VoIP-Based Air Traffic Controller Training

Supreeth Subramanya, Xiaotao Wu, and Henning Schulzrinne, Columbia University
Susan Buriak, Federal Aviation Administration

ABSTRACT

Extending VoIP beyond Internet telephony, we propose a case study of applying the technology outside of its intended domain to solve a real-world problem. This work is an attempt to understand an analog hardwired communication system of the U.S. Federal Aviation Administration, and effectively translate it into a generic standards-based VoIP system that runs on their existing data network. We develop insights into the air traffic training and weigh in on the design choices for building a soft real-time data communication system. We also share our real world deployment and maintenance experiences, as the FAA Academy has been successfully using this VoIP system in six training rooms since early 2006 to train the future air traffic controllers of the U.S. and the world.

INTRODUCTION

Traditionally, voice communication has required dedicated infrastructure — both in the backbone and at the end-points. Then in the mid-1990s, the ubiquitous connectivity of the Internet with its ever increasing bandwidth fuelled the research that made communication over data networks possible. Treating voice as real-time data brought numerous advantages — reuse of the existing data network infrastructure that reduced the deployment and usage costs, digitization of voice that led to superior voice processing and compression techniques, and decoupling of voice service from the underlying infrastructure that fostered innovation in communications. Despite a widespread adaptation of IP communications in commercial environments over the last decade, some specialized agencies continue to use custom-built analog communication systems.

The FAA is an administration of the U.S. Department of Transportation, with the authority to regulate and oversee all aspects of civil aviation in the U.S. Its education and training division, the FAA Academy, provides technical training for the aviation community including future Air Traffic Controllers (ATC) and other aviation personnel in a variety of simulated and real environments. To conduct these trainings, three disjoint networks are employed in every classroom and laboratory — voice network, graphic simulation network, and data network. Graphic simulation and voice networks are custom-built analog infrastructures to support flight simulations and inter-position communications respectively. The data network is a TCP/IP-based Gigabit Ethernet infrastructure with a firewalled connectivity to the Internet.

Equipped with a legacy communication system, the FAA Academy faced tough challenges in keeping up with advancements in training programs: creating new training scenarios, accommodating additional students or even moving students from one position to another required physical rewiring and reinstallation. The voice system was getting prohibitively expensive with its hardware needing custom manufacture. That was when the Interactive Instructional Delivery System (IIDS) of the FAA decided to collaborate with a research lab to seek better alternatives.

This article discusses a novel case-study of successful design and deployment of a VoIP system for the FAA. We gather and analyze the specialized communication requirements of air traffic training and translate these into an IP-based design using standard Internet protocols. We describe our experiences and the lessons learnt in deploying and maintaining the VoIP communication system in six training rooms at the FAA Academy for Initial Terminal Training, Terminal Radar Training, and International Training programs since early 2006. Our work also illustrates the challenges of working in a conservative environment, where extremely high system uptimes are expected and the fact that with due diligence academia can very well create non-toy systems. We are not aware of any other real world attempt to use IP communications in ATC training systems.

BACKGROUND

AIR TRAFFIC CONTROLLER TRAINING

ATCs are the personnel who operate the air traffic control systems to promote a safe, orderly and expeditious flow of air traffic. While at the FAA Academy, the future ATCs learn the techniques of managing air traffic by developing a mental picture of air-space and air-time, by learning to communicate and coordinate with pilots and neighboring ATCs, and by learning to use a multitude of air traffic control displays and devices. The training is conducted in a variety of environments at varying levels of difficulty starting with the low fidelity (instructional games focusing on individual training), then the medi-

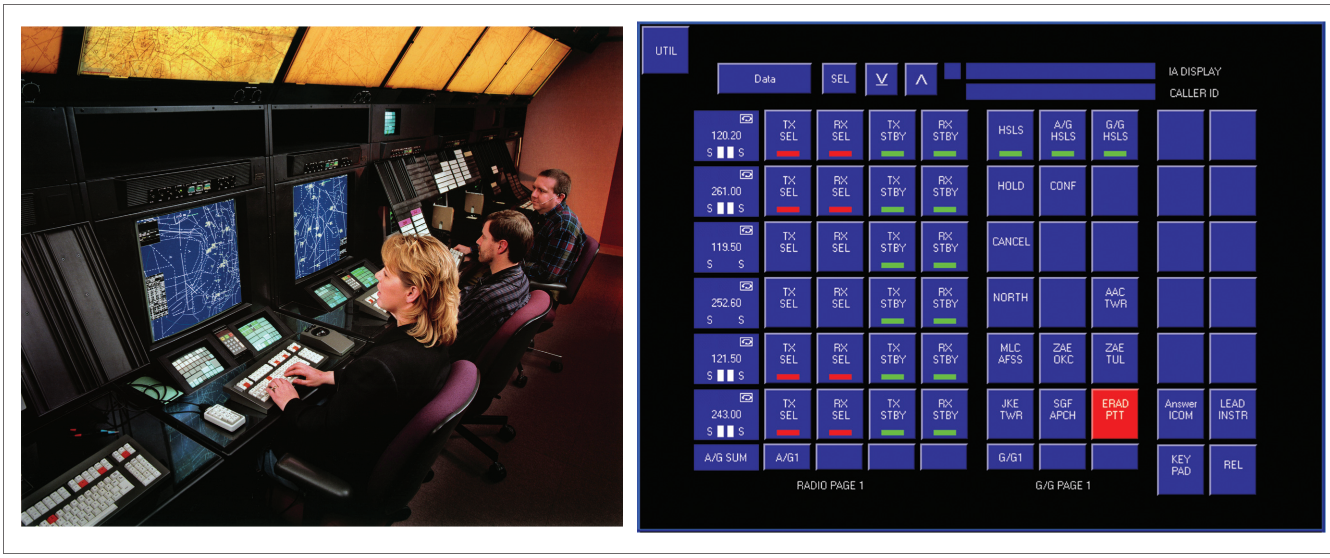

Figure 1. *A high-fidelity training room[1] and one of the communication interfaces of the ATC.*

um fidelity (real-time interactive training), and finally the high fidelity (complex interactions involving real hardware).

IIDS training rooms at the FAA Academy are designed to closely match the real world experiences of ATCs, except that all ATCs are seated in the same room and that pilots are not actually flying the aircrafts (Fig. 1). In the high-fidelity training, every ATC is paired with a position instructor, who provides continuous one-on-one mentoring while the course is taught by the lead instructor. In a typical session, after explaining the conceptual aspects, the lead instructor loads the relevant flight scenario onto the graphical simulation/voice communication systems, instructs pilots to simulate their flight movements, and then lets ATCs and pilots to interact with each other much like the real world.

Air Traffic Communications

Communications in the training environment is comprised of two categories — first, the ones that represent the real world interactions between the air traffic entities and second, those involving the instructors that have been specifically created for an effective conduct of the training.

Radio Lines — Radio lines represent air-to-ground transmission, where transmitters and receivers tune into a selected frequency channel for communication. The channel selection devices Push-To-Talk (PTT) and footswitch, together with touch-screen UI provide an ability to use the channels in receive-only or send-receive modes. These are used for broadcasting messages to all the air traffic entities in a given sector (a sector refers to the air space managed by an ATC).

Ring Lines — These are bi-directional communication channels, similar to traditional telephony. When the caller initiates a call, both the caller and the callee hear the ringing. The call gets established only when the callee explicitly accepts the call by pressing a button, analogous to picking up the handset. These are used for non-emergency one-to-one communication between a pilot and his controller.

Override Lines — Override lines are bi-directional communication channels that do not require an explicit Acknowledgment from the callee for the call establishment. In the training, these are used for both bi-directional communications and passive overhearing of the neighboring ATCs' communication.

Shout Lines — Shout lines have the ability to support both unidirectional and bi-directional communications. When the caller initiates a call, a unidirectional media session gets established and the callee starts hearing the caller. If and only when the callee accepts the call, the media session becomes bi-directional. In the air traffic domain, these are typically used for inter-facility communications.

Channel Status Notification — Communicating entities, especially those not actively tuned into a channel, need to know the current activity on that channel before starting to communicate. Such real-time monitoring of the channel status is essential when channels have to be used exclusively (e.g., override lines) or to reduce crosstalk on collaborative channels (e.g., radios).

Instructor Monitoring/Broadcasting — Instructor monitoring and broadcasting are training-only channels used by classroom instructors for training related communications. Monitoring channels are bi-directional in nature, not involving any visual cues at the student position, such that the instructor can passively listen to all communications of a student and may choose to give occasional feedback. Broadcast channels are unidirectional in nature, and used for classroom wide announcements.

Classroom Recording — For archival purposes, the academy records all the training sessions. The recorded session are played back in VCR-

[1] *The picture on the left is only representative and does not represent the actual training rooms of the FAA Academy. Source: http://www.lockheedmartin.com/data/assets/10307.jpg*

The design goal is to create a voice communication system that makes use of the existing data network infrastructure, and supports all the functionalities present in the proprietary hardwired analog voice system.

	Control	Scalability	Security	Implementation
Centralized conference	Best	Fair	Best	Fair
Cascaded conference	Good	Best	Good	Hard
Full-meshed conference	Fair	Bad	Good	Easy
App. layer multicast	Bad	Fair	Fair	Hard
IP multicast	Worst	Fair	Worst	Easiest

Table 1. *Technologies for conferencing support.*

fashion, typically in conjunction with graphical simulation.

DESIGN AND IMPLEMENTATION

The design goal is to create a voice communication system that makes use of the existing data network infrastructure, and supports all the functionalities present in the proprietary hardwired analog voice system. Additionally, such a real-time IP communication system should offer configurability (i.e., support for creating new scenarios by decoupling student role from their classroom positions) and programmability (i.e., support for adding new communication channels with different rules without any hardware-level changes), both of which were lacking in the hardwired system. Lastly, the new system should at least be as robust and reliable as the existing system.

SYSTEM ARCHITECTURE

Real-time communication is comprised of two aspects — signaling and media transfer. Signaling is the process by which a caller locates the callee, conveys her request to the callee specifying the communication type and attributes, and gets positive or negative Acknowledgment from the callee with her accepted set of parameters. After successful signaling, the communicating entities know exactly where and how to send and receive the media information.

Signaling — Signaling requirements for the air traffic communications discussed in the previous section are more demanding than traditional telephony as every communication channel should be able to specify its own signaling logic. To this end, we decided to use Session Initiation Protocol (SIP) [1], an application-layer control protocol for creation and management of multimedia sessions and conferences that supports a flexible programming interface to signaling. An entity in the SIP paradigm that represents a communicating endpoint is referred to as SIP User Agent (UA), and accordingly our setup has two types, student UA and instructor UA. The UAs communicate with one another with the help of a SIP proxy server, which is responsible for locating users, authentication and authorization, and request/response routing.

The signaling logic in SIP can exist either at the SIP server or the SIP UA or both. In our design, all the call routing logic (determining if a caller is permitted to make that call, and if so, who should receive this call) is placed at the server, while the UA handles the modalities of user interaction with the GUI and the I/O devices. We adapted SIP-CGI to specify the routing logic at the server.

Media Transfer — An intuitive design for media flow is a hybrid architecture where multicast is used for the radio lines and instructor broadcasts, and unicast for point-to-point communications. But when we consider the requirement that every possible communication should be able to be monitored, interjected and recorded, we realize that all communications could be effectively designed as multicast sessions or conferences. There are several ways of handling conferencing and audio mixing in VoIP systems and Table 1 compares the candidate technologies.

In general, it is easier to achieve security handling and conference control, such as floor control and access control at a centralized conference server because it has a single control point. In terms of scalability, since the cascaded conference distributes audio mixing computation to multiple conference servers while not increasing network traffic, it offers best scalability. Implementation-wise, full-meshed conference, application layer multicast, and IP multicast do not require a server implementation, but they all need the support of endpoint mixing, which is simpler than implementing a conferencing server.

To choose a solution, we first analyze the requirements and deployment context of our system. First, there is no specific conference control functions required in our system as the students themselves use PTT devices to switch between listening and talking modes. Second, our system will be used in an intranet that is behind a firewall and will only communicate with entities inside the intranet. So, we do not need firewall traversal capabilities. Third, since all the computers are located in the training rooms and all the users are the academy students, both the users and the communication endpoints are trustable and consequently, no specific security mechanisms are required of the VoIP system.

Last, we address scalability. A training room consists of no more than 27 communicating entities (13 ATCs, 13 pilots, and an instructor in low/medium fidelity training; 6 ATCs, 12 pilots, and 7 instructors in high-fidelity training). Given that in each sector a maximum of six radio lines

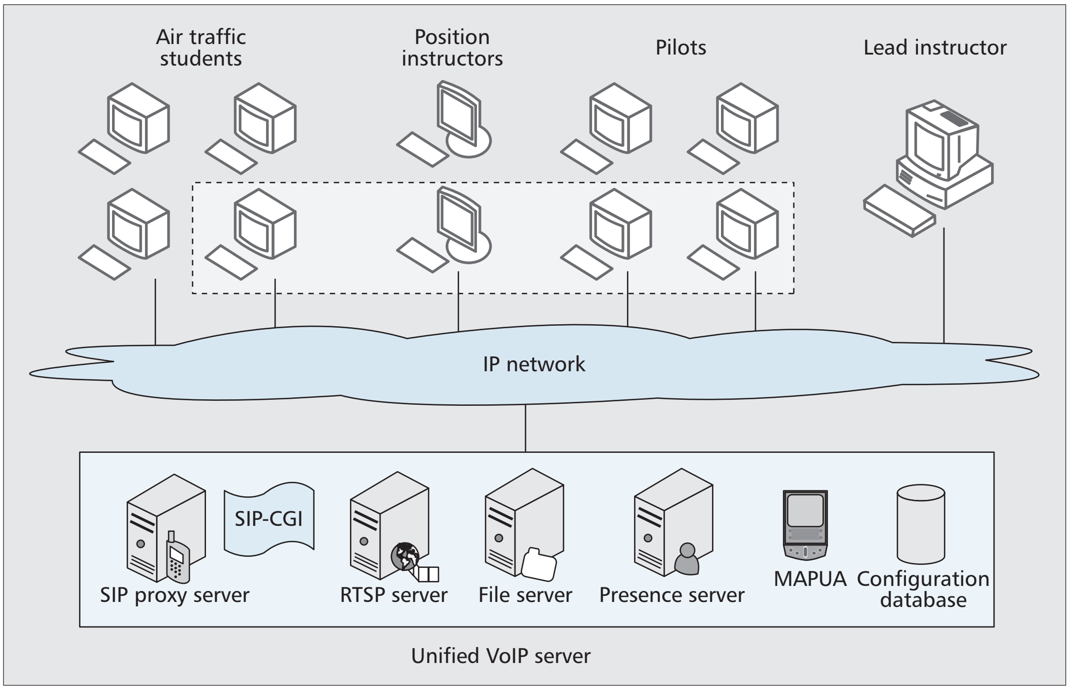

Figure 2. *Communication system architecture of a typical training room.*

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and one point-to-point line can be simultaneously active, there can be no more than 7 conferences per sector or equivalently no more than 91 conferences in any classroom. Also, each conference will have no more than five participants (even when the communication is being both monitored and recorded). This is a theoretical maximum as no training currently requires all the students to be active on all the seven possible channels simultaneously. Even if the number of students were to increase later on, it will only increase the number of conferences, but not the number of participants in a given conference. Since the least scalable of all solutions, namely full-meshed conference and the IP multicast, can support fewer than 10 participants in a conference, scalability is not a differentiating factor for our design choice.

Based on this analysis, we chose to use IP multicast, since it meets all the requirements and is the easiest to build among all the candidate choices. We employ Real-time Transport Protocol (RTP) [2] for packetizing and transmitting digitized voice and Real-time Transport Control Protocol (RTCP) for QoS monitoring. Figure 2 illustrates the system architecture. The dotted box in Fig. 2 indicates a logical grouping of an ATC student, his Position Instructor and the two Pilots of that sector.

Detailed Design

Radio Communication — Radio lines are in-sector broadcast communication channels. Each radio line is assigned a permanent SIP URI (like *sip:radio_120@faa.gov* for the 120 MHz channel) that can be used uniformly by callers in any sector. But internally, every radio line in every sector has to be associated with a unique multicast address, to avoid interference across the sectors. This translation from the generic SIP address of a radio line to its unique multicast address is performed dynamically by the Multicast Address Provider UA (MAPUA). The call flow for establishing a radio line is illustrated in Fig. 3.

Landline Communications — Any non-radio line is typically referred to as landline in air traffic parlance. Despite having seemingly different call behaviors, a central characteristic of ring lines, shout lines and override lines is that they are one-to-one communications. Thus, they can be conveniently grouped under a single umbrella with the differences in their call behavior being handled by the call logic scripts and by conveying the call characteristics like one-way audio and auto-accept through SIP/SDP signaling. As noted earlier, even these apparently one-to-one communications are established as multicast sessions to accommodate dynamic addition/ removal of the monitoring instructors or recording agents. Just like radio lines, land lines are assigned permanent SIP-URIs (like sip:landline_mlcAFSS@faa.gov for a ring line called McAlister Flight Service) that can be used uniformly by callers in any sector or exercise. But internally, every land line in every sector is associated with a unique multicast address.

Notification of Transmissions — SIP event notification framework [3] is used to design a mechanism to provide real-time feedback of the channel status. When the training starts, no channels would be active and every student subscribes to feedback on all the channels. Whenever any student UA gets ready to communicate (either transmit or receive on a channel), it first unsubscribes from channel updates, for it would know the channel status by itself. If a student UA starts transmitting, then it sends a `PUBLISH` request to the Presence server, so that `NOTIFY` messages are sent to everyone not on that channel but subscribed to its status.

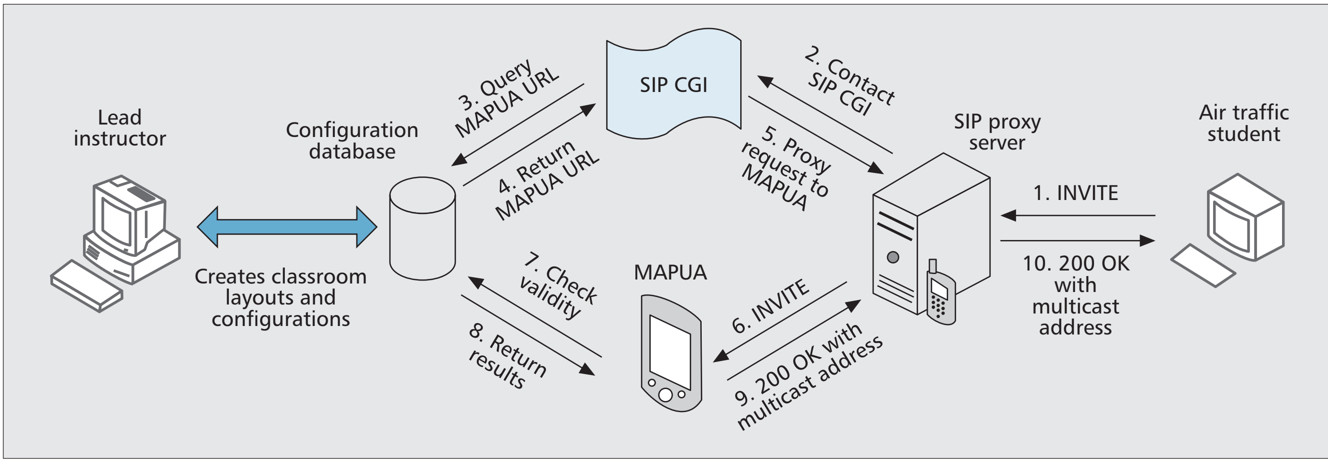

Figure 3. *Call flow diagram for radio line communication.*

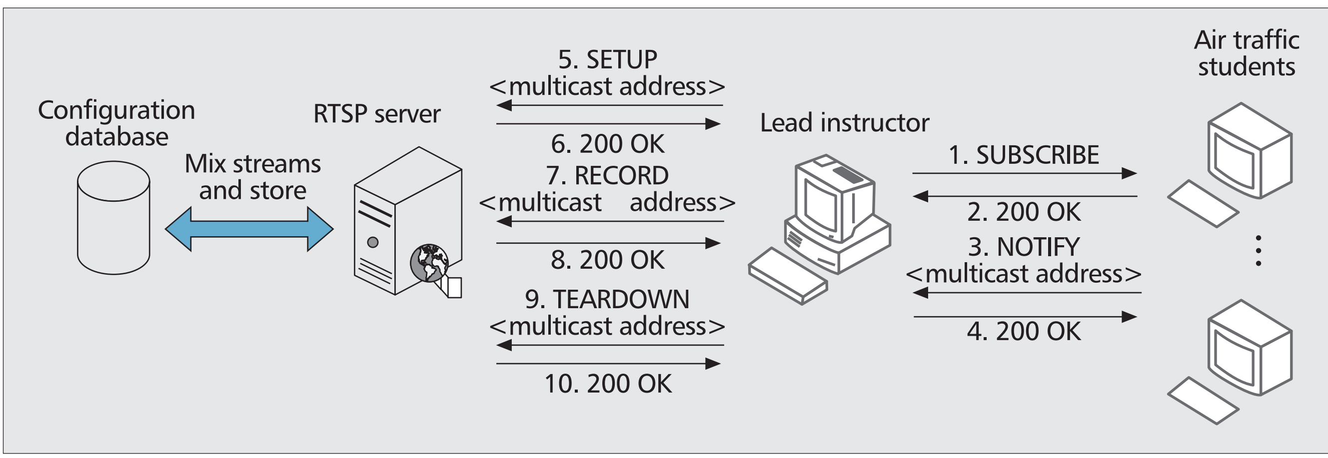

Figure 4. *Call flow diagram for classroom recording.*

Classroom Recording — Classroom recording requires mechanisms to start and stop recording at the granularity of exercises, dynamically add and remove sessions during the course of recording, as well as mechanisms to browse through the recorded streams and play them back in VCR-fashion. All the record and playback functionalities exist at the lead instructor position and are designed using the Real Time Streaming Protocol (RTSP) [4] together with the SIP event notification framework. Figure 4 shows the call flow diagram for classroom recording.

The RTSP client, which runs at the instructor UA, has to get the information about all ongoing communications on a continuous basis. This is achieved by sending `SUBSCRIBE` request to the student UAs; whenever there is any change in their communication status, the student UAs `NOTIFY` the instructor UA. The RTSP client can then issue requests to the RTSP server for adding (`RTSP RECORD`) or removing (`RTSP TERMINATE`) multicast sessions. The RTSP server captures media from all the required multicast sessions, mixes them and saves the unified media stream with the help of the file server. The lead instructor application uses the file server to browse through the recorded files and plays them in VCR-fashion using our archive player.

IMPLEMENTATION

Owing to our limited air traffic know-how, we decided to follow *iterative prototyping* as the system development methodology. Our language of choice was Tcl/Tk, a powerful scripting language well-suited for such rapid prototyping with its platform independence and rich UI development tools. File server, archive player, MAPUA, all the student and instructor applications including their GUI and UA were implemented using Tcl/Tk. User agents were built on top of Columbia's generic SIP UA, SIPc [5]. We bundle all required Tcl/Tk scripts into a single Windows application using Freewrap.

The media engine, running on every student workstation is responsible for all the media encoding and transfer and was implemented in C++ with support for G.711, Speex, iLBC, and GSM. We leveraged CINEMA [6], an umbrella of SIP-based multimedia servers developed at Columbia, for building most of our VoIP server components namely SIP proxy (sipd), RTSP server (rtspd), SIP-CGI framework, and general account management. Including only the relevant components of CINEMA, the project comprises of 70,000 lines of code. Finally, to keep the implementation aspects transparent to the end-users and to ease the deployment, all deliverables are packaged as

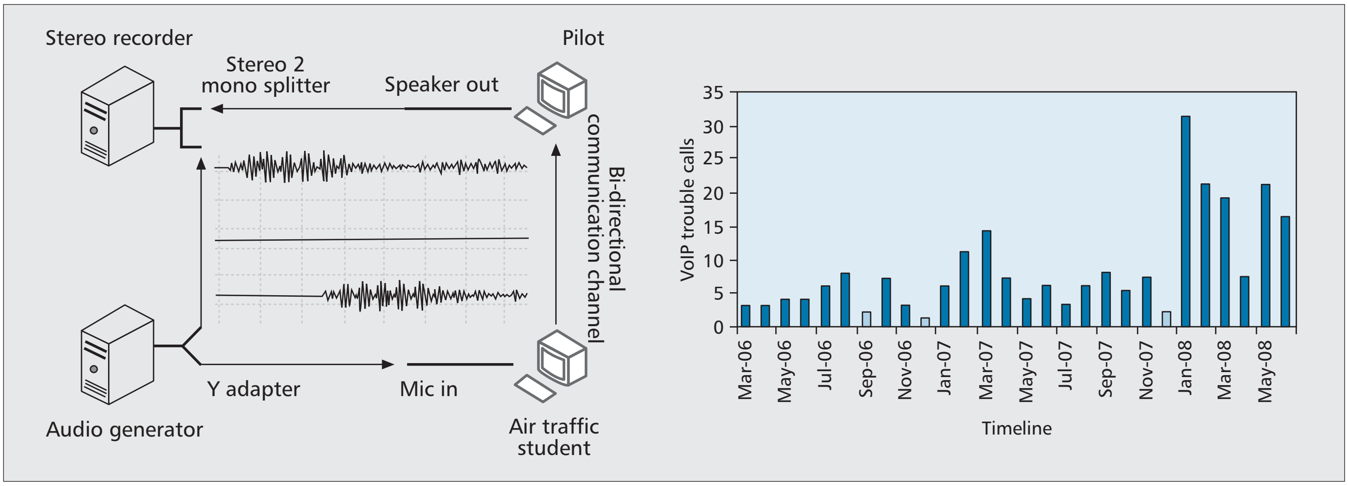

Figure 5. *Evaluations: a) Audio-latency test setup; b) VoIP trouble call distribution.*

RPMs for Linux distributions and as Microsoft installers for Windows environments.

Evaluation

In this section, we discuss the important evaluation criteria including audio latency, packet loss, and system reliability. Figure 5 shows our setup for audio latency measurements. Two air traffic entities, namely an ATC and a Pilot, have established a bi-directional communication channel. The audio generator output is split into two streams such that one of them goes directly to the left channel of the stereo recorder, while the other goes to the ATC student, which then multicasts it on the network to be picked by the Pilot, who then plays it out to the right channel of the stereo recorder. Thus, by calculating the time difference between left channel and right channel inputs at the stereo recorder, we can compute the mouth-to-ear delay. We used Cool Edit Pro and PSEQ to perform these measurements. The results showed that the audio latency was always less than 100 ms. Also, monitoring the network, we observed zero packet loss in the operating environment.

To measure the system reliability, we collected and analyzed all the VoIP-related trouble calls handled by the FAA Academy's IT support staff in the six training rooms over the last 2 years, as shown in the Fig. 6. This data gives an average of 1.8 VoIP trouble calls per room per month. These include a broad range of issues — trivial ones like *VoIP system not started on a workstation* to more challenging ones like the *Electro Static Discharge* (ESD) *problem in PTT*. The graph indicates two peak periods of trouble calls — first during early 2007, due to a change in the instructional and support staff in two training rooms, with its associated learning curve for the new staff. The second peak in 2008 was primarily due to the ESD problem in PTT. Both these issues are discussed in the next section.

Deployment Experiences

With the system being deployed and operational in six FAA Academy training rooms since early 2006 and with two more being currently rolled out, the VoIP system has been a success. Columbia's handling of the project from thought to finish, from design and development to deployment and user training, helped elevate it from an academic research prototype to a production system in a federal environment. Equally appreciable is the openness with which the air traffic instructors and the system administrators embraced the new technology. But the deployment experience has been full of ups and downs; we outline some of the significant operational challenges and reflect on the remedial measures in this section.

Conflicting Mindset of the Stakeholders

"Conflict is inevitable; combat is optional" — Max Lucado

The most prevalent challenge has been to balance the conflicting mindsets of the involved stakeholders, when what was intended as a research prototype got deployed as a production system. The researchers usually have a dislike for anything that is not research; bug fixes do not lead to publications. The academy system administrators, who now had to support a new communication system, were not as confident about its functionalities and fluent about its nuances as they were with the old system. The air traffic students, despite their good intent, may turn unfriendly as any unexpected behavior of the training system would lead to lengthy class breaks.

Continued interaction and collaboration between the stakeholders from the early stages of the project, made sure that the problem of conflicting mindsets was well acknowledged, if not completely solved. Columbia has shown two-fold commitment — first, by visiting the FAA Oklahoma facility seven times in the last four years, to better understand the usage patterns and to identify and fix deployment issues, and second, by organizing hands-on VoIP training sessions for the FAA Academy system administrators, thrice in the last two years. The FAA Academy on its part has shown good product ownership qualities, with the system administrators *hand-holding* the training sessions when the instructors are not fully familiar with the VoIP

Since the FAA Academy operates in a highly-regulated federal environment, security policies prevent any form of remote connectivity to their computing systems. Thus, we had to build a mechanism such that all the interactions of the system with the users, the network and the associated hardware had to be extensively logged.

systems and successfully resolving many usage issues locally and with the managers expressing their long-term commitment with continued research funding.

COMPONENT FAILURES

"The whole is more than the sum of its parts" — Aristotle

Interfacing with a number of hardware I/O devices and software components brings new design challenges, as communication system needs to continue operating despite any malfunctioning of the associated components.

First, some components become non-responsive if not used frequently. For instance, an open connection handle gets automatically closed by the MySQL server after a preset period of client inactivity. Second, explicit care has to be taken while developing device wrappers so that all the critical events are captured and processed. For example, if PTT is inadvertently pulled out and immediately pushed in, the OS may generate a different handler for that device.

Self-Correcting Design — This led us to design component interfaces in a self-correcting fashion. By frequently saving the working snapshot of a given component and by carefully handling the error conditions, the components could be restarted and their state be restored whenever the main application detects a problem with it. For example, our applications now have a seamless plug-and-play support for I/O devices by renewing the device handler upon detecting any hardware pullout/plug-in events.

Faulty Hardware — No design approach would solve the problem of physically faulty hardware. A problem since the early deployment days has been the effect of ESD on the PTT. For user convenience, the PTTs are designed with a metal clip to fix the PTT base to the garment while the student is on the move. This clip unfortunately could serve as an ESD carrier when a student is charged from clothing or humidity levels and has not been cautious to use ESD mats, thus spoiling the internal circuitry. Such issues are not only difficult for the VoIP system to detect but also for the FAA to fix in the short-term.

FAULT DIAGNOSIS

"A problem well stated is a problem half-solved" — Charles Kettering

Lost in Translation — On one hand, replicating the FAA Academy's training room setup at Columbia is impractical due to expensive air traffic equipment and lack of skilled testers. On the other hand, the FAA support staff working in on a tight schedule with demanding uptime requirements cannot always be expected to diagnose the faults precisely; the most commonly reported problem would read "workstation 23 can not hear anything."

The problem was especially evident in a prolonged episode in early 2007, when VoIP was deployed in new training rooms to be managed by self-trained technical support staff. On occasions when a particular workstation experienced a problem with media transmission or reception (either due to inadvertent PTT pullout or the ESD problem with the PTTs, neither of which were known till that point), the VoIP application on all the workstations was being restarted — potentially, disrupting the training. Since PTT-related issues were neither detected by the logging mechanism nor clearly understood by any of us until then, it led to situations where the researchers attributed this to an incorrect administration and the system administrators took it for an incorrect system behavior. Only after a site visit and continued discussions were the issues with PTT interfacing detected and heartbeat mechanisms were integrated into the system along with options to restart the VoIP application on selected workstations.

Remote Debugging — Since the FAA Academy operates in a highly-regulated federal environment, security policies prevent any form of remote connectivity to their computing systems. Thus, we had to build a mechanism such that all the interactions of the system with the users, the network and the associated hardware had to be extensively logged. Using log analyses to reconstruct the chain of events that lead to a failure, is a painfully long process and without a guarantee of success. Such experiences led us to rely heavily on a self-correcting design.

CONCLUSION

This research and development endeavor is a novel case-study of a successful application of VoIP technologies beyond the Internet telephony. The article also shares our experiences in deploying and maintaining the VoIP system over the last two years at the FAA Academy.

VoIP has been identified as one of the voice technologies for the proposed Next Generation Air-space System (NAS) to be designed and deployed by 2020. It is natural to think about the applicability of our VoIP-enabled training system for real world air traffic management. While the nature of the communication channels and the air traffic devices remain almost the same, the underlying network setup invalidates our design assumptions on security and scalability, thus making the training system completely unusable in the field as-is. It is our hope that the success of this training system, combined with existing large-scale deployment of VoIP in the real world serves as stepping stones in this direction.

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BIOGRAPHIES

SUPREETH SUBRAMANYA (sbs2139@cs.columbia.edu) is a doctoral student in the Computer Science Department at Purdue University. He holds dual M.S. degrees in electrical engineering and computer science from Columbia University. Previously, he worked as a research assistant with Prof. Henning Schulzrinne in the Internet Real-Time Laboratory during 2006–2009 and as a software engineer in various industrial environments during 2002–2006. His research focus includes real-time networking and embedded systems.

XIAOTAO WU (xiaotaow@cs.columbia.edu) has been a research scientist at Avaya Labs Research since 2005. He received his B.S. and B.E. degrees from Tsinghua University, Beijing, China, in 1994, and M.S. and Ph.D. degrees from Columbia University, New York, in 1999 and 2007, respectively. His research is on Internet real-time and multimedia communication services and protocols.

HENNING SCHULZRINNE [F] (hgs@cs.columbia.edu), Levi Professor of Computer Science at Columbia University, received his Ph.D. from the University of Massachusetts at Amherst. He was an MTS at AT&T Bell Laboratories and an associate department head at GMD-Fokus (Berlin), before joining the Computer Science and Electrical Engineering Departments at Columbia University. He served as chair of Computer Science from 2004 to 2009. Protocols co-developed by him, such as RTP, RTSP, and SIP, are now Internet standards, used by almost all Internet telephony and multimedia applications. His research interests include Internet multimedia systems, ubiquitous computing, and mobile systems.

SUSAN BURIAK (susan.e.buriak@faa.gov) is with the U.S. Federal Aviation Administration (FAA), where she works as a program manager at the Civil Aerospace Medical Institute, Aerospace Medical Education Division. Previously, she was a program manager for the Interactive Instructional Delivery Systems at the FAA Academy, Air Traffic Training Division. She holds graduate degrees from Purdue University and the University of Oklahoma Health Sciences Center. Her areas of expertise include information technology support for education and training, instructional facility design, and program and quality management.