The accompanying Senior Thesis Addendum, "Updated Acceptance Test Requirements," is submitted in accordance with the requirements of CEEN 4980. I am requesting approval of the latest changes to the Acceptance Test.

Respectfully yours,

Steven J. Eggerling
Senior Thesis Addendum

Updated Acceptance Test Requirements:

VI. ACCEPTANCE TEST

The acceptance test will include:

a) A successful contact of another machine using my Client/Server Browser Application.

b) The Chat Application will be able to initialize voice communication by using the supplied IP address stored on the clipboard by the Client/Server Browser Application.

c) The voice communication can be heard and understood reasonably well on both sides using the telephone/PhoneBridge and headset on second machine.

d) Demonstrate MATLAB DTMF detection from a pre-recorded telephone sound file.

Testing is very straightforward: by successfully completing the four steps shown above, this alone proves without any doubt that the project works as intended by its creator (Steven J. Eggerling).
Voice over Internet Protocol (VoIP)

by

Steven J. Eggerling

A SENIOR THESIS PROPOSAL

Presented to the Faculty of
The Computer and Electronics Engineering Department
In Partial Fulfillment of Requirements
For CEEN 4980 Senior Thesis Proposal

Major: Computer Engineering

The University of Nebraska-Lincoln, Omaha Campus

Fall, 2001
The accompanying Senior Thesis Proposal, "Voice over Internet Protocol," is submitted in accordance with the requirements of CEEN 4980, Senior Thesis Proposal. As stated in the proposal, the project will be done for and funded by myself. I am building this project for the sole benefit and enjoyment of learning how VoIP applications work across networks, specifically the Internet. I worked with Dr. Hamid Sharif to come up with this final proposal.

Respectfully yours,

Steven J. Eggerling
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I. BACKGROUND

Voice over Internet Protocol (VoIP) is one of today’s most sought after e-solutions. Every day of the year, we depend on our national telephone system to send and receive voice communications to and from “long-distance” locations. With broadband Internet Service Providers becoming more available to the average consumer (i.e. Cable, DSL, Wireless, etc.), more people are making calls to distant places using VoIP technology. Cheaper calls can be made today compared to current long-distance telephone rates without sacrificing quality.

Of course, there is a price to pay. Many services are using advertising banners on their services to supplement the cost of bandwidth and other related expenses. Many end users get irritated with this type of payment, but unfortunately, there is no free lunch. In the future, when everything has gone broadband, IP-to-IP will become even more practical than at its current stage now.

II. THESIS OVERVIEW

VoIP can be achieved several ways, not just one method. Due to the costly implementation of the single-user IP-to-Phone network solution for me, I will focus solely on the relatively inexpensive base IP-to-IP solution (direct PC-to-PC communication).

One of the goals for this project will be to implement an “IP dialing” system on a touch-tone telephone. This telephone will be connected through the PC hooked to the Internet. The main user (PC1) will be able to dial another computer’s IP address (PC2) into the telephone keypad and talk to that person if they accept the call.

Other minor things such as data compression and encryption will be looked at and addressed. Compression can relieve unnecessary bandwidth waste, enhance voice quality on slower
connections, remedy network congestion, and other common problems. Encryption will ensure private conversations over the very public Internet.

III. GENERAL DESCRIPTION

Many areas of computers and computer-based networks will have to be researched thoroughly before implementing this project. Several network architectures and Internet protocols will need to be researched in order to achieve a satisfactory end user product. Research of modem and soundcard architectures will also need to be done. Bandwidth and server load problems will be researched and addressed appropriately. Along with the aforementioned, other areas of research may be needed as the project progresses.

One-to-one, IP-to-IP technology will be implemented. This will be achieved with a real-time bi-directional data stream between PC1 and PC2, with a master network server in control of timeouts, logins, and other relevant data. The server load will be minimal due to Client-to-Server polling at predefined intervals rather than Server-to-Client polling (which would require the task to be active on the server at all times).

Research of different data voice compression standards will allow me to implement the best one for any given situation. Also, a special noise-canceling microphone will be used in order to conserve bandwidth and quality on PC2.

In the area of security, “voice encryption keys” will be used to keep conversations private. These “voice encryption keys” will be unique ID’s that when encoded through the streaming data, only people with matching “voice encryption keys” will be able to understand each other. The encryption key system described will utilize at least a 128-bit encryption method.
The client-side application will be developed under Visual Basic and/or Visual C++. This will be initially coded in order to satisfy the Windows-Intel/AMD platform standards. The server-side application(s) will include database technology combined with CGI (Common Gateway Interface) to bind “on-line” users together. PERL, PHP, ASP or another comparable language will be used to run the server-side host or master network server (Apache or IIS).

The figure on the bottom of the page (Figure 1) shows the basic setup of the VoIP system when it is fully implemented. The Internet cloud represents both the server and the connection made between the two machines. A regular touch-tone telephone is connected through a hardware box. This box will house a Digital-to-Analog converter for voice data traveling from PC1 to the telephone. It will also house an Analog-to-Digital converter for voice data traveling from the telephone to PC1’s Serial Port. Voice data will travel from PC1 through the Internet down to PC2, where it will be played on the headset. Voice data will also travel from PC2 through the Internet down to PC1, where it will be played on the telephone receiver.

Figure 1 – VoIP System.
IV. COMPONENTS LIST

I will be funding this specific project myself. For the testing and development part of the project, I have computers, a server, a broadband connection, all programming packages, and noise-canceling microphones. I will also buy the USART, connectors, A-D/D-A converters, telephone equipment, and other required hardware. As stated before, I am doing this project for the sole enjoyment and benefit of learning how VoIP applications work across networks, specifically the Internet.

*Partial Components List (+testing equipment)*

- Telephone Equipment (+microphone) $50
- Serial Port Interface hardware $40
- Internal hardware (other) $100-200
- Software $200
- Router/2 Computers/Broadband $3000
- Server $120

V. TIME SCHEDULE

The research should be done within one month. The rest of the project will need to be put together side by side. I hope to be finished with the main client/server software by late September 2001. The hardware will be finished by mid- to late-November 2001 and the final testing of the whole system will be complete by the beginning of December 2001. Please note that I will be testing this project piece by piece. Therefore, this means that I will be testing all the way through building it also, not just in the end. These are just date estimations, meant to be a guide of where I’m going, not set in stone by any means. Figure 2 (next page) is a rough outline of how my schedule will flow.
VI. ACCEPTANCE TEST

The acceptance test will include:

a) A successful contact of another machine using the IP address dialed by the telephone.

b) The initialization of voice transmission across the Internet.

c) The voice communication can be heard and understood reasonably well on both sides.

Testing is very straightforward: by successfully completing the three steps shown above, this alone proves without any doubt that the project works as intended by its creator.

VII. TEAM MEMBERS

Steven J. Eggerling
VIII. SUMMARY

As stated above, IP-to-IP technology will be implemented. This will be achieved with a real-time bi-directional data stream between PC1 and PC2, with a master network server in control of timeouts, logins, and other relevant data. The server-side application(s) will include database technology combined with CGI (Common Gateway Interface) to bind “on-line” users together. A list of users (friends) will appear along with their current IP address. A regular touch-tone telephone will be connected through a hardware box. This box will house a Digital-to-Analog converter for voice data traveling from PC1 to the telephone. It will also house an Analog-to-Digital converter for voice data traveling from the telephone to PC1’s Serial Port. Voice data will travel from PC1 through the Internet down to PC2, where it will be played on the headset. Voice data will also travel from PC2 through the Internet down to PC1, where it will be played on the telephone receiver. Successful contact and bi-directional transmission of voice data will prove that this project works.

This project will let people have the comfort of using their telephone to talk over the Internet. This could eliminate awkward microphones and unclear computer speakers, which has always been the case for me. There’s something to talk about: “Making long-distance calls on your own telephone for next to nothing with top quality.”