A Telephone Adapter for Internet Telephony Systems

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ABSTRACT

In order to make the Internet telephony system a step closer to being a viable replacement of the conventional telephony system, an Internet telephone adapter (ITA) is proposed to interface a conventional telephone to the host computer. With the ITA, there is no longer the need for the current practice of using the host computer’s microphone as the audio input and the internal or external speaker as the audio output to carry out a conversation. Users are able to dial an Internet telephone number to make a connection, be notified of an incoming Internet call through ringing of the telephone set, and carry out a conversation using the conventional telephone handset. This paper describes the design and development of an ITA which is connected to the parallel port of the host computer and optionally, to an existing sound card in the host computer. This will allow the ITA to be used as a stand-alone device or in conjunction with a sound card to support full-duplex real-time voice communication across the Internet.

Keywords: Internet, Telephony system, Telephone adapter

1. INTRODUCTION

There has been a growing interest in recent years in developing real-time voice communication software for use over the Internet. This has been the result of a multitude of factors which include declining costs of computer hardware, advances in computer technology and phenomenal growth in Internet. The growing enthusiasm also stems from huge potential cost savings by making it possible to make transcontinental telephone calls at prices of local telephone calls plus the nominal standard Internet connectivity charges.

In spite of the numerous Internet telephony systems available today, these are still at its infancy stage and far from being substitutes of conventional telephony systems[1]. This is basically due to three factors: inferior quality of communication, lack of interoperability among systems, and poor mode of handling and operation.

The quality of communication offered by existing products is still not comparable to those offered by telephone companies. The inferior quality is mainly due to the high transmission delay and packets lost in the Internet environment which is characteristic of packet-switched networks without resource reservations mechanisms. With advances in compression algorithms, network technology and higher bandwidth in future, this problem of quality will diminish over time.

As Internet telephony technology is fairly new and recent, almost all existing systems are designed as closed systems (without standards) to support its own community of users. This leads to interoperability problems between systems. Users will not be tempted to buy and use multiple products in
order to communicate with other users. The problem will remain until some form of a standard (such as H.323[2]) emerges or if the whole market is dominated by some particular product.

In terms of handling and operation, existing Internet telephony systems generally require a caller to define the identity of a recipient through some form of cryptic computer input in order to make a connection. This input takes the form of an IP address, recipient’s name, electronic mail address, Universal Resource Locator (URL) or a special Internet telephone number. Apart from the last form (which the authors have yet to detect in any existing system), this mode of operation is totally different from using a conventional telephone. In addition, these systems rely on the host computer’s microphone as the audio input device and the internal speaker as the output device to carry out a conversation. This is both awkward and unnatural. In addition, it does not provide privacy as the speaker’s output is audible to people around unless a low enough volume or a headset is used. Thus, there is a need to ensure that the Internet telephone system can emulate a conventional telephone as closely as possible both in terms of handling and operation. In this respect, this paper proposes that the Internet telephony environment should include an Internet telephone adapter (ITA) to interface a conventional telephone to the host computer.

2. SYSTEM REQUIREMENTS

Figure 1 shows the Internet telephone environment that uses an ITA to connect a conventional telephone set to the host computer. With the ITA, it becomes possible to support both the Public Switched Telephone Network (PSTN) and the Internet telephony systems. With this configuration, voice communication between users is carried out using a conventional telephone set whose handset houses the mouthpiece and loudspeaker. This effectively replaces the host computer’s microphone and internal speaker for audio input and output operations. The software system interfaces with the telephone to accept direct tone connections. Thus, instead of using computer input to define the recipient, the caller can dial up an Internet telephone number, IP address or some other pre-defined number sequence. A look-up table on the host computer is used to translate the numeric input into the required form for connection.

A request for connection can activate the ringing mechanism of the telephone set so that users are informed of incoming calls in the conventional way. A pre-recorded message at the host computer could be played back to the recipient to indicate that it is an Internet call as opposed to a normal trunk call.
Alternatively, the telephone could be set to have a different ringing tone to differentiate calls between the two systems.

The ITA can be extended to incorporate a number of other features. It can be used as a telephone recorder to capture on-going conversation over the Internet telephone system. Conversely, it becomes possible to playback pre-recorded audio during an on-going conversation so that it is audible to both parties. Finally, it is also possible to utilize the DTMF decoder functionality to create applications with automatic answering facilities.

In order to support these desired features, the ITA is designed to satisfy the following system requirements:

• The ITA supports both Internet and PSTN telephony systems. The ITA connects a conventional telephone or speaker phone set to the host computer to support both forms of telephony system.
• The ITA is an external device that connects to the host computer via the parallel port for data exchange and control. This eliminates the need for limited and valuable slots on the host computer.
• The ITA functions with host computers that may or may not have a sound card or in-built audio capabilities. In this instance, some form of audio data sampling and conversion module must exist on the ITA.
• The ITA has the ability to capture touch-tone inputs from the telephone keypad. This enables an Internet telephone number or other numeric sequence to be captured by the Internet telephone software.
• The ITA is capable of ringing the telephone set and utilizing a different ring tone to differentiate conventional calls from Internet calls.
• At the default or power-down state, the ITA ensures the telephone set and the parallel port are used for normal operations.
• The ITA is an externally powered device with a power input rating of 110/230VAC 50-60Hz.
• A set of Application Programming Interfaces (APIs) is provided to allow the complete control of the ITA for extended functions such as record and playback, and to develop any other customized applications.

3. HARDWARE SYSTEM DESIGN

As shown in Figure 2, the ITA comprises a number of modules to support the necessary system requirements of the previous section:

• Subscriber Line Interface Circuit (SLIC) links the ITA to the telephone set and converts the differential signals from the telephone microphone to a single-ended analogue signal and vice versa. It provides the 48V dc to operate the telephone when the ITA is in operation and also detects off-hook signals generated by the telephone when the handset is picked up or replaced. Discrete components, mainly transistors, are used in the design of the SLIC to minimise cost as opposed to the use of commercial SLIC chips such as the Mitel MH88612.

• The CODEC is essentially a combined analogue-to-digital converter (ADC) and a digital-to-analogue converter (DAC) using the \( \mu \)-law companding. It digitises the audio signal for onward transmission via Internet when a call is made and converts the received digital audio into analogue signals when a call is received. The Motorola MC145500 CODEC with both the \( \mu \)-law and A-law companding capabilities is used for design flexibility.
In the presence of a sound card on the host computer, the CODEC is bypassed as its functions can be performed by the sound card itself. A signal conditioning module, comprising essentially operational amplifiers circuitry will condition the signal to the optimum level for interfacing to any sound input-output devices.

As this device has only serial interface, two shift registers are needed, one for input and the other for output.

- Analogue switches are needed to control which analogue signals are to be connected to the CODEC. As recording and playback features are desired, analogue adding and splitting of signals are needed to record and play back the conversation for the parties to listen to. These modules are built using operational amplifiers circuitry.

- DTMF Decoder Unit is used to decode the DTMF tones generated by the touch-tone keypad from the telephone. A simple Motorola MC145436 DTMF receiver is enough to provide 4-bit hexadecimal code for the standard 16 digits tones generated by the telephone. The decoded digits are then transmitted to the host computer for purposes such as IP addresses and control codes recognition.
• DPCO (Double Pole Changeover) Relay 1 directly connects the trunk line to the telephone set in the event of a power failure or when the user chooses to bypass the ITA. When the ITA is in operation, no calls will be forwarded through the trunk line as the calls are conducted via the Internet. As such, the DPCO relays will simply disconnect the trunk line from the SLIC of the telephone set. The incorporation of the relay thus allows dual operation of the telephone set by the user for both normal and Internet telephony.

• DPCO Relay 2 is used to connect the parallel port of the computer to the printer port for normal printing. This is to ensure that in the event of a power failure, the parallel port can be used solely for printing purposes.

• The tone generator produces the basic dial tone from which the ring tone is generated for use by the ringer circuit. The dial tone is a 400 Hz pure sine wave. A Wien Bridge Oscillator is used to generate the dial tone. The busy tone is derived from the dial tone by modulating the sine wave at an on-off interval of 0.4s each, the waveform of which is provided by a 555 timer. The busy tone is used by the ringer circuit to ring the telephone instead of the conventional ring tone. This is to enable user to differentiate incoming Internet calls from ordinary incoming calls.

• The ring current is derived from the busy tone generated above. The busy tone is boosted by a Class AB Push-Pull amplifier to obtain a sufficiently strong current to ring the telephone set. A 1:10 step-up transformer is subsequently used to step up the voltage to achieve 75 Vrms.

• Multiplexer is used to share the bus interface of the parallel port. As the host computer can either read or send data, the multiplexer serves to send and receive the correct set of data to the respective registers. Bus transceivers with logic gates for controlling the direction of data flow are used.

• Signal Conditioning Module comprises two operational amplifiers to condition the signal to the optimum level for interfacing to the audio input-output device on the host computer. One amplifier is used to condition the input audio signal from the microphone and the other is to condition the output signal to the speaker.

• Control Unit is the ‘brain’ of the ITA. It comprises the following modules:

  - Clock Generator Module generates the necessary clock signals for the different parts of the system. This is the main synchronising module for the ITA. This module can be constructed using an oscillator and counters to divide the clock into the various required frequencies.

  - Memory Module stores the state of the device (i.e. active/inactive) as well as protocol information to interface with the software. It may be utilised to store any data or control information needed for any future expansion purposes. Flip flops circuitry is used for this purposes.

  - Digital Speech Input Synchronisation Module controls the reading of the digitised speech signal from the parallel port and channels it to the CODEC for conversion to the analogue equivalent for the telephone speaker. Logic gates are used to produce proper control signals for the different units to operate correctly.

  - Analogue Speech Output Synchronisation Module reads the analogue speech signals from the telephone microphone, directs it to the CODEC for digitisation, and sends the digital speech signals to the parallel port of the host computer. Control signals, derived from logic gates are used to operate the different units.

  - Activation Code Detector Circuit detects codes from the parallel port and determines when to activate the ITA or when to shut it down. In a shut-down condition, it will enable the telephone to be connected to the trunk line and the computer parallel port to function as a normal printer port. These are accomplished via the use of DPCO Relays 1 and 2.
- **Control Codes Decoding Module** decodes and executes instructions from the parallel port. Examples of such instructions include relay on/off, ITA activation/shut down and bypassing the CODEC at the presence of the sound card. This can be easily implemented using decoders/demultiplexers to decode the various commands.

4. **SOFTWARE SYSTEM DESIGN**

![Diagram](image)

**Figure 3. Internet Telephone Environment with ITA**

As shown in Figure 3, the ITA is interfaced to the Internet telephone software system via the parallel port and driven by a set of Application Programming Interfaces (APIs). The set of APIs is derived from a set of basic primitives operations that is supported by the ITA. This is shown in Table 1. There are two main components in the Internet telephone software system, namely, Telephone Exchange and Telephone Software. As the name implies, the Telephone Exchange acts as an exchange to aid connectivity between active Internet telephone users. The Telephone Software component comprises a user interface for session control and audio management processes. This segregation is necessary to cater for the different roles played by each process to support the real-time nature of the communication process.

Session management is concerned with negotiating and monitoring of session parameters. Three processes are utilised in session management: control, negotiation and network communication process. The control process receives a caller’s request for connection with a recipient and attempts to establish a communication path between the two parties. The recipient’s 'Internet telephone number' is input directly from the conventional telephone. The Telephone Exchange is contacted to retrieve the IP address of the recipient. With this information, a call set-up with the recipient is subsequently
The negotiation process takes over to handle the arbitration of communication parameters between both parties before actual communication can take place. With all communication parameters resolved, the negotiation process will signal the audio management process to initiate the conversation. Finally, the network communication process provides the transportation functionality to dispatch messages to and from the recipient using TCP.

<table>
<thead>
<tr>
<th>Primitive</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>detect_device</td>
<td>detect availability of ITA</td>
</tr>
<tr>
<td>send_data</td>
<td>send one byte of data to the ITA</td>
</tr>
<tr>
<td>read_data</td>
<td>read one byte of data from the ITA</td>
</tr>
<tr>
<td>send_command</td>
<td>send selected instruction to ITA</td>
</tr>
<tr>
<td>read_status</td>
<td>read status_register of ITA</td>
</tr>
<tr>
<td>read_DTMF_data</td>
<td>read DTMF decoded data</td>
</tr>
</tbody>
</table>

Audio management is responsible for audio transmission from the recording and playback operations to the transmitting and receiving operations. Audio transmission is periodic with audio sampled at regular time intervals and goes through a series of processes prior to being transmitted. Although, the occasional non-delivery of audio packets will reduce the quality of communication, it will not be disastrous, provided that small audio samples are dispatched each time and that some in-built mechanisms exist to cater for lost data replacement. A number of processes are required to fulfil the role played by audio management:

The Record/Playback (Audio) process provides the necessary audio interface for recording, playing back and adjustment of audio characteristics. It interfaces with the ITA and hence the conventional telephone to capture and playback the audio data. Each packet of audio sample obtained from the telephone is linearly time-stamped by the packet time-stamping process to indicate the instant of sampling. This allows the packets arriving at the recipient’s host computer to be ordered in the correct time sequence before being played back. The audio data is optionally compressed by the compression process and sent to the recipient's host computer through the network communication process using UDP/IP.

At the recipient's host computer, audio data arriving are correspondingly decompressed by the decompression process before being buffered and ordered by the buffer management and packet ordering process. The purpose of buffer management is to cushion the out-of-order, late delivery and jitters experienced by the packets. Audio packets are ordered in sequential order using a ring buffer to store audio samples. A margin of safety is usually set in the total buffer space to take into account of jitters. If a safety factor of 2 is used, then the play-out point occurs when the ring buffer is half full. Prior to playback, the packet lost replacement control process is activated to identify and handle packets lost as a result of unreliable network transmission.

5. SYSTEM OPERATION

The ITA is physically connected to the trunk line, telephone set and the printer port of the PC. When the power of the ITA is not switched on, it is totally bypassed. In other words, the telephone set will function as a PSTN telephone while the printer port will be servicing the printer.

Figure 4 shows the communication process which takes place when the ITA is in operation. To start the Internet telephony application, the user must first run the Telephone Software on the PC. Upon the ITA being powered up, the ITA API residing on the PC will detect its status and signal the Telephone Software to register the ITA’s presence. All calls made on the telephone set will thereafter be channelled via the ITA through the Internet instead of the PSTN.

To make a call, the user simply lifts the handset. The ITA will then connect the dial tone to completely emulate the PSTN environment. The user enters the destination phone number which is then fed
to the Telephone Software. As can be seen in Figure 4, the Telephony Software will query the remote Telephony Exchange for the corresponding IP address and subsequently request for a connection. The call is set up if the connection is successful otherwise a busy tone will be conveyed to the user handset by the ITA to signify that the destination line is busy. On the receiver side, upon receipt of the request for connection, the Telephone Software will instruct the ITA’s API to check if the recipient’s telephone is onhook. If so, the ITA will ring the telephone. If the line is offhook, the ITA will report to the Telephone Software that the connection cannot be put through. When the call is answered, the ITA will inform the Telephone Software to signal successful connection.

Upon successful connection, audio conversation commences. The ITA basically digitises the analogue speech data and transmit the resulting digital signals to the Telephone Software. Either party can terminate the call by setting the telephone onhook. ITA detects the status and conveys it to the Telephone Software which in turn relays the hang-up status to the caller side. The ITA on the caller side will thus send a dial tone to signify that the call is terminated. The users will respectively replace their handsets. To terminate the Internet telephony application, the user just needs to switch off the ITA and terminate the Telephone Software running on the PC.

6. SYSTEM REVIEW AND EVALUATION

The proposed ITA is successfully designed, constructed and interfaced with our prototype Internet Telephone Software System [3-4]. The basic features have been tested and verified. The feasibility of using the ITA to support both conventional and Internet telephony systems has been demonstrated. In both systems, the conventional telephone or speaker-phone is used to carry out a conversation. The ITA connects to the parallel port of the host computer and can be optionally used with an existing sound card. At the default and power-down state, it allows the printer parallel port to be used for printing and conventional telephone operations to be carried out. When in use for Internet telephone operations, the printer port is set to busy status so that the printer (or any device connected to the parallel port) cannot be used concurrently. In this case, new print jobs cannot be sent directly to the printer but is put into the print queue ready to resume once the conversation terminates.

As Internet telephony systems are fairly new, it should not be surprising that there are only a few Internet telephone adapters on the market. However, there is no doubt that this number will increase significantly as Internet telephone becomes commonplace. Three existing ITA products are compared in this section for completeness:

The Internet Telephone Adapter [5] is a low-cost simple interface between a host computer’s sound card and a telephone. It interfaces with a host’s computer sound card via the standard speaker and mike jacks. The inability to fully emulate the functions of normal telephony for more natural Internet telephony operations and its exclusive use of the telephone set for Internet telephony only are its main disadvantages.

The Advanced Internet Telephone Interface [6] is mainly designed for use in an office environment to handle selling through voice mail and recording personalized messages. It connects the host computer’s sound card to a conventional telephone via two separate inputs, one for the telephone’s handset/ headset and the other for the telephone set itself. With this setup, the handset/headset can be used for all sound card applications and a connection is achieved between the sound card and the telephone line. The two somewhat functional similarities with the ITA lie in allowing the handset/headset to be used for Internet conversations and the ability to perform static record and playback. However, the Advanced Internet Telephone Interface also exhibits the same disadvantages as the Internet Telephone Adapter when compared with the proposed ITA.

Finally, SoundXchange [7] is essentially a customised speaker phone system to render Internet telephony easier to use. It comes in various models and is hung alongside the host computer’s monitor to connect to any previously installed sound card or to any computer with built-in audio capabilities through the standard mini jacks. The model which somewhat resembles the proposed ITA has a built-in sound board and plugs into the parallel port of a computer, and a pass-through connector to allow for a parallel port printer. This has the advantage of eliminating the need to install a sound card in the host computer.
and taking up one of its limited slots. It is used for Windows 3.1 applications that do not require any
duplexing. However, it lacks the support for full duplex operations and is hence not optimally designed
for Internet telephony usage. The telephone set is confined to being a stand-alone device for computer-
based audio applications and is not connected to the PSTN for conventional telephone usage.

From these brief reviews, it is apparent that the proposed ITA has been designed to exhibit a host
of additional features and functionality over and above what is currently available.

7. CONCLUSION

This paper has examined the Internet telephony environment and proposed the inclusion of an
ITA to connect a conventional telephone set to the host computer. In doing so, it becomes possible to
improve the handling and usage of Internet telephony system thereby removing one of the factors
inhibiting it from widespread adoption and use. This paper has demonstrated, through the design and
implementation of an ITA, the feasibility of using this approach to connect a conventional telephone set
to the host computer to support both Internet and PSTN telephony systems.

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BIOGRAPHIES

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Figure 4. Communication Process Between Two Internet Telephone Users
Microprocessors and Microsystems
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