Design of an Internet Fax & Voice Gateway

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ABSTRACT

There has been a growing interest in recent years in developing real-time voice and fax communication software for use over the Internet. This paper discusses the design of an Internet Telephony Gateway (ITG) that is used to support the delivery of fax and/or voice services through the Internet to conventional telephones and fax machines on the Public Switched Telephone Network (PSTN). The interface board can offer multiple ports of voice and fax services in a single slot using the EISA bus.

The hardware system comprises two microprocessors and one digital signal processor (DSP). One of the microprocessors is used for the telephone trunk line interface management and system control while the other one is used solely for the fax modem control. The DSP is used to compress and decompress the voice signal for real-time voice communication. Two algorithms to compress the digitized voice signal, namely, Global System for Mobile Communication (GSM) and Code Excited Linear Prediction (CELP), are used in the design. GSM is used to compress primary voice data packets while CELP is used to compress redundancy voice data packets to cater for potential lost data packets during network transmission. The software on the server gateway provides the functionality between two communicating gateways and manages the exchange of fax and voice between senders and recipients by using Internet as the transport medium.

Keywords: Internet, telephony system, Internet telephony gateway, voice communication, fax communication

1. INTRODUCTION

Internet telephony is a general term that applies to the voice and fax services delivered over the Internet. The potential of Internet telephony circumventing long distance rates is certainly appealing as one can fax or talk to someone on the other side of the world at no more than the cost of the local Internet access and connectivity charges. Besides the benefits of cost saving, using the Internet as an alternative backbone for voice and fax traffic makes it possible to
integrate voice, data, graphics and text on the same network. This is certainly beyond the capability of conventional circuit switched telephony.

Currently, the most common application for the Internet telephony is the Internet Phone that provides real-time voice communication over the Internet. Internet Phone requires the use of computers at both caller and recipient ends. Another emerging Internet Phone application is the Internet Telephone Gateway (ITG) that allows the delivery of voice services through the Internet to conventional telephones on the Public Switched Telephone Network (PSTN) or a Private Branch Exchange (PBX), without requiring users to have a computer. One such system currently available is VocalTec’s Internet Phone Telephony Gateway Server [1] that is the earliest commercial product announced in 1996. Recently, another twenty firms have entered or announced intentions to provide Internet voice services for computers and/or conventional telephone users.

Another useful application that an ITG should support is the transmission of fax messages. Due to the large base of traditional stand-alone fax machines that are in use throughout the world, the ITG can be extended to handle these fax calls in much the same way it does for traditional phone calls.

This paper proposes a hardware design for an ITG that can handle both voice and fax communication simultaneously over the Internet. Using this hardware, multiple voice and fax calls can be processed and transmitted over the Internet. Such a solution will enable users anywhere in the world to communicate with one another using a set of conventional telephones and fax machines. It will find particular applications in the area of office communication, especially in large multi-national corporations (MNC) whose operations are geographically dispersed around the world.

2. SYSTEM REQUIREMENTS

Figure 1 Internet Telephony Gateway
Figure 1 shows the ITG environment that will allow callers (i.e. users) to place voice calls or send faxes that are routed over the Internet. The caller first calls the local ITG (caller ITG) number via the PSTN/PBX. When prompted by the Interactive Voice Response (IVR) system of the ITG, the caller enters the destination telephone number or fax number. In the case of a voice call, it will be routed over the Internet to destination ITG (target ITG) serving the specified telephone number in the destination country. From the target ITG, the call is correspondingly activated at the PSTN/PBX to its recipient. In general, the local ITG (caller ITG) will locate and establish contact with the remote ITG (target ITG) through the Internet using the standard Transmission Control Protocol /Internet protocol (TCP/IP) suite [2]. If successful, the sender and recipient would then be logically connected together via the two ITGs to allow real-time voice conversation to take place. In the case of a fax transmission, the ITG will act as a 'store-and-forward' center to cater for higher network reliability requirements. The fax message would eventually be transmitted from the caller ITG to the target ITG and subsequently to the intended fax machine by the target ITG.

In providing such functionality, the ITG essentially integrates the Internet with the traditional telephone network. The hardware of the ITG is thus designed to support the following system requirements:

- The ITG possesses an interface to the trunk line to support the normal call progress functions. For example, the ITG is capable of detecting the ringing signal, receiving and transmitting Dual Tone Multiple Frequency (DTMF) signals and voice signals.
- The ITG has Interactive Voice Response (IVR) function to inform and prompt the caller to enter the recipient's telephone or fax number and issue other instructions.
- The ITG receives and transmits real-time voice data.
- The ITG receives and transmits faxes.
- The ITG processes multiple calls simultaneously.
- The ITG interfaces with the gateway server via the EISA bus and is controlled by a set of Application Programming Interfaces (APIs) that forms part of the gateway server software.

3. HARDWARE SYSTEM DESIGN

Figure 2 shows the system architecture of the ITG that supports a two-port telephony gateway. This architecture is scalable and can be expanded into a multiple -port system up to a maximum of sixteen channels. The architecture combines the signal processing capabilities of a DSP with the decision making and data movement functionality of a general-purpose microprocessor. All operations are interrupt-driven to meet the demands of a real-time system.

The ITG comprises four main modules to support the necessary system requirements of the previous section:

- The Telephone Line Interface Circuit connects the gateway to the trunk lines from the local exchange. It receives the analog voice and telephony signaling information from the telephone network. It comprises a Central Office Interface chip and an integrated DTMF Transceiver chip. The former performs transformerless 2 to 4 wire conversion, between the 2-wire telephone loop and the 4 wire transmit and receive pairs of a voice switching system while the DTMF transceiver has a built-in call progress filter. A call progress filter is selected to allow the microprocessor to analyze call progress tones. It utilizes an Intel micro interface that allows the devices to be connected to a number of popular microcontroller with minimal external logic.
• The System Control Microprocessor controls all operations via a local control bus and interprets and executes commands from the server gateway CPU. This microprocessor handles real-time events, manages data flow to the server gateway CPU to provide faster system response time, reduce CPU processing demands, process the multiple ports DTMF and telephony signaling before passing them to the application. It also manages the IVR system of pre-recorded voice message to prompt the caller to enter the receiver's telephone number.

The Intel 80186 microprocessor [3] is selected as the system control microprocessor. The 80186 is an enhanced version of the earlier 8086 microprocessor. The 80186 family contains a clock generator, a programmable interrupt controller, programmable timers, a programmable DMA controller, and a programmable chip selection unit. These enhancements greatly increase the utility of the 80186 and reduce the number of peripheral components required to implement a system.

• The Fax Modem System is responsible for handling faxes in the ITG. When the system control microprocessor receives a call from one port and judges it as a fax signal, it will communicate with the second microprocessor that controls the fax modem and allows the modem to receive the fax signals. When a fixed amount of fax data is received, the fax modem control microprocessor will request for DMA transportation from the system microprocessor to the server gateway CPU. If both ports contain incoming fax data, the fax modem control microprocessor will control the fax modem to read the fax data from each port at fixed time intervals. As in the system control microprocessor, the Intel 80186
is selected as the fax modem controller in the hardware design.

- The DSP Voice Codec is used to compress the voice data for transmission and decompress it for playback in order to implement real-time voice communication between caller and recipient. Motorola's DSP56001 [4] is used to provide the GSM [5] and CELP [6] voice codecs that the server software needs. When the system control microprocessor receives a call from one port and judges it as a voice signal, it will communicate with the DSP and allow it to receive the voice signals. The system control microprocessor (80186) communicate with the DSP via the DSP's Host Interface. The analog voice signal is first digitized using a PCM codec and subsequently followed by compression by the DSP. The codec is interfaced to the DSP’s Synchronous Serial Interface (SSI) port. When a fixed amount of voice data is received, the DSP will request for DMA transportation from the system microprocessor to the server gateway CPU.

4. SOFTWARE SYSTEM DESIGN

4.1 Driver Software

The driver program controls the interface board operation and provides the APIs for the server gateway software to support for data transmission over the Internet. Programming the Intel 80186 and DSP56001 respectively complements the driver software. The flowchart for the receiving process and transmitting process for a single port is shown in Figures 3 and 4 respectively.

4.2 Gateway Server Software

The system architecture for the gateway server software is shown in Figure 5. It comprises three main components, namely, Session Management, Audio Management and Fax Management to handle the gateway's overall functionality, voice and fax processing respectively [7-9]. These software modules provide the link between the hardware interface card and the Internet to provide connection management, monitoring of quality of service, negotiation of session parameters, audio and fax communication.

Session management is responsible for the connection process, monitoring the quality of services, providing an estimation of characteristic parameters such as the rate of lost packets, duplicated packets and delay, and the negotiation of session parameters between the ITGs. TCP/IP, which is a connection-oriented protocol of the TCP/IP protocol suite with a guaranteed delivery of sequenced data is used to transport control messages used for communication since control messages cannot tolerate packet loss, that will in turn, gravely compromise the functionality of the ITG.

Audio management is responsible for the audio recording, data packets time-stamping, transportation, incoming buffer management, packet ordering, packet lost replacement control and audio playback. Incoming audio is sampled at regular intervals and time-stamped prior to being transmitted to the target ITG. UDP/IP, which is a connectionless protocol of the TCP/IP protocol suite with no guarantee of data delivery or proper sequencing is used to transport the audio packets mainly due to its speed. TCP/IP is not used for audio transmission since the re-transmission of lost packets is undesirable due to the desired real-time nature of the service. This explains why these extra mechanisms are required in the audio management process to achieve real-time communication between caller and recipient. The reverse process is carried out at the target ITG to decompress, arrange and make up for data packets lost so that a digital stream can be sent to the hardware interface for audio transmission over the trunk line.
Call Interrupt from Trunk Line

Yes

Set off-hook port marker

IVR

Read DTMF (to get the fax or voice coder)

Voice or Fax?

Voice

Read DTMF (to get telephone number)

Write DTMF and port Marker to Dual-port RAM

Line Busy?

No

Off-Hook?

Yes

DSP Receiving and Transmitting Voice

Processing End?

No

End

Yes

Read DTMF (to get telephone number)

Write DTMF and port Marker to Dual-port RAM

Read Fax Data to Modem RAM

Data Transmitted in DMA to Dual-port RAM

Receiving End?

No

End

Yes

send Recipient Line Busy to Caller

Fax

End
Figure 3 Software Flowchart for ITG Receiving Process
Figure 4 Software Flowchart for ITG Transmitting Process
Although the occasional non-delivery of voice packets will reduce the final quality of voice, it is not disastrous. This is provided that small voice samples are dispatched each time and that a suitable mechanism exists to cater for packet lost replacement. In the current design, this is achieved by introducing redundancy in each voice data packet that is being transmitted to the server ITG. Thus, each data packet comprises a time stamp to indicate the time of data sampling, primary voice data for a fixed length of time (20ms) in GSM format (that offers toll-grade compression) and redundant voice data for a previous period (not necessary one time period back) in CELP format (that offers sub-standard audio quality but results in lower bandwidth and processing power requirements). If no corresponding lost data packets are encountered then the redundant audio data is simply discarded, else, it is used to make up for the lost data packets.

Fax management is responsible for the fax processing, transportation and receipt. Due to the store-and-forward nature of the fax system that is not real-time, digital data received from the hardware interface will be stored into a binary image file in the standard CCITT fax format ready for transmission to the target ITGs at regular intervals. The reverse is carried out at the receiving ITG so that a digital stream can be sent to the hardware interface for fax transmission over the trunk line. Fax management uses the connection-oriented TCP/IP protocol to guarantee the delivery of an error-free and reliable fax file.

5. EVALUATION

As Internet telephony gateways are fairly new concepts, there are not that many commercial products available as can be seen from the Pulver.com on-line guide on PSTN/IP Gateways [10]. However, this number will increase significantly as Internet telephony gains wider acceptance. This section compares three commercial gateways for voice and fax over the Internet with the ITG in order to evaluate the latter’s design and performance.
VocalTec Telephony Gateway 3.1 [1], launched recently in 1998 by VocalTec Communications Inc., is a commercial Internet telephony gateway based on the Dialogic voice and fax boards [11]. Similar to the ITG, it supports real-time, full duplex conversations over the Internet. When used with VocalTec’s Internet Phone [1], it can support computer to telephone conversations and vice versa. This newest release also supports Internet faxing. Used with its Surf&Call plug-in [1], it can support web to phone conversation and has third party support for billing purposes with full call logging and administration facilities. Target customers include Internet Service Providers offering value-added services, telecommuters and call centre operations.

The ITG when compared to the VocalTec Telephony Gateway, is a very much dedicated and trimmed version. The ITG is aimed at phone to phone and fax to fax communications, allowing users to integrate existing conventional telecommunication equipment into the Internet to circumvent long distance telephony charges. The design philosophy is a frills-free approach to focus on the needs of small and medium business organisations with geographically dispersed offices. The ITG is designed to support two voice or fax lines simultaneously per board with built-in fax modems. VocalTec, on the other hand, supports only voice lines for its 2-channel system. A minimum of six channels is required to support a maximum of 4 voice and 2 fax channels. Moreover, to support fax services, two additional voice/fax modems have to be integrated to the system. From the perspective of enjoying the advantages of basic telephony over Internet, the ITG is designed to simplify its use and is thus more manageable for the users. It is certainly more cost effective on a per-communication-channel basis.

NeTrueLink Fax/Voice Internet Messaging Gateway [12] is a turnkey solution which delivers Internet Fax/Voice access, and a managed global network and account management system. It includes billing/accounting, automated customer support module, network management and centralized settlement. It aims is targeted at Internet Telephony Service Providers (ITSP) to offer Internet fax and voice messaging to their customers, bypassing the international public telephone network. It shares similar basic concepts and objectives as the ITG in reducing cost but is a sophisticated and enormous system targeting at the service provider level.

V/IP (Voice over IP) phone/fax IP gateway [13] is a Micom product. It started off as an overlay voice/fax network on top of any IP network but has since extended its services to include the Internet. It is targeted at corporate users using their own enterprise IP networks. The MICOM gateway is only ideal for supporting faxing and recording/retrieving voice mails over the Internet. It is however, not designed to support real-time voice conversations over the Internet compared to the ITG.

Commercial voice/fax gateways such as the NeTrueLink gateway is a sophisticated system aimed at service providers while VocalTec gateway, though scalable, is less sophisticated than NeTrueLink. Systems such as that of Micom’s category are often designed to support real-time fax [10] and voice mail services only. The ITG therefore bridges the gap between the VocalTec’s and Micom’s categories of commercial products. It is designed to be simple yet efficient, cost effective and yet scalable so that not only is it user-friendly, it is able to deliver the real-time voice and fax communications with minimum hardware add-on. It is simple to use and allows user better control of their system. Given its basic design, its functionality can be easily extended by providing more Application Programming Interfaces (APIs). It is ideal for small and medium businesses.
6. CONCLUSION

The system architecture and hardware design of an Internet Telephony Gateway is proposed and presented. The ITG interfaces with telephone network and a server gateway computer to provide real-time voice communication and fax support to users. The design utilizes two Intel 80186 microprocessors and a Motorola DSP 56001 to capture and process incoming voice and fax data from each of the two ports and redirects it to the telephone recipient or fax machine by using Internet as the transmission medium. With its low running costs and high validity, the ITG offers distinct cost savings and numerous potentials and adds a further dimension in interpersonal communication across the Internet.

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BIOGRAPHIES

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