DEVELOPING A DECENTRALIZED AUDIO CONFERENCING FACILITY OVER VOICE OVER IP (VoIP) NETWORK

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ABSTRACT

VoIP audio conferencing has become one of the major business tools today. Participants of a conference connect to a Multipoint Connection Unit (MCU) that facilitates connectivity, audio mixing and audio packet unicasting. The current and widely used conferencing scheme today uses MCU equipment to facilitate a centralized audio conference. However, these equipment do not reach all areas of the world to allow participants in these areas to join conferences and enable audio conferencing. A solution to this problem is to create a decentralized audio conferencing scheme that will eliminate the limitations of using MCU equipment. However, the current decentralized audio conferencing scheme that requires a component of the MCU, which is the Multipoint Controller, to reside in one of the participants, and probably different equipment, poses more limitations due to its complexity. This paper discusses the current VoIP audio conferencing schemes, their limitations, and the proposed alternative decentralized VoIP audio conferencing scheme that will use VoIP terminals enabled with MCU capabilities. This paper also discusses a methodology for creating an MCU-Enabled VoIP softphone that will be used to start, connect, and maintain a decentralized audio conference.

Keywords:

VoIP, Audio Conferencing, MCU, VoIP Terminals

Chapter 1

INTRODUCTION

1.1 Background

Through the advances in the Internet technology, communication now has developed into a higher level. Instead of using phone lines to communicate, we can now use the Internet through Voice over Internet Protocol (VoIP) technology. In VoIP technology, a VoIP device (hardware or software or a combination of both) converts analog sound signals into packets, breaks them down to pieces, and then sends them to the destination.

Using VoIP to make calls has many advantages over traditional phone networks, besides the fact that VoIP calls are cheaper, and in some cases free. One of its advantages is mobility: wherever in the world you may be as long as you and the one you are calling has a connection to the Internet, communication can be possible.

The inherent capabilities of VoIP make it a leading tool for communication nowadays. VoIP technology is now integrated in traditional phone networks, making different kinds of calls like PC to Phone line calls and vice versa. Another area of communication that VoIP has majored is the audio conference call service. Audio conferencing using VoIP works similarly as traditional conferencing using analog telephones, wherein the clients connect to a common Multipoint Control Unit (MCU) to be able to communicate with each other. If this is the case, conference calls through VoIP may be experiencing the same limitations experienced by the traditional conference calls through analog phone lines. Conference calls, which are hosted by a Multipoint Control Unit (MCU), can only accommodate a fixed number of clients predetermined before the conference starts. Limitations on bandwidth can also be a problem on the client side upon connection to the MCU server. Further, the availability of audio conferencing equipment makes it impossible for clients in some areas to participate in audio conferencing. The proponents found this area as a challenging research problem and looked into the merits of a decentralized approach as a possible solution.

1.2 Statement of the Problem

This study aimed to address the general problem: How is a decentralized audio conferencing facility for VoIP audio conferencing designed?

Specifically, this study aimed to address the following questions:

1. How does a Multipoint Control Unit device (MCU) facilitate audio conferencing?
2. How are softphones enhanced with MCU functionalities to facilitate or join a decentralized audio conference?
3. How is a decentralized audio conferencing facility without negative effects on audio quality designed?
4. How is a decentralized audio conferencing functionality that will not require VoIP terminals to have a physical one-to-one connection to n-1 other connections designed?

1.3 Objectives of the Study

The general objective of this study was to develop a decentralized audio conferencing facility for VoIP conferencing.

The specific objectives of this study were:

1. To identify specific MCU device audio conferencing functionalities that would be enhanced on a soft-phone.
2. To enhance a softphone with MCU functionalities to facilitate or join a decentralized VoIP audio conference.
3. To design a decentralized audio conferencing scheme without negative effects on sound quality.
4. To design a decentralized audio conferencing facility that would require a VoIP terminal to connect only to at least one of the currently participating VoIP Terminals.

1.4 Scope and Limitations of the Study

This study defined an alternative decentralized audio conferencing scheme by enhancing a VoIP softphone with MCU capabilities used in audio conferencing. The proponents used the current ITU-T standard protocol for multimedia conferencing over packet-based networks, which is H.323. It defines the standards on audio conferencing methods that include call connection, audio compression and decompression, codec synchronization, and packet distribution.

This study was limited to audio conferencing only. Moreover, it did not include video, text and other data manipulation and distribution. This study only focused on single conferencing and not on multi-conferencing. Further, this study did not include the implementation of a VoIP gatekeeper that facilitates IP address translations, from H.323 aliases to network addresses. This entails the participants (VoIP terminals) to know the network addresses of the other participants of the conference. Lastly, this study used the available OpenPhone and OpenMcu components from OpenH323.org, and Mr. Phyll Astorga's OpenH323.ocx in the development of the software facility.

1.5 Significance of the Study

The success of this study would be beneficial on areas that have demands on audio conferencing, but has limited audio conferencing equipment. It would also be beneficial for participants that have limited connection speeds since they can now have multiple choices on where to connect based on their connection speeds.

VoIP conference calls can no longer be limited by equipment and would instead make VoIP and its technologies accessible to all areas of the world. Future VoIP developments would no longer focus on server equipment development but on the client's software enhancements instead, giving more control and power to the clients.

1.6 Definition of Terms

1.6.1 **Packet switching** is the now-dominant communications paradigm in which packets (units of information carriage) are routed between nodes over data links shared with other traffic.

1.6.2 The **Internet** is the worldwide, publicly accessible network of interconnected computer networks that transmit data by packet switching using the standard Internet Protocol (IP).

1.6.3 A **protocol** is the special set of rules that end points in a telecommunication connection use when they communicate.

1.6.4 The **Internet Protocol (IP)** is the method or protocol by which data is sent from one computer to another on the Internet.

1.6.5 **VoIP (Voice over IP)** is a technology that allows audio communication through the internet. VoIP allows conversion of analog audio signals to digital audio signals that will travel through the internet.

1.6.6 **H.323** is a protocol standard for multimedia communications. H.323 was designed to support real-time transfer of audio data over packet networks like IP.

1.6.7 **H.323 Terminal** can either be a Personal Computer (PC) or a stand-alone device that runs a H.323 application.

1.6.8 **VoIP Call** is a point-to-point audio communication between two H.323 Terminals.

1.6.9 A **softphone** is a H.323 application for making VoIP calls over the Internet using a general-purpose computer, rather than using dedicated hardware.

1.6.10 A **codec** is a device or program capable of performing encoding and decoding on a digital data stream or signal. The word *codec* may be a combination of any of the following: '**Co**mpressor-**Dec**ompressor', '**Co**der-**Dec**oder', or '**Co**mpression/**Dec**ompression algorithm'.

1.6.11 **Multipoint Control Unit (MCU)** is H.323 device that enables three or more H.323 terminals to participate in a multipoint conference. The MCU includes a mandatory Multipoint Controller and optional Multipoint Processors.

1.6.12 **Multipoint Conference** is a conference between three or more terminals, which may be on the LAN or on the Circuit Switched Network.

1.6.13 **Centralized Multipoint Conference** is call conference in which all participating terminals communicate in a point-to-point fashion through a single MCU device.

1.6.14 **Decentralized Multipoint Conference** is a call conference in which the participating terminals communicate with-out connecting to a single MCU device.

1.6.15 **Multipoint Controller (MC)** is an entity that provides for the control of three or more terminals in a MCU.

1.6.16 **Multipoint Processor (MP)** is an entity that provides for the processing of audio, video, and/or data streams in a multipoint conference. The MP provides for the mixing, switching, or other processing of media streams under the control of the MC.

1.6.17 A **gatekeeper** is a management tool for H.323 multimedia networks. A single gatekeeper controls interactions for each zone, which comprises the terminals, multipoint control units (MCUs), and gateways within a particular domain. Although the gatekeeper is an optional component, when it is included, it becomes the central administrative entity.

1.6.18 **PChannel** is a base class of an object defined in PWLib that is used to facilitate audio data processing. PChannel contains unimplemented functions for Write and Read of audio data.

1.6.19 **PSoundChannel** is an inherited an object from PChannel that is used by H.323 to access a VoIP Terminal's sound card resource. It is a serves as a medium for the travel of audio data to and from the VoIP Terminal's sound card. It has an implementation of the functions Read and Write.

1.6.20 **MCUPhone** is the term used in this paper to identifiy the enhanced OpenMCU software that is now capable of communicating among participants in an audio conference.