Reprint

ATM AAL2 protocol efficiency for supporting Voice over DSL

A. Papadopoulos, T. Antonakopoulos and V. Makios

The 8th International Conference on Advances in Communications and Control – COMCON'8

RETHYMNON, JUNE 2001

ATM AAL2 protocol efficiency for supporting Voice over DSL

Athanasios Papadopoulos[1], Theodore Antonakopoulos[2] and Vassilios Makios[2]

[1]*Computers Technology Institute, Riga Feraiou 61, 26100 Patras, Greece*

[2]*Department of Electrical Engineering and Computers Technology,*

University of Patras, 26500 Rio Patras, Greece

Tel: +30 (61) 997 286, Fax: +30 (61) 997 342, e-mail: V.Makios@ee.upatras.gr

Abstract

Voice-over-DSL (VoDSL) technology provides a major boost to competition in local telephony market by greatly reducing the cost of local access. VoDSL leverages the high-speed transmission of DSL technology to deliver multiple telephony circuits plus high-speed data over a single loop. Although the data services delivered over xDSL are usually IP-based, native ATM is the ideal technology choice for offering data and voice services, since it enables multiple priority levels to be specified, allowing the highest priority to voice traffic. The AAL2 protocol has been recently standardized for bandwidth efficient transmission of delay sensitive, low bit-rate voice services. In this paper, we present comparative results using simulations techniques of the efficiency and performance of AAL2, AAL1 and IP/AAL5 to support voice in a DSL environment.

1 Introduction

DSL brings a combination of increased transmission speed and packet-based transport to the existing copper loop access infrastructure, the only universal medium of connectivity for residential and small business subscribers. Currently, the principal application of DSL is for Internet access, hence the majority of traffic seen today on DSL access networks is based on the Internet Protocol (IP). Voice-over-DSL (VoDSL) is a new technology that allows service providers to offer multiple telephone lines over a single subscriber access line in addition to high-speed data transmission services.

The DSL connection to the customer makes use of a packet network, such as ATM, to support voice and data services. ATM is more efficient than other packet-based networks in supporting voice and data since it provides an efficient mechanism for multiplexing data traffic and real-time multimedia services. The most common approach to encapsulate voice over ATM is to use Circuit Emulation Services (CES) based on the AAL1 encapsulation method. Unfortunately, CES does not offer the statistical gains required for maximizing network utilization and it has high overhead. At the same time, Voice-over-IP (VoIP) applications are increasingly being used to provide a cheaper and more flexible means of communication, in the enterprise LAN and also over the Internet. Voice packets over an IP network are usually encapsulated in AAL5 cells to provide low transmission delay.

The AAL2 protocol has been recently standardized by ITU-T [1] and ATM-Forum [2] in order to transfer short variable-length low-rate packets in delay-sensitive applications. AAL2 enables multiple user channels on a single ATM virtual circuit and supports voice compression, silence detection/ suppression, and idle channel removal.

Bandwidth efficiency is the major issue for packet voice over DSL, since voice traffic shares the bandwidth available on a DSL connection with data traffic. In this paper, we evaluate the efficiency of AAL2 for voice transport across DSL networks using a simulation approach. Initially, we present the generic network topology for supporting VoDSL and a concise description of AAL1 and VoIP/AAL5 protocols. Then, we give the general characteristics of the AAL2 protocol. Finally, we determine the maximum number of voice sources of a given bit rate that can be supported on the DSL bandwidth for AAL1, AAL2 and VoIP/AAL5 protocols, under specific QoS requirements.

2 Generic architecture for VoDSL

DSL delivers transmission speeds from some hundred kbps right up to 6 Mbps over a single copper pair. The most important members of the DSL family currently are Asymmetric DSL (ADSL) and Symmetric DSL (SDSL). ADSL provides unequal amounts of bandwidth in each direction of transmission, with a maximum of 640 kbps upstream and 6 Mbps downstream. SDSL is used most often for business customers where greater upstream bandwidth is needed, since it may be capable of delivering as much as 1.544 Mbps in each direction.

A generic network topology for VoDSL is illustrated in Figure 1. DSL service is delivered over conventional copper loops from DSL Access Multiplexers (DSLAMs) in the Central Office. For those customers who receive only data services over DSL, these loops are terminated at the customer premises with a DSL modem or router. For combined voice and data services, the DSL loop is terminated typically by a device that provides integrated voice and data access (IAD).

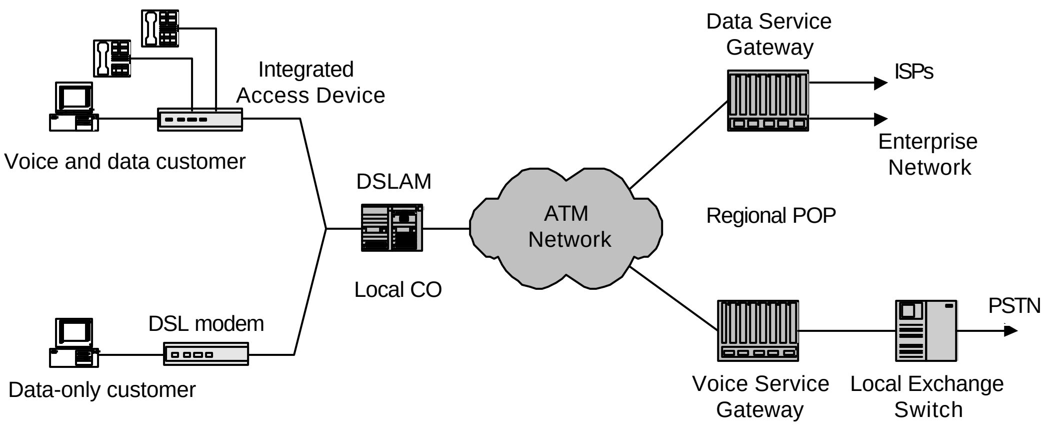


Figure 1: Generic architecture for packet voice and data over DSL

The DSLAM serves as a packet concentrator, delivering traffic from multiple customers over a high-speed uplink to a metropolitan or regional packet network. The principle data service that is offered to DSL customers is Internet access, so the packet network is connected to the Internet, typically through a device known as a Data Service Gateway. Connections to enterprise data networks may also be present.

Voice services are delivered to DSL customers by means of a voice gateway which connects the public switched telephone network (PSTN) to the packet network. Digital voice streams are converted into packet format for transport over the packet network between the voice gateway and the integrated access device on the customer premises. The voice gateway connects to the PSTN via a Class 5 switch. Since the voice gateway represents a digital access network from the point of view of the Class 5 switch, the connection between the gateway and the Class 5 switch typically makes use of a standard interface for digital loop carrier systems, such as GR-303, TR-008 or V5 [3].

End-to-end transmission delay for voice is a critical factor in the perceived quality of the voice service. Packetizing voice introduces substantial high delay. When delay reaches above 25 ms in a voice connection between the integrated access device and the voice gateway, echo canceller circuits are required to control the echo.

3 Voice over ATM solutions

3.1 AAL1 protocol - Circuit Emulation Services

Traditional solutions for voice over ATM have been based on a technique known as Circuit Emulation Services (CES), where a frame structure such as T1 or E1 is converted into a stream of ATM cells that is transmitted at a constant rate (CBR). There are two methods that can be used for the transport of voice traffic over circuit emulation: unstructured and structured CES. Unstructured CES allows the user to establish an AAL1 ATM connection to support a circuit, such as a full Tl (1.544Mbps) or El (2.048Mbps), over the ATM backbone. Structured CES establishes an AAL1 ATM connection to support N x 64 Kbps circuits, such as a fractional Tl or El, over the ATM backbone.

The size of information payload for AAL1 is 47 octets. Therefore, in the case of unstructured CES each cell must wait for 47 sample times (47 octets of voice or data) until it is filled with data. Using Structured CES the cells are partially filled with data and the remainder becomes overhead. There is no standard mechanism in the AAL1 structure for silence detection/suppression and idle channel removal. Bandwidth is used even when there is no traffic. Also, AAL1 does not support compressed voice traffic due to the higher packetization delay as shown in Table 1.

Voice Source Rate (Kbps)	**Packet Fill Delay (ms)**	
	47 bytes payload (98% efficiency)	**24 bytes partial fill (50% efficiency)**
64	5.875	3
32	11.75	6
16	23.5	12
8	47	24

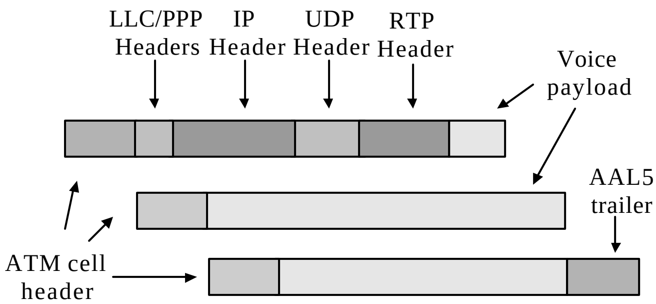
Table 1. Cell assembly delay

In Unstructured CES, an AAL1 circuit that transports six channels requires an ATM connection at a rate of 433 kbps. Using Structured CES with partial cell fill would result in a lower delay, but requires higher bandwidth. If partial fill is used with a fill rate of 24 samples, or octets, the result is an ATM bandwidth requirement of 848 kbps for six channels. Therefore, the use of partial cell fill results in either less channels in a given bandwidth or greater bandwidth requirements for a given number of channels.

3.2 Voice over IP/AAL5

For Voice-over-IP, the most widely accepted protocol stack is based on the Real-Time Protocol (RTP) running over the User Datagram Protocol (UDP) [4]. RTP provides identification of the voice encoding format within the voice packet. UDP provides a port number which enables multiple voice streams to be multiplexed between two IP end-points. IP provides the source and destination address that enables the IP network to switch packets from one end-point to another. AAL5 is the usual ATM adaptation layer for packet-mode data.

In order to control the delay, the voice packet is typically sized to hold 10 ms of voice media stream. Standard uncompressed voice at 64 kbps requires an 80-bytes packet to hold 10 ms of voice. The RTP, UDP and IP protocol headers add respectively 12, 8 and 20 bytes. For transmission over ATM and DSL, a Point-to-Point header (PPP-2 bytes) and a Logical Link Control header (LLC - 4 bytes) have to be added [5], [6]. The 8-bytes AAL5 trailer is then added to the end of the packet, which is then padded so that the entire packet is an integral number of 48-bytes payloads. Figure 2 shows the packet format for VoIP/AAL5.



Field	Bytes
RTP Header	12
UDP Header	8
IP Header	20
LLC/PPP Headers	6
ATM Headers	15
Voice Payload	80
AAL5 Trailer	8
AAL Padding	10
Total	159

Figure 2. Packet format for VoIP/AAL5 connection with 80-byte voice payload

Therefore, in order to transmit 80 bytes of voice it is necessary to transmit a total of 159 bytes (3 cells) including ATM cell overhead. This is equivalent to a bandwidth efficiency of almost 50%. Carrying a 64 kbps voice stream using 80-byte voice packets would actually consume 128 kbps of DSL bandwidth. When a 44-byte voice packet is used with the IP stack, efficiency falls to around 40%.

3.3 AAL2 protocol

AAL2 is the new ITU-T specification to support low-bit-rate and delay-sensitive applications such as voice over ATM. AAL2 is subdivided into a service-specific convergence sublayer (SSCS) and common part sublayer (CPS). The CPS receives the SSCS protocol data unit (PDU) and converts it into a CPS-Packet, which includes a 3-byte header and a variable size 0-64bytes payload with a default size of 45 octets. The CPS-Packets from single or different users are segmented (if necessary) and packed into 47 octets, and converted into a CPS-PDU with a single-octet header.

If an arriving packet cannot fit in the remaining space of the working cell, the bytes that can be fit in are put into the available space, and the cell is dispatched. The rest of the packet is put in the next cell. The CPS-PDU is then mapped into an ATM cell payload at the ATM layer and is transmitted to the remote AAL2 peer entity. Figure 3 illustrates a scenario of voice packets from three sources being packed into cells.

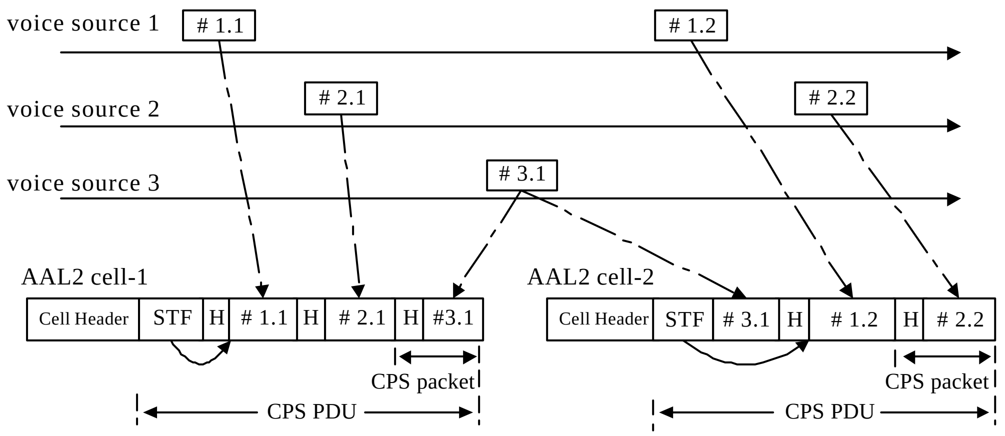


Figure 3: AAL2 cell packing

The format of AAL2 cells is also shown in Figure 3. Every cell has a standard 5-byte ATM header. Following the header is the Start Field (STF - 1 byte) that indicates where the next complete packet starts. For every CPS packet, there is a 3-byte mini-header (H) which includes the Channel ID (CID), Length Indicator (LI), User-to-User Indication (UUI) and Header Error Control (HEC) of the packet. The CID field within each AAL2 header identifies the voice call with which the AAL2 packet is associated. It can identify up to 248 voice connections.

The CPS packet size has to be set appropriately since it significantly affects the link efficiency. Optimum usage of bandwidth can be achieved by ensuring that each ATM cell contains a single CPS packet whose length is chosen to ensure that the CPS packet exactly fills the entire cell payload. The CPS packet payload size that meets this condition is 44 octets. However, operating with this CPS packet size would essentially defeat the purposes of AAL2, which are the multiplexing of many voice channels on a single ATM connection and the voice compression. With a 12-bytes CPS packet payload (15-bytes total size of CPS-Packets), just over 3 CPS-Packets can fit in each ATM cell.

Moreover, a parameter called Timer_CU can be used to avoid high delay in packetizing the cells. Suppose the first CPS-PDU packet is put into a cell and is waiting for the arrival of other packets to complete the cell. When the first packet is put in the cell, a timer is set to Timer_CU parameter value. If the cell is not completely packed within Timer_CU time (no other packets arrive), the cell is sent even though it is partially packed.

4 Comparative results

Simulations results are used to analyze the performance of different VoDSL methods. We compare the efficiency of the AAL2 protocol to AAL1 and VoIP/AAL5 under homogeneous voice sources. Human voice consists of alternating talkspurts and silence intervals. Therefore, each voice source is modeled as an ON - OFF source with independent exponentially distributed ON and OFF times. In a commonly accepted model [7], ON times have an average length of 420 ms and OFF times have an average length of 580ms (42% voice activity). In AAL1 and VoIP/AAL5 cases, we report results without silence elimination.

Bandwidth efficiency is the most critical factor that characterizes VoDSL performance. Thus, we extensively investigate the number of voice sources that can be supported on a certain channel rate for DSL connections with given voice coding rate and speech activity factor. The basic system model for all simulations consists of a number of voice sources that are connected to an access device (IAD). Voice sources send voice packets to the IAD that implements AAL1, AAL2 or IP/AAL5 protocol. The ATM cells go through the DSL link and arrive at the DSL access multiplexer (DSLAM).

Five basic values of bandwidth are used for the DSL connection: 256, 640, 768, 1152 and 1544Kbps. Also, three voice bit rates are considered: 16, 32, and 64 kbps. Table 2 shows the basics parameter values for all VoDSL simulation models.

Parameter	Values
DSL Channel Rate (kbps)	256, 640, 768, 1152, 1544
Voice Coding Rate (kbps)	16, 32, 64
Voice Activity Factor (%)	42, 100
CPS Packet payload (bytes)	8, 16, 20, 32, 40, 44, 50
Timer_CU (ms)	1, 2
VoIP payload (bytes)	80

Table 2. VoDSL simulation parameters

For voice transport applications, AAL1 and VoIP/AAL5 require an ATM virtual channel for each voice source, while at the AAL2 simulation model, voice sources send CPS packets to the AAL2 multiplexer. The simulation tool that was used for defining the most appropriate method is Compuware's COMNET III.

Transmission delay in packet voice is largely a function of voice packet size and voice coding rate. Echo cancellers are usually required when the total 1-way delay in a connection between the customer access device and the voice gateway exceeds 25 ms. Therefore, the cell fill delay in the AAL2 multiplexer must be kept small enough, since other network queuing delays, propagation delay, and coder/decoder delay are also included in the end-to-end delay.

We consider a maximum value of 10 ms for the packetization delay. Then, for each voice bit rate, we run multiple simulations with different CPS Packet Sizes and different numbers of voice sources to find the optimal value of CPS Packet Size. The Timer_CU has two different values: 1 and 2 ms. Figure 4 shows the maximum number of users for each CPS Packet Size when the available bandwidth is 1544 kbps.

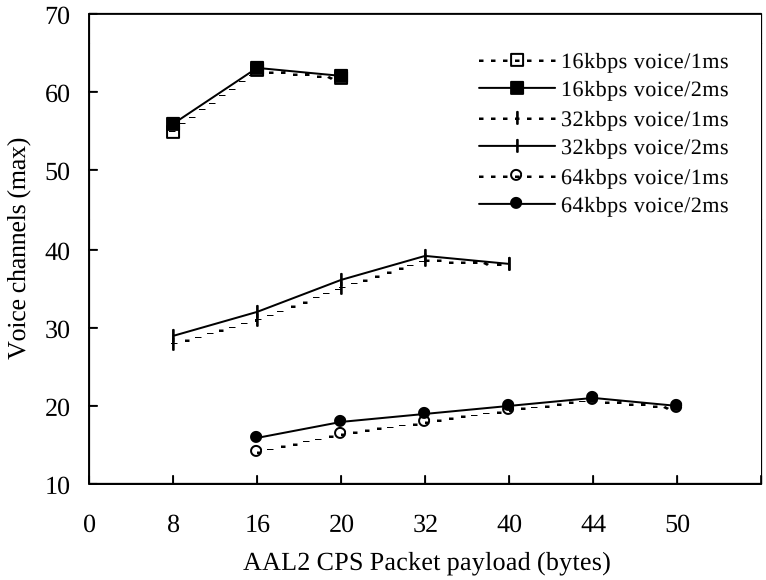


Figure 4. Number of voice channels for different size of AAL2 CPS packet payload

Figure 4 illustrates that the size of CPS packet plays an important role to maximum number of supported voice channels, especially at low voice coding rates. AAL2 with an 8-octet CPS Packet Size can support approximately 56 16-kbps voice users while a 16-octet CPS Packet Size allows approximately 63 16-kbps users. As shown in Figure 4, as the number of the users increases, the Timer_CU value has small effect in the packing process. In AAL2 protocol, the cell fill delay decreases as the number of users increases. If we consider a value of 2 ms for the Timer_CU, then the optimal values of CPS Packet Size for each voice coding rates are 16 octets for 16 kbps, 32 octets for 32 kbps, and 44 octets for 64 kbps voice rate.

Figure 5 shows the number of voice channels that can be carried over different DSL line rates as a function of voice coding rate for the three considered methods. In AAL2 model, we use the optimal CPS Packet Size for each voice coding rate. Using AAL1 structured CES service interfaces for voice support would result to a maximum of 23 voice channels loaded on 64 kbps connections simultaneously. The maximum of 23 voice channels is based on fully filled cells with 47 samples. This is the best case for efficiency using AAL1. For 64 kbps coding, the maximum number of voice channels is nearly the same for all three methods.

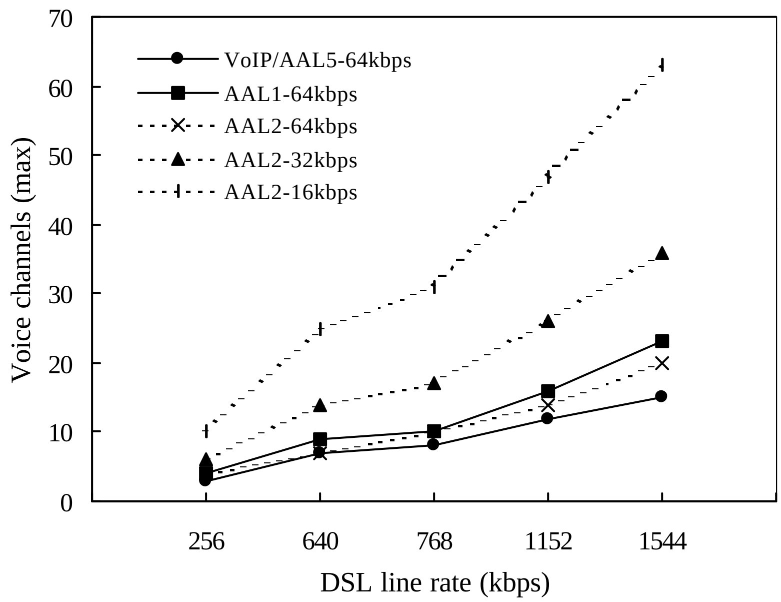


Figure 5. Number of voice channels for different DSL line rates

Actually, Figure 5 shows that using 64 kbps with AAL2 is less efficient than AAL1. However, the advantage of AAL2 emerges as the voice coding rate decreases. The maximum number of voice users with AAL2 increases significantly as the voice coding rate decreases.

Significant improvements in bandwidth efficiency can be gained by not transmitting during the silent intervals, which is feasible using AAL2. Figure 6 illustrates the case where silence suppression is included.

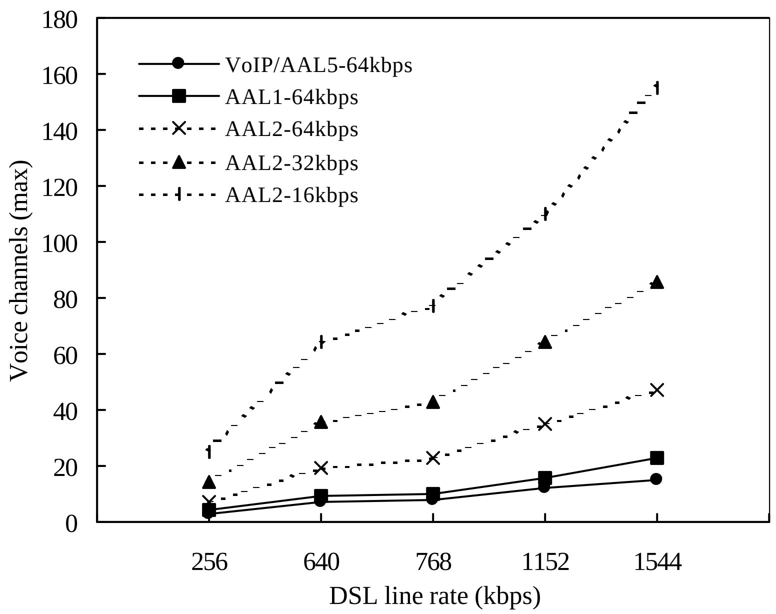


Figure 6. Number of voice channels with silence suppression for AAL2

We assume that a voice source contains 42% silence that can be removed from the AAL2 protocol. Although VoIP can support silence suppression, the bandwidth efficiency gets even worse when it is used. Figure 6 shows that the number of 64 kbps voice channels supported by AAL2 now exceeds the capability of AAL1. These results verify the effectiveness of AAL2 versus AAL1 and VoIP/AAL5 when the silence suppression is included. The number of voice channels supported over a DSL connection with AAL2 32 kbps is almost four times the number supported by AAL1 and VoIP/AAL5.

5 Conclusions

Bandwidth efficiency is a major concern for VoDSL since the bandwidth available on a DSL connection must be shared between voice and data traffic. AAL2 can provide higher bandwidth efficiency than AAL1 and IP/AAL5 protocols, due to its capability of supporting multiple voice sources per ATM connection together with the support of compressed and silence-suppressed voice traffic. Simulation results verify the effectiveness of AAL2 protocol to support packet voice in bandwidth limited DSL connections.

References

[1] ITU-T Recommendation 1.363.2, "B-ISDN ATM Adaptation Layer Type 2 Specification", September 1997

[2] ATM Forum, “ATM Trunking using AAL2 for Narrowband Services”, <af-vtoa-0113.000>, February 1999

[3] ATM Forum, “Voice and Multimedia Over ATM - Loop Emulation Service Using AAL2”, <af-vmoa-0145.000>, July 2000

[4] H Schulzrinne, J Rosenberg, "Internet Telephony: architecture and protocols - an IETF perspective", IEEE Computer Network, pp. 237-255, Feb 1999

[5] T.C.Kwok, “Residential Broandband Architecture over ADSL and G.Lite (G.992.2): PPP over ATM”, IEEE Communications Magazine, Vol. 37, No.5, pp. 84-89, May 1999

[6] G.Gross et al., “PPP over AAL5”, IETF RFC 2364, July 1998

[7] Shuang Deng, “Traffic characteristics of packet voice”, IEEE International Conference on Communications, Vol. 3, pp. 1369-1374, 1995