A voice over IP solution for mobile radio interoperability

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

Abstract- **While performing their duties, 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.**

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 authors wish to thank the US DOJ for funding Project54 through grant #1999-DD-BX-0082

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 agency, 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. All clients have the ability to perform conference creation or deletion. Transmission will begin at the mobile unit and travel to its base station. At its base station, 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.

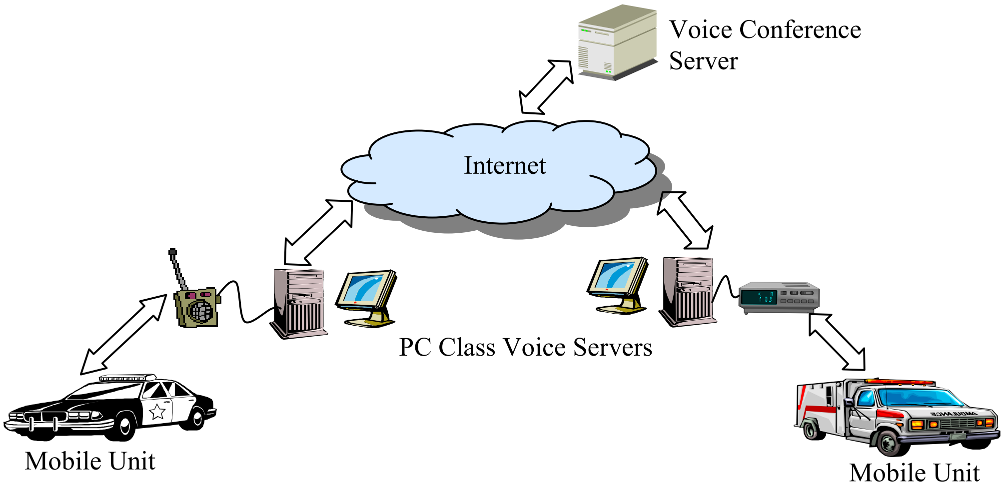

Figure 1- System for voice Over IP solution for mobile radio interoperability

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 can choose to receive this mixed stream or choose a stream from a single selected endpoint. Conference Server supports the following audio codecs: G.711, G.723, and G.729A. The mixing of these during a conference is also supported which means that every endpoint is not required to use the same codec. When the incoming audio streams are mixed and retransmitted, they are encoded using the same codec as the destination endpoint. 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 administration web page. This gives clients the ability to perform conference management.

The hardware used on the client-side consists of a desktop computer. 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 2. 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].

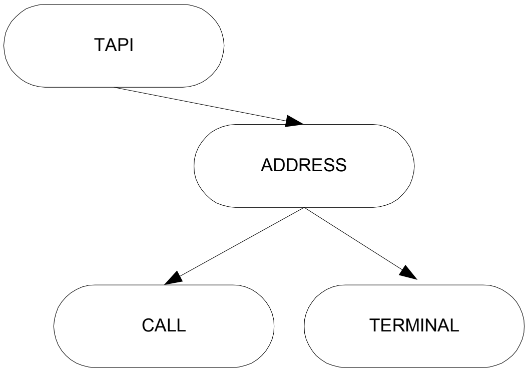

Figure 2 - TAPI objects

The window-based graphical user interface (GUI) of the application provides a simple interface for joining conferences. Figure 3 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 or name of the conference server and the names of the conferences that are available. The clients can view available conferences by using the Conference Server administration web-based GUI. In the Destination address type field the user chooses to use either a domain name or IP address to connect to the conference server. In the Conference Name field, a valid conference name must be entered before connecting to the server. When all fields are filled with valid entries, the Connect button is used to establish the connection with the server. The Disconnect button is used to leave a conference and disconnect from the server. While participating in a conference, it is not possible to join a different conference without disconnecting from the server. Once disconnected, the Conference Name field may be changed and a new connection to the server can then be established. Future enhancements to the client software may add the functionality of querying the server for the names of available conference. Future enhancements will also add the capability to create and delete conferences, but currently these functions are also performed through the web-based GUI.

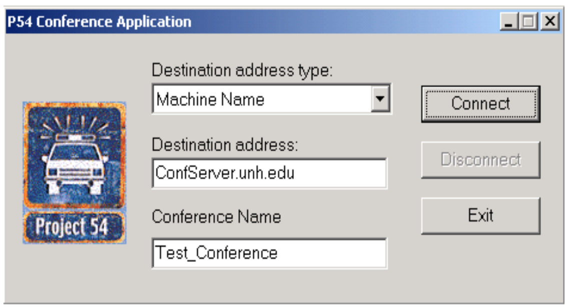

Figure 3 - Graphical user interface

III. VERIFICATION AND TESTING

The testing of this prototype system was performed using a NIST Net Router to emulate real network conditions. The NIST Net network emulator is a general-purpose tool for emulating performance dynamics in IP networks [7]. The tool is designed to allow controlled, reproducible experiments with network performance sensitive/adaptive applications and control protocols in a simple laboratory setting. By operating at the IP level, NIST Net can emulate the critical end-to-end performance characteristics imposed by various wide area network situations. NIST Net is implemented as a kernel module extension to the Linux operating system and an X Window System-based user interface application. 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 4. Clients A and B are configured to use the NIST Net router to reach client C and likewise client C is configured to use the router to reach clients A and B.

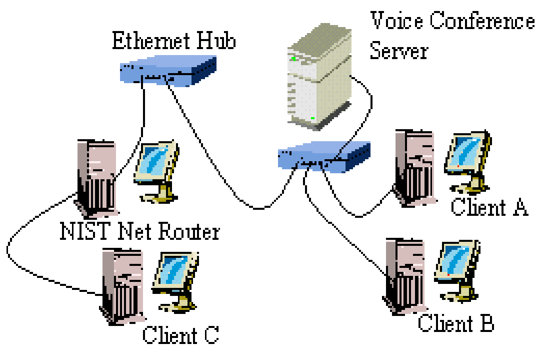

Figure 4 - Prototype system test setup

Preliminary testing of this prototype system consisted of ensuring basic audio conferencing functionality among three endpoints and subjectively testing audio quality under limited bandwidth conditions. All of the preliminary testing was performed with all clients using the G.729A 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. All three endpoints were able to join a conference hosted by the Conference Server application 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, congestion, and packet reordering. 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], Perceptual Evaluation of Speech Quality (PESQ) [10]. If testing of the prototype system is successful, an interface to the client-side desktop computers will be developed and the mobile radios will be incorporated into the system. This completed system will then be put through the same tests as the prototype. 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;
- Congestion;
- Packet reordering/duplication.

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

REFERENCES

[1] A. L. Kun, W. T. Miller, W. H. Lenharth, *"Modular System Architecture for Electronic Device Integration in Police Cruisers,"* IEEE Intelligent Vehicle Symposium, Versailles, France, June 18-20, 2002

[2] "CUseeMe Conference Server technical overview," CUseeMe Networks, 2001

[3] "Packet Based Multimedia Communications Systems," ITU-T Recommendation H.323, November 2000

[4] R.J.B. Reynolds, A.W. Rix, "Quality VoIP – an engineering challenge,",*BT Technology Journal*, April 2001, pp. 24

[5] S. Yamarthy, "Subjective and objective evaluation of voice quality over internet telephony," *MS Project, ECE Department*, University of New Hampshire, 2000

[6] "IP telephony with TAPI 3.0," MSDN library, April 2000

[7] www.antd.nist.gov/itg/nistnet/

[8] "Objective quality measurement of telephone-band (300-3400 HZ) speech codecs," ITU-T Recommendation P.861 Appendix II, February 1998

[9] "Objective quality measurement of telephone-band (300-3400 HZ) speech codecs," ITU-T Recommendation P.861 Appendix II, February 1998

[10] "Perceptual evaluation of speech quality (PESQ), an objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs," ITU-T Recommendation P.862, February 2001