TECHNICAL REPORT

DSL Forum TR-039

Requirements for Voice over DSL

Version 1.1
March 2001

Abstract:
This Technical Report specifies an interoperable end-to-end architecture to support broadband voice and data service over DSL systems operating in either packet-mode or ATM-mode.

Notice:
The DSL Forum is a non-profit corporation organized to create guidelines for DSL network system development and deployment. This Technical Report has been approved by members of the Forum. This document is not binding on the DSL Forum, any of its members, or any developer or service provider involved in DSL. The document is subject to change, but only with approval of members of the Forum.

Table of Contents

1. Introduction

This document provides architectural requirements and recommendations for the deployment of voice services on a DSL broadband access network.

1.1. Scope

1.1.1. General

This document serves as the foundation for the family of Voice-Over-DSL Technical Reports published by the DSL Forum. It contains descriptions of the architectures and service models considered for supporting Voice over DSL as well as a reference model for protocol standards. Compliance with the family of Voice-Over-DSL Technical Reports is defined, and the identification of public, private and standard protocols is included.

This document is composed of a main body and annexes. The Main Body addresses functional requirements that are independent of the voice transport mechanisms. Annexes describe the specifics of implementation options for particular Voice over DSL mechanisms.

1.1.2. Key Concepts

Given a broadband access network – as defined in TR12 [3] – the intent of this document is to leverage this broadband infrastructure to allow service providers to derive additional service revenue, namely from voice services.

Voice-Over-DSL (VoDSL), described here, is distinguished from conventional baseband voice in that the voice channels are digitized within DSL codewords and transported through the Access Node at frequencies above the POTS physical layer of the DSL link, in order to provide derived telephony services. Therefore, for ADSL, this is via the high-pass filter path of the POTS splitter as opposed to the low-pass filter path.

An access network conforming to the requirements set out in this document enables voice communication between users on a point-to-point, or point-to-multipoint (conference call) basis. Voice services in this environment rely on Interworking Functions (IWF) located in the Access Network, and at the Customer Premises to derive telephony services.

By adding voice capabilities to a DSL broadband network, it is likely that many access networks will inter-mix voice channels with traditional data services at the "U" interface. Although such voice/data multiplexing may be common, it is not required.

The delivery of end-to-end voice services may be by an integrated facilities based service provider, or by a combination of access provider and dedicated service provider. An access network supporting derived voice services, regardless of the ownership and organization, common management functions are required for voice services across the various reference points.

The term VoDSL encompasses a number of telephony and data communications technologies and applications of these technologies. In order to satisfy certain applications or cost goals, as well as the need to accommodate rapidly changing technology, several transport mechanisms are defined in the family of DSL Forum VoDSL Technical Reports. These VoDSL recommendations are defined for a variety of DSL physical media.

1.2. Terminology of Requirements

This document uses several words to signify the specification requirements. These words are capitalized. This section defines these words as they should be interpreted. The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described below.

1. MUST	This word, or the terms "REQUIRED" or "SHALL", mean that the definition is an absolute requirement of the specification.
2. MUST NOT	This phrase, or the phrase "SHALL NOT", mean that the definition is an absolute prohibition of the specification.
3. SHOULD	This word, or the adjective "RECOMMENDED", mean that there may exist valid reasons in particular circumstances to ignore a particular item, but the full implications MUST be understood and carefully weighed before choosing a different course.
4. SHOULD NOT	This phrase, or the phrase "NOT RECOMMENDED" mean that there may exist valid reasons in particular circumstances when the particular behavior is acceptable or even useful, but the full implications should be understood and the case carefully weighed before implementing any behavior described with this label.
5. MAY	This word, or the adjective "OPTIONAL", mean that an item is truly optional. One vendor may choose to include the item because a particular marketplace requires it or because the vendor feels that it enhances the product while another vendor may omit the same item. An implementation which does not include a particular option MUST be prepared to interoperate with another implementation which does include the option, though perhaps with reduced functionality. In the same vein an implementation which does include a particular option MUST be prepared to interoperate with another implementation which does not include the option (except, of course, for the feature the option provides).

2. References

The following publications contain provisions, which, through reference in this text, constitute provisions of this Technical Report. At the time of publication, the editions indicated were valid. All references are subject to revision.

[1] ISO/IEC 7498-1, "Information Technology – Open Systems Interconnection – Basic Reference Model: The Basic Model", 1994

[2] DSL Forum Technical Report 1, "ADSL Forum System Reference Model"

[3] DSL Forum Technical Report 12, "Broadband Service Architecture for Access to Legacy Data Networks over ADSL, Issue 1", June 1998

[4] Telcordia GR-303-CORE Issue 3, "IDLC Generic Requirements, Objectives, and Interface", December 1999

[5] ITU-T Recommendation G.964 "V-Interfaces at the Digital Local Exchange (LE) - V5.1-Interface (based on 2048 kbit/s) for the support of Access Network (AN)", June 1994

[6] ITU-T Recommendation G.965 "V-Interfaces at the Digital Local Exchange (LE) – V5.2 Interface (based on 2048 kbit/s) for the support of Access Network (AN)", March 1995

[7] ANSI Standard T1.508, "Network Performance - Loss Plan for Evolving Digital Networks" July 1992

[8] Telcordia TR-TSY-000064, "LATA Switching Systems Generic Requirement (LSSGR) Issue 2"

[9] ANSI Standard T1.401, "American National Standard for Telecommunications – Interface between Carriers and Customer Installations"

[10] Internet Engineering Task Force RFC 1157, "A Simple Network Management Protocol (SNMP)", J. Case, M. Fedor, M. Schoffstall, and J. Davin, May 1990

[11] Internet Engineering Task Force RFC 1189, "The Common Management Information Services and Protocols for the Internet (CMOT and CMIP)", U. Warrier, L. Besaw, L. LaBarre, B. Handspicker, October 1990

[12] ITU-T Recommendation I.366.2 "AAL type 2 service specific convergence sublayer for trunking", February 1999

[13] ATM Forum af-vmoa-0145.000 "Voice and Multimedia Over ATM - Loop Emulation Service Using AAL2", 2000

[14] Telcordia TR-NWT-000057 Issue 2 "Functional Criteria for Digital Loop Carrier Systems" January 1993

[15] Telcordia TA-NWT-000909 "Generic Requirements and Objectives for Fiber in the Loop Systems"

[16] European Standard (Telecommunications series) Attachments to Public Switched Telephone Network (PSTN) ETSI EN 300 001 V1.5.1; "General technical requirements for equipment connected to an analogue subscriber interface in the PSTN", October 1998

[17] Telcordia TR-TSY-000008 Issue 2, "Digital Interface Between The SLC-96® Digital Loop Carrier System And A Local Digital Switch", August 1987

[18] ITU-T Recommendation H.323 "Packet Based Multimedia Communications Systems", January 1998

[19] DSL Forum Technical Report 17, "ATM over ADSL Recommendation", March 1999

[20] ITU-T Recommendation V.18 "Operational and interworking requirements for DCEs operating in the text telephone mode", February 1998

[21] ITU-T Recommendation G.114 "One-way transmission time", February 1996

[22] ITU-T Recommendation G.711 "Pulse code modulation (PCM) of voice frequencies", November 1988

[23] ITU-T Recommendation G.726 "40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)", December 1990

[24] ITU-T Recommendation G.729A "C source code and test vectors for implementation verification of the G.729 reduced complexity 8 kbit/s CS-ACELP speech coder", November 1996

[25] ITU-T Recommendation G.723.1 "Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s", March 1996

[26] ITU-T Recommendation G.992.1 "Asymmetrical Digital Subscriber Line (ADSL) Transceivers"

[27] ITU-T Recommendation G.992.2 "Splitterless Asymmetrical Digital Subscriber Line (ADSL) Transceivers"

[28] ITU-T Recommendation G.131 "Control of talker echo", August 1996

[29] ITU-T Recommendation G.165 "Echo cancellers", March 1993

[30] ITU-T Recommendation G.168 "Digital network echo cancellers", April 1997

[31] DSL Forum Technical Report 3, "Framing and Encapsulation Standards for ADSL: Packet Mode", September 1997

[32] ITU-T Recommendation I.366.1, "Segmentation and Reassembly Service Specific Convergence Sublayer for the AAL type 2", July 1998

[33] Internet Engineering Task Force RFC 2427, "Multiprotocol Interconnect over Frame Relay", C. Brown, A. Malis, September 1998

[34] Internet Engineering Task Force RFC 2684, "Multiprotocol Encapsulation over ATM Adaptation Layer 5", D. Grossman, J. Heinanen, September 1999

[35] Frame Relay Forum Implementation Agreement FRF.11.1, "Voice over Frame Relay", December 1998

[36] Frame Relay Forum Implementation Agreement FRF.8, "Frame Relay/ATM PVC Service Interworking", April 1995

[37] ITU-T Recommendation I.363.2, "B-ISDN ATM Adaptation Layer Type 2 Specification", September 1997

[38] ETSI ETR 275, "TM: Considerations on transmission delay and transmission delay values for components on connections supporting speech communication over evolving digital networks", April 1996

[39] Telcordia SR-2275, "Notes on the Networks", December 1997

[40] Frame Relay Forum FRF.12, “Frame Relay Fragmentation Implementation Agreement,” FRF.12, December 1997.

[41] ITU-T Recommendation M.3010, “Principles for a Telecommunications Management Network”, May 1996

[42] Telcordia GR-909-CORE, “Generic Criteria for Fiber in the Loop Systems”, March 2000

[43] ITU-T Recommendation I.363.5 (1996): “B-ISDN ATM Adaptation Layer Specification: Type 5 AAL”.

[44] ITU-T Recommendation I.610 (1998): “B-ISDN Operation and Maintenance Principles and Functions”.

[45] ATM Forum, *af-saa-0088.000,* “FUNI-to-ATM Interworking Function”, July 1997.

[46] ATM Forum. *af-saa-0109.000.* “Multi-Service Extensions to FUNI 2.0 Specification”, February 1999.

3. Definitions

The following definitions apply for the purposes of this Technical Report:

Access Network – One or more network entities interworking together to provide the transport services between Access Nodes and Service Provider Equipment. Thereby the Access Network normally contains the Access Node, the Regional Broadband Network, and VoDSL Access Network Interworking Functions. The Regional Broadband Network may institute different transport protocols such as ATM, Frame Relay or IP.

Access Node – Refers to the DSLAM in the Access Network

AN-IWF – Generic Access Network Interworking Function for VoDSL

ADSL – One of the copper access-transmission technologies with data rate of 1.5 – 9 Mbps for downstream and 16 – 640 Kbps for upstream. Specific line coding is not assumed.

Architecture Coexistence – One or more network dependent protocols e.g., ATM and Frame Relay coexist in the Core Network architecture

BLES – Broadband Loop Emulation Service (See Annex A)

Call Progress Signals – Voiceband tones and announcements used to inform the calling customer of the progress or state of a call.

Class 5 Switch – Also referred to as Local Exchange

CO-IWF – Central Office Interworking function for Broadband Loop Emulation VoDSL.

CP-IWF - Customer Premises Interworking function for VoDSL.

CPE Architecture -- An architecture that defines the access behavior within the Customer Premises Network and the interface to the Access and the Core Network. T and R reference points are considered part of the CPE architecture.

DTMF – Dual-Tone Multifrequency is a tone signaling method of transmitting address and other information where a set of dual-tone pulses is used to represent a set of characters. Each DTMF tone consists of two components: one component from a group of four low frequency tones, and the other component from a group of four high frequency tones, resulting in 16 possible combinations.

Downstream -- The direction of transmission from the Access Node to the Network Termination

Flash – A short on-hook interval during a prolonged off-hook period that indicates a desire to activate a service function, or a custom-calling feature.

LATA -- Local Access and Transport Area is a US regulatory term defining a geographic area covered by one or more local telephone companies, or local exchange carriers. A connection between two local exchanges within the LATA is referred to as *intraLATA*. A connection between a carrier in one LATA to a carrier in another LATA is referred to as *interLATA*. InterLATA is long-distance service.

MBN – Multiservice Broadband Networks (See Annex B)

Regional Broadband Network – The transport and switching interconnections among Access Nodes and service provider equipment in a geographic area. The switching system is capable of data rates at or above 1.5/2.0 Mbps.

SDSL – A term used to refer to symmetrical DSLs

Service Provider – A collective terminology for PSTN Service Provider, Internet Service Provider, Corporate Network and Locally Hosted Content provider.

Service Provider Equipment – This includes Class 5/4 switches, Media Gateways, Signaling Gateways and Media Gateway Controllers.

Upstream -- The direction of transmission from the Network Termination to the Access Node

VDSL – A term used to refer to Very High Speed DSL

4. Abbreviations

AAL	ATM Adaptation Layer
ADSL	Asymmetric Digital Subscriber Line
AN-IWF	Access Network Interworking Function
ATM	Asynchronous Transfer Mode
ATU-C	ADSL Terminal Unit – Central
ATU-R	ADSL Terminal Unit – Remote
BLES	Broadband Loop Emulation Service
B-NT	Broadband Network Termination
CLASS	Custom Local Area Signaling Service
CLE	Customer Located Equipment
CMIP	Common Management Information Protocol
CO	Central Office
CO-IWF	Central Office Interworking Function
CPDN	Customer Premises Distribution Network
CPE	Customer Premises Equipment
CP-IWF	Customer Premises Interworking Function
DSL	Digital Subscriber Line
DSLAM	DSL Access Multiplexer
DTMF	Dual-Tone Multifrequency
E&M	Ear and Mouth
FXO	Foreign Exchange Office
FXS	Foreign Exchange Station
IDLC	Integrated Digital Loop Carrier
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
IWF	Interworking Function
LAN	Local Area Network
LATA	Local Access and Transport Area
MEGACO	IETF Working Group developing the Media Gateway Control Protocols
MGCP	Media Gateway Control Protocol
MIB	Management Information Base
MBN	Multi-service Broadband Network
NSP	Network Service Provider
NT	Network Termination
PBX	Private Branch Exchange
PDN	Premises Distribution Network
POTS	Plain Old Telephony Service
PPP	Point-to-Point Protocol
PSTN	Public Switched Telephone Network
PVC	Permanent Virtual Circuit
QoS	Quality of Service
RSVP	ReSource reserVation Protocol
RTP	Real-time Transport Protocol
SNMP	Simple Network Management Protocol
SVC	Switched Virtual Circuit
VAD	Voice Activity Detection
VC	Virtual Circuit
WAN	Wide Area Network
VMOA	Voice and Multimedia over ATM

5. VoDSL Reference Model

The reference model shown below in Figure 1 is derived from the DSL Forum System Reference Model defined in TR-001 [2] and TR-012 [3]. Two intermediate interfaces (A10a, and Ta) are added in order to support an Interworking Function in the Access Network and at the Customer Premises. These Interworking Functions MAY be stand-alone equipment or physically integrated into adjacent equipment. This choice is made according to the architectural model defined in this document, and implementation decisions.

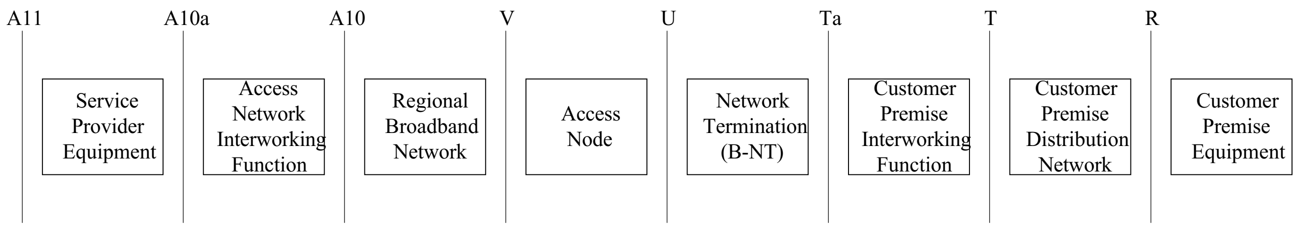

Figure 1: **VoDSL Architectural Reference Model**

5.1. VoDSL Reference Model Functional Block Definitions

5.1.1. Service Provider Equipment

The Service Provider Equipment provides the telephony services to the subscriber. These services MAY include Class 5 and Class 4 type features. Class 5 type features include regular telephony services, custom calling features, CLASS[SM] features and CENTREX services. Class 4 type features include trunking, and tandem switch applications. Service Provider Equipment MAY be implemented using traditional connection orientated telephone switch technology, or MAY utilize connectionless technologies, possibly without guaranteed Quality of Service of which examples include the H.323 [18] suite of standards.

5.1.2. Access Network Interworking Function

The Access Network Interworking Function (AN-IWF) performs the translation from signaling and bearer methods used by existing telephony equipment, to the signaling and bearer methods defined in this document. This MAY be a “null” function where telephony equipment inherently supports VoDSL services. The reference model does not preclude the integration of the AN-IWF with other Access Network functions. The Access Network Interworking Function is referred to as a Central Office Interworking Function (CO-IWF) for BLES Networks, described in Annex A.

5.1.3. Regional Broadband Network

The Broadband Network MAY include switching and/or routing equipment. The reference model does not preclude the integration of the Broadband Network equipment with other Access Network functions. In the case where both VoDSL and traditional broadband data services are deployed, the switching and routing equipment within the Regional Broadband Network perform the separation of these sessions, to their appropriate destinations.

5.1.4. Access Node

The Access Node provides concentration of multiple physical “U interface” ports, and concentration of bandwidth to the Access Network. This device is often referred to as a DSL Access Multiplexer (DSLAM). The reference model does not preclude the integration of the Access Node with other Access Network functions.

[SM] CLASS is a service mark of Telcordia Technologies, Inc.

5.1.5. Network Termination

The Network Termination performs the functions of terminating the DSL signal entering the Customer Premises. The reference model does not preclude the integration of the Network Termination with other Customer Premises functions.

5.1.6. Customer Premises Interworking Function

The Customer Premises Interworking Function (CP-IWF) performs the translation from signaling and bearer methods used by existing telephony equipment, to the signaling and bearer methods defined in this document. This MAY be a “null” function where telephony equipment inherently supports VoDSL services. The reference model does not preclude the integration of the CP-IWF with other Customer Premises functions.

5.1.7. Customer Premises Distribution Network

The Customer Premises Distribution Network MAY include switching and/or routing equipment to deliver voice (and data) to the Customer Premises Equipment. In the case where both VoDSL and traditional broadband data services are deployed, the switching and routing equipment within the Customer Premises Distribution Network perform the separation of these sessions, to their appropriate destinations.
The reference model does not preclude the integration of the Customer Premises Network with other Customer Premises functions.

5.1.8. Customer Premises Equipment

Voice (and possibly data) services are delivered across the Customer Premises Distribution Network to Customer Premises voice (and possibly data) devices. Examples of these devices include analog telephones, digital telephones, and Ethernet telephones. The reference model does not preclude the integration of the Customer Premises Equipment with other Customer Premises functions.

This recommendation draws no distinction between Customer Located Equipment, CLE, and CPE. (Note: some operators distinguish between CPE and CLE. CPE in their definition refers to equipment that is customer-owned and managed and CLE refers to equipment that is operator-owned and managed, but located at the customer premises.)

5.2. VoDSL Reference Model Interface Definitions

5.2.1. A11 Interface

The “A11” interface connects the Service Provider to the PSTN or Internet. This interface is beyond the scope of this document.

5.2.2. A10a Interface

The “A10a” interface connects the Service Provider Equipment to the AN-IWF. Examples of this interface include GR-303 [4], TR-008 [17], V5.1 [5] and V5.2 [6] when connecting to a Class 5 switch. This interface MAY be an internal function when the AN-IWF is integrated with other access network equipment.

5.2.3. A10 Interface

The “A10” interface connects the AN-IWF with the Regional Broadband Network. Examples of this interface include AAL2 ATM over OC3 for BLES Networks as described in Annex A, and AAL 1/2/5 ATM and IP over DSL Forum Technical Report TR-012 [3] networks for Voice over MBN as described in Annex B. This interface MAY be an internal function when the AN-IWF is integrated with other access network equipment.

5.2.4. V Interface
The “V” interface connects the Regional Broadband Network with the Access Node. Examples of this interface include AAL2 ATM over OC3 for BLES Networks as described in Annex A, and AAL 1/2/5 ATM and IP over DSL Forum Technical Report TR-012 [3] networks for Voice over MBN as described in Annex B. This interface MAY be an internal function when the Regional Broadband Network is integrated with other access network equipment.

5.2.5. U Interface
The “U” interface connects the Access Node with the Network Termination at each Customer Premises. An example of this is ATM over ADSL.

5.2.6. Ta Interface
The “Ta” interface connects the Network Termination with the CP-IWF. This interface MAY be an internal function when the CP-IWF is integrated with other Customer Premises Equipment.

5.2.7. T Interface
The “T” interface connects the CP-IWF with the Customer Premises Distribution Network. Examples of this interface include E&M, or this interface MAY be an internal function when the Customer Premises Distribution Network is integrated with other Customer Premises Equipment.

5.2.8. R Interface
The “R” interface connects the Customer Premises Distribution Network to the Customer Premises Equipment. Examples include FXS, FXO, and Ethernet. This interface MAY be an internal function when the Customer Premises Equipment is integrated with other Customer Premises Equipment.

5.3. VoDSL IWF Reference Model

VoDSL networks MUST provide one or more channels of derived telephony services. Interworking Functions (IWF) MUST be added in the Access Network and at the Customer Premises in VoDSL networks to perform the translation of signaling and bearer methods used by existing telephony equipment, to the signaling and bearer methods required to transport each derived voice channel across a DSL "U" interface. VoDSL IWF-IWF communication transparently transports the telephony bearer traffic between Service Provider switching equipment and Customer Premises locations, and between Customer Premises locations in the case of MBN. Telephony signaling is carried transparently between Customer Premises Equipment and Service Provider equipment.

The location of the Interworking Function in the Access Network and at the Customer Premises MAY be incorporated into other equipment elements or distributed across multiple elements.

5.3. General VoDSL Service Model

VoDSL enables multiple virtual derived telephony channels to be extended from Service Provider Equipment to the Customer Premises and between Customer Premises. A VoDSL network transports the service functionality of a line as provided by Service Provider Equipment through the Regional Broadband Network to the Customer Premises. Examples for services that may be transported over a VoDSL network include POTS, ISDN and Voice-over-IP enabled services. The VoDSL network provides a transparent means of transporting existing as well as new services over the Regional Broadband Network and the "U" interface so that it is possible that both the Service Provider Equipment and the Customer Premises Equipment may find the service indistinguishable compared to transport over a traditional local loop or PSTN trunk.

In addition to basic POTS and ISDN service, there are many CLASS features, custom calling features, and CENTREX services that can be offered to subscribers to enhance the telephony service and to support specific telephony applications. CLASS and CENTREX features use hook-flash, DTMF and MF or equivalent signaling techniques to support these features. VoDSL networks emulate or generate these signals so that these features continue to work transparently to the user.

Users will receive calling services from VoDSL networks that are at least equivalent to services provided by the PSTN. In particular, these calling service parameters include, but are not limited to, end-to-end delay, echo cancellation, dial tone delay, hook-flash signaling, and call teardown delay. This includes the support for fax machines and modems using derived channels. These services are transported transparently by bearer services without compression or silence suppression. Recovery of Network Timing MUST be supported for Fax or Modem transport to ensure that the Customer Premises timing is locked to the Central Office timing. Fax or Modem relay is beyond the scope of this technical report.

VoDSL compatible systems MUST support Class 5 functions. Class 4 services are provided transparently. VoDSL derived services may emulate a residential service, or business services such as CENTREX, PBX trunks, or off-premises PBX extensions. These services include, but are not limited to, dial tone, making a call, accessing network services, etc.

It is intended that VoDSL networks be able to fit into new as well as existing PSTN environments without necessitating the modification of legacy equipment.

Provider service offerings may have requirements that are dictated by regulatory environments. There may be local requirements such as E-911 services to be accessible over an analog POTS line even if power is out to the subscriber. Traditionally this has resulted in the loop powering of analog phones. Today's DSL services generally require locally powered Network Termination equipment. Baseband services attached through a splitter on a Voice over ADSL network for lifeline services MUST continue to be provided during power outages.

Subscribers of traditional analog POTS and ISDN lines connect many products to utilize the services including analog phones, ISDN terminals, key systems, PBXs, fax machines, Point Of Sale (POS) devices, dialup modems, TTY terminals according to V.18 [20], etc. A reference for functionality and capabilities expected from these devices can be found for North America in Telcordia publications TR-NWT-000057 [14], and TA-NWT-000909 [15], and in Europe ETSI EN 300 001 [16]. End-users expect to connect these terminals into derived lines.

The Interworking Function in the Access Network and at the Customer Premises MAY be incorporated into other functions or distributed across multiple functions and across multiple interfaces. A CP-IWF MAY be integrated into the Network Termination for an Integrated Access Device, where the traditional RJ11 interface is provided to the Customer Premises Equipment. In addition, a variety of PDN architectures may allow for the distribution of derived services over alternative residential or business networks.

6. General Requirements for VoDSL

6.1. General DSL Requirements

Events in ADSL (G.992.1 [26] and G.992.2 [27]) such as Dynamic Rate Repartitioning (DRR), Dynamic Rate Adaptation (DRA) and fast retrain all have the potential to interrupt voice service.

DRR and DRA will disrupt voice service if the available bandwidth falls below that allocated to the voice channel. DRR may also interrupt service on ADSL channels even when sufficient bandwidth remains. TR17 [19] recommends mechanisms to reduce service interruptions when DRR or DRA events occur. These mechanisms, however, might not permit voice solutions to meet their quality requirements. In this case DRR and DRA events SHOULD be controlled.

Fast retrain, a condition specific to G.992.2 [27], occurs when the line characteristics change because a local telephone connected to an ADSL baseband channel has gone on- or off-hook. The condition can be mitigated by the introduction of distributed (microfilters) or centralized splitters.

6.2. General System Requirements

6.2.1. Multi-Line Capability

A single “derived voice channel” may be attractive for some applications of VoDSL, but many applications contemplate multiple simultaneous, independent conversations. The maximum number of derived channels is limited by the bandwidth of the DSL link. An implementation SHOULD support multiple simultaneous VoDSL channels at the “U” interface.

6.2.2. Over-subscription

It is possible that a greater number of Customer Premises Equipment terminals be attached to the network than the maximum number of channels that can be simultaneously active. This includes PBX and distributed CENTREX examples. Channel over-subscription MAY be implemented. In the event that the call experiences blocking due to over-subscription of derived channels at the A10 interface, blocked call treatment of that call MUST be performed, which could include reorder (fast busy) to the originator of the call.

In addition, over-subscription of bandwidth in the network between the “U” interface and the “A10” interface is possible. This is at the discretion of the Service Provider. In the event that the call experiences blocking due to over-subscription, blocked call treatment of that call MUST be performed.

6.2.3. Dynamic Bandwidth Allocation for Voice

It is expected that in most cases, a DSL circuit will be used for both voice and data services. Since voice usage is sporadic, and bandwidth on the DSL link is limited, each voice channel SHOULD consume bandwidth only when a conversation is active. Required voice channel bandwidth MUST be reserved and maintained during any given active conversation. Voice bandwidth MUST have precedence over data bandwidth. Data bandwidth MAY dynamically utilize all available bandwidth (subsequent to voice channel requirements being met) according to the link bandwidth.

Where a large number of voice calls are active, requiring use of voice-compression to adhere to the guaranteed bandwidth allocated to the voice channel, voice quality MAY be optimized by dynamically minimizing the use of compression when the channel allocated to data is being under-utilized. Such functionality, which optimizes the use of bandwidth across different services and different channels, MAY require additional cross-layer functionality to be defined. For example, one implementation would be a Cross-layer Coordination Control Function sharing information between layers (e.g. xDSL PHY, ATM, Voice, etc) across a Coordination Control Interface.

Other benefits of cross-layer communication applied to the optimization of bandwidth across services MAY be implemented, such as use of voice compression and silence suppression on voice calls to help clear data-congestion.

6.2.4. Voice Quality

Service Providers and end-users generally do not want to be forced to compromise on the quality of voice connections. Technology is available to ensure that the voice quality is indistinguishable from that provided by a conventional wireline connection to the PSTN. Maximum delay and echo cancellation requirements SHOULD be supported according to T1.508 [7], G.114 [21], or G.131 [28]. Examples include G.165 [29] or G.168 [30]. G.114 [21] provides a number of delay allocation objectives for national and international voice connections.

G.114 [21] specifies end-to-end one-way delay in an international connection to be less than 150 ms to be acceptable for most user applications. It further recommends a one way processing time of no more than 50 ms in each of the national systems and for the international chain of circuits.

This allocation includes the delay introduced in the CP-IWF(s), DSL network(s), AN-IWF(s), and the transport network(s) between the call originator and the destination. Appendix 2 of Annex A further describes delay issues.

The CP-IWF and AN-IWF implementations SHOULD take all measures to minimize the delay. In addition the delay in the DSL network itself can be significant. For example, with ADSL (G.992.1 [26] and G.992.2 [27]), network operators could consider using the fast path or setting minimal interleave depth.

Note: that when the ADSL physical line bandwidth is sub-divided (e.g. into ADSL dual latency or other reserved channels) this will affect delay by reducing the “line rate” at which cells are inserted into the channel.

6.2.5. Service Feature Transparency

To the extent that the DSL connection merely serves as an alternative connection mechanism to an existing service, it MUST not impede the operation of any aspect of the service. Dialing, call management, and display capabilities MUST operate on a VoDSL network just as they would if the user were on a wireline connection to the PSTN. Further details are defined in section 6.3.

However, if physical-layer actions of the DSL modem (such as retraining when the margin is low) take place without regard for the state of the higher layer traffic (such as active derived voice calls), this could result in voice calls being dropped. This behavior compromises service feature transparency. The effect of such actions on voice calls in progress MAY be minimized by suitable cross-layer cognizance and communication, such as discussed in 6.2.3. For example, when voice calls are in progress, the DSL modem MAY hold-off retrain until the last possible moment, waiting for an opportunity when no voice calls are in progress. Additionally, if it is known that no voice calls are in progress, retrain MAY be performed to improve margin, even though normal retrain margin threshold has not been crossed. Such selective retraining, which relies on the physical layer management associated with the xDSL modem understanding the nature and state of the payload traffic, could significantly reduce the impact of retraining on derived voice service-quality, whilst subjecting to the interruptions only data traffic with a greater latency/retransmission tolerance. Suitable cross-layer communication MAY therefore also help meet the reliability requirements (section 6.2.11).

6.2.6. Continuous Availability

In conventional public and private telephone networks, the telephone equipment is always connected. The user SHOULD be able to make and receive calls without service interruption on a VoDSL network. VoDSL Customer Premises Equipment is not always line powered; therefore availability MAY depend on local power assurances. Note that battery backup at the customer premises is a possibility.

6.2.7. Performance

The dial-tone delay and call setup performance on a VoDSL network SHOULD be comparable to the performance experienced by users of conventional POTS telephone service.

6.2.8. Management

To the extent that deploying a VoDSL network requires additional functionality in both the Access Network and the Customer Premises, the equipment providing this functionality needs to be easily installable by the user or network operator. The operator SHOULD be able to remotely monitor, and maintain the VoDSL equipment including the Customer Premises Equipment, and MAY be able to troubleshoot the VoDSL equipment.

6.2.9. Network Security

Third parties SHOULD not be able to eavesdrop into conversations held over DSL. Mechanisms should prevent denial of service and theft of service. In addition, mechanisms SHOULD prevent third parties from imitating other users, so that calls cannot be diverted from their proper destination, or originated by an imposter with another user's identity.

6.2.10. Network Compatibility

The installed base of DSL network infrastructure is growing rapidly. VoDSL solutions SHOULD be compatible with this base. At the same time, new network capabilities SHOULD be anticipated and leveraged as they become available.

6.2.11. Reliability

The telecommunications industry has established exceptionally high expectations regarding network reliability. Each individual item of Service Provider Equipment supporting VoDSL SHOULD have a minimum reliability of 99.999%.

Cross-layer communication MAY be applied to manage retraining at the PHY layer to help meet the reliability requirements (see section 6.2.5)

6.3. General Service Requirements

The following table is a list of generally required features. Further details of mechanisms for implementation are left to the appropriate Annex. "O" indicates an optional feature, where as "R" indicates a recommended feature.

Capability	*Derived Line*
Multiple Bearer Channels	O
64Kbps μ/A-law	R
Low bit rate CODEC	O
Compression	O
Silence Suppression	O

Dynamic Bandwidth Allocation	O
Over subscription	O
QoS	R
End to end DTMF signaling	R
Bearer Capabilities Negotiation	R
Fax & Modem Transmission	R
Fax & Modem Relay	O
64K Clear Channel	O
Hook-Flash Support	R
Distinctive Ringing Control	R
Analog Loop Control	O
TTY (TDD) Terminal Support (V.18 [20])	R

Table 1: **Capabilities Matrix for Voice Services over DSL**

6.3.1. VoDSL Bearer Requirements

It MAY be appropriate to apply bandwidth-reduction techniques to the voice channel(s), making more of the link available for data or for additional derived voice channels. The encoding techniques MUST support and default to G.711 [22] when common capability negotiation of the encoding scheme is not possible. uLaw or Alaw MUST be selectable as appropriate for the country or local operating environment. Examples of commonly used vocoders include G.726 [23], G.729A [24] and G.723.1 [25] with silence suppression. The use of compression and silence suppression SHOULD be negotiated and managed through the VoDSL signaling mechanisms.

The AN-IWF and CP-IWF bearer mechanisms MUST be able to recover gracefully from intermittent impairments that disrupt the bearer data flow.

6.3.2. VoDSL Signaling Requirements

Line supervision signaling MUST support loop-start, and MAY support ground-start modes.

Signaling MAY include mechanisms for registration to validate the authenticity of the subscriber.

The use of compression and silence suppression SHOULD be negotiated and managed through the VoDSL signaling mechanism.

VoDSL signaling MUST support the establishment of incoming calls and outgoing calls, and the allocation of the associated bearer channel, and the termination of calls and the de-allocation of the channel.

The signaling mechanism MUST be able to recover gracefully from intermittent impairments that disrupt the signaling data flow. VoDSL signaling SHOULD provide a keep-alive mechanism, for each end to determine when connectivity has been lost.

VoDSL signaling SHOULD provide for the delivery of value-added features and services such as CLASS, CENTREX, and custom calling features. Customer Premises Equipment invokes and controls features by generating DTMF tones and flash hook indication or equivalent indication. Event notification from the Service Provider Equipment is provided through the generation of MF (multi-frequency tones) or FSK (frequency shift keying) and CCS messages. This MAY be through in-band signaling on the bearer channel provided that bearer compression does not adversely affect transport of this signaling.

6.3.3. Class 5 Services

Class 5 Services provide voice access services for a customer from the network. This service includes the ability to make telephone calls to other subscribers, and the ability to receive calls from other subscribers. This section defines the minimum and supplementary features that VoDSL SHOULD deliver via the Customer Premises Equipment.

6.3.3.1. Regular Telephony Features

The features described here are generally included as regular telephony features. These features are a group of services that MUST be implemented by a VoDSL network to a user. This feature set includes Outgoing Call Establishment, Incoming Call Request, Call Release, and Basic Service characteristics.

The text in these subsections is stated as if the voice service is delivered by a traditional circuit switch. It should be understood, however, that these subsections also include so-called “soft switches” and other distributed switching mechanisms (e.g., as in Annex B).

6.3.3.1.1. Outgoing Call Establishment

The establishment of a call is initiated by the subscriber operating the Customer Premises Equipment. The dialed number MAY include local, national, international or special dialing sequences (emergency, carrier identification codes, custom calling feature activation, facility codes, etc.).

The remote Service Provider Equipment SHOULD respond to a call attempt with a call progress message. Call progress messages indicate that the call is being processed. These call progress messages include audible ringing, busy tone, reorder tone, and recorded announcements.

6.3.3.1.2. Incoming Call Request

Call request is the ability to receive incoming calls from other telephone users. Alerting is generated from the call destination switch to the destination CPE. All varieties of Call Waiting are transparently supported. If Call Waiting is disabled, the service provider equipment will not attempt any signaling over the Access Network to a terminal located at the CPE that is busy or is already being alerted.

6.3.3.1.3. Call Release

Call Release can be initiated either by the user, remote network elements, or equipment located within the Access Network upon a call error condition.

6.3.3.1.4. Basic Service Characteristics

VoDSL telephony implies availability and performance meeting service characteristic criteria.

Availability characteristics rely on a combination of arrangements, facilities, and procedures. These items include alternate routing, automatic switching to spare units for critical equipment, standby facilities for power equipment, rapid access to restoral equipment, etc. These items are beyond the scope of this report.

Performance characteristics include both speed of call establishment, and transmission quality. Overall call establishment performance include dial tone delay, and call setup time. These parameters are defined in North America by LSSGR: Service Standards, TR-TSY-000064 [8], and in Europe by ETSI EN 300 001 [16]. Detailed signaling timing parameters include address signaling timing, cessation of dial tone timing, partial dial timing, hit and flash timing, and disconnect timing. These parameters are defined for North America networks by the ANSI Standard T1.401 [9] specification, and in Europe by ETSI EN 300 001 [16].

6.3.3.2. Supplementary Features

Supplementary features are a group of services that MAY be implemented by the Customer Premises Equipment. These feature sets include Custom Calling features, CLASS features, and CENTREX features.

6.3.3.2.1. Custom Calling Features

Custom calling features are a group of services that MAY be provisioned by the serving end office. These features include speed dialing, call waiting, three-way calling, and call forwarding.

6.3.3.2.1. CLASS Features
CLASS (Custom Local Area Signaling Service) features are a set of call management features that when provisioned make use of the calling as well as called party telephone directory number that is forwarded to the call terminating Central Office.

CLASS features include Type I and Type II Calling Number Delivery, Calling Number Delivery Blocking, Distinctive Ringing/Call Waiting, Selective Call Forwarding, Selective Call Rejection, Customer Originated Trace, Automatic Callback, and Automatic Recall

6.3.3.2.2. CENTREX Features
CENTREX features MAY delivered by the Service Provider Equipment. General CENTREX features include Call Forwarding, Call Waiting, Call Hold, hunt groups, and 3 way calling. The control of these general CENTREX services are all activated and controlled through the transport of a flash hook or equivalent indication along with DTMF digits. The access or facility codes to activate and control these features MAY vary between carriers. These dialed number sequences MUST be transmitted in an enbloc or overlapped manner by the signaling protocol. These numbers MUST include 0-9, *, #, and Flash Hook, and MAY include A-D.

Additional CENTREX features MAY be implemented including Direct-Inward-Dialing, Calling Number Display, Distinctive Ringing, and Message Waiting.

6.3.4. Class 4 Services
Class 4 switch services normally reside in the core of a telephone network, and therefore do not normally traverse the local-loop (whether DSL or traditional POTS). It is therefore beyond the scope of this work to directly support Class 4 services, which remain available through the Class 5 switching function.

7. VoDSL Management

To the extent that VoDSL requires additional functionality in Customer Premises Equipment, that equipment needs to be easily installable by the carrier or the user. The network operator MUST be able to easily and remotely monitor, maintain and troubleshoot the CPE.

7.1. VoDSL Management Functions

Management functions relate to users' needs for facilities that support the planning, organization, supervision, control, protection and security of VoDSL network resources, and account for their use. These facilities MAY be categorized as supporting the functional areas of configuration, fault, performance, security and accounting management as defined in ITU-T M.3010 [41]. These can be summarized as follows.

Configuration management provides for the identification of VoDSL network resources, initialization, reset and closedown, the supply of operational parameters, and the establishment and discovery of the relationships between resources. The configuration management function SHOULD include the ability to perform the upgrading of any firmware and software.

Fault management provides for fault prevention, detection, diagnosis, and correction.

Performance management provides for evaluation of the behavior of VoDSL network resources and of the effectiveness of service delivery. This could include for example monitoring of such performance parameters as peak and average delay to dial tone for originating calls, end to end voice path latency, call establishment success/failure ratio.

Security management provides for the protection of resources.

Management facilities in VoDSL equipment will address some or all of these areas, as appropriate to the needs of that equipment and the environment in which it is to be operated.

7.2. VoDSL Management Architecture

The management facilities defined in the DSL Forum VoDSL recommendations are based on the concept of managed objects, which model the semantics of management operations. Operations upon a managed object supply information concerning, or facilitate control over, the process or entity associated with that object.

Operations upon a managed object can be initiated by mechanisms local to the equipment being managed (e.g. via a control panel built into the equipment), or can be initiated from a remote management system by means of a general-purpose management protocol carried using the data services provided by the DSL interface to which the equipment being managed is connected.

There are two general-purpose management protocols of relevance to the management of VoDSL equipment:

-- The Simple Network Management Protocol (SNMP), as described in RFC 1157 [10].
-- The OSI common management information protocol (CMIP), as described in RFC 1189 [11].

SNMP or CMIP MAY be used to manage VoDSL equipment.

Annex A

Requirements for Voice over DSL access facilities to Broadband Loop Emulation Service (BLES)

Annex A: Broadband Loop Emulation Service (BLES)

Broadband Loop Emulation Service (BLES), described here, is one method of building and deploying VoDSL networks. There are 2 approaches described that make use of different transport environments. These are:

- A method using ATM with AAL2 Adaptation Layer encapsulation based on to ATM Forum af-vmoa-0145.000 ("AAL2 over ATM");

- A method using ATM Forum *af-saa-0088.00* framing (FUNI) with AAL2 Adaptation Layer based on ATM Forum *af-vmoa-0145.000* ("AAL2 over ATM");

- A method for carrying packets over frame relay based on Frame Relay Forum FRF.11 ("IP-based FRF.11").

Each of these approaches transport one or more channels of Narrowband Services, including the transport of voice, voice-band data and fax traffic over a broadband subscriber line connection (including ADSL, SDSL and VDSL) between Customer Premises and the Service Provider switched telephone network. The BLES includes support for the VoDSL reference model for functional blocks and interfaces as defined in the reference model below (see Figure A-1). In addition, BLES includes support for the VoDSL IWF-IWF reference model for both a POTS and ISDN delivery methods, and the services supported by a Service Provider Class 5 Switch and Customer Premises Equipment.

BLES networks are expected to operate in environments that may inter-mix voice channels with traditional data services at the "U" interface. Although such multiplexing may be common, it is not required.

BLES network topologies rely on Interworking Functions (IWF) located in the Access Network, and at the Customer Premises to derive narrowband services. A DSL interface provides a broadband access between the Customer Premises and a network capable of supporting these derived narrowband services.

BLES includes the support for compressed voice and non-compressed voice together with or without silence removal.

This annex includes details of the architecture, a list of requirements, and recommended protocols and interfaces based on (where possible) existing standards that are specific to Broadband Loop Emulation Services (BLES). In particular, the annex references work performed in the ATM Forum VMOA working group for Loop Emulation Services.

A.1 BLES Reference Model

The general reference model for a VoDSL network is described in section 5 of the main body of this Technical Report. The example network for a BLES Model in Figure A-1 shows the intermediate interfaces (A10a, A10, V, U and T) that support an Interworking Function in the Central Office and at the Customer Premises. These Interworking Functions MAY be stand-alone equipment (as depicted between the A10 and A10a interfaces) or physically integrated into adjacent equipment (as shown in the Customer Premises Network Termination). This choice is made according to the architectural model defined in this document, and implementation decisions.

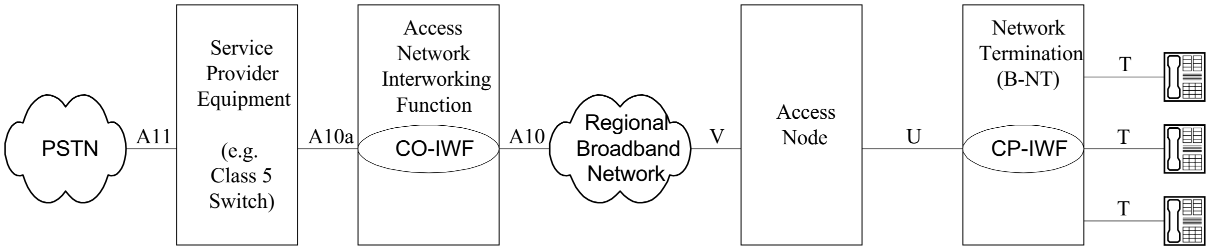

Figure A-1: **Example BLES Architecture**

A.1.1 BLES Reference Model Functional Block Definitions

A.1.1.1 Service Provider Equipment

The Service Provider Equipment provides the telephony services to the subscriber. These services are typically implemented using traditional connection-orientated Class 5 telephone switching technology and include regular telephony services, custom calling features, CLASS[SM] features and CENTREX services.

A.1.1.2 Central Office Interworking Function

The CO-IWF performs the translation from signaling and bearer methods used by existing telephony equipment, to the signaling and bearer methods defined in this annex (refer to section A.1.3.1 and A.1.3.2). This MUST include at least one of POTS or ISDN/BRA signaling methods.

A.1.1.3 Access Node

The Access Node (DSLAM) provides concentration of multiple physical "U interface" ports, and concentration of bandwidth to the Access Network.

A.1.1.4 Network Termination

The Network Termination performs the functions of terminating the DSL signal entering the Customer Premises. The CP-IWF performs the translation from signaling and bearer methods used by existing telephony equipment, to the signaling and bearer methods defined in this annex (refer to section A.1.3.1 and A.1.3.2.).

A.1.1.5 Customer Premises Equipment

Examples of Customer Premises equipment include analog phones, key systems, PBXs, fax machines, Point Of Sale (POS) devices, ISDN telephones/terminal adapters and dialup modems.

A.1.2 BLES Reference Model Interface Definitions

A.1.2.1 A10a Interface

The "A10a" interface connects the Service Provider Equipment to the CO-IWF. Examples of this interface include GR-303 [4], TR-008 [17], V5.1 [5] and V5.2 [6].

A.1.2.2 A10 and V Interfaces

The "A10" and "V" interfaces connect the CO-IWF with the Access Node through the Regional Broadband Network. Examples of functions/capabilities supported over the interface include AAL2 ATM over DS3, AAL5 ATM over DS3, or IP based FRF.11 encapsulation. The V interface MAY be different from the A10 interface.

[SM] CLASS is a service mark of Telcordia Technologies, Inc.

(Note that in the DSL Forum reference model the V interface separates the DSLAM from the core network, whereas in the ITU-T voice reference model, the V interface separates the Service Node (Class 5 function) from the attached equipment).

A.1.2.3 U Interface

The "U" interface connects the Access Node with the Network Termination at each Customer Premises. Examples of this interface include ATM over ADSL, and ATM over SDSL. BLES networks MUST have at least one dedicated virtual circuit that can support one or more derived voice calls between the Service Provider Equipment and the Customer Premises

A.1.2.4 T Interface

The "T" interface connects the CP-IWF with the Customer Premises Equipment. Examples of this interface include FXS, FXO, and E&M.

A.1.3 BLES IWF-IWF Reference Model

BLES networks MUST provide one or more channels of derived telephony services. Interworking Functions (IWF) MUST be added in the Service Provider Equipment and at the Customer Premises in BLES networks to perform the translation of signaling and bearer methods used by existing telephony equipment, to the signaling and bearer methods required to transport each derived BLES channel across a DSL "U" interface. BLES IWF-IWF communication transparently transports the telephony signaling and bearer traffic between the Service Provider switching equipment and the Customer Premises.

Each CP-IWF utilizes at least one Virtual Circuit (Permanent or Switched) to the CO-IWF for signaling, bearer services, and management. BLES VCs MUST be assigned a Quality of Service such that the requirements of the selected signaling and bearer channels can be met. Note that QoS requirements, especially those involving delay/latency and delay variation, are often needed to meet specific A10a interface and regulatory needs (e.g. signaling times from physical off-hook to dial-tone delivery). The BLES VCs MUST be provisioned as a higher priority through the network than data VCs. Broadband data Virtual Circuits exist independently according to DSL Forum TR-012 [3], or DSL Forum TR-003 [31].

For AAL2 over ATM, the BLES IWF-IWF definition is performed according to ATM Forum af-vmoa-0145.000 [13]. This work in progress is based upon ATM Adaptation Layer 2 (AAL2) as defined in the I.366.2 [12]. The AAL2 VCs SHOULD be configured as a Variable Bit Rate – Real Time (VBR-RT) circuit, but MAY be configured as a Constant Bit Rate circuit.

For BLES over Frame User Network Interface (FUNI), the usage of AAL2 MUST follow the principles set out in ATM Forum *af-vmoa-0145.00*. Transport for BLES over FUNI MUST conform to the Frame User Network Interface (FUNI) framing defined in ATM Forum document *af-saa-0088.00* and encapsulation described in DSL Forum *TR-003 v1.0*. A FUNI-to-ATM interworking function MAY exist in the voice path at either the V interface or the A10 interface to facilitate transparent interoperability with BLES CO-IWF. Specific modes of operation for BLES over FUNI implementations that meet this requirement are detailed in section A.1.3.3 below. The AAL2 VCs SHOULD be configured as a Variable Bit Rate – Real Time (VBR-RT) circuit, but MAY be configured as a Constant Bit Rate (CBR) circuit.

For IP-based BLES with ATM or Frame Relay as the transport, voice payloads and Channel Associated Signaling MUST be formed as in the FRF.11.1 [35] to form the information field of a packet. This information field MUST be carried as the payload of an IP/UDP packet using IANA assigned UDP port 21590. Packets are carried over the VCs using standard multiprotocol interconnection over the media as in RFC 2427 [33] or RFC 2684 [34]. The VCs themselves are configured in any manner to meet the above requirement that voice VCs receive preferred treatment through the network. Fragmentation as in FRF.12 [40] MAY be supported on any data VCs in the network in order to minimize the delay of voice traffic.

For interoperability, the packing factor M of FRF.11 must be chosen. This is the integer number of 10ms intervals that will constitute the voice information field from one line. The value of M=1 corresponding to 10ms samples MUST be supported; values of M>1 MAY be supported. In addition to the size of each of information field, it may be desirable for some implementations to limit the number of voice payloads in each packet to limit the serialization delay due to data link queuing. Both the CO-IWF and the CP-IWF SHOULD be configurable to limit the number of voice payloads to be inserted into each frame.

BLES derived loop characteristics including ringer equivalence, hybrid balance, and others must be addressed in compliance with Telcordia TR-NWT-000057 [14].

A.1.3.1 BLES IWF-IWF Bearer Requirements

It MAY be appropriate to apply bandwidth-reduction techniques to the voice channel(s), making more of the link available for data or for additional derived voice channels. The encoding techniques MUST support and default to G.711 [22] in the absence of other configuration or negotiation possibilities. uLaw or Alaw MUST be selectable as appropriate for the country or local operating environment. Examples of commonly used vocoders include G.726 [23]. Silence suppression MAY be used. For AAL2 over ATM, the BLES IWF-IWF bearer requirements is according to ATM Forum af-vmoa-0145.000 [13]

One method of reducing packetization delay is to use more than one AAL2 mini-cell per ATM cell; for example using two 20-byte G.726 [23] mini-cells (as defined in I.366.2 [12]) reduces latency by 6 ms to 5 ms compared to using the maximum payload of 44 octets for a single cell as in ATMF profile 7. This method is, of course, bandwidth inefficient if there is only a single active call, but this may be outweighed by the reduced latency.

A.1.3.2 BLES IWF-IWF Signaling Requirements

BLES IWF-IWF signaling MUST be transported using either bit-oriented Channel Associated Signaling (CAS), or message-based Common Channel signaling (CCS) methods. For AAL2 over ATM, the BLES IWF-IWF signaling requirements are according to ATM Forum af-vmoa-0145.000 [13].

For IP-based FRF.11 encapsulation, the BLES signaling requirements are according to the encapsulation of robbed-bit signaling as specified in FRF.11.1 [35].

A.1.3.3 BLES IWF-IWF Operation with BLES over FUNI

When BLES is implemented with BLES over FUNI, the CP-IWF and the CO-IWF MAY natively support BLES over FUNI. Otherwise, a FUNI-to-ATM Interworking function MUST exist at the V interface or the A10 interface. BLES IWFs that are connected via the ATM Forum *af-saa-0088 Frame User Network Interface (FUNI 2.0)* FUNI-to-ATM Interworking Function MUST use the method described below.

A.1.3.3.1 BLES over FUNI with FUNI-to-ATM Interworking

The method described here uses the ATM Forum AAL2 LES procedure to form AAL2 packets and places them into a FUNI frame instead of cells. When the FUNI frame is interworked to ATM, the AAL2 packet(s) are placed into cells using AAL5 CPCS. This produces n+1 AAL5 cells, where n is the number of AAL2 packets in the FUNI frame. The one additional cell carries the AAL5 pad and trailer.
In this method, AAL2 is an application riding on the standard FUNI layer. It is not a new adaptation layer for FUNI.

At the Frame User Network Interface, AAL2 packets MUST be formatted as shown in Table 1:

Length (octets)	Content	Reference
1	Start Field	ITU-T I.363.2, section 9.2.1 (see note 1)
3	CPS-Packet Header	ITU-T I.363.2, section 9.1 and ITU-T I.366.2
40+4 or 44	AAL2 payload + padding (see note 2)	ATM Forum *af-vmoa-0145.00*

Note 1: the value of the Offset Field, which is part of the Start Field, MUST be zero.
Note 2: for Type 3 packets as defined in I.366.2, the packet must be followed by padding consisting of an integral

number of zero value octets to a total packet size of 48 octets. ATM Forum LES specification allows either 44 bytes with no padding or 40 bytes with 4 bytes of padding.

Table 1: AAL2 Packet Format for BLES over FUNI

In FUNI-to-ATM Interworking, the payload of frames traveling in the direction user-to-network is converted into the payload of an ATM AAL5 PDU. CP-IWFs MUST use an AAL2 packet size of 48 octets (which includes the Start Field and CPS-Packet Header and either 40 octets plus 4 bytes of padding or 44 octets of AAL2 payload), and MAY insert one or more AAL2 packets concatenated together in each frame that is to be sent. CP-IWFs MUST insert an integral number of 48-octet AAL2 packets in each frame that is to be sent, and MUST be capable of receiving frames that contain an integral number of AAL2 packets of 48 octets each. CP-IWFs SHOULD discard incoming frames that are not an integral multiple of 48 octets in length. Implementations MUST NOT place more than one AAL2 type 3 packet in a single FUNI packet to preserve the redundancy aspects of the type 3 packet.

CO-IWFs implementing AAL2 over FUNI MUST support the transmission and reception of AAL5 PDUs that contain one or more AAL2 packets of 48 octets each, concatenated together. CO-IWFs implementing AAL2 over FUNI SHOULD discard incoming AAL5 PDUs which are not an integral multiple of 48 octets in length.

A.1.3.3.4 Management Considerations for BLES over FUNI

CO-IWFs that support the Loop Emulation Service (LES) using AAL2 in accordance with *af-vmoa-0145.00* may originate F5 OAM loopback cells to verify the integrity of virtual circuits that carry voice traffic. The FUNI-to-ATM Interworking function MUST handle F5 OAM cells that are incoming on ATM VCCs in accordance with ATM Forum *af-saa-0088.000* FUNI-to-ATM Interworking Function.

A.1.3.3.5 Jitter and Delay Management for BLES over FUNI

CP-IWFs that support the BLES over FUNI based upon the ATM Forum FUNI 2.0 *af-saa-0088.000* FUNI-to-ATM Interworking Function for FUNI-based multi-service transport SHOULD conform to the recommendations in ATM Forum *af-saa-0109.000 Multi-Service Extensions to FUNI 2.0 Specification.* Implementations that utilize packing of multiple AAL2 payloads within a single FUNI packet still MUST meet voice quality requirements per TR-036 section 6.2.4. Consideration should be given to informative Appendix A-2 with regards to delay.

A.1.4 General BLES Service Model

BLES enables multiple virtual derived telephony channels to be extended from Service Provider Class 5 Switching Equipment to the Customer Premises over a DSL “U” interface. A BLES network transports the service functionality of a line as provided by Service Provider Equipment through the Regional Broadband Network to the Customer Premises. Services that may be transported over a BLES network include POTS, and ISDN services, in a way that both the Service Provider Equipment and the Customer Premises Equipment may find the service indistinguishable compared to transport over a traditional local loop.

In addition to basic POTS and ISDN service, there are many CLASS features, custom calling features, and CENTREX services that can be offered. CLASS and CENTREX features use hook-flash, DTMF and MF signaling techniques to support these features. BLES networks emulate these signals so that these switch-provided features continue to work transparently. A BLES network implementation may be completely invisible to the Service Provider and Customer Premises Equipment with regard to these services.

The introduction of a derived voice channel on a broadband access network should be transparent to users. That is, the PSTN services should be indistinguishable whether the local loop is a copper pair, a derived TDM channel (DLC), or a derived broadband channel (BLES). In particular, these calling service parameters include, but are not limited to, end-to-end delay, echo cancellation, dial tone delay, hook-flash signaling, and call teardown delay. End-to-end service MUST conform to Telcordia TR-NWT-000057 [14] in North America and in Europe ETSI EN 300 001 [16].

A.1.4.1 BLES Service Provider Equipment Services

The generic services that are supported in Class 5 Service Provider switching equipment should be supported by a BLES compatible system. BLES derived services may emulate a residential service, or business services such as CENTREX, PBX trunks, or off-premises PBX extensions. These services include, but are not limited to, dial tone, making a call, accessing network services, etc. The Service Provider equipment should not require modification and should not need to be aware that a derived VoDSL line serves the subscriber. Functionality and capabilities of the BLES network to support this Service Provider Equipment MUST be according to Telcordia publications TR-NWT-000057 [14], and TA-NWT-000909 [15] in North America, and in Europe ETSI EN 300 001 [16].

A.1.4.2 BLES Customer Premises Equipment Services

Subscribers of traditional analog POTS lines connect many products to utilize the services including analog phones, ISDN terminals, key systems, PBXs, fax machines, Point Of Sale (POS) devices, dialup modems, TTY terminals according to V.18 [18] etc. Functionality and capabilities expected from the BLES network to support these devices MUST be according to Telcordia publications TR-NWT-000057 [14], and TA-NWT-000909 [15] in North America, and in Europe ETSI EN 300 001 [16].

Appendix A-1: TR-NWT-000057 Compatibility

This appendix is informative only

This appendix provides insight into appropriate parameters for functionality and conformity to TR-NWT-000057 [14]. Note that there are similar specifications according to TA-NWT-000909 [15]. Of all the requirements in TR-NWT-000057 [14], the following were viewed to have special or specific relevance for VoDSL networks using Class 5 PSTN Service Provider Equipment, including interfaces such as GR-303 [4] and TR-008 [17].

Section 5.1.6 Sending Loop Current Feed from the RT
Applying of loop current feed has specific timing requirements (R-11) that must be maintained.

Section 5.1.8 Reverse Loop Current Feed Applied to the COT
Applying of reverse loop current feed has specific timing requirements (CR-16, CR-17 and CR-18) that must be maintained.

Section 5.1.10 Sending Loop Current Feed Open from the RT
Applying of loop current feed open has specific timing requirements (R-21) that must be maintained.

Section 5.1.12 Ringing Requirements
Applying of ringing timing has specific requirements (R-23, R24, R-25, R-26) that must be maintained.

Section 5.1.13 Transmission Path
Applying of Off-hook transmission criteria (R-27) and On-hook transmission criteria (R-28) have specific requirements that must be maintained.

Section 5.1.14 Dial Tone Delay
Applying of dial-tone delay (R-34) has specific requirements that must be maintained.

Section 5.1.16 Showering Lines
Protection against rapid on-off hook sequences resulting from power-line shorts being represented to the Class 5 switch must be maintained by the Customer Premises Equipment (R-35)

Section 5.1.17 System Failures - COT
If transmission is known to be inactive, the CO-IWF should guard against indicating continuous loop-closed to the Class 5 switch (R-37)

Section 5.2.5 Sending Ring Ground from the COT
Applying of Ring Ground has specific timing requirements (R-11 and O-12) that must be maintained.

Section 5.2.6 Application of Loop Closure at the COT
Applying of Loop Closure has specific timing requirements (R-13) that must be maintained.

Section 5.2.9 Sending Loop Current Feed from the RT
Applying of Loop Current Feed has specific timing requirements (R-28, R-29 and O-30) that must be maintained.

Section 5.2.11 Reverse Loop Current Feed Applied to the COT
Applying of Reverse Loop Current Feed has specific timing requirements (CR-37, CR-38 and CR-39) that must be maintained.

Section 5.2.15 Transmission Path
Off-hook transmission requirements have specific timing (R-43, and R-44) that must be maintained.

Section 5.2.17 System Failures - COT

System Failure requirements at the COT (R-46) must be maintained.

Section 5.2.18 System Failures - RT
System Failure requirements at the RT (R-47) must be maintained.

Section 5.3.4 Loop Open/Loop Closure Distortion
Applying of timing of Loop opens and closures has specific requirements (R-10) that must be maintained.

Section 5.3.5 Loop Open/Loop Closure Delay
Applying of timing of Loop open and closure delays has specific requirements (R-11) that must be maintained.

Section 5.3.9 DLC Generated Loop Open Intervals at the RT
Applying of timing of Loop open intervals has specific requirements (R-24) that must be maintained.

Section 5.3.10 Loop Current Feed Open Intervals
Applying of timing of Loop current feed open intervals has specific requirements (R-25, R-26, R-27, and R-28) that must be maintained.

Section 5.3.11 Dial Pulsing
Applying of dial pulse timing has specific requirements (R-29) that must be maintained.

Section 5.3.12 DTMF Signaling Requirements
Applying of DTMF Signaling has specific requirements (R-30 and R-31) that must be maintained.

Section 5.4 Sending Ringing from the RT
Applying of ringing timing has specific requirements (R-5, R-6 and R-7) that must be maintained.

Section 5.4.3 Ring Trip
Applying of ring trip timing has specific requirements (O-9) that must be maintained.

Section 5.4.4 Ring Trip Reporting Delay
Applying of ring trip reporting delay timing has specific requirements (R-10) that must be maintained.

Section 5.4.6 System Failures
System Failure requirements (R-13) must be maintained.

TR-NWT-000057 Section 6.1 describes all the analog service provided by a DLC system that apply to VoDSL systems that connect to Class 5 Circuit Switched Service Provider Equipment.

Section 6.1.8 DLC System Loss
A VoDSL network is normally a loss-less system, and care should be taken to address proper setting of gain (R-8, O-9, R-10 and O-11)

Section 6.1.10 60 Hz Loss
Customer Premises Equipment should be consider control of 60Hz encoding into the bearer bit stream (R-15)

Section 6.1.21 Frequency Offset
A frequency reference may be needed to maintain an accurate sampling rate (R-35)

Appendix A-2: Bearer Channel and Signaling Delay in BLES Systems

This appendix is informative only

A-2.1 Introduction

Voice over DSL systems that use packet protocols such as ATM and frame relay to transport voice introduce significant delay into the transmission path for both voiceband payload and signaling. Excessive delay in the bearer transmission path may result in degradation of the perceived quality of voice communication, while excessive delay in the signaling transmission path may result in signaling errors that lead to degradation of the service.

This informative appendix identifies the principle sources of delay in the bearer and signaling transmission paths of VoDSL networks, with reference to the BLES architecture, and assesses the contribution of each delay source to the total end-to-end delay. It also identifies the upper limits on delay, beyond which perceptible service degradation may be experienced.

A-2.2 Network Model

This discussion of network delay refers to the network model illustrated in Figure A-2.1. This model is intended as an example of a practical implementation of a VoDSL network, and it differs from the reference architecture for BLES in that it includes sources and sinks of data traffic, which is carried over the DSL access network alongside voice. Data traffic must be considered because its presence may have a significant impact on the delay experienced by the voice traffic.

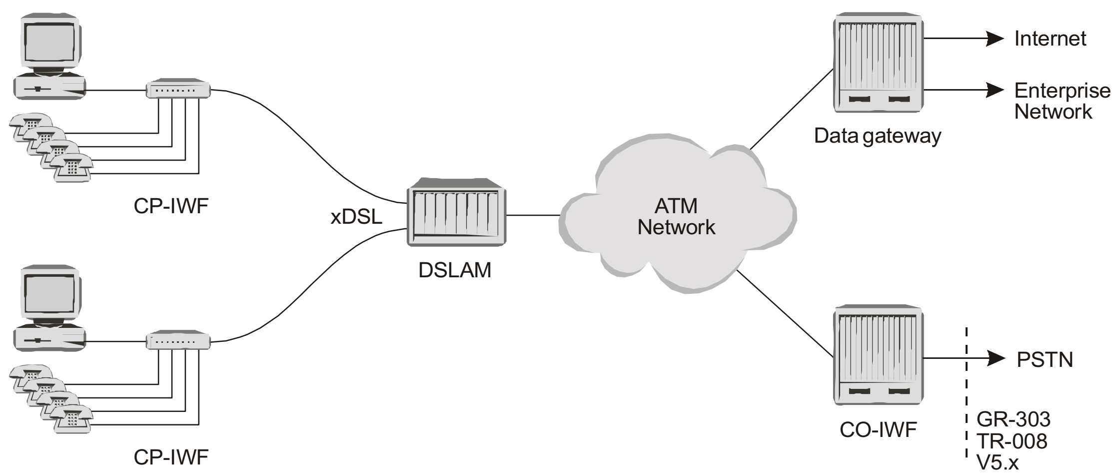

Figure A-2.1: VoDSL Network Model for Discussion of Voice and Signaling Delay

The CP-IWF that is shown in this diagram includes the following functions that are described in the BLES architecture:

- Network Termination (B-NT)
- Customer Premise Interworking Function
- Customer Premise Distribution Network

The CP-IWF also includes additional functions that are related to the provision of data service, including a data interface such as 10BASE-T Ethernet, and a bridging or routing function that supports the movement of data between this interface and the DSL connection.

In this example, data and voice are carried over the same physical DSL connection to the Access Node (DSLAM), but are carried in separate virtual circuits. Voice and data virtual circuits traverse the ATM network from the DSLAM to the CO-IWF and to a data gateway function respectively.

The data link layer operating over the DSL connection in this model is ATM.

The CO-IWF represents a voice gateway that converts between voice over AAL2 and TDM voice, and is connected to a CO Class 5 circuit switch using a digital access network interface such as GR-303 [4], TR-008 [17], V5.1 [5] or V5.2 [6].

A-2.3 Voice Path

A-2.3.1 Upstream Voice Path

A-2.3.1.1 Voice Encoding

Analog voice input to the POTS ports at the CP-IWF is converted to digital form and then encoded as a serial stream whose data rate depends on the encoding method. For many VoDSL systems, the encoded voice stream will have a data rate of 64 kbps, 32 kbps, or 16 kbps.

Voice encoding delay depends on the encoding method. Examples of voice encoding delay are given in Table A-2.1.

Encoding Method	**Encoding Delay**
G.711 PCM 64 kbps	0.75 ms
G.726 ADPCM 32 kbps	1 ms
G.728 LD-CELP 16 kbps	2 ms

Table A-2.1: Voice Encoding Delay

A-2.3.1.2 Voice Packetization

The digitized voice stream is accumulated to create a voice packet. The size of this packet depends on the transport protocol that is used for voice. Examples of packet sizes for VoDSL systems are 20 bytes, 36 bytes, 40 bytes and 44 bytes. The time taken to accumulate sufficient encoded voice to fill a packet (packetization delay) is the first major element of transmission delay in the voice path.

Packetization delay is proportional to packet size and inversely proportional to the data rate of encoded voice. Examples of packetization delay are shown in Table A-2.2.

	Packet Size and Packetization Time			
Encoding Method	20 bytes	36 bytes	40 bytes	44 bytes
G.711 PCM 64 kbps	2.5 ms	4.5 ms	5 ms	5.5 ms
G.726 ADPCM 32 kbps	5 ms	9 ms	10 ms	11 ms
G.728 LD-CELP 16 kbps	10 ms	18 ms	20 ms	22 ms

Table A-2.2: Packetization Delay

A-2.3.1.3 Link Layer Mapping

The voice packet is then mapped to the underlying link layer. The delay introduced by this depends on the nature of this mapping. Mappings that may be seen in real VoDSL systems include the following:

- Simple AAL2 mapping, where one AAL2 packet occupies the entire payload of a single ATM cell. This mapping involves negligible incremental delay.

- Sub-cell multiplexed AAL2 mapping, where a stream of AAL2 packets is packed into ATM cell payloads, with no fixed relationship between packet boundaries and cell boundaries. The delay involved in this mapping depends upon the value chosen for the "combined use" timer in the AAL2 common part process defined in I.363.2 [37]. In this application, the combined use timer will normally be set to a value that is substantially less than the packetization time, in which case the incremental delay due to the AAL2 common part process will be equal, in the worst case, to the value of the combined use timer.

- IP trunking mapping, where multiple RTP packets are packed into a single IP packet payload. The delay involved in this mapping depends upon the degree of synchronization between the RTP packet generation processes on each voice channel.

In the case of IP-based solutions, further processing is required for lower layers of the protocol stack. IP packets are typically carried over PPP over ATM AAL5. This additional protocol processing typically introduces negligible delay.

The incremental delay due to link layer mapping varies between negligible and a value that is just less than the packetization delay.

A-2.3.1.4 DSL Link Queuing

The data link Protocol Data Unit (PDU) – whether it be an ATM cell or a frame relay frame – now has to be queued for transmission on the DSL link. There are two components of DSL link queuing delay. The first component arises from the sharing of the DSL link between voice and data, while the second component is concerned with the queuing of voice packets to be sent over the link.

A-2.3.1.4.1 DSL Link Queuing – Data

Voice packets should normally always be given priority over data packets. Hence if there are both voice and data packets queued for transmission, the queuing algorithm will always select a voice packet for transmission ahead of data packets. This means that data packets are only selected for transmission if there are no voice packets currently in the queue.

However, if a data packet is in the process of transmission at the moment a voice packet arrives in the queue, the transmission of the data packet must continue until it is complete before the transmission of the voice packet can commence.

The sharing of the DSL connection between voice and data therefore introduces a variable delay to the voice path. This delay can vary between zero and the transmission time for the maximum size data packet supported on the link. The transmission time for any given size of data packet depends on the transmission rate of the DSL connection. See Table A-2.3.

	Data Packet Size and Transmission Time		
DSL Line Rate	53 bytes (ATM)	260 bytes (FR)	1504 bytes (FR)
256 kbps	1.6 ms	8.1 ms	47 ms
384 kbps	1.1 ms	5.4 ms	31 ms
768 kbps	0.6 ms	2.7 ms	16 ms

Table A-2.3: Packet Transmission Time or Serialization Delay

A-2.3.1.4.2 DSL Link Queuing – Voice

In addition to the variable queuing delay that may be introduced due to the sharing of the DSL connection between voice and data, there is a further variable delay that may arise from the queuing of voice packets at the DSL link. There will be no long-term build-up of voice packets in the DSL link queue, because the DSL bandwidth available for voice should always exceed the aggregate bandwidth of the active voice channels, but at any given instant there may be multiple voice packets in the queue.

Each active voice channel generates a voice packet at regular intervals. For instance, a voice channel that is generating AAL2 packets containing 44 bytes of PCM-encoded voice will generate a voice packet every 5.5 ms. If all the active voice channels generate packets in phase with one another, then multiple packets will arrive in the DSL link queue at roughly the same instant. Thus these packets may experience a queuing delay that varies between zero and the inter-packet interval.

For example, let us suppose that we have 8 active voice channels generating 44-byte packets containing ADPCM 32 kbps voice at 11 ms intervals, and that we have a DSL link with 384 kbps upstream bandwidth. If all the voice packets arrive in the DSL link queue at roughly the same instant, then the first packet to arrive will start transmission with negligible delay. It will take 1.1 ms to transmit this packet, and then the second packet can start transmission. The eighth and last packet will thus experience a queuing delay of 7 * 1.1 ms, or 7.7 ms.

To minimize the incremental queuing delay that arises from this effect, it is desirable for the voice channels to generate voice packets in a distributed phase relationship with one another.

A-2.3.1.5 Transmission Over DSL

Once a voice packet has reached the front of the DSL link queue, it must be physically transmitted on the link. This process takes appreciable time. Examples of packet transmission times with various packet sizes and link speeds are given above in Table A-2.3.

With ADSL, there is a choice of “fast” or “interleaved” paths for packet transmission. The fast path provides immediate serial transmission of waiting packets, and the packet transmission times given in Table A-2.3 apply. The interleaved path buffers a substantial amount of data, and performs interleaving of this data over a specified interleave depth in order to improve immunity to burst noise on the DSL link. The interleave depth is specified in milliseconds, and the effect of interleaving is to add delay to the voice path equal to the magnitude of the interleave depth.

The interleaved path of ADSL has value primarily for the transport of highly compressed video streams, where relatively low bit error rates can cause substantial impairments to the received picture. The fast path is preferred for both data transmissions and voice over DSL.

The line coding for both fast and interleave paths use Forward Error Correction algorithms that require block processing, and therefore incur fixed delays. G.992.1 defines the ATM payload transfer delay to be (4 + (S-1)/4 +SxD/4) ms, where S is 1 for the fast path and 1, 2, 4, 8, or 16 for the interleave path, and D is 1, 2, 4, 8, 16, 32, 64 for the interleave path (not applicable for the fast path).

Thus the lower bound of delay for ADSL line encoding is 4 ms. The upper bound, using the interleave path, is as much as 263.75 ms (S = 16 and D = 64). Note that G.lite (G.992.2) uses only the interleave path and thus has a lower bound of 4.25 ms.

The ADSL line encoding time is additive to the packet transmission time.

A-2.3.1.6 Transit Through Packet Network

On arriving at the DSLAM, the voice packet will be sent to an uplink that connects the DSLAM into the packet network. This uplink may be ATM or frame-based. To assure satisfactory voice quality, the bandwidth on this uplink must be sufficient to support all active voice calls through this DSLAM. Furthermore, the voice packets must be prioritized over any data packets that are contending for bandwidth on the uplink.

The uplink from the DSLAM into the packet network, and the links between the switches that make up the packet network, generally offer bandwidths that are orders of magnitude greater than that of a DSL connection. As a result, the queuing delays and transmission times experienced by voice packets on these links are typically orders of magnitude less than those seen on the DSL link.

For example, the packet transmission time for a single ATM cell making a single hop on a 45 Mbps DS-3 uplink is 9.4 microseconds. Switching times through ATM switches are of the order of a few microseconds. Propagation delay across a metropolitan area ATM network is also small in relation to other delays in the system. For example, the delay involved in transmission over 25 miles of fiber cable is around 0.2 ms.

It is difficult to add further detail about the delays experienced by voice packets in transit through the packet network between the DSLAM and the CO-IWF, because so much depends on the architecture of the network and the traffic load that exists within it.

A-2.3.1.7 Voice Decoding/Transcoding

When the voice packet reaches the CO-IWF, if the payload of the packet is not PCM then it must first be decoded before it can be converted to a serial PCM stream for onward transmission to the CO switch. For the encoding schemes identified in Table A-2.1 above, decoding delays are typically similar to encoding delays.

A-2.3.1.8 De-jittering

After the payload of the voice packet is converted to PCM format, the PCM data is placed in a FIFO buffer. The output of this buffer is clocked out one byte at a time, at a nominal rate of 8000 bytes per second, and sent to the TDM interface. The purpose of the FIFO buffer is to store the decoded content of the voice packet until it is ready to be sent to the TDM interface, and to compensate for the variability of arrival times of the voice packets. Hence this buffer is often known as the “jitter buffer.”

If packets arrived at exactly equal intervals, then the instant that the packet content has been decoded, the first byte would be ready to send to the TDM interface. The next packet would arrive just as the last byte of the previous packet was being transmitted out of the buffer.

In real networks, voice packets do not arrive at exactly equal intervals because of the variable delays that are experienced at multiple points in the network. Hence the jitter buffer must always contain some slack to allow for the late arrival of a packet, so that the PCM stream sent to the TDM interface remains continuous.

The average amount of slack (or “buildout”) that is present in the jitter buffer represents an additional delay in the voice path. The buildout that is required in the jitter buffer is a function of the variability of delay in the entire voice path, e.g., variation caused by DSL link queuing and transit through the packet network. To ensure clean reproduction of the voice content, the buildout must accommodate worst-case lateness of a voice packet relative to the mean arrival instant.

The jitter buffer in the upstream voice path must allow for the possibility that the voice samples being transmitted from the CP-IWF were generated according to derived network timing, sometimes know as adaptive clocking. With derived timing, the CP-IWF clock may drift slightly relative to the clock at the CO-IWF, although it should track the CO-IWF clock accurately over a long period of time. The jitter buffer in the CO-IWF should therefore be dimensioned to allow for short-term clock frequency deviations as an additional component of delay variation in the upstream voice path.

A-2.3.2 Downstream Voice Path

The downstream voice path contains all the same elements of delay and variability as the upstream path. The only significant difference that may arise between the delay characteristics of the upstream and downstream paths is when the bandwidth of the DSL connection is asymmetric. If there is more bandwidth available in the downstream direction, then the DSL link queuing delays will be correspondingly shorter.

A-2.3.3 Summary of Voice Path Delays

Examples of typical VoDSL application scenarios are provided here to illustrate the total voice path delay characteristics of VoDSL systems.

The tables in the following pages include the following elements of delay:

- Voice encoding and decoding – fixed delay, dependent on coding scheme.
- Voice packetization – per Table A-2.2 above.
- Link layer mapping – zero when one voice packet is mapped to one frame or cell. If multiple voice packets or packet fragments are mapped into a frame or cell, then the link layer mapping delay may be significant.
- DSL link queuing – values for data queuing and voice queuing must be estimated based on probable mixes of traffic and relative arrival times of packets at the DSL link queue.
- DSL transmission time – fixed time, dependent on packet size and DSL link speed.
- Packet network transit – it is very hard to provide accurate estimates of packet network transit since so much depends upon network architecture and loading. For the purpose of these examples, an average delay of 3 ms with delay variation of 1 ms is assumed. Real networks typically demonstrate substantially better performance than this.
- Adaptive clocking allowance – estimated typical value is used.
- Jitter buffer buildout – calculated as sum of delay variations.
- Total delay – the sum of all path delay elements, including jitter buffer buildout.

AAL2 over ATM, 44 byte packets, PCM, 384/384 kbps ADSL, 4 active channels		
Component of Delay	**Average delay**	**Delay variation**
Voice encoding	0.8	0
Voice packetization	5.5	0
Link layer mapping	0	0
DSL link queuing (data)	0.2	1.1
DSL link queuing (voice)	2	2
Serialization Delay	1.1	0
Transmission over DSL (assuming interleave depth of one)	4	0
Packet network transit	3	1
Voice decoding	0.8	0
Adaptive clocking allowance	0	1
Maximum total delay variation		5.1
Jitter buffer buildout	5.1	
Total voice path delay	22.5	

Table A-2.4a: Example VoDSL Voice Path Delay

AAL2 over ATM, 44 byte packets, ADPCM 32 kbps, 384/384 kbps SDSL, 4 active channels		
Component of Delay	**Average delay**	**Delay variation**
Voice encoding	1	0
Voice packetization	11	0
Link layer mapping	0	0
DSL link queuing (data)	0.2	1.1
DSL link queuing (voice)	2	2
Transmission over DSL	1.1	0
Packet network transit	3	1
Voice decoding	1	0

Adaptive clocking allowance	0	1
Maximum total delay variation		5.1
Jitter buffer buildout	5.1	
Total voice path delay	24.4	

Table A-2.4b: Example VoDSL Voice Path Delay

A-2.3.4 Voice Path Delay Limits

The introduction of delay into the voice path through a public telephony network has two kinds of negative impact: audible echo and impairment of conversational interactivity.

A-2.3.4.1 Audible Echo

Audible echo arises most noticeably from the signal reflection that occurs in the line hybrid device, which couples the separate digital paths for each direction to the single pair of wires connected to the telephone instrument. Echo may also arise from the acoustic path between mouthpiece and earpiece in the telephone handset. When the round trip delay (the sum of the transmission delay in the forward and reverse paths) is short, echo is not noticeable to the speaker. But when the round-trip delay exceeds a certain threshold, the echo can clearly be heard, and the speaker finds this disconcerting.

ANSI T1.508 [7] provides a guideline for acceptable delay performance in digital access networks for the purpose of echo control:

> "There is a widely accepted guideline for dealing with incremental delays caused by the introduction of digital technology. The guideline is that any new system, network or component that, by itself, adds more than 5 ms round trip delay should provide echo cancellation."

The examples shown above in Table A-2.4 indicate a range of round trip delay (twice the one-way delay) between about 40 and 100 ms. This is far in excess of the guideline defined in ANSI T1.508 [7], hence echo cancellation is always a requirement for VoDSL systems.

A-2.3.4.2 Impairment of Conversational Interactivity

Even if echo is adequately controlled, excessive voice path delay can have a serious negative impact on the subjective experience of interactive conversation. Let's say we have a voice path between speaker A and speaker B that has a 300 ms one-way delay. Speaker A is talking, and comes to the end of what he has to say. He waits for an answer from B. Speaker B hears the pause at the end of A's sentence 300 ms after A stops speaking. If speaker B starts speaking immediately, then A will hear the start of what B has to say 600 ms after A stopped speaking. During this silent period of 600 ms, A may start to wonder if B is actually going to say something, and may decide that B is really waiting for him to continue. So he starts speaking again. The result is a collision, at which point both speakers back off.

ITU-T G.114 [21] provides a recommendation for limits on one-way delay times to provide acceptable performance for conversational interaction. The recommendation is summarized as follows:

> "One-way delay 0 to 150 ms: Acceptable for most user applications. Note: some highly interactive voice and data applications may experience degradation for values below 150 ms. Therefore, increases in processing delay on connections with transmission times even well below 150 ms should be discouraged unless there are clear service and application benefits."

A-2.3.5 Voice Path Delay Budget

The values for one-way delay given as examples in Table A-2.4 above are all well below the maximum recommended one-way delay of 150 ms defined in G.114 [21]. However, the delay introduced by a VoDSL network is only one component of end-to-end delay in a typical voice call.

There are two additional major components of delay in the voice path in typical telephone call scenarios:

- Propagation delay over copper or fiber cables.
- Access network delay at the other end of the connection: significant if the access network is a wireless network or another packet network.

The propagation delay over copper is 0.0079 ms per mile, and over fiber is 0.0084 ms per mile, according to Telcordia Notes on the Networks [39]. Hence for a coast-to-coast call in the US, the one-way propagation delay would be about 25 ms, while for a call from San Francisco to London, it would be about 50 ms.

The ITU-T recommendation G.114 [21] suggests that a national network incur no more than 50 ms of delay to an international gateway. VoDSL systems that introduce delay of the order of 20-30 ms will typically meet this requirement, if we can assume that the delay from the originating Class 5 switch to the international gateway is no more than around 20 ms.

Voice path delay over public wireless access networks may be as high as 95 ms one-way, depending on the wireless technology used. According to ETSI ETR 275 [38], delays for wireless access networks are as follows:

DECT	14 ms
GSM	95 ms

If assume a one-way delay of 30 ms in a VoDSL access network, a call from a user of this network over a transatlantic connection from San Francisco to London that terminates in a GSM cell phone may experience a one-way delay of 175 ms, which slightly exceeds the maximum recommended by G.114 [21].

A-2.3.6 Voice Path Delay Conclusions

As we have seen, typical VoDSL implementations introduce of the order of 20 to 50 ms one-way delay into the voice path. This amount of delay represents around 15 to 30% of the maximum recommended one-way delay for calls in the public switched telephone network. Other major components of delay, including propagation delay and access network delay at the other end, may be of similar magnitude to the VoDSL delay.

In general, therefore, phone calls made over networks that include VoDSL either at one or both ends will not experience a level of delay that exceeds the ITU-T guidelines. However, it would be undesirable for VoDSL systems to introduce substantially more than 50 ms one-way delay, since this would risk infringing ITU-T guidelines for a number of common call scenarios.

Note that a lower delay contributes to a subjectively better quality of communication, even if the overall delay is well within the maximum value permitted by the guidelines.

A-2.4 Signaling Path

In general, signaling information in a VoDSL system is carried over the same path as voice and is subject to the same magnitude of delay and delay variation. In systems based on AAL2, Channel-Associated Signaling (CAS) is sent in the same AAL2 channel as the voice channel with which it is associated, whereas Common Channel Signaling (CCS) is sent in a single AAL2 channel for all the voice channels that exist on the ATM virtual circuit connection. In both cases, the delay and delay variation experienced by the signaling will be the same as that experienced by the voice channels on the same virtual circuit.

A-2.4.1 Impact of Transmission Errors

DSL connections are subject to interference from various sources of noise in the local loop. This noise generates errors exhibited as bit errors or burst errors due to impulse noise and may result in significant error rates. In systems based on ATM AAL2, the content of the voice packet is not protected by a checksum, and packets with bit errors in the voice payload are therefore not discarded. This is desirable for voice transport, since it is better for overall subjective voice quality to deliver a voice packet with one or two bit errors in it than it is to drop an entire packet that may contain several milliseconds of speech.

Packets that contain signaling information must be delivered without errors; hence the payload of signaling packets is protected by a checksum or redundancy. This means that when a signaling packet is subjected to one or more errors, the system must discard the packet. DSL connections are subject to interference from various sources of noise in the local loop and may be subject to significant impulse noise from electrical equipment such as motors, streetlights, etc. Impulse noise errors may cause the generation of errors lasting several milliseconds.

Therefore the protocol used to transport signaling over the VoDSL system must provide robust mechanisms for handling such packet loss. The design of these mechanisms may have an impact on delay and delay variation.

A-2.4.1.1 Impact of Transmission Errors on CAS

The standard method for CAS transport in systems based on ATM AAL2 is specified in I.366.2 [12]. It involves the triple redundant transmission of "type 3" packets that contain ABCD bit information, each time the CAS state of the voice channel changes.

A type 3 packet with CAS signaling contains 5-bytes, including the ABCD bit values, a timestamp and a checksum. Whenever the line state changes, a type 3 packet containing the new value of the ABCD bits is transmitted 3 times, at 5 ms intervals. A redundancy information field in the type 3 packet indicates whether this packet is the first, second or third transmission on any given state change. Note that a bit error in the cell header will cause the cell to be discarded. So the protected portions of Type 3 packets in a cell consist of 10 octets: 5 for the CAS packet and 5 for the cell header.

The packet containing CAS information is small (only 5 bytes) which means that the probability of bit errors within the packet is much lower than the probability of a bit error in the ATM cell as a whole. The checksum provides the means to determine if a bit error has occurred, and packets with errors are discarded. The redundancy scheme ensures that even if 2 out of 3 signaling packets are discarded, the signaling state change will be propagated without excessive delay, and the timing of the state change can be accurately reproduced by the receiving end using the timestamp information contained in the packet. The timestamp is specified in milliseconds, hence the CAS scheme is capable of reproducing signaling timing with millisecond accuracy even in the face of severe error rates.

The duration of the impulse noise may be about the same as the duration of a triple redundant transmission. Since the CAS methodology has no means of detecting lost packets (there is no timer for re-transmission in the CAS method), once the data is lost, the CP-IWF and the CO-IWF will be out of synchronization with respect to the information being transmitted. I.366.2 defines a refresh period of five seconds for CAS. In the default case, the out of synchronization state may exist for five seconds.

In IP based FRF.11 systems, the impact of bit errors are ameliorated using both redundancy and oversampling. The method is to sample ABCD bits every 2ms for 10 samples in a 20ms period, and to repeat the previous 2 sample sets, all contained in the current packet. This results in 15 bytes of ABCD information, with a 5-byte 'current' sample set and two5-byte redundant sample sets, in a packet. With this high rate of sampling and triple redundancy the receiver of the packet can greatly reduce the probability of error.

A-2.4.1.2 Impact of Transmission Errors on CCS

The standard method for CCS transport in systems based on ATM AAL2 is specified in I.366.2 [12] and I.366.1 [32]. In this scheme, the CCS message has an 8-byte trailer appended to it, which contains packet length and checksum information, and is then segmented into AAL2 packets for transmission over a specific AAL2 channel.

V5 PSTN signaling is the CCS signaling protocol adopted by the ATM Forum.

This protocol provides message formats that support the communication of line state changes from one end of the system to the other. For instance, an off-hook event at a CP-IWF is communicated to the CO-IWF by means of a SIGNAL message with the "steady signal" information element having the value 5 to designate "loop closed."

In general, V5 PSTN messages are quite compact and in all practical situations a single message will fit inside a 40- or 44-byte AAL2 packet, occupying a single ATM cell. Nevertheless, such a message is substantially longer than a type 3 packet containing CAS signaling information and therefore is more likely to experience one or more bit errors than a type 3 CAS packet at any given level of noise on the DSL link. If a CCS signaling message is discarded because of a bit error, the link layer that supports the transport of the CCS signaling messages is responsible for requesting a re-transmission. The link layer used to support the V5 PSTN signaling protocol is the LAP-V5 (Link Access Protocol), which is similar in operation to the LAPD protocol used for the transport of ISDN signaling messages. LAP-V5 specifies a timeout of 150 ms for signaling message acknowledgements, hence a discarded packet will result in the receiving end requesting re-transmission of the message if it does not receive an acknowledgement after 150 ms.

For this reason, CCS-based signaling cannot guarantee to reproduce the timing of signaling state changes to better than 150 ms in VoDSL environments where the bit-error rate is anything other than negligible. Since some signaling state transitions may take place over much shorter timescales than 150 ms (pulse dialing, for example), it is necessary to allow for this in the overall design for end-to-end signaling in the VoDSL system.

If a CCS signaling message is lost due to an impulse noise error, the link layer that supports the transport of the CCS signaling messages is responsible for requesting a re-transmission. For this reason, CCS-based signaling guarantees the transmission of state changes in the IAD even in VoDSL environments where impulse noise is present.

A-2.4.2 Signaling Timing Requirements

Telcordia spec TR-57 [14] provides useful information about the precision with which signaling timing needs to be reproduced by digital access networks in North America. TR-57 [14] is widely accepted as the standard by which VoDSL systems should be judged for their ability to support POTS with acceptable service quality.

Following are some of the requirements specified in TR-57 [14] that relate to signaling timing. In applying these requirements to the VoDSL case, it should be noted that the Central Office Terminal (COT) is an integral part of the CO switch, that the Remote Terminal (RT) is effectively the CP-IWF, and that the VoDSL system as a whole emulates a Digital Loop Carrier (DLC).

The majority of TR-57 [14] signaling timing requirements are concerned with the precision with which short signaling intervals are reproduced. Those requirements that are concerned with total end-to-end delay in the signaling path are relatively relaxed. For instance, the time permitted to signal a trunk seizure in ground start circuits to minimize occurrences of a glare condition is 200 ms.

5.1.10 Reminder Ring
When one 500 millisecond burst of ringing signal is applied to the COT, the DLC must produce one 500 $\pm$ 50 millisecond burst of ringing at the output of the RT.

5.1.11 Distinctive Alerting
When a ringing voltage meeting the requirement of 5.5.2 is applied to the COT in any one of the following nominal 4 or 6 second period patterns, the pattern shall be reproduced at the output of the RT with the duration of the ringing and silent intervals within $\pm$ 50 ms of the duration of the intervals applied to the COT.

5.4.8 On-Hook/Off-Hook Distortion
With the DLC line off-hook caused by a loop closure resistance $\leq R_{dc}$ at the output of the RT, the duration of temporary changes to on-hook at the RT shall be reproduced at the COT with a

tolerance of ± 10 milliseconds. Note: the dial pulsing requirements of 5.4.9 may be more stringent.

5.4.9 Dial Pulsing
Dial pulses at the output of the COT shall be uniform and between 46% and 75% break when dial pulses at 8 to 12 pulses per second in the range of 58% to 64% break are applied at the RT through any loop with a loop resistance ≤ R_{dc}. As a demonstration that this criterion is met, the percent break specified above shall be met at the COT when the following conditions are applied at the input to the RT:
1. 12 pulses per second at 64% break.
2. 12 pulses per second at 58% break.

To determine if a VoDSL system can meet the dial pulse timing requirements, these tolerances on break periods need to be converted into time values:

With a pulse frequency of 12 pps and a 64% break interval, the break period is 53.3 ms, while with a 58% break interval the break period is 48.3 ms. These represent the extremes of break interval into the analog port of the CP-IWF.

The signaling that is reproduced at the CO-IWF needs to exhibit break intervals in the range of 46% to 75% break, equivalent to a range of 38.3 ms to 62.5 ms. The tightest tolerance here is the difference between a minimum 48.3 ms break at the CP-IWF and a permitted minimum break period of 38.3 ms at the CO-IWF, which requires a signaling timing precision of 10 ms. This is the same level of precision as the requirement defined in section 5.4.8 of TR-57 [14].

There are a number of other requirements of TR-57 [14] that relate to signaling timing, but none are more demanding than the examples given here.

A-2.4.3 Signaling Path Delay Conclusions

A-2.4.3.1 CAS-based Signaling

The end-to-end delay for CAS signaling is the same as that for the voice band, and is in the range of 20 to 50 ms as discussed above in section 3.6. This end-to-end delay is within the limits permitted by TR-57 [14] for signaling propagation delay.

The timestamp information contained within the type 3 CAS packets, and the triple redundancy procedure for transmitting CAS packets, ensure that signaling state changes can be reproduced in a VoDSL system by both CP-IWF and CO-IWF with a precision of around ± 1 ms. At this level of precision, the requirements of TR-57 [14] can be met.

A-2.4.3.2 CCS-based Signaling

The end-to-end delay observed with CCS signaling is the same as that for CAS-based signaling, and the requirements of TR-57 [14] for absolute signaling delay can be met.

CCS signaling packets containing V5 PSTN messages do not contain timestamp information. The accuracy of reproduction of signaling timing over a VoDSL network is therefore directly dependent on the delay variation introduced by the network.

A-2.5 Conclusions

The end-to-end delay introduced in the voice path by typical VoDSL implementations falls within an acceptable range, taking into account the limits suggested by G.114 [21] and the allowances that must be made for other components of delay in a range of different call scenarios. The end-to-end delay introduced into the signaling path by typical VoDSL implementations is also within acceptable limits, at least in terms of the requirements for North America defined by TR-57 [14].

Both CAS and CCS systems are subject to disruption under certain error conditions. These conditions are the responsibility of Service Providers to ameliorate.