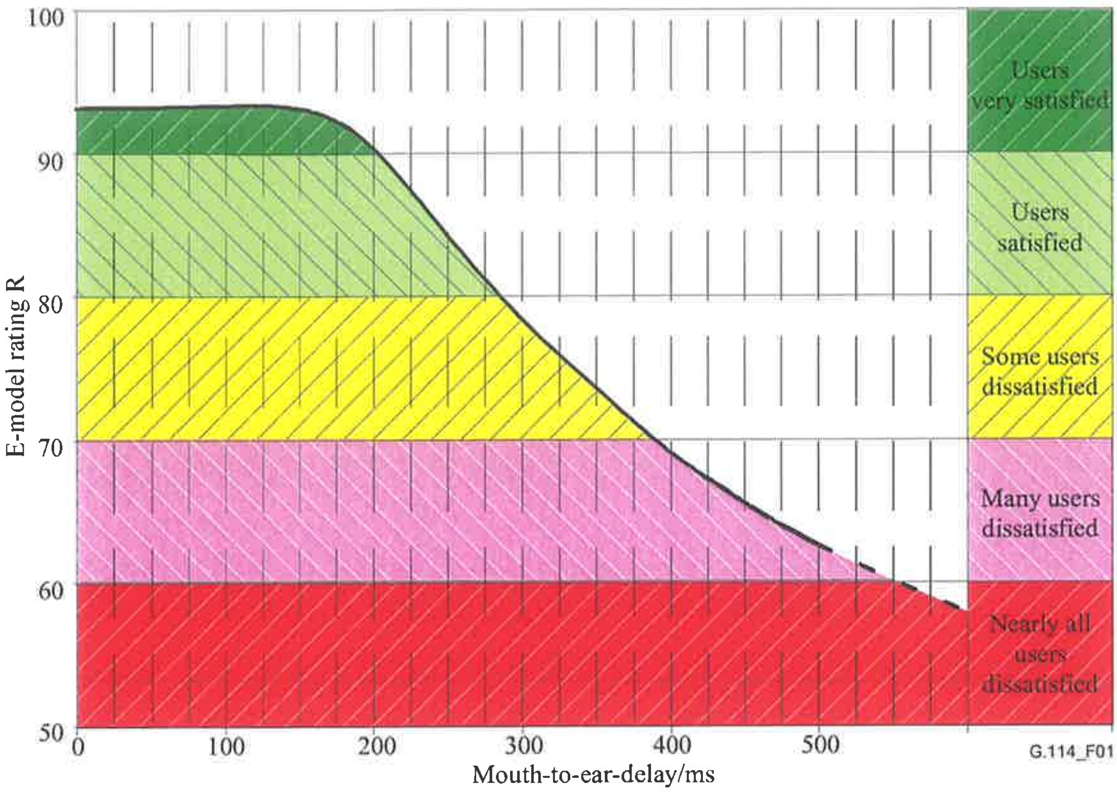


FIGURE 9

Impact of mouth to ear delay on R value

NOTES:

1. The curve in Figure 9 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 an 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.

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).

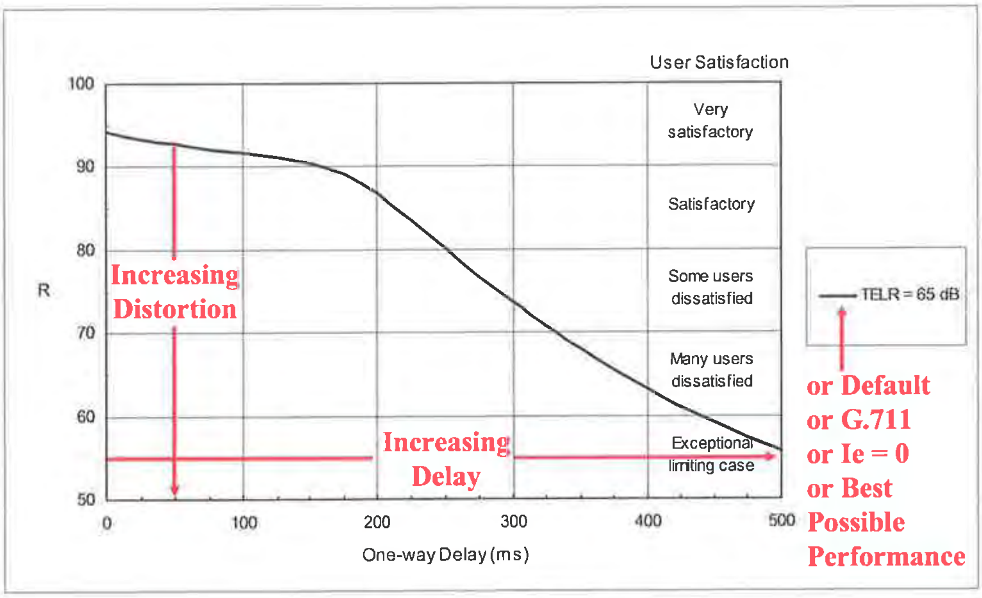


FIGURE 10

ITU-T G.107 Default Delay Impairment

C.5.2 Effect of Increasing Echo

Figure 11 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.*

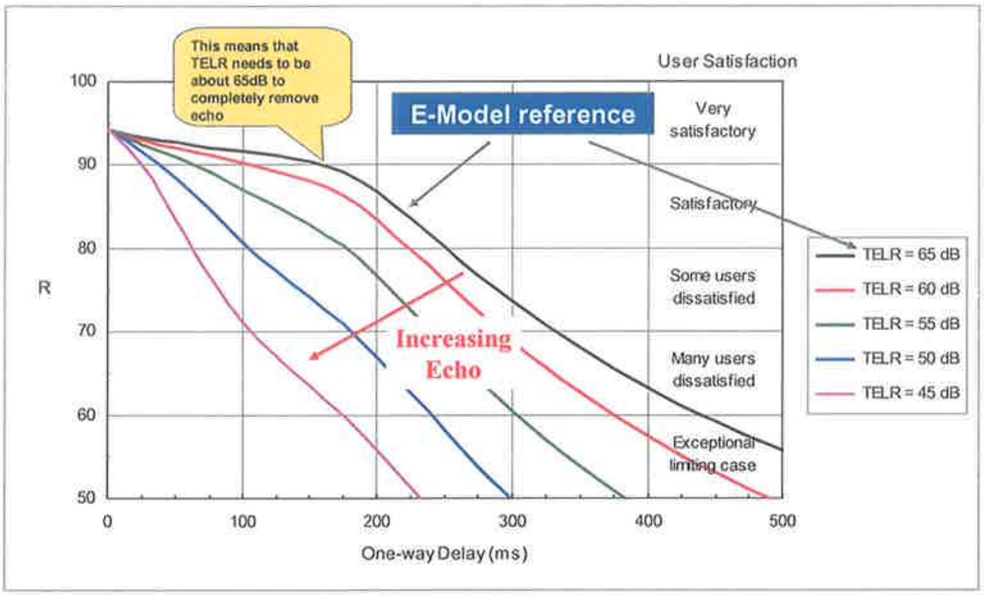


FIGURE 11
E-Model, Echo Impairment

C.5.3 Effect of codec change or transcoding

Figure 12 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.

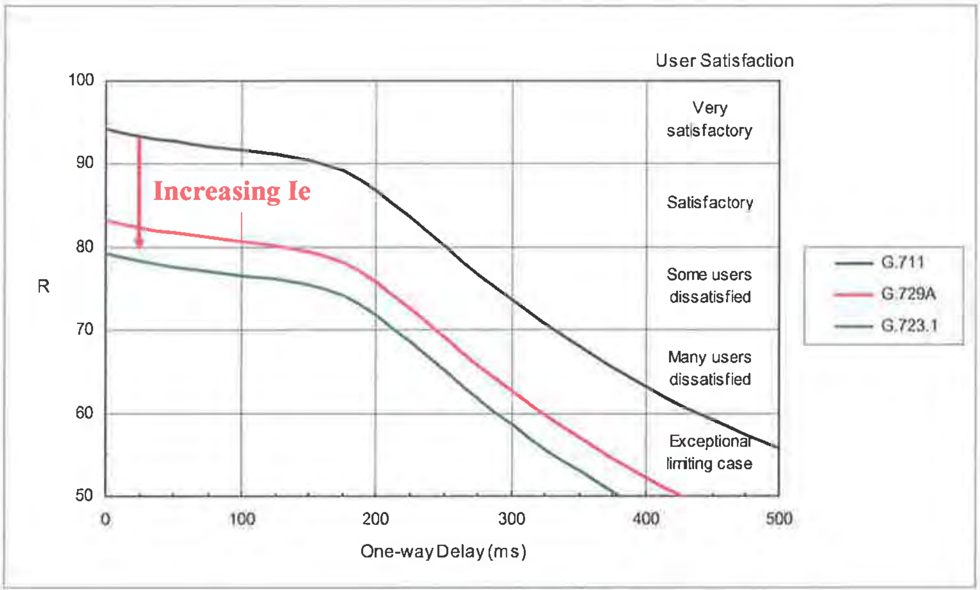


FIGURE 12

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 13 shows an example for the G.729a codec.

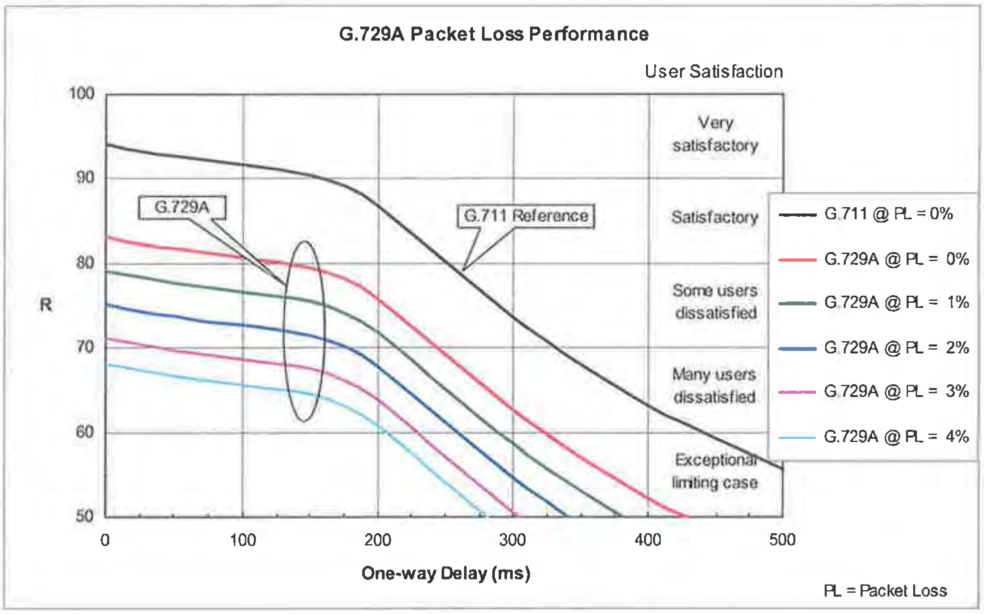


FIGURE 13

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.

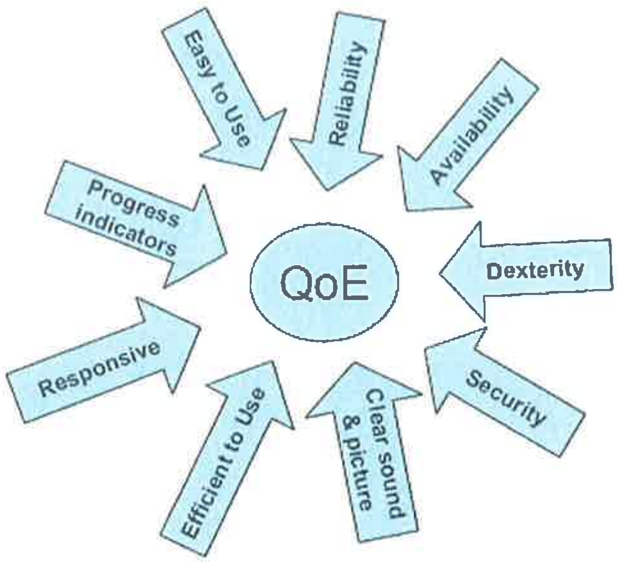


FIGURE 14

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 behavioral science techniques. QoE can be measured in a laboratory setting or in the field, through user ratings, surveys, or observation of user behavior. 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.

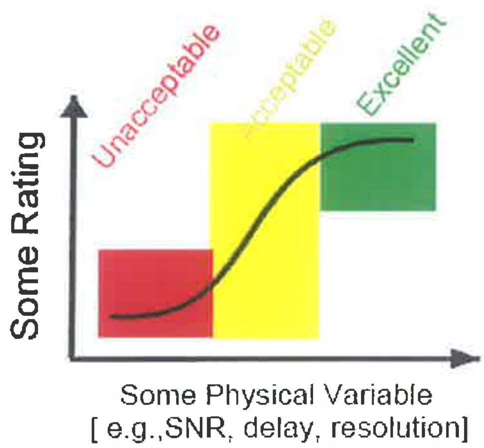


FIGURE 15
QoE variation as function of a variable

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The Working Committee that developed the Guideline consisted of the following organisations and their representatives:

Organisation	Membership	Representative
ACCC	Non-Voting	Rowan Groves
AAPT	Voting	Peter Crosby
AAPT / PowerTel	Non-Voting	Peter Kohlmayer
ACMA	Non-Voting	Noel Buchanan
Alcatel-Lucent	Voting	Evan Stanbury
Cisco Systems	Voting	Kim Yan
Nortel	Voting	Julio Cadena
Pacific Internet	Voting	Gary Marshall
SingTel Optus	Voting	James Dam
TEDICORE	Voting	Barry Dingle
Telstra	Voting	Glenn Colville
Telstra	Non-Voting	Chris Hill
Vodafone	Voting	Davorka Karacic

This Working Committee was chaired by Gary Marshall. James Duck of Communications Alliance provided project management support.

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.

Communications Alliance seeks to facilitate open, effective and ethical competition between service providers while ensuring efficient, safe operation of networks, the provision of innovative services and the enhancement of consumer outcomes.

It is committed to the achievement of the policy objective of the Telecommunications Act 1997 - the greatest practicable use of industry self-regulation without imposing undue financial and administrative burdens on industry.

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Care should be taken to ensure the material used is from the current version of the Standard or Industry Code and that it is updated whenever the Standard or Code is amended or revised. The number and date of the Standard or Code should therefore be clearly identified. If in doubt please contact Communications Alliance

The lives of your customers may be at stake. So it is absolutely critical that you allocate the correct CLI and ABC codes as well as ensure that you provide the necessary information to the Integrated Public Number Database.

By taking these actions you will be making emergency calls over the Internet a much safer option for your customers.

For further information, feel free to call Communications Alliance on 02-9959 9111.

An overview of the regulatory regime
The communications industry in Australia has an effective system of protection for its customers. Protective measures exist on a number of levels and all service providers should be aware of their obligations. These measures are the responsibility of the following bodies:

Communications Alliance Ltd
The 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. The Communications Alliance has its genesis in the merger of ACIF and SPAN.

Australian Competition and Consumer Commission (ACCC)
The ACCC promotes competition and fair trade in the market place to benefit consumers, business and the community. It also regulates national infrastructure services. Its primary responsibility is to ensure that individuals and businesses comply with the Commonwealth competition, fair trading and consumer protection laws.

Australian Communications and Media Authority (ACMA)
The ACMA licenses telecommunications carriers, ensures compliance with carrier licence conditions and service provider rules, and monitors service performance and quality. The ACMA also administers legislative provisions relating to powers and immunities of carriers in the construction of telecommunications facilities, and protection of consumers through safeguards and service guarantees.

Telecommunications Industry Ombudsman
The Telecommunications Industry Ombudsman (TIO) is a free and independent alternative dispute resolution scheme for small business and residential consumers in Australia who have a complaint about their telephone or Internet service.

We encourage you to visit the Communications Alliance website www.commsalliance.com.au.au or call us on 02-9959 9111 to understand your obligations under ACIF* Codes or just to discuss the information you should provide to your customers.

* ACIF is a division of Communications Alliance Ltd

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ATTACHMENT D

Access to emergency services for users of VoIP and Internet Telephony

Your obligations as a service provider

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Being able to use the phone to get help in an emergency is one of the most important benefits of modern communications.

That's why, as a provider of VoIP or Internet Telephony services, you have a number of obligations that are of vital importance in helping your customers to rely on their phone in emergencies.

In fact, if the service you are offering is considered to be a Standard Telephone Service (check this with the Australian Communications and Media Authority), it is mandatory that you provide access to emergency services through 000 and 106.

The reasons why it is so critical for you to meet these requirements are mainly related to the fact that phone calls made over the Internet are not as easily pinpointed as traditional fixed line calls or even mobile phone calls.

So emergency call persons and the emergency services need extra help in locating the source of a VoIP call to ensure a correct connection and guarantee a timely response.

How emergency calls are handled

In Australia, emergency call-taking is a two-stage process.

All calls to 000 are routed to one of two dedicated Telstra emergency call centres, and for 106 calls they are routed to one of two Australian Communication Exchange emergency call centres.

From these centres, each call is then transferred to the appropriate emergency service organisation (police, fire, or ambulance) in the relevant state.

The key to the success of this process is that calls are transferred to the correct service location as soon as possible.

For fixed services, the decision on where to transfer the call relies on the 000/106 emergency call centres having the correct location of the caller.

This is achieved by using the incoming calling number and the registered address of the phone service, and then mapping this information to the corresponding emergency service organisation for connection.

However, the connection may be made incorrectly because the service address of a VoIP caller may not accurately reflect the location of the caller.

To avoid this occurring, the call centre operators for 000/106 need to have a way of knowing that the call is being made over the Internet.

There has to be a "trigger" indicating to the 000/106 call centre operator that the caller must be asked to verify the location of the emergency. This will ensure correct connection of the call.

How you can help locate emergencies

To help emergency services locate the source of incoming VoIP calls, service providers need to do a number of things.

ABC Codes

Firstly, you have to make sure that when your customers make emergency calls, the correct CLI and the correct 3 digit ABC code are attached.

This alerts the 000/106 call centre operator that it is a VoIP call and that they must verbally request from the caller their location to ensure a correct connection to an emergency service organisation.

If you are unfamiliar with the ACIF* ABC Codes (typically in the form '98x' for VoIP), visit the Communications Alliance website and click on the following link:

http://www.commsalliance.com.au/ACIF_documents/guidelines/G557

In addition, the VoIP Location Indicator signaling specification is available at

http://www.commsalliance.com.au/ACIF_documents/specifications/G629

IPND Data

Secondly, you must provide information to the Integrated Public Number Database (IPND) which can further assist the 000/106 call centres in connecting calls correctly.

This information will include your Carriage Service Provider Code as well as an Alternate Address Flag, to indicate to the emergency services that the calling address may not reflect the physical location of the caller.

The data you provide to the IPND is explained in G619:2005 IPND Data Guideline available at

http://www.commsalliance.com.au/ACIF_documents/guidelines/G619

Specifically, you should look at Section 6.8 for information about your Carriage Service Provider Code and Section 6.9 regarding the Alternate Address Flag.