AC 2010-1397: USING VOIP AS A COMMON FRAMEWORK FOR TEACHING A SECOND COURSE IN COMPUTER NETWORKS

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Using VOIP as a Common Framework for Teaching a Second Course in Computer Networks

Abstract

Although the value of experiential learning is increasingly widely recognized, it is sometimes hard to provide such experiences to students studying computer networks because of the breadth of the material that must be covered in the short time available. A second course in networks introduces even more wide-ranging and seemingly disparate topics further compounding this problem. The aim of this study was to integrate many of these topics in a second course on networks using a common framework, the Voice Over Internet Protocol (VOIP) application. The course also provides students with a very hands-on and practical learning experience; students actually implement a fully functional VOIP Public Automated Branch Exchange (PBX) that can inter-operate with a traditional Public Switched Telephone Network (PSTN). The VOIP project provides the context for introducing new networking topics as needed. Traditional lecturing is interspersed with hands-on project sessions in the classroom. Student comments on conclusion of the course were very positive and and indicated that the project-based approach gave them new insights and enthusiasm for learning about computer networks.

I. Introduction

The National Research Council's (NRC) reports “How Students Learn”[1] and “How People Learn”[2] and related publications stress the importance of experiential learning. Students demonstrate superior acquisition and retention of new information if the information is presented in an interactive manner using a hands-on approach. Unfortunately, rigidly structured course plans tend to overlook this important point. Traditional lecturing followed by laboratory sessions does help in consolidation of the material learned in class. Indeed, from the instructor's perspective, such an arrangement is time-efficient in covering the most amount of material in the short time available. However, due to the time lag involved between covering a topic in the classroom and the associated laboratory session, students do not remain engaged in the classroom. Many students see no real purpose in acquiring this 'theoretical' knowledge. The connection between the theory taught in class and laboratory experiments becomes tenuous in the students' minds.

The NRC reports also stress the importance of placing student learning in a common contextual framework. A unifying theme to teach a wide and loosely correlated selection of topics has been employed in fields such as psychology[3] and biology[4]. However, the most commonly taught topics in networks such as network protocols (TCP, UDP, RTP and SIP), bandwidth, latency and their effect on applications, network address translation (NAT), packet filtering, firewalls, security and inter-networking are taught in an isolated manner. Topics covering the practical aspects of network administration and debugging are left out altogether. The compartmentalization of this information has the effect of losing the 'big picture.' This is unfortunate especially since in a computer network, all these pieces need to inter-operate consistently in order for applications to work as intended.

Therefore, this study aims to address the abovementioned problems in our “Advanced Topics in Networks” course, which is a second course in networks, offered as an elective. The VOIP application theme is used as the 'glue' to integrate many, otherwise distinct topics, by introducing them as pieces of the larger VOIP puzzle. As the semester-long VOIP project progresses, students learn each topic as needed. They gain practical experience in configuring a VOIP PBX, writing dial-plans for call-connections, audio-conferencing, call processing and forwarding using pattern-matching, voice mail-to-email forwarding, and VOIP-PSTN interconnection for call-bridging. They also learn testing and trouble-shooting skills through the use of diagnostic tools such as packet analyzers (also called as 'packet sniffers'). They also experience the necessary trade-offs between voice quality and bandwidth when using the different prepackaged voice codecs. An additional benefit is that students are also introduced to the world of open source software in the form of the Asterisk VOIP PBX running on the Linux operating system. The end result of the course is a fully functioning VOIP PBX which students can use as their own mini telephone switch with functionality such as voice calls, voice mail, auto-emailed voice mail messages, audio-conferencing and VOIP-PSTN call routing.

The rest of the paper is organized as follows. Section II discusses the course objective and style of teaching. Section III provides details of the semester long VOIP project and the theory and skills that each hands-on task targets. Section IV makes some observations on the benefits and challenges of teaching this course in the current format. Section V concludes the paper.

II. Course Objectives and Teaching Style

A. Course Details

The Advanced Topics In Networks course (numbered ECE 5470) is offered by the Department of Electrical and Computer Engineering at Villanova University as an elective, three-credit second-level course in networks. It is primarily targeted toward undergraduate students in their senior year.

The main goals of this course are to impart to students:

- A good theoretical understanding of selected transport and application layer protocols.
- A familiarity with at least one network packet analyzing tool.
- The practical skills in using the packet analyzing tool to verify protocol behavior and to aid in debugging.

The main topics for study are:

- UDP and TCP: behavior and performance.
- Selected application protocols such as VOIP, RTP/RTCP, SIP, Firewalls and Network Address Translation (NAT).
- Wireless networking with IEEE 802.11.
- Quality of Service (QoS) and its impact on applications.
- Parallels between traditional telephony and VOIP, and their interoperability.
- A semester-long VOIP project that ties-in most of the topics above.
- The *wireshark* packet analyzer for analyzing and debugging protocols used by VOIP.

The above schedule is somewhat ambitious at first glance. However, since this is an undergraduate course, we strive for breadth rather than depth. Therefore, (non-essential) protocol details and mathematical treatment of performance analysis (using queuing theory) is left to a higher-level (graduate) course. Two textbooks are recommended: a traditional textbook on computer networks for the study of protocols from a theoretical perspective,[5] and a more hands-on textbook specifically geared toward the implementation of an actual VOIP PBX.[6] The latter is available as a free, legally downloadable electronic copy from O'Reilly Media Inc.

B. Instruction Style

Classes are scheduled twice a week in two-hour sessions. The general mode of instruction is as follows. The instructor starts the class by outlining the goals of the current session. The discussion begins with the instructor identifying a technical challenge that is related to voice-calling on the Internet. Then, the students brainstorm for possible solution approaches, with the instructor's guidance. Usually, with modest effort, the discussion can be steered toward a topic in the course syllabus. Then, traditional lecturing begins with an explanation as to where the subtopic fits in the overall VOIP framework. After the essential subtopics are taught, students are assigned to work in groups of two on a hands-on experiment or task. Such interspersing of lectures with hands-on tasks occurs throughout the semester.

III. Hands-on VOIP Project

The VOIP unifying theme as selected due to its popularity among students, from past experience. Students frequently use the commercially available *skype* or *googletalk* VOIP services and are naturally curious about their operation. Therefore, they display great enthusiasm and inquisitiveness when given the opportunity to design a working PBX and in effect, become their own service provider.

A. Project Platform

Since the PBX is expected to be operational 24 hours a day, 7 days a week, a low-power (10W or less) system is desirable. In addition, for robustness, the system must have no moving parts such cooling fans or spinning drives. The net4801 and net5501 embedded systems supplied by Soekris Engineering, CA meet all these requirements, and are very affordable as well (under $300). The systems are based on the AMD Geode processor and employ passive cooling. The boards have a Compact Flash (CF) card slot with a CF-IDE interface that allows the usage of a CF card as a hard drive. The operating system and the PBX software can be loaded on this card.

We use the open-source AstLinux software[7] which bundles the Asterisk open source PBX software with a very efficient, stripped down flavor of Linux. AstLinux provides both a command line interface (CLI) as well as a secure web interface for PBX programming and maintenance. The PBX hardware is housed in an access-controlled room, connected to an Ethernet switch. One PBX machine is provided to each group of two students. Physical access to the hardware is not required after the initial configuration. Thereafter, all programming and maintenance is performed remotely over University-provided students' personal laptops.

B. Hands-on Tasks

Students are assigned tasks to accompany a lecture session. Each task is composed of a number of smaller specific subtasks. Only the main tasks/subtasks are listed below for brevity.

Task 1: Install the AstLinux software on the Soekris target and assign it a static IP address.
Significance: Students learn how to use the Linux command line interface (CLI) to flash the CF card with a card reader. After plugging the card into the target machine, they learn how to monitor bootup messages from the target's com port (since the target does not have a video port). They observe the IP address acquired by the target, and learn how to connect to it over an ssh tunnel, change the default password, and configure the target with a static IP address, Domain Name Service (DNS) and gateway addresses, all using the Linux CLI. This gives an opportunity for the instructor to review IP routing, addressing and subnetting.

Task 2: Download, install and configure a VOIP softphone application – Xlite (on Windows laptops) or Ekiga (Linux laptops). Register the softphone with the VOIP PBX.
Significance: Students learn how to set up phone accounts on the remote PBX, authenticate the softphone client running on their laptop and register its IP address to the PBX. The accompanying lecture discusses the session initiation protocol (SIP) and packet format/fields, the concept of a registration server (PBX), its authentication mechanisms and debugging techniques. Students install and run the *wireshark* packet analyzer and log the traffic between their softphone and PBX. Registration failures are debugged by interpreting the error messages in PBX responses using *wireshark* and also by examining the messages on the Asterisk CLI in debug mode.

Task 3: Program dial-plans for call routing and voice mail.
Significance: The lecture introduces the concepts of call signaling and call routing – the signaling system 7 (SS7) used in traditional telephony is compared with the SIP signaling scheme for call establishment and teardown. With the help of examples from the textbook,[6] students assign extension numbers for their softphones, and write simple PBX dial-plans for routing calls among themselves. They learn how to associate a single extension number with two or more softphones, ring them all concurrently or sequentially (the 'follow-me' call option). Unanswered calls are sent to voice mail boxes. After a brief discussion of the SMTP protocol, students configure the PBX to email voice mail sound files to call recipients. Students use w*ireshark* is used to examine SIP signaling messages between the phones and the PBX and to debug connection failures. Lectures on real time transport (RTP/RTCP) are coupled with *wireshark* sessions to observe voice media traffic to/from the softphones.

Task 4: Study the effects of firewalls and network address translation (NAT) on VOIP traffic.
Significance: Unlike lower-level undergraduates who live in dormitories on campus, all our senior students live off-campus close to the University. They are asked to test if VOIP calls work from their dwelling – they don't! The students have to report why this might be the case, and most of them do manage to infer the reason: the PBX is on the campus network, which is firewalled; the off-campus laptop-based softphones cannot register

with the PBX since the SIP registration ports numbered 5061/62 are blocked. The class then studies the various firewall schemes, NATs and port-forwarding. Again, *wireshark* is used to examine VOIP traffic for evidence of port-blocking or one-sided conversation (a side-effect of NAT).

Task 5: Impact of bandwidth, latency and jitter on VOIP communication.
Significance: The lecture examines the concept of QoS in general, and VOIP QoS requirements, with the discussion focusing primarily on bandwidth, latency and delay jitter. A synthetic traffic generator software is activated on student laptops generating a large amount of spurious traffic. This has the effect of reducing the effective bandwidth available for VOIP traffic and increasing latency. The distribution of the traffic generated can be adjusted to provide varying levels of delay jitter experienced by VOIP packets. The experiment demonstrates the relative intolerance of VOIP to such environments. Students learn that 'voice breakup' can be lessened by selecting lower bandwidth codecs such as GSM, G729a instead of the much higher bandwidth default PCM μ-law or A-law codecs. The Asterisk CLI debugging mode is activated to see the codec negotiation process between the softphones and the PBX by means of SIP/SDP protocols.

Task 6: Implement macros and advanced call processing.
Significance: Students soon learn that hand-editing configuration files to add new subscribers and dial-plans to the system is tedious and error-prone. With the help of textbook examples, macros are implemented to simplify general call-handling. An extension is set up and dial-plans are written for audio-conferencing. The caller ID fields of incoming calls are processed for specialized handling. For instance, calls without valid caller ID (telemarketers) can be blocked, undesirable callers can be 'blacklisted' and favored callers (friends) trigger the 'follow-me' dial plan wherein the call is forwarded to *all* the callee's phones, in a preferred sequence.

Task 7: Terminate VOIP calls over the PSTN and vice versa.
Significance: The class lecture on VOIP—PSTN interoperability begins with an explanation of the North American Numbering Plan (NANP) for PSTN and a high-level discussion of the analog Foreign Exchange Subscriber (FXS), Foreign Exchange Office (FXO) ports and how a VOIP—PSTN gateway operates. The actual interfacing of the PBX switch with a PSTN service can be done either by means of i. the inexpensive Linksys-Sipura 3000 single-line VOIP-PSTN gateway. This does require considerable expertise and time in configuring both the Internet side and the PSTN side of the device, as is therefore not recommended, or ii. Subscription to a commercial gateway provider such as Gizmo[8], which do provide detailed instructions and dial-plan statements to set up a call trunk between the PBX and the gateway service. On the PBX, Dialed calls are pattern-matched to determine if the called number conforms to the NANP. If so, the call is routed to the commercial gateway service. Pattern matching is also used to block calls to '900' numbers.

At the end of the semester, most student groups manage to implement the complete VOIP PBX. Furthermore, students can copy the entire project code to a bootable USB stick running

AstLinux. The resulting installation allows any X86 machine to boot off the stick and act as a fully functional PBX switch.

IV. Some Observations

As the course progressed, it was apparent that the students had embraced this style of instruction interspersing lecture and theory. The instructor received a steady stream of emails from students seeking clarifications and help on assigned homework tasks all week long, far in advance of due dates. Students sought help even during semester breaks, and outside the instructor's office hours, usually a very rare occurrence. In addition, barring one group, all groups turned in *all* homework assignments on time. This compares very favorably with past experience in the traditional lecturing style, wherein most students would miss or do really poorly on at at least one or two homework assignments. The end of semester course evaluations filled out by students made many positive remarks about the course format and instruction style, the practical skills and the many insights on network protocol behavior that they got through experimentation.

The added work does place a fairly large burden on faculty time. In order to make efficient use of time, the University-wide WebCT environment was used to host tutorials, FAQs and online technical discussions, with the instructor acting as facilitator and moderator. This approach fostered collaborative learning, with most student groups contributing significant amounts of self-learned information, tip and tricks. Some of the student-contributed information was incorporated in the instructor's class notes.

In addition to providing help outside class, significant in-class input was required from the instructor. Since the project tasks build and previous work and get progressively difficult as the semester progresses, without close supervision, the weaker student groups fall behind in their work with little hope of catching up. Thus without additional help from qualified teaching assistants both in as well as out of class, this approach might not have been as successful for larger class sizes (above 15 students).

V. Conclusion

The NRC reports on student learning demonstrate the value of a hands-on approach in teaching. The reports also stress on placing topics under study within a common contextual framework. This study shows how a second course in computer networks that covers seemingly unconnected topics can be taught using the common unifying VOIP framework. The interspersing of lectures with hands-on tasks increased students' enthusiasm for learning, and increased faculty-student interaction even outside class. The use of VOIP as a practical unifying theme for an 'Advanced Topics in Networks' course was therefore very successful. The approach requires very little capital expenditure beyond the standard computing resources that are standard at most universities, and is very portable to other institutions.

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