

A HYBRID OF TRADITIONAL TELEPHONE SERVICE AND COMPUTER BASED INTERNET TELEPHONY

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ABSTRACT

A Hybrid of Traditional Telephony and Internet Telephony is a communications network that integrates the traditional Public Switched Telephone Network (PSTN) and the Internet Protocol (IP) based packet switched network. In this work various types of hybrid PSTN-IP telephony connections including the currently available service are explored. Selected hardware and software tools have been used to develop a hybrid Internet telephony service in LAN. The Microsoft product TAPI 3.0 is used to develop all the interfaces and the ITU-T H.323 standard is used as the voice over IP. The software developed for this work has been tested and found working satisfactorily. The problem associated with full duplex on line communication is also addressed. This system has been implemented successfully.

INTRODUCTION

In the past, dedicated telecommunication networks have been specially designed for different services. The public switched telephone network has been developed for voice, data networks for computer communications, and broadcast networks for television [1]. They are designed in such a way that their network performance is optimal for their intended services. The network requirements such as bandwidth, holding time, end to end delay, and error rate are different for each network. However the current information technology is bringing a breakthrough on both the hardware and software to integrate these different networks to utilize some of the advantages associated with each network. The PSTN-IP integration is one of such instances to interoperate these two networks for real time voice communication.

The telephone network used by the public/ PSTN is a circuit switched network. In PSTN a link created between subscribers or stations is dedicated until the line is released after the session is

complete. The dedicated link in PSTN increases reliability of the service for real time voice communication. However because the dedicated link will not be shared, the bandwidth will not be used efficiently [2]. Packet switching network is used primarily for data transmission. Here the connection is established temporarily at intervals during the call to transmit a discrete packet of data each time. The obvious part of the economic advantage of IP networks is shared by all packet networks that multiple packetized data streams can share a circuit. Comparing with the circuit switched networks, packet switched networks will on average double the bandwidth and improves the efficiency by 30%, thus halving operational costs [2]. Hence the integration of these two networks brings paramount advantages for voice communication which either network can not offer by itself.

This paper is divided into four sections. In the first section various types of PSTN-IP based networks are discussed. In second section the Telephone Application Programming Interface (TAPI 3.0) and the H.323 ITU-T standard are introduced. In third section the Hybrid PSTN-IP telephony software developed using TAPI 3.0 for IP based LAN is presented. Finally, the last section is concluded with test results and recommendations for future scope of study.

PSTN-IP Telephony

The Public Switched Telephone Network (PSTN) is a circuit switching network that renders the Plain Old Telephone Service (POTS). However to exploit the cost advantage of using IP data network for voice communication it has become imperative to integrate PSTN with Internet. The telephony service over the Internet is done by digitizing the analogue voice signal, compress and convert it into IP packets like any data to be transmitted over the Internet. Gateways are used to interconnect the PSTN and the Internet.

The IP telephony which is sometimes referred as Internet Telephony is classified in to three major classes in accordance with the type of connection between the PSTN and IP based network [3].

- 1) Traditional International/long distance telephone service using Internet technology.
- 2) Totally computer/Internet based Internet telephony.
- 3) Hybrid of traditional telephone service and computer based Internet telephony.

In the first class the IP network replaces the long distance carrier in international subscriber dialing. In such connection gateways are required at both ends of the connection and the subscriber loop is still PSTN. The user follows two step dialing. First the special service code (telephone call through Internet) is entered to access the local gateway and then the destination subscriber number including the country code is entered. The gateway at the destination side is selected based on the country code. In the second class the entire path between the end terminals is IP network. The end terminals are also computers instead of simple telephone apparatus. Here the user simply enters an IP address or domain name of a computer to place the call. In this case both the end terminals should be connected to Internet. Both terminals or machines are also required to be loaded with the necessary telephone software. In the third class the IP network will take part in one side (call origination or call termination side) and the PSTN will take part on the other side. In general to get the maximum cost advantage the distance between the call termination point and the gateway should be as short as possible. In this work the third class is being realized.

Hybrid of Traditional Telephone Service and Computer Based Internet Telephony.

According to the study made by Clark [3], the discussion in European Telecommunications Standards Institute (ETSI) and the proposals in Internet Engineering Task Force (IETF) the potential connections of IP Telephony are divided in to several sub classes in terms of the interconnection between Switched Circuit Network

(SCN) and the Internet. The two possible configurations are presented as follows.

1. *Connection of Phone Terminal in SCN to PC in the Internet or Connection of PC in the Internet to Phone Terminal in SCN.*

In this type of connection, a calling user on telephone terminal in SCN is connected to a called user on computer terminal in the Internet or a calling user on computer terminal in the Internet is connected to a telephone terminal in SCN as shown in Fig. 1. The gateway of calling and called user is required for this connection at the interconnection point between the network of calling user and called user as the two calling parties are with in two different networks.

A calling user is assigned E.164 number (standard telephone number) if he/she is calling from with in SCN [4] and IP address if he/she is calling from within internet. The gateway of calling and called user is assigned E.164 number in SCN and IP address in the Internet. Similarly a called user is assigned IP address if he/she is with in the Internet and E.164 number if he/she is within SCN. If the called user is within the Internet, the called user needs to have their computer terminal powered up and ready to receive a call. The path in this type of connection is considered as composed of the following two portions, shown in Fig. 1. The first portion is within the SCN network and the second portion is within the Internet being connected to the SCN via the server of IP telephony and the IP telephony gateway.

2. *Connection of PC in the Internet to PC in the separate IP-based network.*

In this type of connection, the path is set between the user on Internet and user on separate IP based network via SCN as shown in Fig. 2. Here both calling user and called user are on computer terminals. The network of a calling user is in the Internet while the network of called user is the other independent IP-based network separate from the Internet. The network of called user does not have the direct connection to the Internet and can be connected with the Internet only via SCN. The addressing or routing system of the network of called user is IP-based but proprietary. It is independent from the address of the Internet. This

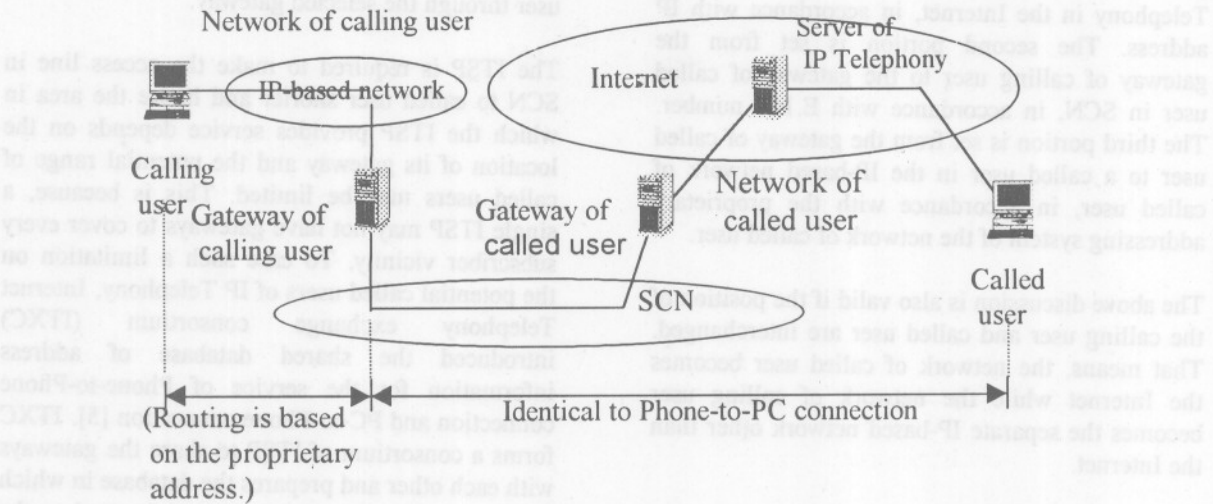


Figure 2 Connection of PC in the separate IP-based network-to-PC in the Internet

type of connection requires the gateway of calling user at the interconnection point between the Internet and SCN, and the gateway of called user at the interconnection point between SCN and the network of called user.

A calling user is assigned IP address in the Internet. Similar to the computer terminal of the calling user in PC-to-Phone connection, the computer terminal of the calling user in this type of connection may be directly connected to the Internet or connected to the Internet by the dial-in

access through SCN using modem. The gateway of calling user is assigned IP address in the Internet and E.164 number in SCN. The gateway of called user is assigned E.164 number in SCN and IP-based but proprietary address in the network of called user. A called user is also assigned IP-based but proprietary address in the network of called user.

The path in this type of connection is considered as composed of the following three portions. The first portion is set from a calling user to the

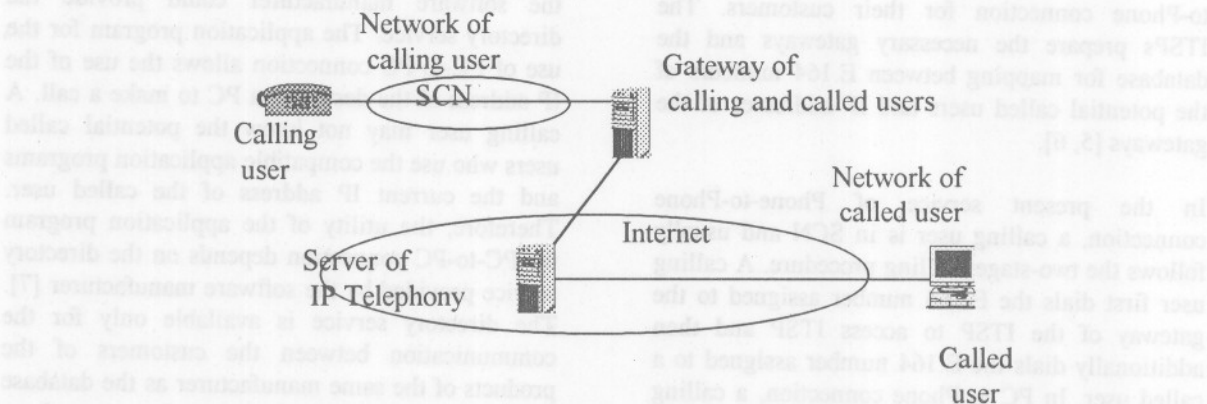


Figure 1 connection of phone-to-PC in the Internet

gateway of calling user via the server of IP Telephony in the Internet, in accordance with IP address. The second portion is set from the gateway of calling user to the gateway of called user in SCN, in accordance with E.164 number. The third portion is set from the gateway of called user to a called user in the IP-based network of called user, in accordance with the proprietary addressing system of the network of called user.

The above discussion is also valid if the position of the calling user and called user are interchanged. That means, the network of called user becomes the Internet while the network of calling user becomes the separate IP-based network other than the Internet.

Trend of the present IP Telephony services:

Some survey of the existing IP telephony services at the present days has been carried out. Currently some of the classes of connection discussed are available for the practical service of IP Telephony. Several programs are made available that will let any one, equipped with the right computer hardware, an Internet connection, and required software, to speak (voice) in real-time over the Internet. Some of the programs available for voice on the net are free and others are commercial. Most of these products are ready for use, but others are still very experimental. Some Internet Telephony Service Providers (ITSPs), such as Access Power, Delta Three, Network Telephony Corporation and TouchWave, Vocal Tech. provide the services of phone to phone connection and PC-to-Phone connection for their customers. The ITSPs prepare the necessary gateways and the database for mapping between E.164 numbers of the potential called users and IP addresses of the gateways [5, 6].

In the present service of Phone-to-Phone connection, a calling user is in SCN and usually follows the two-stage dialling procedure. A calling user first dials the E.164 number assigned to the gateway of the ITSP to access ITSP and then additionally dials the E.164 number assigned to a called user. In PC to Phone connection, a calling user which is in the Internet directly access the ITSP through the Internet. The ITSP selects the appropriate gateway as a result of the mapping according to E.164 number of the called user given

by the calling user and sets the path to the called user through the selected gateway.

The ITSP is required to make the access line in SCN to called user shorter and hence the area in which the ITSP provides service depends on the location of its gateway and the potential range of called users may be limited. This is because, a single ITSP may not have gateways to cover every subscriber vicinity. To ease such a limitation on the potential called users of IP Telephony, Internet Telephony exchange consortium (ITXC) introduced the shared database of address information for the service of Phone-to-Phone connection and PC-to-Phone connection [5]. ITXC forms a consortium of ITSP to share the gateways with each other and prepares the database in which E.164 numbers of called users are mapped to the IP addresses of shared gateways. When any ITSP, member of the consortium, uses a shared gateway that do not belong to itself, the user ITSP pays fee for the usage of the gateway to the owner ITSP. ITXC mediates the payment between the ITSPs. This ensures the efficient usage of the available gateways without the need to have gateways for each ITSP.

The user of product for PC-to-Phone connection is required to subscribe to the ITSP. This is because the service of IP Telephony in PC-to-Phone connection is provided by the ITSP using the product. On the other hand, the PC-to-PC connection is not required to subscribe to the ITSP. However, the user is strongly recommended to make the registration for the use of product so that the software manufacturer could provide the directory service. The application program for the use of PC-to-PC connection allows the use of the IP address of the destination PC to make a call. A calling user may not know the potential called users who use the compatible application programs and the current IP address of the called user. Therefore, the utility of the application program for PC-to-PC connection depends on the directory service provided by the software manufacturer [7]. The directory service is available only for the communication between the customers of the products of the same manufacturer as the database is managed by the individual manufacturer and not shared with others. Although the services of Phone-to-Phone connection, PC-to-Phone connection and PC-to-PC connection are provided

now, the service of Phone-to-PC connection is not yet available globally for commercial purpose.

Telephone Application Programming Interface (TAPI 3.0)

TAPI is a programming tool that provides a uniform set of commands for telephony application. Using TAPI functions, a windows based application program for telephony service can be developed. This helps to concentrate on the development of functionally rich application layer, rather than on implementing expensive and difficult proprietary interface solutions. The set of commands provided by TAPI enable various telephony devices to be accessed by an application program regardless of the programming language used to develop the application. Any TAPI-supported activity to access a telephony device attached to our computer involves three layers of software [8]. The first one is the application layer. This is the application software developed to carry out telephony related service. The second layer is TAPI itself. This comes as part of windows operating system package. It is located in windows system folder as tapix.dll file (x is to indicate the TAPI version). The third layer is where the Telephony Service Provider Interface (TSPI) resides. A TAPI service provider translates the commands for a telephony device or telephony protocol. TAPI service providers for modems and several telephony protocols are installed with the Windows operating system, and others are provided by independent hardware vendors. For this work the Unimodem service provider, TAPI remote service provider and the H.323 TAPI service providers are used. the application interacts with the telephony dynamic-link library (DLL) by means of TAPI. The telephony DLL is part of the Windows operating system and implements TAPI support [8]. The telephony DLL interacts with service providers through the Telephony Service Provider Interface (TSPI). Hence TAPI consists of two interfaces: an API that developers use to write applications and the service provider interface (SPI) that applications use to establish the connection to the specific telephone network. Service providers can be thought of as one thinks of printer drivers—they are developed by telephony hardware vendors to support operating within Windows.

TAPI has been under continual upgrading to accommodate the various needs of current communication technology. The first version was structural, limited to few service providers, and was difficult for developing applications. The successive versions (TAPI 1.3, TAPI 1.4, TAPI 2.0, TAPI 2.1) appeared with improved qualities than the previous one. TAPI 3.0 is the latest version packed with windows 2000 and NT 5.0 operating system. TAPI 3.0 has included new service providers like H.323 and IP Multicast which help to develop applications for IP telephony and audio conferencing. It is also a component object model instead of structural which enables it to be programmed using any of object oriented programming languages easily [8].

There are four major components to TAPI 3.0:

- TAPI 3.0 Component Object Model (COM) API
- TAPI Server
- Telephony Service Providers
- Media Stream Providers

The TAPI Server process (TAPISRV.EXE) abstracts the TSPI (TAPI Service Provider Interface) from TAPI 3.0 and TAPI 2.1, allowing TAPI 2.1 Telephony Service Providers to be used with TAPI 3.0, maintaining the internal state of TAPI.

Telephony Service Providers (TSPs) are responsible for resolving the protocol-independent call model of TAPI into protocol-specific call-control mechanisms. TAPI 3.0 provides backward compatibility with TAPI 2.1 TSPs. Two IP telephony service providers (and their associated media stream providers) are packed by default with TAPI 3.0: the H.323 TSP and the IP Multicast Conferencing TSP.

TAPI 3.0 API objects and their relationship

There are five objects in the TAPI 3.0 API [8]

1. TAPI
2. Address
3. Terminal
4. Call
5. CallHub

TAPI

The *TAPI* object is the application's 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 and remote addresses.

Address

An *Address* object represents the origination or destination point for a call. Address capabilities, such as media and terminal support, can be retrieved from this object. An application can wait for a call on an Address object or can create an outgoing call object from an Address object.

Terminal

A Terminal object represents the sink, or renderer, at the termination or origination point of a connection. The Terminal object can map to hardware used for human interaction, such as a telephone or microphone, it can also be a file or any other device capable of receiving input or creating output.

Call

The *Call* object represents an address's connection between the local address and one or more other addresses (This connection can be made directly or through a CallHub). The Call object can be imagined as a first-party view of a telephone call. All the call control is done through the Call object. There is a call object for each member of a CallHub.

CallHub

The *CallHub* object represents a set of related calls. A CallHub object cannot be created directly by an application—it is created indirectly when an incoming call is received through TAPI 3.0. Using a CallHub object, a user can enumerate the other participants in a call or conference, and possibly (because of the location independent nature of COM) perform call control on the remote Call objects associated with those users, subject to sufficient permissions

H.323 and TAPI 3.0

H.323 is a comprehensive International Telecommunications Union (ITU) standard for multimedia communications (voice, video, and data) over connectionless networks that do not provide a guaranteed quality of service, such as IP-

based networks and the Internet [9]. It provides for call control, multimedia management, and bandwidth management for point-to-point and multipoint conferences. The H.323 is the heart of PSTN-IP integration in TAPI 3.0.

The H.323 Telephony Service Provider (with its associated Media Stream Provider) allows TAPI-enabled applications to engage in multimedia sessions with any H.323-compliant terminal on the local area network. Specifically, the H.323 Telephony Service Provider (TSP) implements the H.323 signaling stack. The TSP accepts a number of different address formats, including name, computer name, and e-mail address.

The H.323 Media Service Provider (MSP) is responsible for an H.323 connection (including the RTP, RTP payload handler, codec, sink, and renderer filters). To counter the effects of LAN latency, H.323 uses as a transport the Real-time Transport Protocol (RTP), an IETF standard designed to handle the requirements of streaming real-time audio and video over the Internet [10].

The H.323 protocol is specified so that it interoperates with other networks. The most popular H.323 interworking is IP telephony shown in Figure 3, when the underlying network of H.323 is an IP network and the interoperating network is SCN. SCN includes PSTN and ISDN networks. H.323 is compatible with various other H.32x networks. The routing from the PSTN to the IP network gateway or from the IP gateway to the PSTN subscriber loop is done by the common channel signaling system (CCSS). This is the extension of intelligent network capabilities to and from the IP network [11], the detail is beyond the scope of this paper.

Integration with the Windows 2000 Active Directory

H.323 telephony is complicated by the fact that a user's network address (in this case, a user's IP address) is highly volatile and cannot be counted on to remain unchanged between H.323 sessions. The TAPI H.323 TSP uses the services of the Windows 2000 Active Directory to perform user-to-IP address resolution. Specifically, user-to-IP mapping information is stored and frequently refreshed using the Internet Locator Service (ILS)

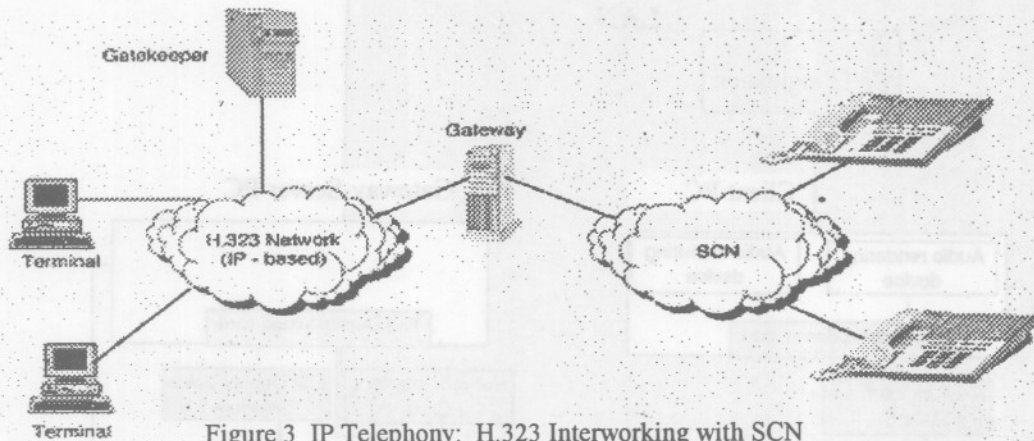


Figure 3 IP Telephony: H.323 Interworking with SCN

Dynamic Directory, a real-time server component of the Active Directory [8].

THE PSTN-IP TELEPHONY IN LAN

The overall architecture is as given in Fig. 4. The two networks involved in the hybrid network are the PSTN and the LAN. The LAN carries the voice over IP part of the hybrid system. The PSTN caters the voice in circuit switched mode. A PSTN telephone call placed from the LAN terminates at the PSTN end passing through both networks. For this work two separate programs are developed. The first one is a client program to be loaded on one of the LAN PC and the second one is a Gateway-Server program to be loaded on another PC with in the same LAN. The client PC is where the call is to be originated, and the Gateway-Server is a PC with in the LAN on which a telephony device (Modem) is fixed. The telephone call originated from the client PC access the PSTN through the Gateway-Server. The complete path from the call origination point to the call destination point along with the hardware involved is shown in Fig. 4.

The TAPI based application program developed for the Gateway-Server PC uses two TAPI service providers. One is the H.323 TAPI service provider and the other is the Unimodem TAPI service provider. The H.323 TAPI service provider is used by the application to access the network interface card of the Gateway-Server to communicate with the Client PC. The Call establishment, control signaling, Media Streaming and Control, call release are all executed by the H. 323 protocol.

Similarly the Unimodem TAPI service provider enables the program to access the modem and utilise it to communicate with a telephony terminal over PSTN. The call setup signal, call progress signal, and the call release signal are generated by the modem during call attempt to PSTN by the server-gateway. The "telephony snap-in" a component of telephony server on windows 2000 is used to monitor the Gateway-Server and the users that access it over the LAN. Hence only the users registered on the "telephony snap-in" can access the Gateway-Server for dialling. The TAPI based application program written for the client PC uses two TAPI service providers from the windows 2000 service package. The first one is the TAPI remote service provider and the second one is the H.323 TAPI service provider. TAPI remote service provider is automatically loaded by the operating system when a PC with in a LAN is configured as client PC. The H.323 TAPI service provider performs the same task that it carries on for the Gateway-Server. It enables the client PC and the Gateway-Server to communicate over the LAN with the H.323 protocol. Before a user can use the application program, the TAPI server must be specified on the client computer. When a program on the client makes a TAPI request, the Remote Service Provider residing on the client PC interprets the request and sends it across the computer network to the TAPI service running on the Gateway-Server and accesses the application program. The application program on the Gateway-Server then uses the Unimodem TAPI service provider to access the telephone device (voice modem) installed on the server.

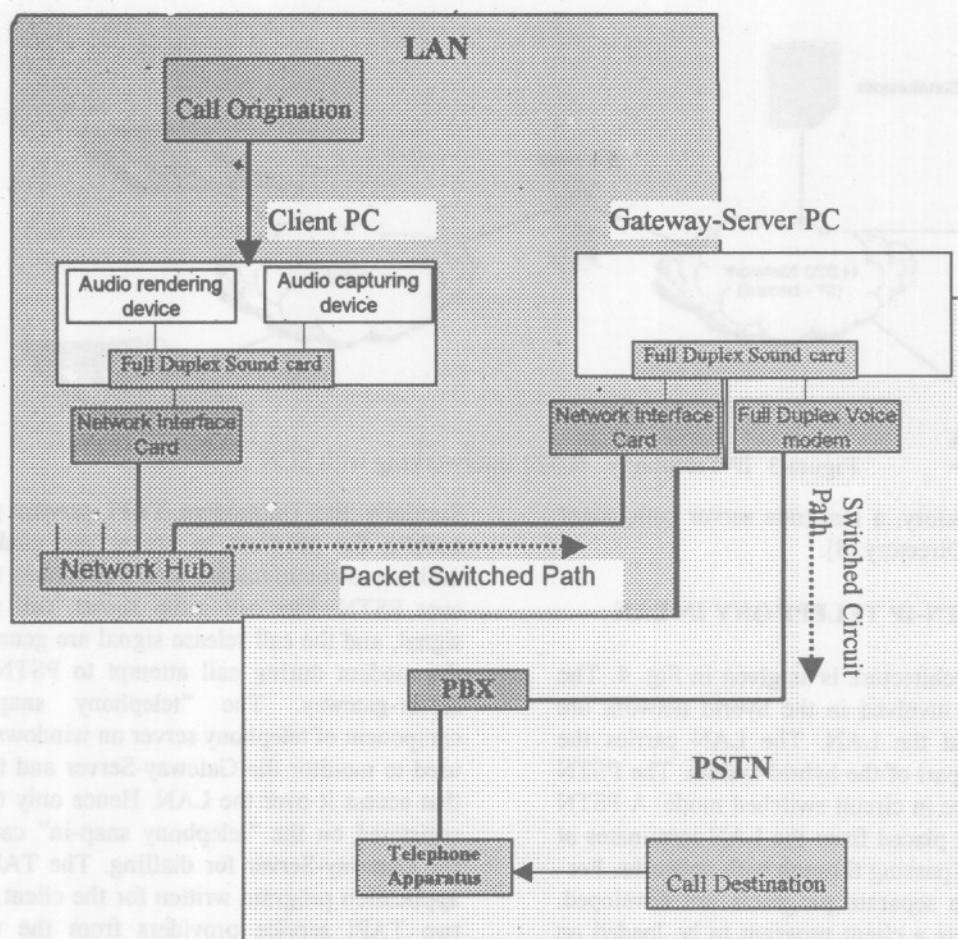


Figure 4 PSTN - IP Telephony

The fact that voice modem is used as a telephony device for this work, the number of possible calls at any time is limited to one. Hence only one client can access the Gateway-Server and make a call at a time. Using a PBX interfaced to the Gateway-Server or dialogic card [12] instead of a voice modem can extend the number of calls to be made at a time.

The Hybrid IP Telephony Software:

The Hybrid IP Telephony software subsystems, the Client and the Gateway-Server programs, work together. Before the client PC attempts a PSTN call, the Gateway-Server should be registered for receiving a call. This is done by clicking the "receive call" control on the server's telephony switchboard. Then the telephone server will be listening for any H323 call from client PCs which

request a PSTN call. Placing the PSTN call from the client PC is initiated by pressing the "MakeCall" control to show the dialing window and then pressing the "dial" control after entering the destination telephone number. Once the request from the client is detected on the server, the offer is notified to the application program loaded on the server and a full duplex audio path is established between the server and the client PC. Then application program on the client PC initiates a separate UDP (User Datagram Protocol) path towards the server. This is done by clicking the "T-server" and "Close" controls respectively on the client's telephony switchboard. The application program on the server extracts the destination address from the client PC using the temporarily established UDP connection. Once the destination address is extracted the server automatically places the PSTN call with out the

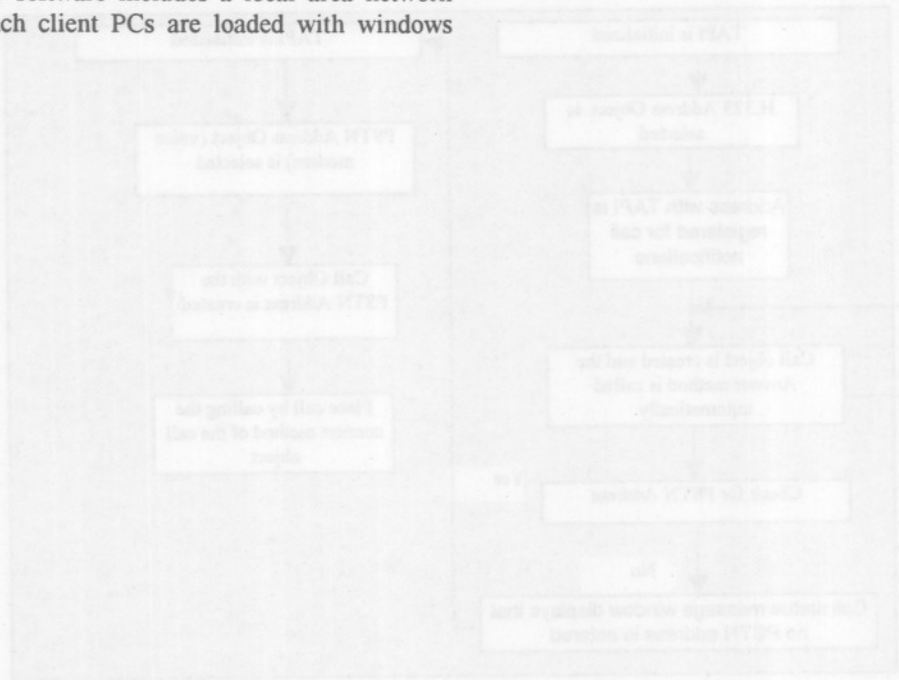
need for any manual intervention. If the called party receives the call, the voice will be extracted from the modem to the sound card of the server. This extracted voice will automatically be passed to the call object created by the server during the initial communication with the client PC. Basically the server PC carries out the protocol conversion using the network interface card and the voice modem to emulate the commercial IP-PSTN gateways. The flow chart for the Client and the Gateway-Server programs is shown in Fig. 5. Both the client and the server parts of the software are provided with message windows to display TAPI call-status messages for the user. By observing the message on this window, the user knows whether TAPI is initialized successfully, call is on progress, call is connected, or call is disconnected. Additional call-status indicators are also developed and included with in the message window. They are three coloured circular shapes: green, yellow and red. The yellow colour blinks when the call is in progress, the green colour blinks when the call is connected, and the red colour blinks when the call is disconnected. The user interface for the developed IP-PSTN telephony software is shown in Fig. 6(a) and (b).

The associated hardware for the above hybrid IP telephony software includes a local area network where each client PCs are loaded with windows

2000, a full duplex sound card on each client PC, a full duplex voice modem with a unimodem full duplex audio device so that the audio path between the modem and the sound card is full duplex, a telephone line, Mic fitted with echo canceller and speaker.

CONCLUSION

The developed software and set-up has been implemented in laboratory. Calls have been repeatedly placed and worked successfully. The full duplex unimodem audio device which connects the modem with the Gateway-server PC's sound board was not available and hence only a half duplex communication was possible. However this problem will be solved if the unimodem full duplex audio device is used. The direction of placing the call is limited to calling from the LAN to PSTN. This is because each client PC is required to have E.164 at the server and also there should be a mapping mechanism between the E.164 addresses and domain names of the client PC's. In this work, Only one call at a time can be handled, this is because modems can handle only one telephone call at a time. For multiple calls, PBXs or other telephone devices like dialogic cards may be utilized.



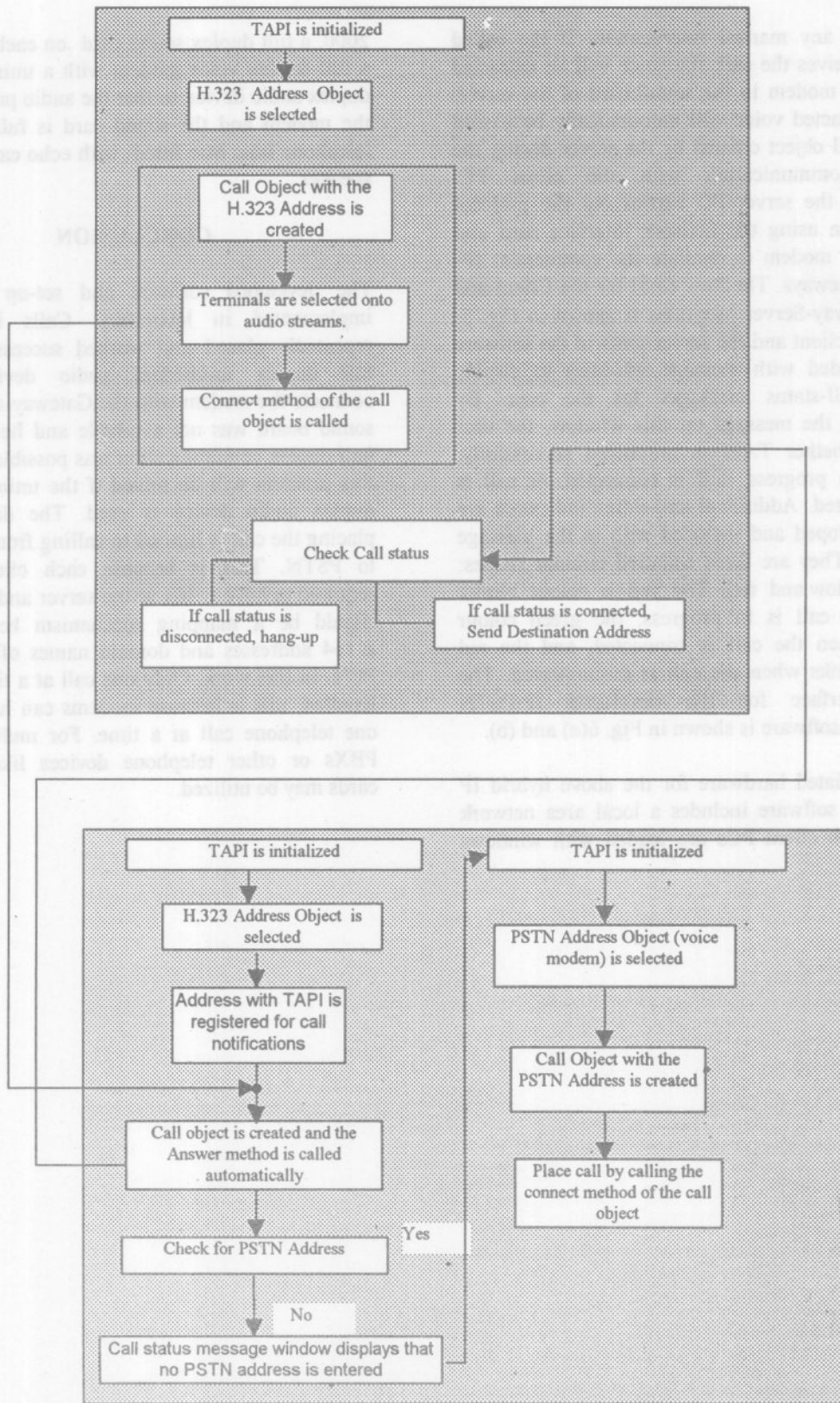


Figure 5 Flowchart of the Hybrid IP telephony

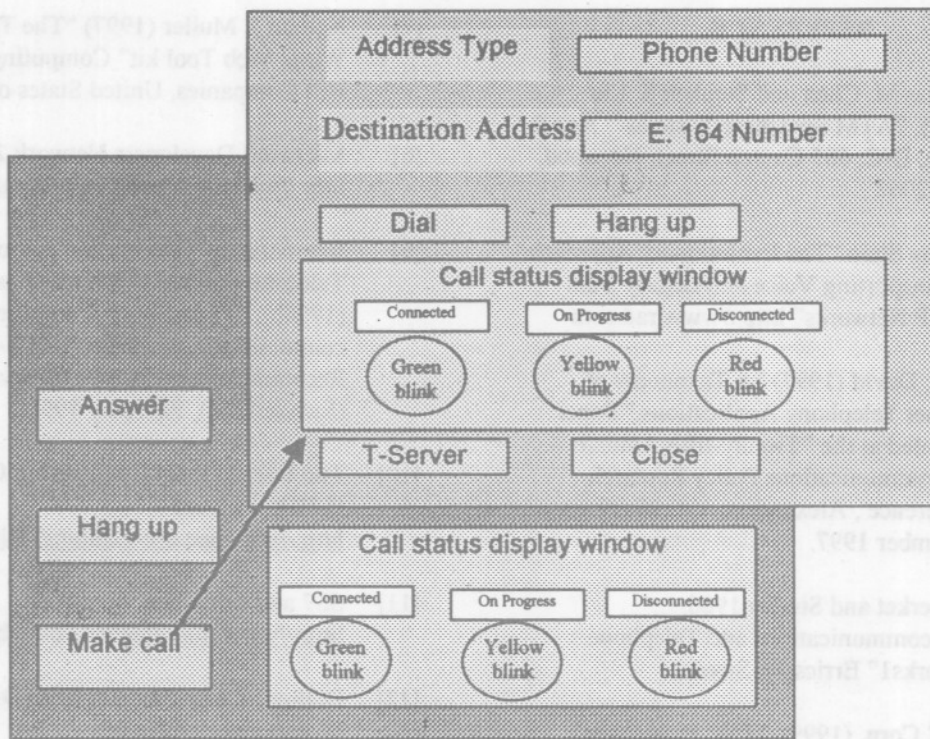


Figure 6(a) User Interface for the hybrid IP Telephony on the client PC

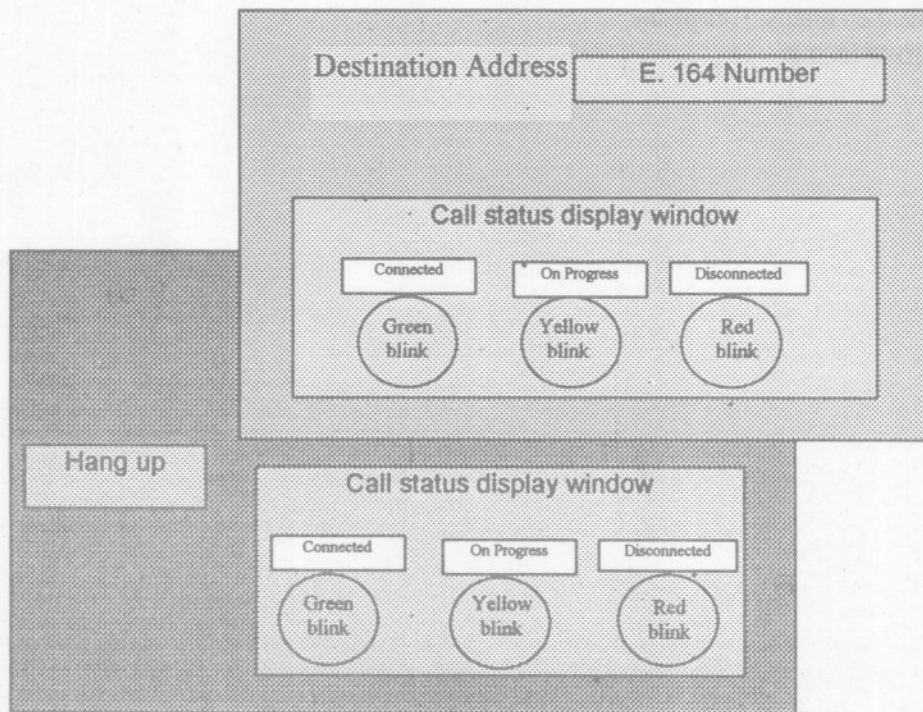


Figure 6(b) User Interface for the hybrid IP Telephony on the Gateway-server PC

REFERENCE

- [1] Thomas M. Chen and Stephen S. Liu (1995), "ATM Switching Systems" Artech House, INC. 685 Canton Street, Norwood, MA 02062
- [2] Yaakov Stein "The Evolutionary Approach to Transporting Voice and Legacy Data over IP Networks" <http://www.rad.com>
- [3] Clark, David (1997) "A Taxonomy of Internet Telephony Applications." Paper presented at the "Twenty-fifth Annual Telecommunications Policy Research Conference , Alexandria, VA, 27-29 September 1997.
- [4] Televerket and Studen 1986, "Telecommunications and Telephone networks1" Ericsson Sweden.
- [5] ITXC Corp. (1999), ITXC Homepage, <http://www.itxc.com/>
- [6] The International Engineering Consortium, Document on Internet Telephony, <http://www.iec.org>
- [7] Nathan J. Muller (1997) "The Totally Wired Web Tool kit" Computing McGraw-Hill Companies, United States of America.
- [8] Microsoft Developers Network 2000 <http://msdn.microsoft.com/library>
- [9] International Telecommunications Union Telecommunication Standardization Sector (1998b), "Packet-based multimedia communications systems", ITU-T Recommendation H.323, Geneva (Switzerland), February 1998
- [10] The International Engineering Consortium, H.323document <http://www.iec.org/exhibits/trillium>
- [11] SS7 and Intelligent networks <http://www.microlegend.com/aboutss7.ht>
- [12] Dialogic Corporation (1999), Dialogic Worldview Homepage, <http://www.dialogic.com/>