Implementation of LAN-Based IP Telephony Simulator

Firas Abdullah Thweny Al-Saedi[1], Jihan Salah [2]

[1,2]Computer Engineering Department, Al-Nahrain University, Baghdad, Iraq

Abstract — *In the past few years, Internet telephony has gained tremendous attention due to the growing use of the Internet. Internet telephony is likely to substitute traditional telecommunication technology because the functionalities of Internet Phone systems are more efficient and better than those of traditional telephone systems. The most significant advantage of Internet Phone systems is that they provide an economical way to make international communication, which is vital to corporations with multi-international sites. This project covers the issues related to implementing a Voice Over Internet Protocol (VOIP) system to provide real-time voice communication over a common Local Area Network (LAN) infrastructure. The LAN based IP telephone system was built depending on the functionality of the NetMeeting Software Development Kit (SDK) and the Visual Basic (VB) programming language. The objects of the ActiveX control were used for embedding the functionality of the NetMeeting on the VB codes. LAN based IP telephone system was tested on a LAN network with 12 Personal Computers (PCs) and the results were satisfactory from the point of view of voice quality and time latency under different conditions of LAN traffic.*

Keywords — *VOIP, LAN, VB, SDK, NetMeeting.*

I. INTRODUCTION

More than 30 years ago Internet didn't exist. Interactive communications were only made by telephone at the Public Switched Telephone Network (PSTN) line cost. Data exchange was expensive (for a long distance) and no one had been thinking to voice interactions [1].

The rapid growth of the internet protocol in the past few years has promoted many aspects of web development such as real-time interactive systems. PSTN is no longer the only means for transmitting voice data. Internet telephony is booming and is becoming one of the fastest moving trends. It allows users to make phone calls to others using the Internet Protocol, just as if they were using an ordinary telephone. The advantage of Internet Telephony is that it provides a more economical way for people to have interactive real-time voice communication with friends or relatives overseas [2].

VOIP is a kind of technology that allows voice communication IP data networks rather than the PSTN. It uses the IP to transmit voice as packets over an IP network. So, VOIP can be achieved on any data network that uses IP, like Internet, Intranets and LAN. VOIP sends voice information digitally in discrete packets rather than using the dedicated circuit-switched protocols of the PSTN [3].

IP telephony is a term used to describe a suite of products and solutions used to transport voice traffic over a data network. Utilizing IP as a transport mechanism, IP telephony allows the user to create a converged network in which all communications (voice, video, or data) share the same infrastructure [4]. Voice over IP (VOIP) is not a new network, but a new application on a larger technology category, IP telephony, which encompasses transmission of FAX, video, and any other form of data traditionally transmitted over the PSTN. The idea of IP telephony is itself part of a larger trend toward convergence of computer, telephone, television, security monitoring, and related technologies [5].

The advantages of VOIP are many, one of them is reducing long distance telephone costs which is always a popular topic and provides a good reason for introducing VOIP. Today flat rate long distance pricing is available with the Internet and can result in considerable savings for both voice and facsimile. This requires the routing of calls to the lowest-cost network, depending on the time of day and destination, and it is referred to as Least-Cost Routing (LCR). In order to achieve greater savings, the network that use the IP protocol routes calls to destinations by re-dialing them through the lowest cost alternative carrier/terminator available [6]. Also Internet technology makes available to anyone with a personal computer and modems the ability to bypass the long distance PSTN to provide lower cost [7].

The sharing of equipment and operations costs across both data and voice users can also improve network efficiency since excess bandwidth on one network can be used by the other, thereby creating economies of scale for voice (especially given the rapid growth in data traffic) [8]. In some respects, IP networks also offer the potential for higher reliability than the circuit-switched network because IP networks automatically re-route packets when any problem happened such as damaged lines [7]. In addition to all the above advantages, VOIP services have compelling technical advantages over circuit switching. VOIP networks are based more on an open architecture than that of their circuit-switched contemporaries. This open, standards-based architecture means that VOIP services are more interchangeable and more modular than that of a proprietary, PSTN. Open standards also translates into the realization of new services that the user can rapidly develop and deploy rather than waiting for a particular vendor to develop a proprietary solution [9].

To build VOIP, regardless of what's in between, a phone conversation between two people requires that each have both a microphone and a speaker. In the traditional telephone, the microphone is located in the mouthpiece and the speaker is located in the earpiece. In an analog telephone, the voice signal produced by the mouthpiece is sent directly along the wire to a telephone exchange or a local PBX [10].

II. ARCHITECTURE OF THE LAN BASED IP TLELHPONE SYSTEM

The overall LAN based IP telephone system was implemented on the computer lab using of 12 PCs connected to each other via TCP/IP LAN networks (as shown in Fig. (1)). The communication process was done based on using the objects of the NetMeeting SDK Active X control.

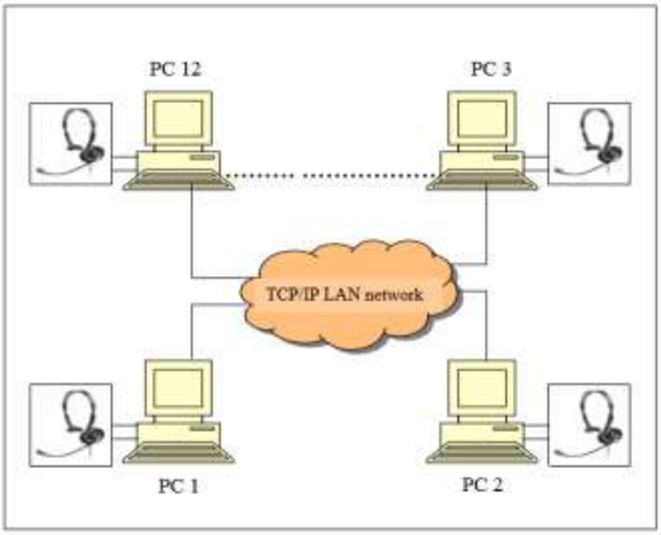

Fig. 1 Architecture of overall LAN based IP telephone system

Each PC has its own IP address to make sure that the message gets to the right place on the TCP/IP LAN network. This allows the user to transmit information to a specific PC on the network excluding all the other PCs. The addresses are stored in specifically defined parts of the IP packet and the LAN frame. The consistent position of addresses is a key factor that allows software to interpret addressing information correctly. In the other words, if the user puts the address in the wrong format the communication software will not work correctly. Also the addresses that are used in this implementation are class C because the number of the PCs that used in this experiment was just 12 PCs and because it's not expensive like class A and class B.

The following points have to be considered. The first point is how to make a phone call between any two PCs, each PC must have its own sound card and an audio device represented by a microphone and a headphone or a speaker. Second, the communication of this LAN phone system is built by exchange of packets between the end-points. On the sending side, the voice information is fragmented and encapsulated into packets. Each packet is then sent to the destination. On the receiver side, the packets are received and assembled to the original voice information for playback. In this project, the NetMeeting ActiveX control is chosen to establish the communication.

III. SYSTEM WORKFLOW

LAN Based IP telephone is the core for this system, which provides real-time voice communication between users. Implementing the NetMeeting ActiveX control using VB programming language does the LAN telephone system. In general there are two steps that must be followed when writing the VB program, the first step is the form design. The form is a container class, which is, consists of one or more pages that behave like a standard window or dialog box. The form contains controls for displaying and editing data. The control is a graphical object, such as a text box, a rectangle, or a command button, that can be place on a form to display data and perform an action in order to make the form easier to read. It's possible to draw these controls on a form using the form controls toolbar in the form designer.

The second step is writing the codes that are responsible for program's drawing controls that are found on the form. The flowchart of the overall LAN Based IP telephone system is shown in Fig. (2), which describes how to make a voice communication between any two PCs of the LAN network of this system.

The system workflow of making a voice communications between any two PCs is explained as follows:

1. The program of this project is build depending on the use of the NetMeeting object. So the first step is to install the NetMeeting SDK (Microsoft NetMeeting SDK can be found at www.microsoft.com/windows/netmeeting or on the owner MSDN CDs). the next step is to change the option of the NetMeeting by define first name, and last name in order to use them when talking with other members. When the installation is complete the program will be ready to use all the NetMeeting objects. Figure (3) shows the LAN based IP telephone form after the installation of the NetMeeting.
2. The NetMeeting object exposes a simple ActiveX control which is used for embedding the NetMeeting on the owner code. The conference manager object is an important object on the ActiveX control objects because this object uses its manager interface to expose methods that

activate and initialize the conferencing system and create calls. Fig. (4) shows the LAN based IP telephone form after the initialization of the conference manager object.

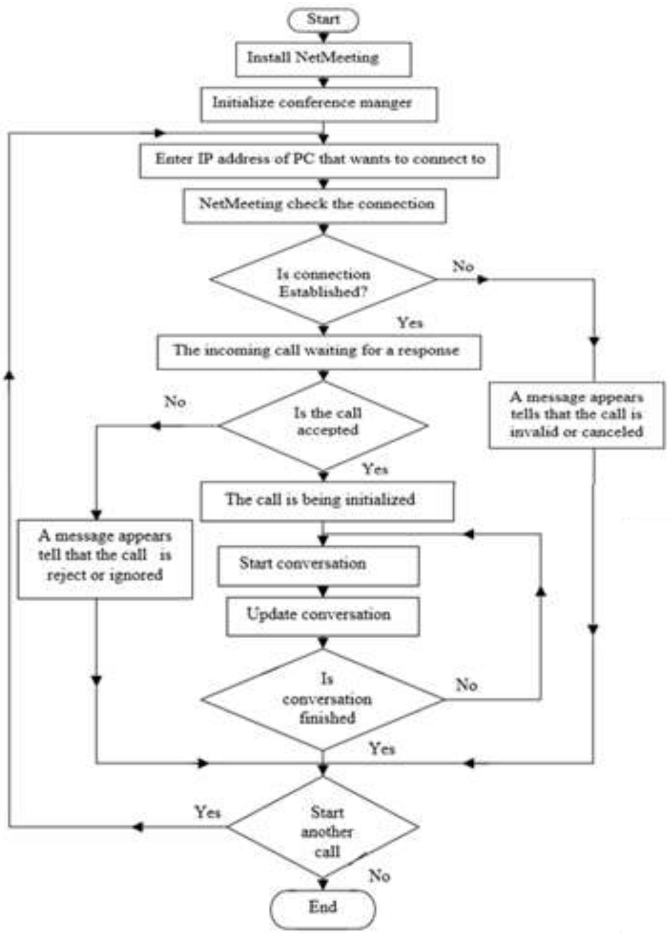

Fig. 2 Flowchart of the overall voice communication process

Fig. 3 The LAN based IP telephone form after NetMeeting being installed

Fig. 4 The form after the conference manager initialization

3. To initialize a call between any two members, the IP address, which is correlated to the PC address that want to connect to, must be typed on the text address that found on the LAN based IP telephone form and then click on the call button to initialize the call. At this moment, if there is any error found on the connection of the LAN network or the IP address of the remote computer was not correct, then the call would be canceled, otherwise two other events can be fired from the conference manager object, and they are the conference object and the call object.
4. When the call is initialized then an index box will appear asking the remote computer to accept or reject the call, and the following Fig. (5) shows the LAN based IP telephone form and the index box.

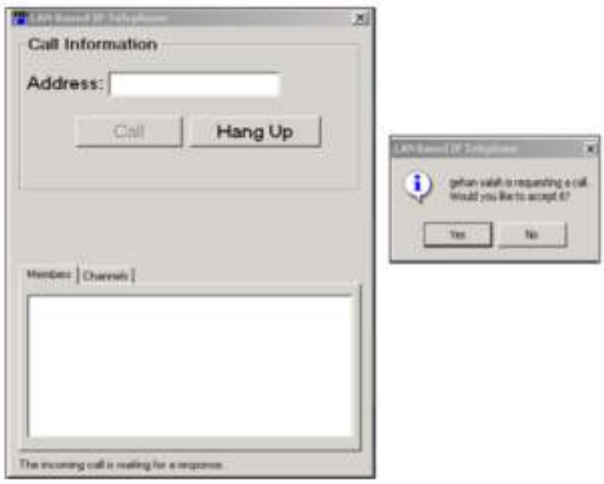

Fig.5 Form showing the index box

5. When the remote computer rejects the call, then the call will be canceled. But if it accepts the call,

then the conversation will start between the two members by using the NetMeeting audio channel also the conversation will be updated until the conversation finished. Figure (6.a) shows the objects of the audio channel which contain the incoming and the outgoing computers, and Fig. (6.b) shows the member objects.

(a) The audio channel object (b) The member objects
Fig. 6 The initialization of the conversation process

6. When the conversation is finished or when one of the users want to end the call, then just click on the Hang Up button that is found on the form in order to remove the connection between the two PCs and to show on the form that the call is not active, it contents no channels or members.

Fig. 7 Form when the conversation was finished

IV. SYSTEM VERIFICATION

As mentioned before, 12 PCs were connected using a LAN Network to build the LAN Based IP phone system, and in order to recognize the response of this system, a simple test was done to calculate the Signal to Noise Ratio (SNR). Here the SNR is the ratio of the signal peak power level to the total noise level and is expressed in decibels (dB). The SNR is computed by searching the entire spectrum to find the peak frequency and then calculating the total noise power in the remaining spectrum. The SNR is then computed as the ratio of the peak power to the noise power and expressed in decibels. The test was done by using three PCs, as shown in Fig. (8) the first and the second PC were connected peer-to-peer by using LAN cable and instead of make a voice communication between these two PCs, a sine wave with 44.1 Kbps was generated on the PC1 (using a software program named Spectra PLUS-FFT Spectral Analysis system version 2.32.04 copyright 1995- 2001 by CA 95008 USA) and transmit this signal to the PC2 through the LAN cable by using the VB program that used to implement the telephone system. The PC3 was used like an oscilloscope that shows the average spectrum of the transmitted and the received sine wave by recording them using the same program that was used for the generation of the sine wave. In this test, a sine wave was used instead of the voice wave because the voice wave is random and cannot be recognized also it’s not easy to see the differences between the transmitted and the received wave. And because the shape of the sine wave is known because it's standard so it's easy to recognize what's happened to the signal when it's transmitted through the LAN cable from PC1 to PC 2. The test was done at two stages the first stage represented the measurement of the SNR ratio for different range of frequencies when there is no traffic between the PC1 and PC2 and the results were shown in Table (1).

The second stage was done by measure the SNR, but with traffic by using a file, its size is 250 MByte transmitted from two directions of the LAN cable and the results shown in Table (2).

The term traffic meaning that a file of size of 250 MByte was transferred between PC1 and PC2, when there is calling between them in order to load the telephone system to recognize the differences between the two cases when there is traffic and when there is no traffic.

Figures: Fig. (9), to Fig. (19), shows the average spectrum for the transmitted and the received waves for a range of frequencies from 100 Hz to 3400 Hz when there is no traffic where the upper channel represented the transmitted signal and the lower channel represented the received signal.

Table (1) Shows the SNR when there is no file transfer

Frequency (Hz)	SNR for the transmitted signal (dB)	SNR for the received signal(dB)
100	61.150	57.978
400	59.150	57.254
800	58.750	55.123
1600	58.930	55.771
2000	59.385	56.000
2400	57.195	54.622
2800	56.650	53.600
3000	55.904	52.778
3200	55.249	51.220
3300	54.892	50.130
3400	54.965	49.611

Table (2) Shows the SNR when there is a file transfer of about 250 MByte in size

Frequency (Hz)	SNR for the transmitted signal (dB)	SNR for the received signal (dB)
100	60.376	56.646
400	58.898	55.689
800	59.779	54.087
1600	58.525	53.093
2000	58.032	53.713
2400	56.982	52.130
2800	55.327	51.528
3000	55.750	51.166
3200	54.297	50.819
3300	53.387	49.997
3400	53.115	49.542

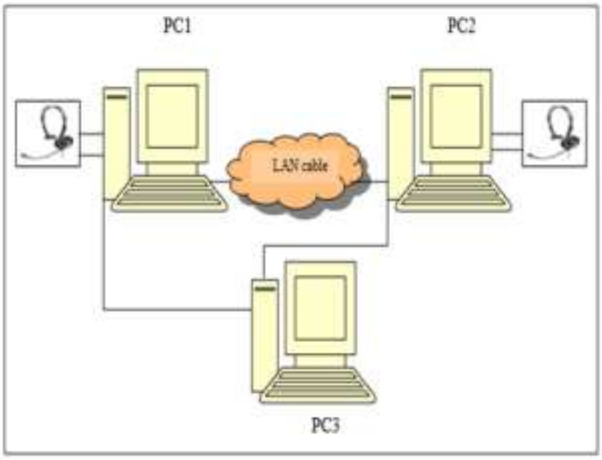

Fig. 8 The three computers that were used on the sine wave test

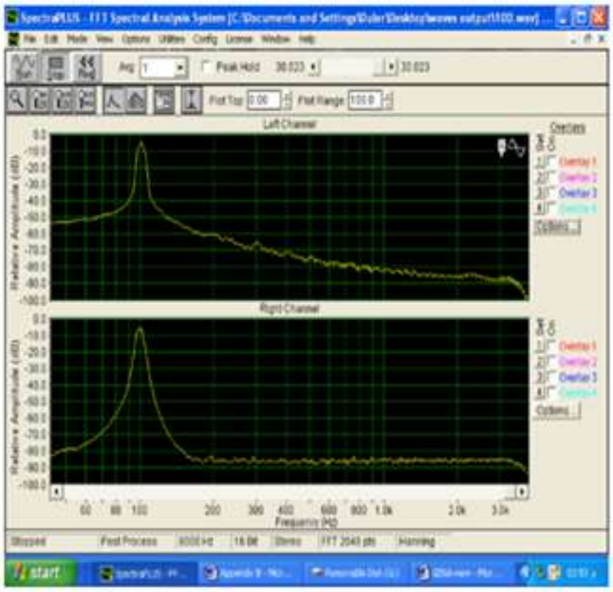

Fig. 9 The average spectrum for the transmitted and the received sine wave at frequency 100Hz when there is no traffic

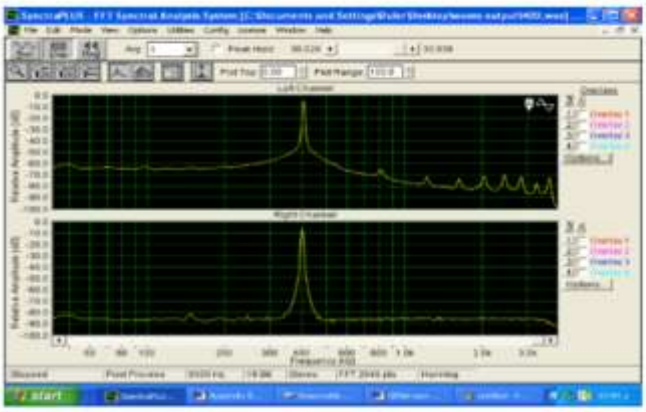

Fig. 10 The average spectrum for the transmitted and the received sine wave at frequency 400Hz when there is no traffic

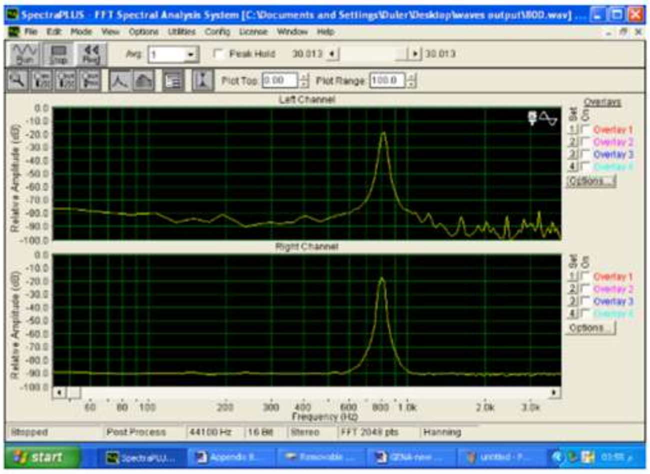

Fig. 11 The average spectrum for the transmitted and the received sine wave at frequency 800Hz when there is no traffic

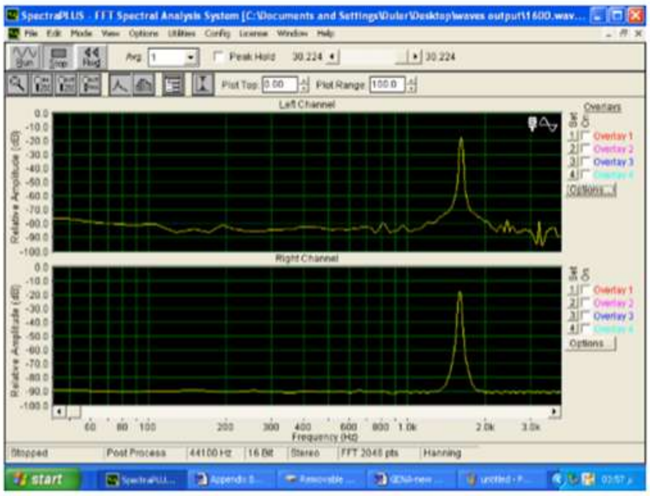

Fig. 12 The average spectrum for the transmitted and the received sine wave at frequency 1600Hz when there is no traffic

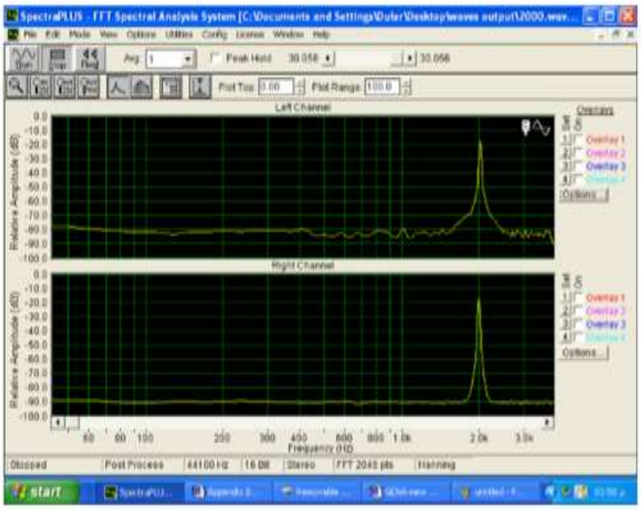

Fig. 13 The average spectrum for the transmitted and the received sine wave at frequency 2000Hz when there is no traffic

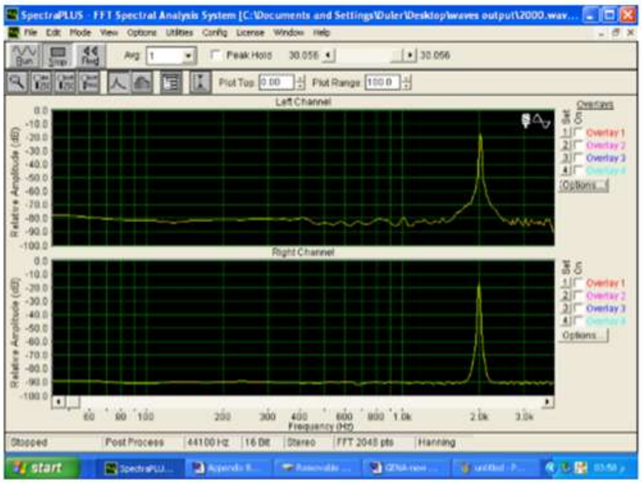

Fig. 14 The average spectrum for the transmitted and the received sine wave at frequency 2400Hz when there is no traffic.

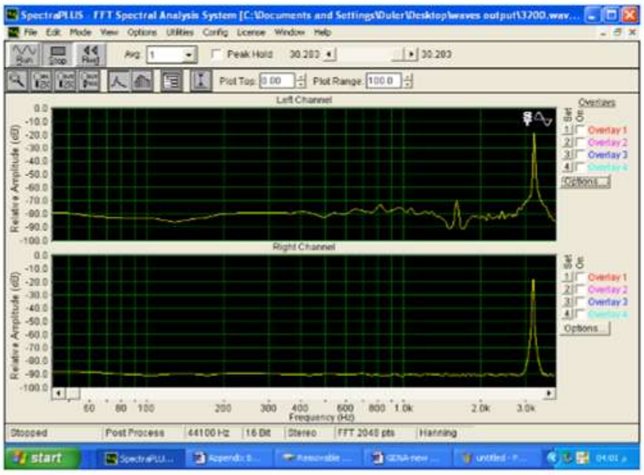

Fig. 15 The average spectrum for the transmitted and the received sine wave at frequency 2800Hz when there is no traffic

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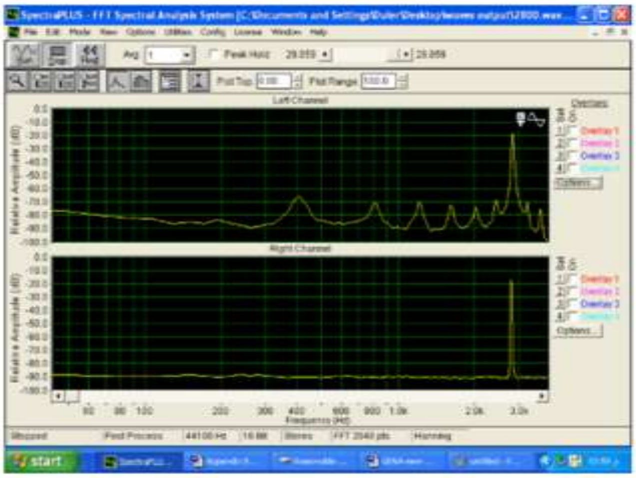

Fig. 16 The average spectrum for the transmitted and the received sine wave at frequency 3000Hz when there is no traffic

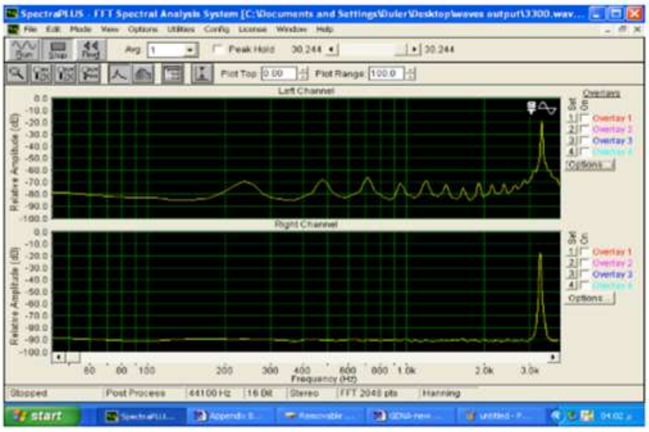

Fig. 17 The average spectrum for the transmitted and the received sine wave at frequency 3200Hz when there is no traffic

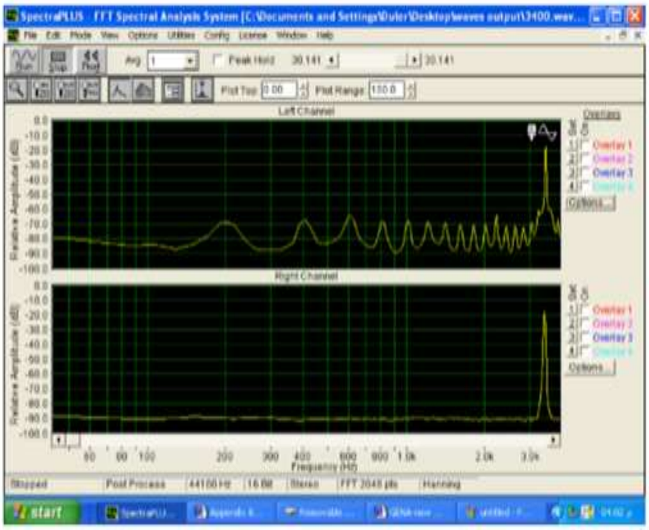

Fig. 18 The average spectrum for the transmitted and the received sine wave at frequency 3300Hz when there is no traffic

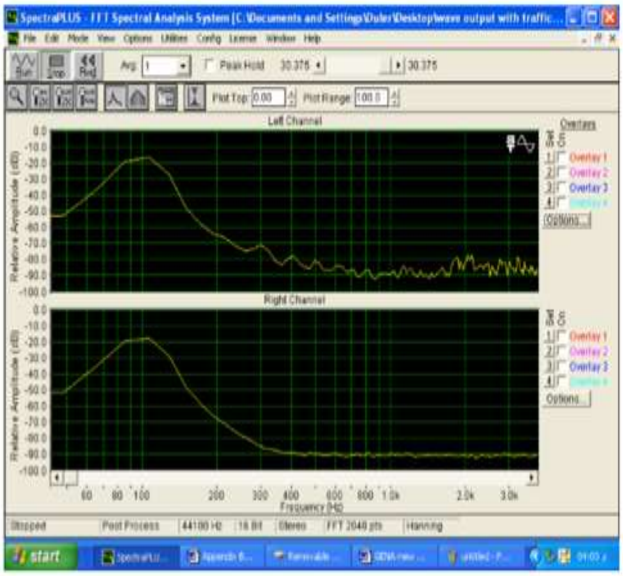

Fig. 19 The average spectrum for the transmitted and the received sine wave at frequency 3400Hz when there is no traffic

Moreover, Figures: Fig. (20), to Fig. (30), show the average spectrum for the transmitted and the received waves for a range of frequencies from 100 Hz to 3400 Hz when there is traffic where the upper channel represented the transmitted signal and the lower channel represented the received signal.

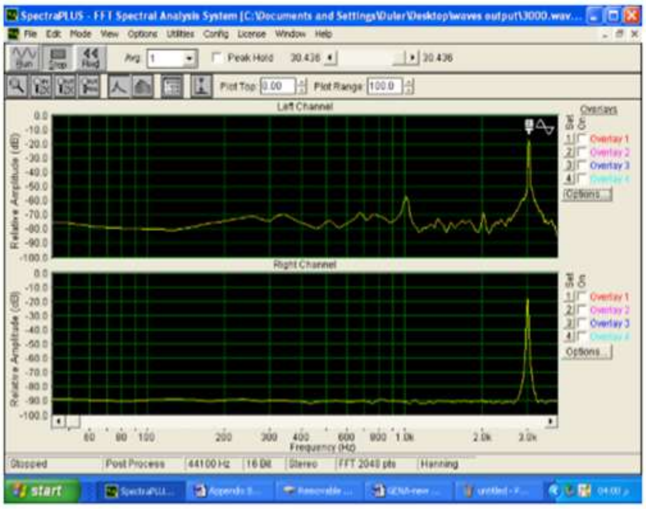

Fig. 20 The average spectrum for the transmitted and the received sine wave at frequency 100Hz when there is traffic

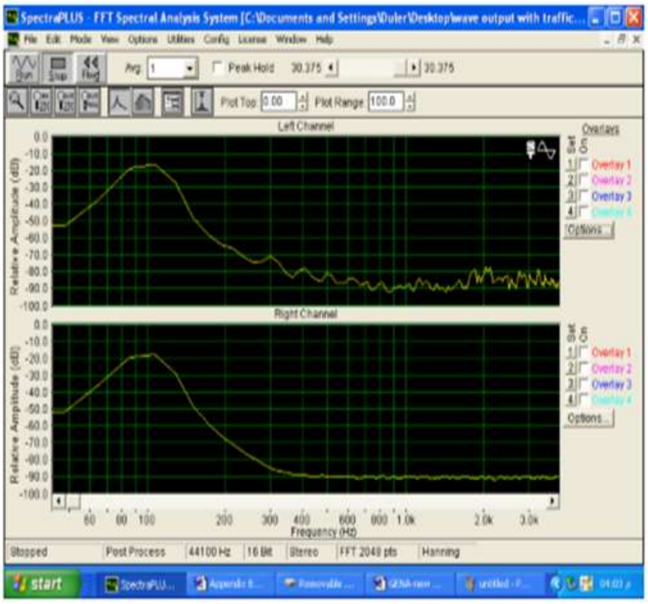

Fig. 21 The average spectrum for the transmitted and the received sine wave at frequency 400Hz when there is traffic

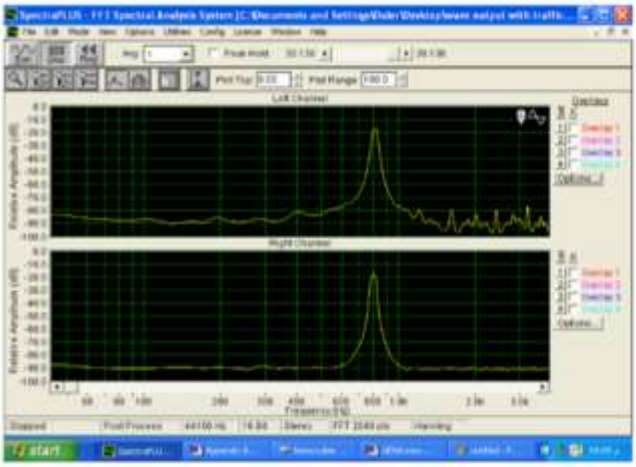

Fig. 22 The average spectrum for the transmitted and the received sine wave at frequency 800Hz when there is traffic

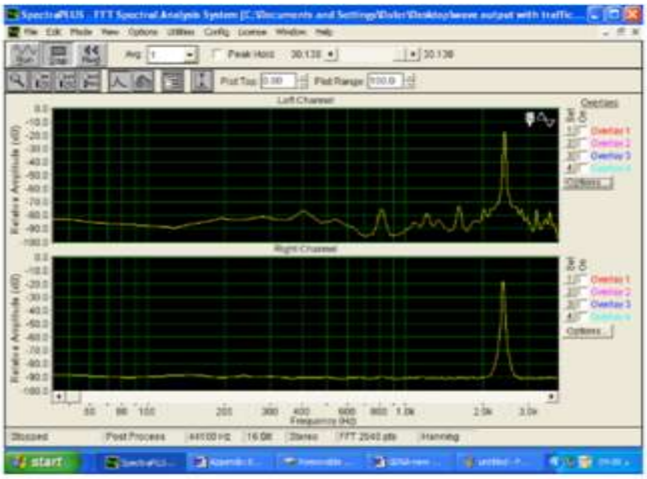

Fig. 23 The average spectrum for the transmitted and the received sine wave at frequency 1600Hz when there is traffic

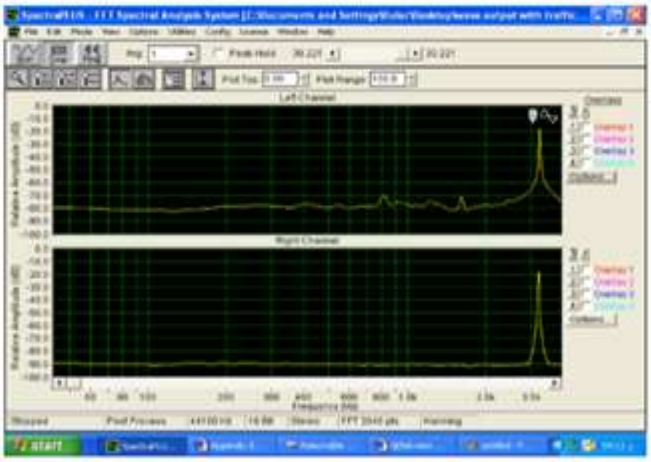

Fig. 24 The average spectrum for the transmitted and the received sine wave at frequency 2000Hz when there is traffic

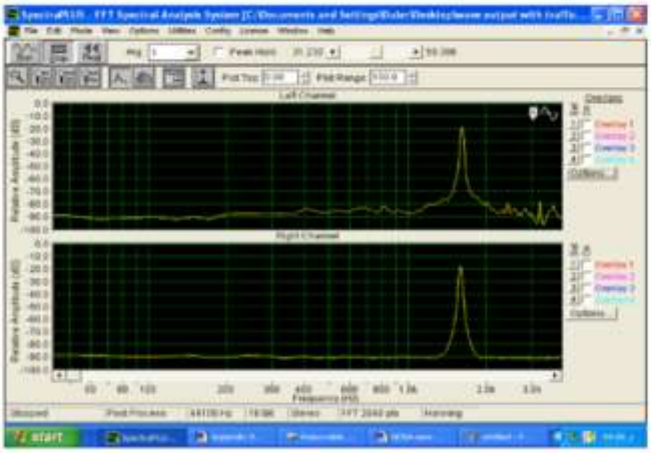

Fig. 25 The average spectrum for the transmitted and the received sine wave at frequency 2400Hz when there is traffic

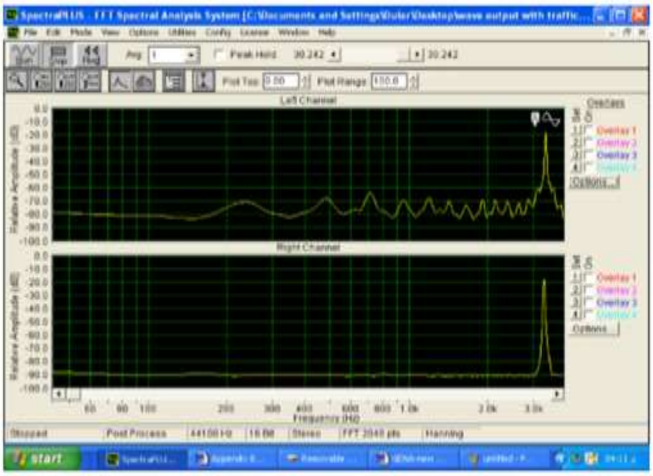

Fig. 26 The average spectrum for the transmitted and the received sine wave at frequency 2800Hz when there is traffic

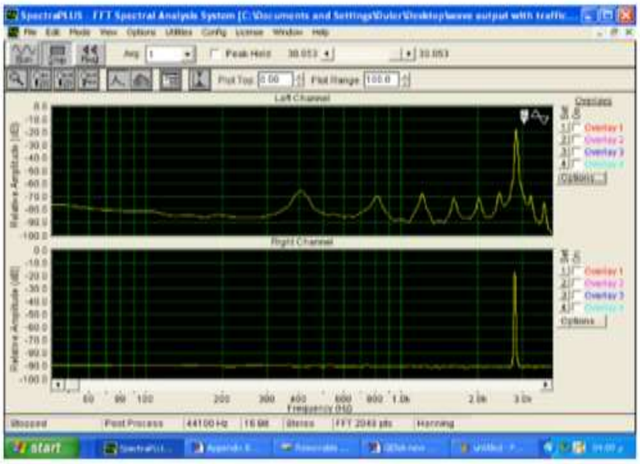

Fig. 27 The average spectrum for the transmitted and the received sine wave at frequency 3000Hz when there is traffic

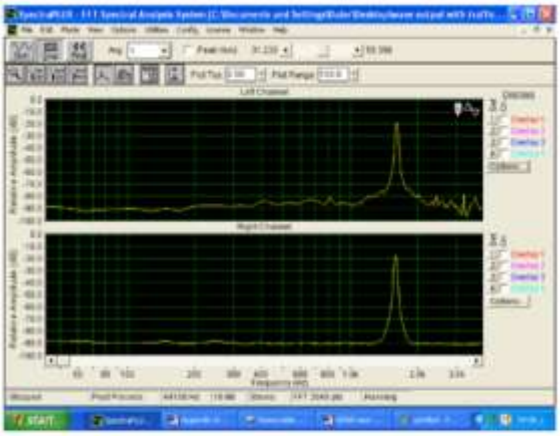

Fig. 28 The average spectrum for the transmitted and the received sine wave at frequency 3200Hz when there is traffic

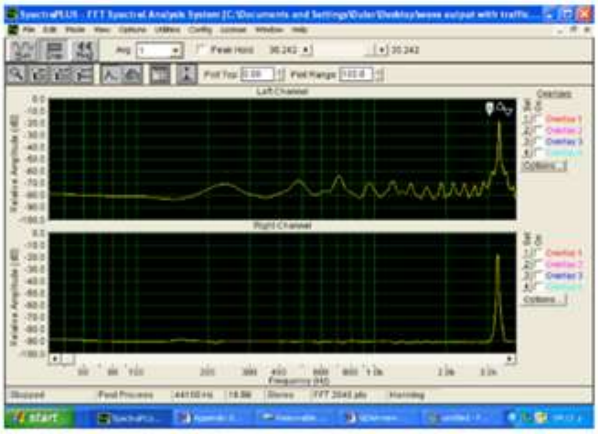

Fig.29 The average spectrum for the transmitted and the received sine wave at frequency 3300Hz when there is traffic

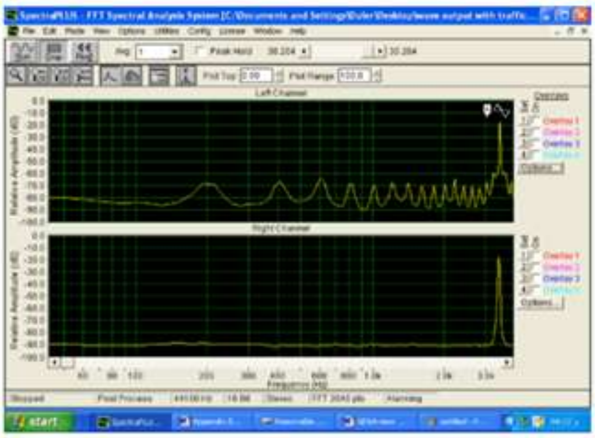

Fig. 30 The average spectrum for the transmitted and the received sine wave at frequency 3400Hz when there is traffic

V. CONCLUTION

The LAN based IP telephone represent the development of VOIP over a LAN network in making a successful phone call.

The Microsoft NetMeeting SDK provides a simple, yet powerful component object model that helps for embed the NetMeeting applications functionally on the VB cods that was used to build the system of the telephone. The VB was used as a programming language, and the LAN based IP telephone system uses the ActiveX control, (which is a component object model), for embedding the functionality of the NetMeeting SDK on the owner code and that provide real-time voice communication between users.

The system was tested using 12 PCs and the overall results of the system were found to be satisfactory for quality of the voice over the network. The SNR test shows that there is a very small difference between the SNR values during traffic and no traffic cases because the channel bandwidth is about 100 Mbps.

But there is a difference between the SNR results for the transmitted and the received signals where the SNR for the transmitted signal is more than that of the received sine wave and this is because of the attenuation that was result from the LAN cable that is connecting between PC1 and PC2.

As a results from this paper, good quality of voice for the LAN based IP telephone system is expected to be executed by corporation LAN network with a high bandwidth reserved. Therefore, the LAN network of this system was running at 100Mbps. Running at this bit rate, the voice quality is very good.

Also accept and reject the call before communication is established, the local computer needs to send a request to the remote computer and then waits for acknowledgment. This mechanism provides the remote computer with the capability of accepting or rejecting a call. On the other hand, the local computer can also have the right to drop the call while waiting for the response.

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