

A voice over IP system to solve the problem of mobile radio interoperability

John H. Mock, W.Thomas Miller
University of New Hampshire, ECE Department
Kingsbury Hall, Durham, NH 03824

Abstract- While responding to emergency situations, personnel from different departments need to communicate with each other. Currently, unless every department has the same brand of radio, this is not possible due to lack of radio interoperability.

This research addresses the need for these non-interoperable mobile units to communicate with one another. It is not financially possible for these agencies to each purchase new, matching radios; instead a solution is being developed using Voice Over IP (VoIP). This solution involves establishing a network consisting of a single central server and clients located at each of the departments. Interfaced to each client is a radio that is compatible with the individual department's mobile radio units. Each client runs an H.323 compliant application that was developed using Microsoft's Telephony Application Programming Interface 3.0 (TAPI 3.0).

A prototype system has been developed and preliminary testing has proven the system operates successfully. Further testing methods for determining voice quality are being evaluated.

I. INTRODUCTION

Project54 is a collaborative research and development effort between the University of New Hampshire and the New Hampshire Department of Safety and is supported by the U.S. Department of Justice. The goal of the project is to apply high technology to law enforcement so that officers can perform their duties more safely and efficiently [1].

While performing their duties, police officers and other emergency service personnel in mobile units from various departments need to communicate with each other. Currently, unless every department has identical radios, this is impossible due to lack of radio interoperability. The motivation for the work presented here was the need for these non-interoperable mobile units to communicate with one another. It is not financially possible for these agencies to each purchase new, matching radios; instead a solution is being developed using Voice Over IP (VoIP). This solution involves establishing a statewide VoIP network consisting of a single central server and a client located at each of the participating departments.

The central server will host a voice conference that follows the H.323 standard for audio, video, and data communications across the Internet. A H.323 compliant application was developed using Microsoft's Telephony Application Programming Interface 3.0 (TAPI 3.0) to run on a desktop computer at the client side. A radio that is compatible with the individual departments' mobile radio units will be interfaced to the desktop computer at the client.

When a mobile unit wishes to communicate with a mobile unit from a different department, instead of doing so directly, the transmission will traverse the established network. A voice conference must first be created on the server if there is not one present. Transmission will begin at the mobile unit and travel to its headquarters. At its headquarters, the transmission will be processed by the client and sent to the conference on the server. The server will then distribute the transmission to whoever has joined the conference. Each client computer at the different agencies then sends the transmission out using a radio compatible with its mobile units. The entire system described is depicted in Fig.1.

II. SYSTEM DESCRIPTION

In keeping with the goal of being cost-effective, this system is being designed using mostly off-the-shelf components. The hardware used to host the voice conference is a server-class computer. The server uses First Virtual Communications' (FVC) Conference Server software based multipoint control unit (MCU). Conference Server enables the hosting of multiple simultaneous conferences with audio and video mixing [2]. The video features are not currently used; buy may be incorporated into the system in the future. While Conference Server supports the H.323, SIP, and T.120 conferencing standards, only H.323 endpoints will be used. H.323 is a recommendation from the International Telecommunications Union (ITU) that sets standards for multimedia communications over IP based networks that do not provide a guaranteed Quality of Service (QoS), such as the Internet [3]. To counteract the effects of network latency and lack of guaranteed QoS, H.323 uses the Real-time Transport Protocol (RTP) as a foundation.

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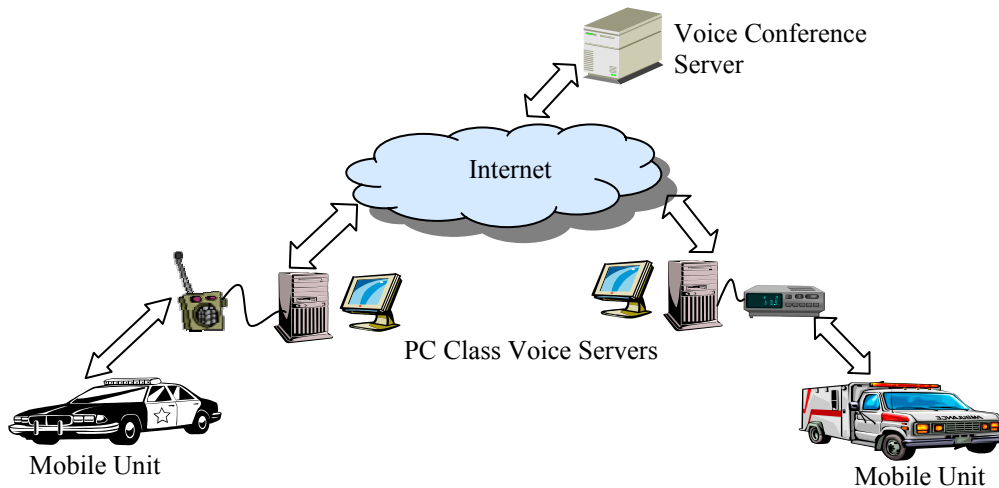


Figure 1- System for voice Over IP solution for mobile radio interoperability

RTP is an application layer protocol that provides a degree of QoS over IP based networks. The conference server collects audio streams from all connected clients and mixes them into a single stream. Each client then receives this mixed stream. During a conference all clients must use the same audio codec, which is determined at conference creation. Conference Server supports the following audio codecs: G.711, G.723, and G.729A. Most endpoints will choose to use either the G.723 or G.729A codecs because their maximum bit rate is much lower than that of G.711, 6.3 kbit/sec and 8 kbit/sec for G.723 and G.729A respectively versus 64 kbit/sec for G.711 [4]. Sound quality is better using G.711 over an error free channel with guaranteed bandwidth, but the Internet rarely meets these constraints. The lower bit-rate codecs perform much better when errors are present and bandwidth is restricted [5]. Conference management is performed by using either a set of Telnet commands or an HTML based interface. An application has been developed to act as a user-friendly interface for issuing server configuration Telnet commands. Figure 2 shows the GUI for the Project54 Conference Configuration Application. This application allows the user to create a conference by specifying a conference name, number, and audio codec. A conference may also be deleted from the server by specifying its number. At any time the user may also refresh the list of available conferences on the server.

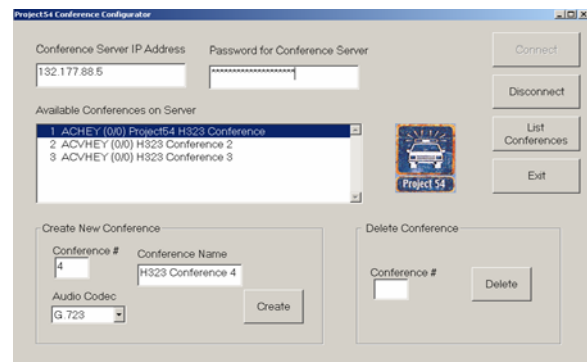


Figure 2 - GUI of conference configuration application

The hardware used on the client-side consists of a desktop computer, a radio compatible with the mobile units from that department, and an interface between the two. The interface between the computer and the radio is used to pass audio between the two and to activate the radio's push-to-talk button. An H.323 compliant voice conferencing application (Project54 Voice Conference Application) was developed and is used on the client-side desktop computer. The application was developed using Microsoft's Telephony Application Programming Interface 3.0 (TAPI 3.0). The TAPI objects used in this application are shown in Figure 3. The TAPI object is the applications entry point to TAPI 3.0.

This object represents all telephony resources to which the local computer has access, allowing an application to enumerate all local addresses. An ADDRESS object represents the origination or destination point for a call. The application uses the ADDRESS object to create an outgoing CALL object. The CALL object represents the connection between the local address and a remote address; in our case this is the client and server, respectively. All call control, such as connecting and disconnecting, is done through the CALL object. The TERMINAL object represents the sink, or renderer, at the termination or origination point of the connection. In our application the TERMINAL object is mapped to hardware, such as the speakers and microphone, but can also be mapped to a file [6].

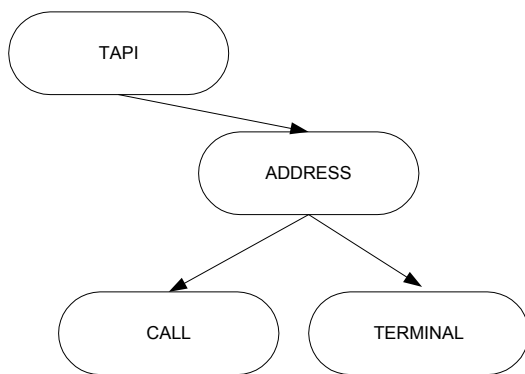


Figure 3 - TAPI objects

The window-based graphical user interface (GUI) of the application provides a simple interface for joining conferences. Figure 4 shows the GUI used by the Project54 Voice Conference Application. In order to join a conference, the following information must be known a priori; the IP address and password of the server. If this information is used, a connection can be made with a conference server. Once connected, a list of conferences available to join is displayed. The user can then select and join a conference from the list. The list can also be refreshed at any time.

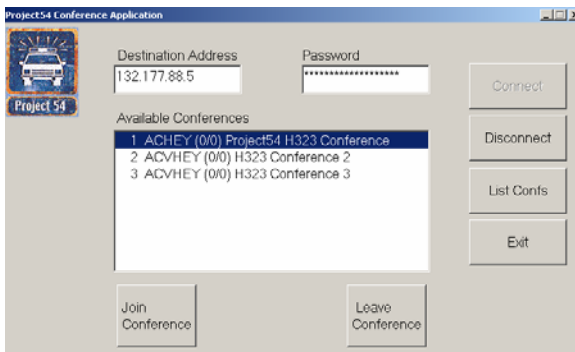


Figure 4 – GUI of voice conference application

III. VERIFICATION AND TESTING

The testing of this prototype system was performed using a NIST Net Router to emulate real network conditions. The National Institute of Standards and Technology Network Emulation Tool (NIST Net) is a general-purpose tool for emulating impairments in IP networks. It is designed to allow controlled, reproducible experiments with network performance sensitive applications. NIST Net is implemented as a kernel module extension to the Linux operating system and is configured through a set of commands or an X Window based GUI. [7]. The tool can be used to emulate numerous network impairments, including: variable packet delay distributions, congestion and packet loss, and bandwidth limitation. The X Windows interface allows the user to select and monitor specific traffic streams passing through the router and apply selected impairments to those streams. The tool allows an inexpensive PC-based router to emulate numerous performance scenarios, including: tunable packet delay distributions, congestion, bandwidth limitation, and packet reordering / duplication. The configuration of the testing system is shown in Figure 5. Client A is configured to use the NIST Net router to reach client B and likewise client B is configured to use the router to reach client A.

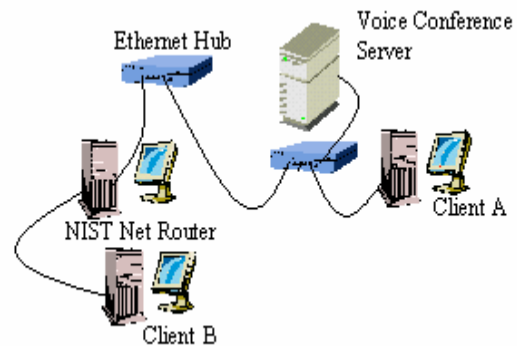


Figure 5 - Prototype system test setup

Preliminary testing of this prototype system consisted of ensuring basic audio conferencing functionality among two endpoints and subjectively testing audio quality under limited bandwidth conditions. All of the preliminary testing was performed with the clients using the G.723 audio codec. The bandwidths used for testing are: 10,000, 100, 57.6, 38.4, 28.8, 19.2, and 9.6 (all values have units of kilobits per second). This testing has proven successful. The two endpoints were able to join a conference hosted by the conference server and audio quality remained acceptable with bandwidth restrictions imposed.

IV. FUTURE WORK

Before this system can be deployed, further testing must be performed. Future testing will introduce the following network impairments into the system: delay, packet loss, and jitter. The goal is to simulate the conditions the final system will face when operating over the Internet. In addition to subjective, objective testing of voice quality will be performed. The objective testing algorithms that are currently being evaluated for use are: Measuring Normalized Blocks (MNB) [8], Perceptual Speech Quality Measure (PSQM) [9], and Perceptual Evaluation of Speech Quality (PESQ) [10]. Once testing is complete and the system is found to perform adequately under real-world conditions, it will be deployed across the state of New Hampshire.

V. CONCLUSION

A prototype system has been developed to address the issue of mobile radio interoperability. A server class computer with FVC Conference Server software based MCU is used to host voice conferences. A TAPI based voice conferencing application with a user-friendly GUI was developed for use on the client-side desktop computer. A test network for the prototype system was developed with the ability to introduce the following conditions:

- Packet delay
- Packet loss
- Jitter

Initial testing, which included ensuring basic audio conferencing functionality among two endpoints and subjectively testing audio quality under varied bandwidth conditions, has proven successful.

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