Assessment of Voice over IP as a Solution for Voice over ADSL
Abhishek Ram†, Luiz A. DaSilva‡ and Srinidhi Varadarajan†
(aram@vt.edu) (ldasilva@vt.edu) (srinidhi@cs.vt.edu)
† Department of Computer Science
‡ Bradley Department of Electrical and Computer Engineering
Virginia Polytechnic Institute and State University
Blacksburg, VA

Abstract – This paper compares VoATM and VoIP in terms of their suitability for carrying voice traffic over DSL. ATM is currently the preferred protocol due to its built-in QoS mechanisms. Through simulations we show that IP QoS mechanisms can also be used to achieve comparable performance for voice traffic over DSL. Our performance metrics are the ETE delay of voice packets across the DSL access network and the bandwidth requirements of a voice call. We also propose an implicit signaling mechanism to provide admission control for individual voice calls over DSL. We implement a simulation model that uses our mechanism and perform simulations to verify its effectiveness. We conclude that by incorporating appropriate QoS mechanisms and implicit signaling, it is possible to achieve performance for VoIP comparable to that provided by ATM. In this case, the ubiquity of IP makes it a very attractive candidate for future deployments of VoDSL.

I. INTRODUCTION

Digital Subscriber Line (DSL) [1] refers to a collection of technologies used for the transmission of high-speed data over copper wires. Asymmetric Digital Subscriber Line (ADSL) is the most widely deployed DSL technology. ADSL provides higher bandwidth from the network service provider to the subscriber (downstream) than from the subscriber to the network service provider (upstream). The Customer Premises Equipment (CPE) is connected by ADSL to a DSL Access Multiplexer (DSLAM) located in the Central Office of the network service provider. The DSLAM aggregates traffic from different customers and sends it over a high-speed link towards the core of the network. In addition to data, DSL can also be used to deliver voice traffic over the same copper wires.

The current trend is to use Voice over ATM (VoATM) technology for carrying voice over DSL (VoDSL) [2,3,4]. Voice and data are carried over different virtual circuits (VC) from the CPE to the DSLAM. Voice is carried directly over ATM Adaptation Layer 2 (AAL2) using Constant Bit Rate (CBR) service. Data is carried over IP ATM Adaptation Layer 5 (AAL5) using Unspecified Bit Rate (UBR) service. At the DSLAM, AAL2 voice traffic is switched to a voice gateway and AAL5 data traffic is switched towards the core IP network. Through the use of ATM's signaling mechanism, voice flows that get admitted are guaranteed a portion of the bandwidth and hence experience low delays across the network.

Another technology for carrying voice traffic over DSL is Voice over IP (VoIP) [2,5]. In this case, both voice and data packets are carried over IP from the CPE to the DSLAM, which then routes them towards the core. Voice packets are encapsulated by Real Time Protocol (RTP) and User Datagram Protocol (UDP) before being sent to the IP layer. The Layer 2 protocol may be ATM, Frame Relay or Serial Line IP (SLIP). To avoid long packetization delays and long transmission times over low-speed links, it is best to keep the payload size of voice packets small. RTP, UDP, IP and the Layer 2 protocol together impose considerable overhead on these small packets. Solutions such as header compression [5] and IP trunking [2] help alleviate the bandwidth problem. VoIP may use IP QoS mechanisms [6,7,8] to provide quality of service to voice packets [12,13,14,15].

Using VoIP in both the core and the access network paves the way for end-to-end IP telephony. Hence, if IP QoS mechanisms could be used to improve the performance of VoIP and make it comparable to that of VoATM, then VoIP would be the preferred VoDSL technology. In this paper, we make a case for using VoIP for VoDSL. We first compare the performance of VoATM and VoIP. Then, we study the improvement in VoIP performance using IP QoS mechanisms. We also propose our own admission control based on implicit signaling. Section II describes the simulation models and the metrics used for comparing VoIP and VoATM. The experiments performed for the comparison are discussed in Section III. Section IV discusses the need for admission control in providing QoS for voice flows and proposes an implicit signaling mechanism to perform admission control. Conclusions and the scope for future work are discussed in Section V.

II. SIMULATION MODEL

![Figure 1: Simulation Model](image)

The system of interest in our study ranges from the end-hosts at the customer premises to a traffic sink located just beyond the DSLAM. As shown in Figure 1, the key components of this system are the end-hosts, the CPE, the...
The ADSL link, the DSLAM and the traffic sink. The metrics used to evaluate the performance of voice traffic are the end-to-end delay of voice packets (ETE Delay) across the access network and the bandwidth required for the voice calls.

The ADSL link in our simulations has an upstream bandwidth of 128Kbps and a downstream bandwidth of 1.5 Mbps. The upstream link is the bottleneck. For the VoATM scenarios, the end hosts consist of Ethernet-based clients generating data traffic and AAL2-based clients generating voice traffic. The CPE has an Ethernet interface for data traffic and AAL2 interfaces for voice traffic. The interface connecting the CPE to the DSLAM is ATM-based. The DSLAM is an ATM switch that switches voice and data packets to their respective destinations. In our model, these destinations are traffic sinks connected directly to the DSLAM. For the VoATM scenarios, the end-hosts are all Ethernet based clients that can generate both voice and data traffic. The CPE is an IP router, which has an Ethernet interface for the all the traffic originating from the end-hosts. The interface connecting the CPE to the DSLAM could be ATM, SLIP or Frame Relay. The DSLAM is another IP router that routes the voice and data packet towards the traffic sinks.

We used OPNET Modeler™ for our simulations. OPNET Modeler™ supports several existing IP and ATM QoS mechanisms. However, in modeling our proposed admission control mechanism, we added additional functionality to the IP layer at the CPE and to the interface between the application layer and the transport layer at the end-hosts. The simulation time in our experiments was chosen such that the system was in steady state for more than 90% of the time. This was determined by conducting a few preliminary simulation runs.

### III. COMPARISON OF VOATM AND VOIP

In all our experiments, voice calls use G.729 encoding. One end-host on the customer premises generates FTP traffic by uploading a 2 MB file. This FTP traffic competes with voice traffic for the upstream bandwidth. The experiments and the results are summarized in Figure 2.

![Figure 2: Comparison of VoATM and VoIP](image)

In the first VoATM experiment, both voice and data receive UBR service. The FTP application generates traffic that competes with the voice calls for the upstream bandwidth. ETE Delay of the voice traffic is very high as a result. The same experiment is repeated with voice traffic receiving CBR service and data traffic receiving UBR service. Since ATM’s built-in signaling and admission control mechanisms ensure that the network does not over-commit the available resources, there is call blocking. The admitted voice calls experience low ETE delays and low delay variation.

The next set of experiments study VoIP with ATM as the Layer 2 protocol. Again, the FTP application generates traffic that competes with the voice traffic. There is severe congestion at the CPE and the voice packets suffer considerable ETE delays when no QoS mechanisms are used. We then investigated the improvement in performance resulting from the use of IP QoS mechanisms. The Weighted Fair Queuing (WFQ) [9] algorithm is implemented at the CPE for sending packets over the bottleneck upstream link. Voice and data packets are marked with different Type of Service (TOS) values and placed in separate queues. The bandwidth allocated to the voice queue is sufficient to support the number of calls in progress during the experiment. Results show that the end-to-end delay is greatly reduced with the introduction of WFQ.

However, the ETE delays of the packets are still high compared to those for VoATM. The end-hosts are all Ethernet-based, so the default MTU for their output interfaces is 1500 bytes. A 1500 byte IP packet takes approximately 94 ms to be transmitted over the 128 Kbps upstream link. Voice packets that arrive during the transmission of a data packet have to wait for an unacceptably long time in the queue. IP layer fragmentation of large data packets is a possible solution to this problem, but it also results in an increase in protocol overhead, since the IP header is replicated in each fragment.
A better solution is fragmentation and interleaving of packets at Layer 2 [5]. Multilink-PPP (MP) [10] provides one such solution. We focus on the improvement in performance achieved by fragmentation at the IP layer. The results for voice traffic would be similar if link layer fragmentation were used. We fragment large data packets by setting the MTU of the upstream interface at the CPE to 328 bytes. Packets of this size fit exactly into 7 ATM cells and need only 23 ms for transmission over the 128 Kbps upstream link. The results show that incorporating WFQ and IP layer fragmentation makes ETE delay of VoIP comparable to that of VoATM.

We also studied VoIP with SLIP as the Layer 2 protocol. Due to the reduced bandwidth requirements of a voice call in this case, we generate more voice calls but reserve less bandwidth as compared to the experiment with ATM at Layer 2. The ETE delays experienced by voice packets with SLIP at Layer 2 are slightly lower than what was observed in the other experiments. In the VoATM scenario, voice packets are carried directly over AAL2 and encapsulation of voice packets by AAL2 is a complex process [11]. Apart from this, the AAL2 PDU may undergo segmentation and reassembly. The processing delay of these operations might be causing the ETE delay for VoATM to be slightly higher than that with VoIP (SLIP at Layer 2). Figure 3 summarizes the results of this section.

IV. ADMISSION CONTROL

The IP QoS mechanisms considered so far do not perform any form of admission control for voice flows. In a situation where too many flows belonging to a certain class compete for limited bandwidth reserved for that class, none of the flows gets a satisfactory level of service. Admission control is thus necessary to provide guaranteed service.

End-point admission control [16,17,18,19] is an approach where the end-hosts themselves make admission decisions. The end-hosts first send probe packets into the network to measure the quality of service that these packets receive. Based on these measurements, they decide whether or not to admit the flow. Another approach is to have the network make the admission control decisions [20, 21, 22, 23]. This is typically done by means of a signaling mechanism to carry the reservation request from the end-host to the network. The network checks the resources requested against the resource availability and the flow is admitted if there are sufficient resources.

In this paper, we propose an implicit signaling mechanism by means of which the CPE can admit individual voice calls originating from the end-hosts and provide a guaranteed level of service to those calls. We classify voice calls into two categories - regular voice calls and premium voice calls. Our admission control mechanism does not reject any voice calls as such. Instead, it admits only a limited number of voice calls as premium voice calls and admits the rest as regular voice calls. Thus, we do not allow the arrival rate to exceed the service rate for premium voice calls. This guarantees low ETE delays for these calls. On the other hand, there is no limit to the number of voice calls that can be admitted as regular calls. Hence, these calls may receive satisfactory service at light loads, but the quality degrades as the load increases. The CPE needs to store some information for each voice call admitted in the premium category. It uses a hash table for this purpose, with the flow ID of the voice call as the hashing function. If we assume that each end-host runs not more than one voice call, then the flow ID would comprise just the source and destination IP addresses. The number of simultaneous voice calls at each subscriber is

<table>
<thead>
<tr>
<th>Scenario</th>
<th># Voice calls generated</th>
<th>Upstream b/w resv for premium voice (Kbps)</th>
<th># Voice calls admitted as premium</th>
<th>Mean ETE Delay for regular voice (s)</th>
<th>95% Conf Interval for regular voice (s)</th>
<th>Mean ETE Delay for premium voice (s)</th>
<th>95% Conf Interval for premium voice (s)</th>
</tr>
</thead>
<tbody>
<tr>
<td>VoIP - SLIP@L2</td>
<td>4</td>
<td>-</td>
<td>-</td>
<td>0.0509</td>
<td>(0.0483, 0.0535)</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>No Admission Ctrl</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>VoIP - SLIP@L2</td>
<td>4</td>
<td>35</td>
<td>2</td>
<td>0.0658</td>
<td>(0.0600, 0.0717)</td>
<td>0.01355</td>
<td>(0.0135, 0.0136)</td>
</tr>
<tr>
<td>With Admission Ctrl</td>
<td>4</td>
<td>35</td>
<td>2</td>
<td>0.0658</td>
<td>(0.0600, 0.0717)</td>
<td>0.01355</td>
<td>(0.0135, 0.0136)</td>
</tr>
</tbody>
</table>

Figure 3: Mean ETE Delays for various VoDSL scenarios

Figure 4: Performance of the Admission Control Algorithm
remarked to indicate that it belongs to a premium call and removed from the list of premium calls. The packet is the hash table is reset to indicate that the call has been a premium call. If so, then the bandwidth availability is checks the hash table to see whether the packet belongs to "Invite" messages. When the CPE receives this message, it the same pieces of information as the code used for the marks the "Bye" message, used by SIP to establish the session, with a special code in the Type of Service (TOS) field. The TOS octet encodes the following pieces of information: (i) whether the packet is an application layer signaling message, and if so, what kind of message it is; (ii) the voice encoding scheme; and (iii) the number of voice frames included in a packet. These TOS codes allow the CPE to distinguish between packets that carry application layer signaling messages and those that carry voice payload. The CPE can also figure out the bandwidth requirement for a voice call using items (ii) and (iii) above. Even if silence suppression were used, the CPE would at least get an upper bound of the bandwidth requirement. Upon receiving this message, the CPE checks the bandwidth requirements for this call against its bandwidth availability. If sufficient bandwidth is available, it adds this call to the list of premium voice calls and updates its bandwidth availability to reflect the admission of the new call. In the hash table entry for this call, it sets a flag to indicate that the call has been admitted as premium. It then remarks the TOS field to indicate that this packet belongs to the premium category. If sufficient bandwidth is not available, the CPE simply remarks it as belonging to a regular voice call and forwards it.

After the end-host has established a session, it starts sending voice packets. The end-host does not know whether its session has been classified as premium or regular, since the admission control procedure is transparent to the end-hosts. For each voice packet received, the CPE checks the hash table to verify if the packet belongs to a premium call. If so, the voice packet is remarked to reflect this and then forwarded towards the DSLAM. Otherwise, the voice packet is remarked as belonging to the regular category.

When the end-host wishes to end the voice call, it marks the "Bye" message, used by SIP to close the session, with a special code in the TOS field. This code also carries the same pieces of information as the code used for the "Invite" messages. When the CPE receives this message, it checks the hash table to see whether the packet belongs to a premium call. If so, then the bandwidth availability is increased to reflect the closing of this session. The flag in the hash table is reset to indicate that the call has been removed from the list of premium calls. The packet is remarked to indicate that it belongs to a premium call and forwarded. If the call to which the "Bye" message belongs is not a premium call, then the CPE simply remarks the packet as one belonging to a regular voice call and forwards it. This also takes care of duplicate "Bye" messages that might have been sent by the same call. In such cases, the CPE would not mistakenly update its bandwidth availability twice, since the second time the call would no longer be in the list of premium calls.

It is possible for a premium voice application running on the end-host to crash without sending the "Bye" message indicating the end of the session. To de-allocate the bandwidth that had been reserved by the premium call, our mechanism uses the following implementation of soft state. For each premium call, the CPE keeps track of the time when a packet from the call last arrived. There is a maximum silence period defined for premium calls. If a particular premium call has not sent a packet for a period longer than the maximum silence period, then the call can be preempted from the list of premium calls.

An additional feature of our resource management scheme is the upgrade of existing voice calls from the regular to the premium category if resources become available during a call. If there is more than one regular call in progress and there are enough resources for only one, then our scheme selects one of the regular calls at random and upgrades it to a premium service. The upgrade feature requires that the end-hosts use the TOS field to encode the bandwidth requirements of the call in not just the "Invite" and "Bye" packets, but in each packet carrying voice payload.

We incorporated our resource management scheme in the VoIP scenario (SLIP at Layer 2) with WFQ and IP Fragmentation and studied the resulting improvement in performance. The results are shown in Figure 4. In the scenario with no admission control, there is no concept of premium and regular calls - all calls are treated the identically (we call them regular calls in Figure 4). 50% of the upstream bandwidth is reserved for voice traffic. There are four voice calls in progress. With G.729 encoding and 2 voice frames per packet, each voice call requires 19.2 Kbps. Hence the voice traffic load exceeds the bandwidth reserved for voice. As a result, we see that all voice calls experience high ETE delays. In the scenario with admission control, 35% of the upstream bandwidth is reserved for premium calls and 15% for regular calls. The bandwidth reservation allows up to two voice calls to be admitted as premium calls. The other two voice calls are admitted as regular calls. Note that the load offered by premium calls does not exceed the reserved bandwidth whereas the load offered by regular calls is allowed to do so. The packets belonging to the premium voice calls experience ETE delays much lower than those belonging to the regular voice calls. Thus our admission control
mechanism is able to protect premium calls. Figure 5 illustrates the effectiveness of our mechanism.

![ETE Delay (s) vs Scenario and Call Type](image)

**Figure 5: Effect of Admission Control**

V. CONCLUSIONS AND FUTURE WORK

In this paper, we compared the suitability of VoATM and VoIP for carrying voice over DSL. We found that when appropriate QoS mechanisms are used, their performance - in terms of ETE delay and bandwidth requirements - is comparable. Given the predominance of IP networks, this result clearly shows the suitability of VoIP for VoDSL, which enables end-to-end IP telephony. We recommend the use of a lightweight protocol such as SLIP, PPP or Frame Relay at Layer 2 of the VoIP protocol stack. Our results show that the use of ATM at Layer 2 does not result in any gain, and only causes poor bandwidth due to increased overhead. In this paper, we also proposed a simple admission control mechanism to limit VoIP calls, avoiding inter-call interference. Our mechanism consists of an implicit signaling protocol between the end-hosts and the CPE. Results in the paper show significant reduction in ETE delays due to the admission control mechanism.

More detailed results of the use of implicit signaling for admission control combined with IP QoS for VoDSL, including different traffic mixes, are described in [23].

Future research plans include investigating the performance of data traffic over DSL in the presence of competing voice flows, to determine how TCP utilizes to the residual bandwidth of the link. We recognize that our implicit signaling mechanism is specifically suited to our needs, i.e. admission control for voice calls over DSL. A possibility for future work is to make the mechanism more general and to devise an end-to-end solution based on it. The coexistence of voice, video and data traffic over DSL is another issue that could be investigated further.

REFERENCES


