# AC 2009-192: A VOICE OVER IP INITIATIVE TO TEACH UNDERGRADUATE ENGINEERING STUDENTS THE FUNDAMENTALS OF COMPUTER COMMUNICATIONS

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# A Voice over IP Initiative to Teach Undergraduate Engineering Students the Fundamentals of Computer Communications


## Abstract

The purpose of this paper is to disseminate simple strategies to adapt undergraduate laboratories on computer networks to the teaching of Voice over IP (VoIP) protocols. Teaching a new technology and updating our curriculum with VoIP was our main goal initially. From the response of our students to this VoIP initiative, we have learned that we are not only introducing our students to a new technology but we also are helping them to better understand basic concepts of computer communications.


### 1 - Introduction

Video-conferencing and voice over IP (VoIP) phones are popular among young and old. At home or work, VoIP has become a cost-efficient way of making phone calls. Is VoIP[1] a telephone service, or a data application? The answer: both. In other words, VoIP applications combine requirements of traditional telephony and data applications. Therefore, they encompass all aspects of communication protocols, such as call signaling, routing, and quality of service (QoS). This combination offers a great opportunity to advance the knowledge and understanding of communication protocols to our undergraduate engineering students.

To introduce VoIP to our students, a VoIP initiative has started, in which new labs are being added to a class on Local Area Networks and to follow-up classes:

- One of the labs has the objective to show the differences between real-time (UDP-based) traffic and non-real-time (TCP-based) traffic. Using a network analyzer tool, students inspect the VoIP packets and all the signaling needed to make the calls, while other applications such as web browsing generate TCP-based traffic.

- Another lab allows students to experiment with different audio encoders/decoders (codecs) and verify their bandwidth requirements, including the overhead added by underlying communication protocols.

- To combine the student's interest in wireless networks and VoIP, a third lab allows them to implement a VoIP system over a point-to-point wireless link. By generating impairments in this wireless link, the students verify the performance of several audio codecs.

Initial results of our VoIP initiative were presented in a conference sponsored by the National Science Foundation (NSF), and organized by the National Center for Information and Communications Technologies (ICT)[2]. In addition, related work to this VoIP initiative was a student project[3] in which we combined network simulation and laboratory experiments in a network modeling and simulation class.

In this paper, we first provide an overview of the Electronics and Telecommunications Engineering Technology (EET/TET) program at Texas A&M and more specifically we address our work in one of our classes: ENTC 315, which is a class on Local and Metropolitan Area Networks. The following section describes the laboratory experiments including the equipment needed for the experiments and their network setup. Some feedback from our students is also presented. Section 4 presents additional laboratory experiments that are being adopted in a more advanced telecommunications class (ENTC 345). Finally, we conclude this paper by evaluating this VoIP initiative and discussing future plans to keep integrating VoIP into our curriculum.

## 2 – Moving towards IP-based communications

The environment where this VoIP initiative is taking place is the Electronics and Telecommunications Engineering Technology program at Texas A&M. This four-year engineering program offers several courses on telecommunication networks. One of them is a class on Local and Metropolitan Area Networks (ENTC 315), which is a required class to both electronics and telecommunications engineering majors. In addition, ENTC 315 is a required class to students from the Liberal Arts College, which offers a major on Telecommunications Media Studies (TCMS). This mix of engineering and non-engineering students varies from semester to semester, as shown in Figure 1.

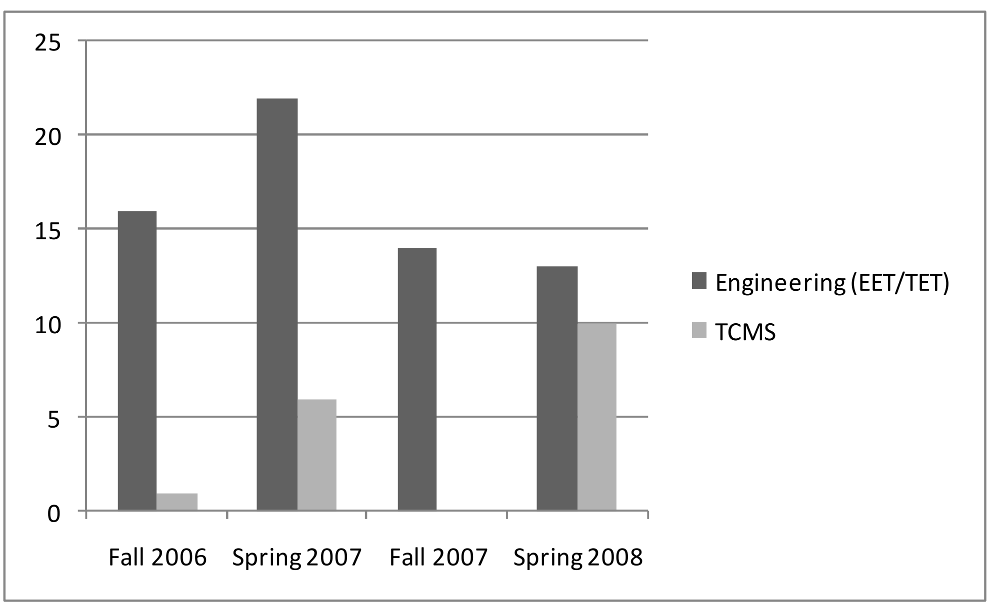


**Figure 1 - Sample distribution of students in ENTC 315.**

In order to keep both engineering and non-engineering majors motivated and engaged in the lectures and labs, classes such as ENTC 315 need to be constantly updated. But one might ask: why voice over IP and not other new technology? First, from a teacher's perspective, our efforts are to modernize the telecommunications classes to teach packet-based communications (including real-time voice and video) (Figure 2). Internet Protocol (IP)-based communications has had great penetration in the telecommunication markets. There is a need in the industry for students, and recent developments have shown VoIP applications for emergency communications[4] such as in Next Generation-9-1-1 (NG-9-1-1). Additionally, this VoIP initiative has the potential for undergraduate research[5], by allowing undergraduate students to replicate research paper's experiments in the laboratory.

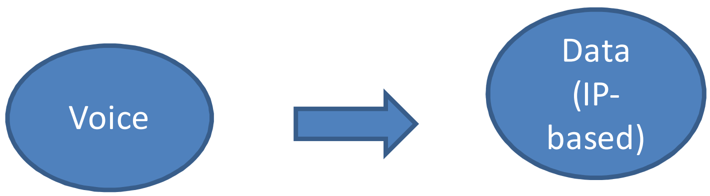


**Figure 2 - Changing the emphasis of the telecommunications program.**

Now, from a student's perspective, would the teaching of VoIP make the lectures and laboratories more engaging? We do not know the answer yet. We have not formally assessed the results of our VoIP initiative, but we have been recording students' expectations in their first day of classes. Basically, this is a way for us to learn about their interests and to put the topics that will be covered in the semester in a context in which they will relate to their interests. Thus, we ask every semester for the students to write down (anonymously) what are the things they find most curious about computer networks and the Internet. In other words, what are the things that they really want to understand well about how computers communicate.

The answers from this "first-lecture survey" vary to a certain degree. After collecting the results of five semesters (Fall 2006 to Fall 2008), we were able to classify their interests into five main areas. Questions related to how the networks interact, or about how packets find their way to destination, or about how large the Internet is and how efficiently it works were classified in the *IP/Routing* category, which received 46.5 percent of the answers. Questions related to home networks, Ethernet, and Wi-Fi were classified as *LAN/Wireless LAN*, which received 26 percent of the answers. *Security* is another major category (with 12.5 percent of the answers) and it includes questions about firewalls, encryption mechanisms, and wireless LAN security. Interest in social networks, file sharing, and differences in web browsers were classified as *Web Applications*, which received 7.5 percent of the answers. Other questions related to other protocols in the OSI layers, such as the difference between TCP and UDP or other link-layer technologies, received about 7.5 percent of the answers. These results are summarized in Figure 3.

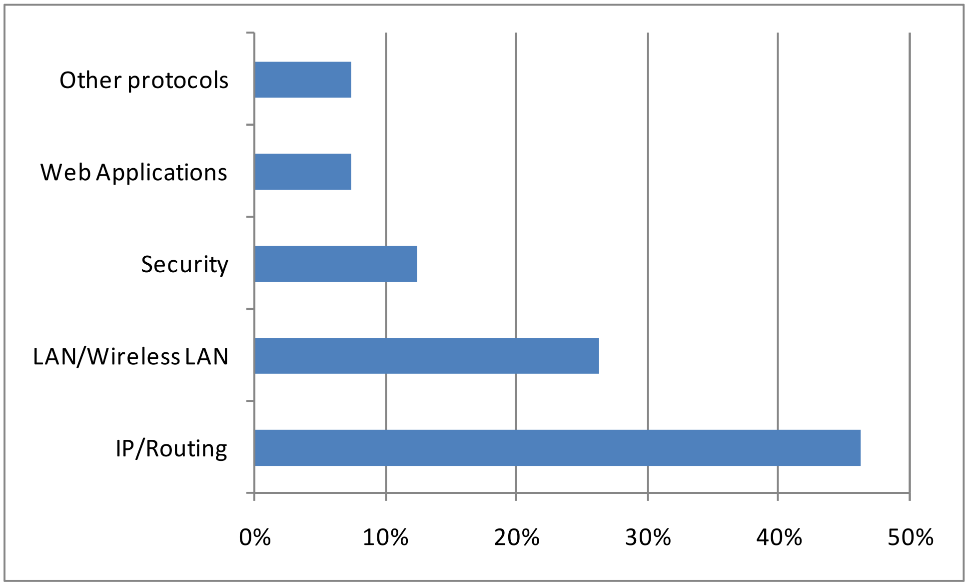


**Figure 3 - Measurements of student interest in network topics (from Fall 2006 to Fall 2008).**

On the category of *IP/Routing* there were a few questions from students about the streaming of voice and video over the Internet. Thus, the topic of VoIP shows potential for engaging students on the understanding of IP and routing protocols, as well as other categories such as wireless LANs (e.g., VoIP over Wireless LANs) and security (e.g., VoIP security issues). They represent innovative areas and in some cases are still being researched.

## 3 - First laboratory on Voice over IP

In this section, we present our open-source approach to implement a VoIP soft-switch, the infrastructure of our laboratories, and the objectives and tasks of our first VoIP lab in the local and metropolitan area networks class (ENTC 315).

### 3.1 – An open source approach

Asterisk[6] is an open source program that implements not only the functionality of a Session Initiation Protocol (SIP)[7], but also the features of a class-5 Private Branch Exchange (PBX) system. With Asterisk, you configure all the hardware and most of the software yourself. It has features like voicemail, call waiting, call forwarding, and many others. It supports numerous protocols and codecs, and translates between different protocols and codecs if needed.

Asterisk, by itself, is a Linux-based system, where everything is created and written in Linux. For the non-Linux users, there is a graphic-based Asterisk called Trixbox that uses Linux, but can be accessed by Windows, or another graphic-based operating system, via the internet using a web graphic user interfae (GUI). This is done by entering the IP address of the computer with Trixbox into the address line of the browser.

## 3.2 – Laboratory setup

The laboratory is set up with six workstations each with three computers and four of the six having two VoIP phones. The phones and computers are connected through a router to a firewall and to the proxy, which is where the Internet is accessed.  To ensure all the phones can talk to the Asterisk server, we placed Asterisk in the same network as the firewalls.

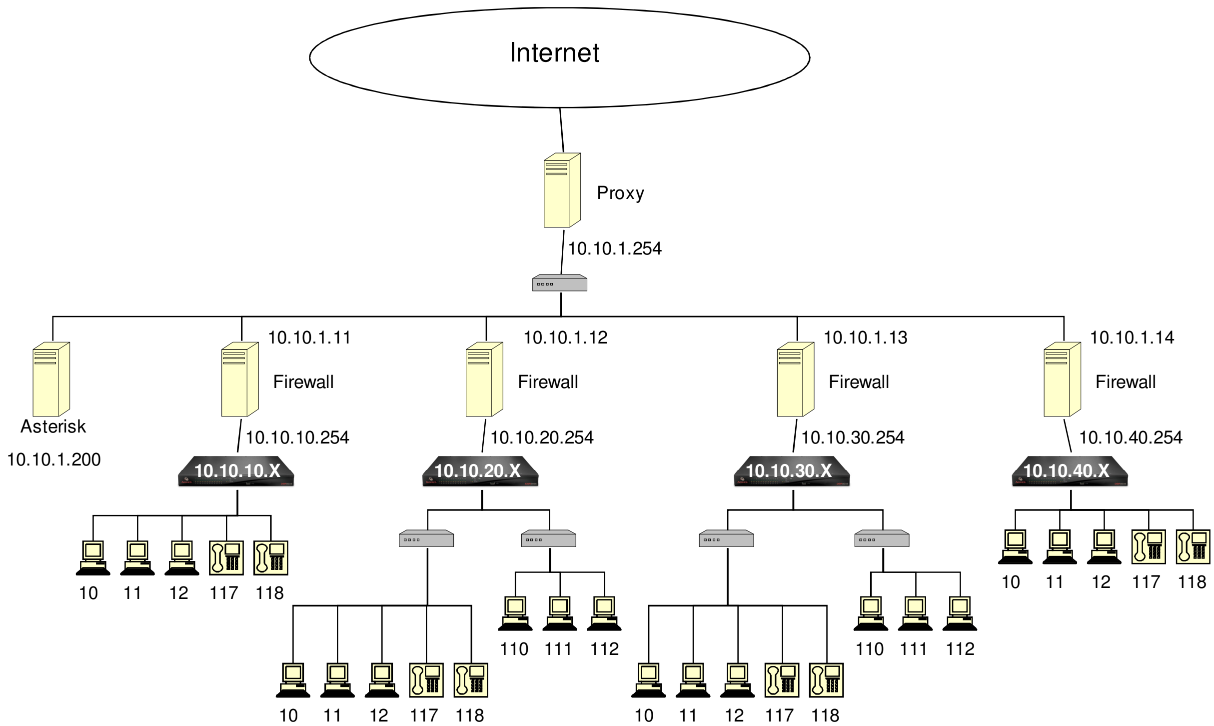


**Figure 4 - Network architecture.**

Additional software that is needed in each workstation is a packet network analyzer. We are using Ethereal (also known as Wireshark) to allow students to observe the packet flows and examine the signaling packets as well as the real-time protocol (RTP) packets which carry the audio samples. In addition, Ethereal has special features to analyze statistics of VoIP flows.

These are the current laboratory experiments in our ENTC 315 class:

Lab 1: Cisco and Ethernet cables; Physical layer
Lab2:  Introduction to Cisco 2900XL Switch Configuration and VLANs
Lab3: General Router Configurations, Point-to-Point Serial Connections
Lab4: Ethereal network analyzer
Lab5: DCHP
Lab6: IP Routing (RIP)
Lab7: IP Routing (OSPF)
Lab8: VOIP

As one can see, the VoIP was added as our last experiment. The main reason is that we are using

VoIP application to demonstrate the difference between UDP and TCP transport (OSI Layer 4) protocols. Our semester basically introduces the OSI layers protocols from the ground-up, starting from the physical layer (Layer 1) to the transport layer (Layer 4).

During the VoIP lab, the students learn how to configure the VoIP phones and how to make them communicate with the Asterisk server. This is done by configuring the IP address, subnet mask, default gateway, and entering the SIP server information into account 1, or line 1 of the phones via the web interface (Figure 5). Once the phones are registered to Asterisk, the students can call one phone with the other and capture the traffic with Ethereal. The students are then asked to differentiate between the TCP packets and the UDP packets, and draw the flow chart of the SIP call signaling message (e.g., caller requests a call, callee receives request and acknowledges, etc).

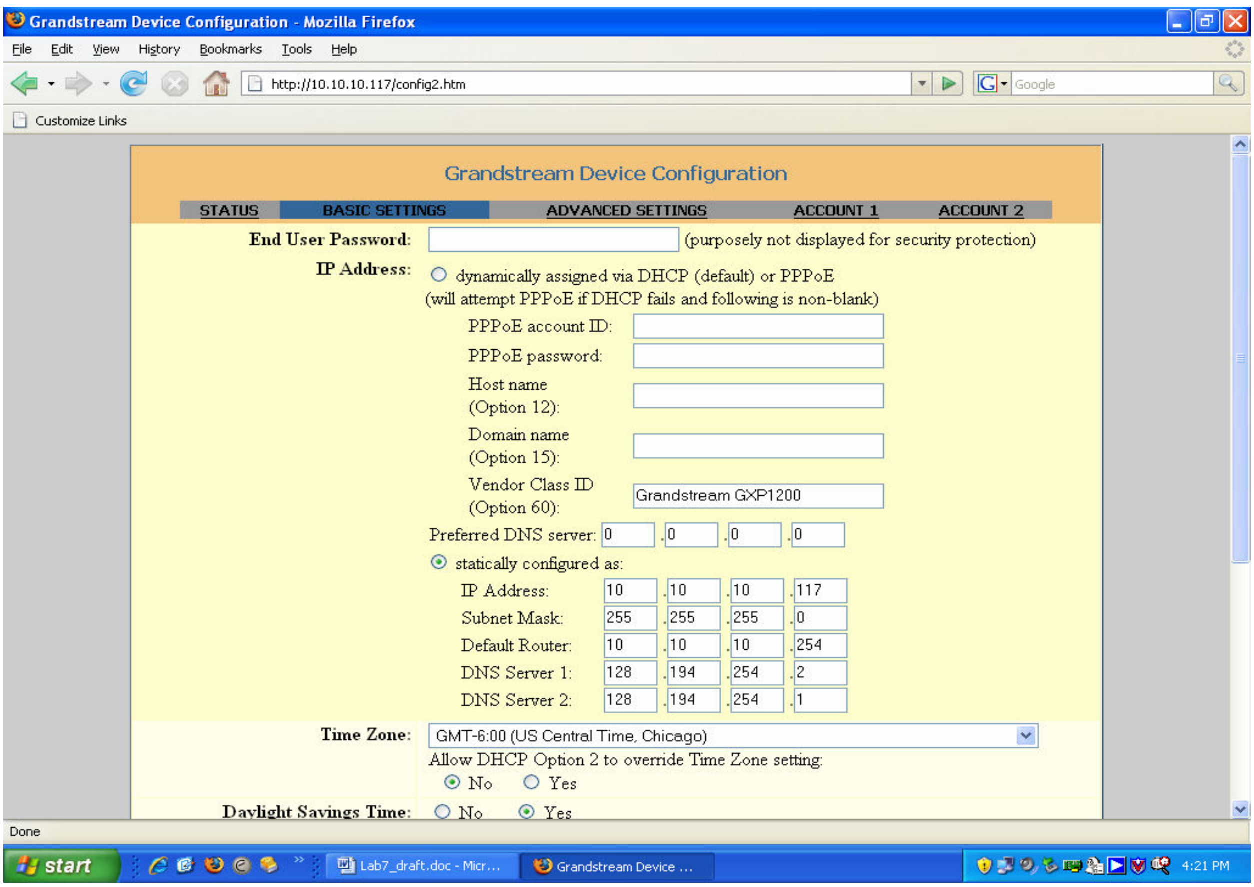

**Figure 5 - Configuring VoIP phones using the Web interface.**

One of the most interesting student feedback was from a student who said the following: "… the phones appeared just like a normal node on the network, requiring the same IP configuration as the desktop computers. This was interesting because usually phones are not thought of as configurable client/server devices."

## 4 – New ideas to integrate Voice over IP

A follow-up class to ENTC 315 is a class on Network Simulation and Modeling (ENTC 345). We use the same textbook in both classes: Computer Networks, a Systems Approach, from Larry Peterson and Bruce Davie[8]. In this Network Simulation and Modeling class, we address more advanced networking concepts such as quality of service (QoS) and congestion control. Hence, by using the current laboratory setup configuration shown in Figure 4, students perform two other VoIP experiments.

In the first experiment, students set up the VoIP phones to different codecs (e.g., G.711, G.729, and G.723) and take measurements of required bandwidth, latency and jitter[9]. With the help of the packet analyzer (i.e., Ethereal), they observe the packet sizes of the audio samples. With this information and knowing the codec's sampling rate, students calculate the bandwidth requirements of VoIP applications over Ethernet links[10]. For instance, the nominal rate of a G.711 codec is 64 kbits/sec; however, when we consider all the overhead added by the protocols under the audio application, the actual bandwidth over the network is approximately 90 kbits/sec.

After understanding the bandwidth requirements of each codec, the next experiment is about the performance of VoIP over a wireless point-to-point link. The main objective is for the students to analyze the impact of network impairments on VoIP QoS parameters, such as throughput, packet loss rate, and jitter. The network configuration for this lab is shown in Figure 6, which illustrates the links between two separate rooms (T-101 and T-105). Note that the Asterisk server is located in T-105, which is the network represented in Figure 4.

The wireless link is a point-to-point T1 link (1.5Mbps). In this experiment, the two laboratories were connected by the wireless link, and the students studied the performance of three different audio codecs (G.711, G.723, G.729). With this network setup, students were able to cause interference (or impairments) in the wireless link while a VoIP call was active. Using data collected from Ethereal, they plotted the performance of different codecs. For instance, when a G.729-based call is tested, the students can experience the latency and packet losses during the experiment, by calling each other and observing the quality of the call. Then, they quantitatively analyze these results, as shown in Figure 7.

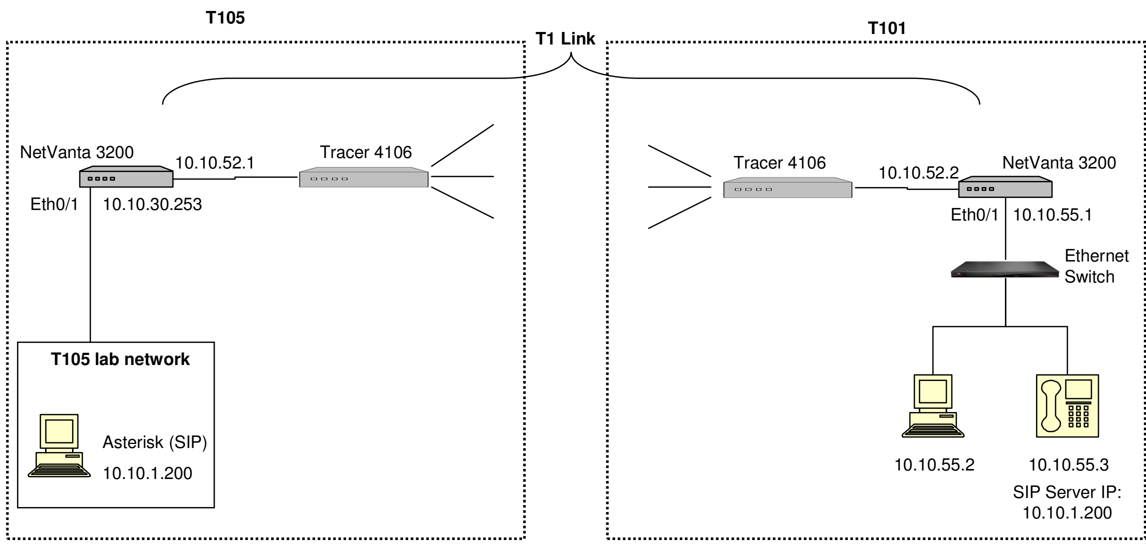


**Figure 6 - Laboratory setup to test VoIP over a wireless link.**

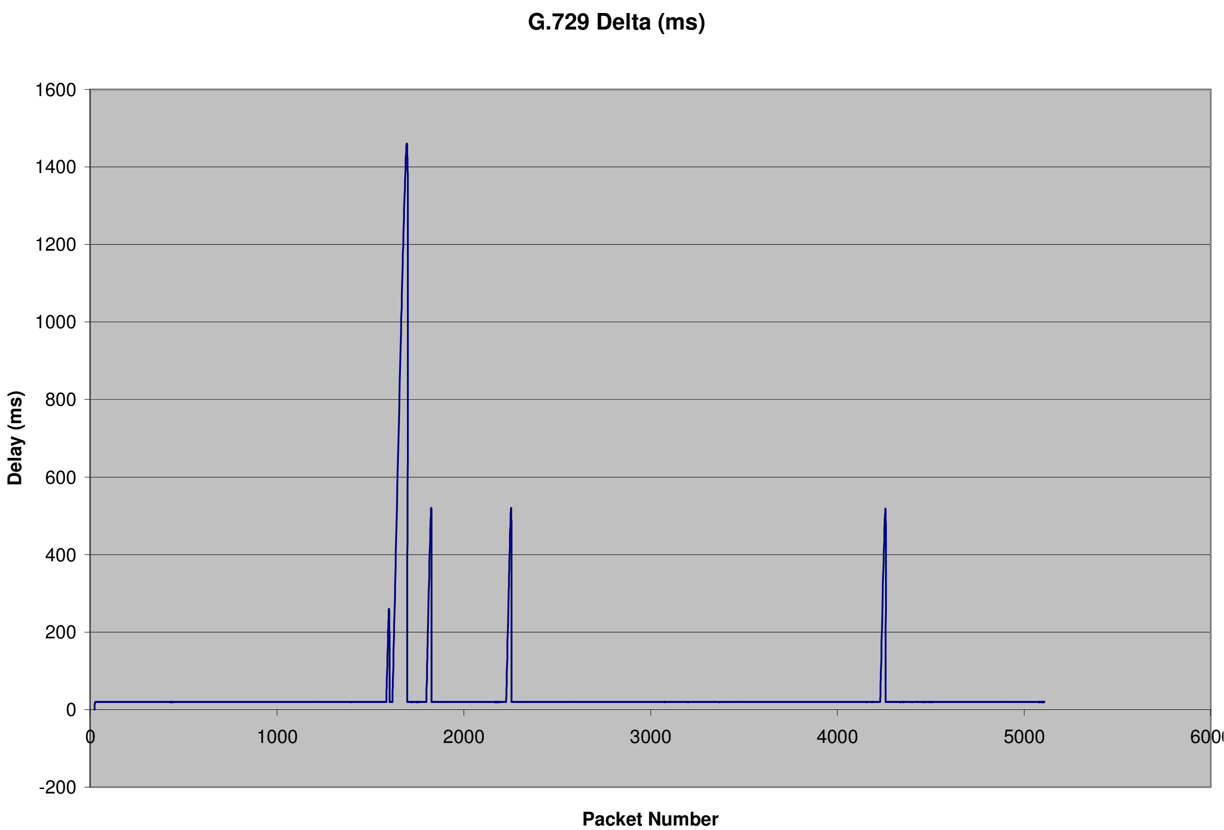


**Figure 7 - Example of delay measurements for a G.729 call over a wireless link.**

## 5 - VoIP Security considerations

VoIP systems are prone to the same security threats as other Internet applications. However, because it is relatively newer than other applications, sometimes we do not take all the security issues in consideration. As Asterisk is an open-access platform, free to be used by the Internet community, we recommend that users promptly change the default password of Asterisk to a "strong" password. Moreover, a dedicated firewall to this server should be used as well, with proper access control to UDP ports. Note from Figure 4 that our current laboratory is a private network, protected by a firewall and with limited access to the public network.

## 6 – Conclusions

We have presented some ideas to implement VoIP experiments in typical computer networks laboratories. This work represents the initial steps of our VoIP initiative, which has been driven by the need in our program to update its electronics and telecommunications curriculum to teach packet-based communications. We have discussed the different areas that students are mostly curious about, and as future work we plan to develop these other areas as well. Other experiments (along with updated lecture contents) will include wireless networks and security issues under this VoIP initiative umbrella.

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