Exploiting VoIP Telephony in IP-Pbx Solution.

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ABSTRACT

This study entails the simulation and implementation of a VoIP telephony system using an IP PBX solution. A new technology called Voice over Internet Protocol (VoIP), or Internet telephony means that your voice is carried over the IP network, otherwise known as the Internet. Voice, which is an analogue signal, is converted to digital data, which is then disassembled and transmitted through the Internet only to be re-converted back to an analogue signal on the other end using an IP-PBX solution which is a Linux base server that supports a GUI web interface. This service can be properly managed and deployed over a network with less stress and expenses. The IP PBX main server also has integrated in it other communication services such as instant messaging using an Openfire server, electronic mail using a postfix server and FTP service all embedded in the IP PBX SYSTEM

This technology promises an evolutionary leap beyond the standard telephone service we have been accustomed to, as well as a host of benefits for individuals. The new technology transmits voice signals the same way email is sent, using the Internet's data-transfer protocols to break conversations into digital packets that can be sent on lower-cost, more efficient "packet-switched" networks. That innovation makes many other innovations possible, from eliminating the distinction between local and long distance calls, to easily maintaining several telephone numbers in a single account, to sorting and storing voice messages on your computer, FTP between users and instant messaging. This project was able to address the persistent communication problem which existed in the department by allowing users to communicate with the services the solution provided with less stress and comfort.

(Keywords: voice over internet protocol, telephony)

INTRODUCTION

The structure of the wire line telephone network has not fundamentally changed since Alexander Graham Bell invented the telephone 130 years ago. Calls travel along dedicated phone lines and are switched in various places to maintain an electronic circuit leading from the phone where the call is made to the phone with the number being called—hence the phrase "circuit-switched network" [1].

While digital technologies have increased the functionality of the phone system, any intelligence in the system is largely engineered into the circuit switches of the telephone company's central office. Contrast to that the broadband Internet world. Consumers use a computer to type an email that is then separated into digital packets (groups of ones and zeros) that are transported via a broadband "pipe" onto high-speed backbone networks. The packets are then reassembled at the destination account, which can be configured for access from any Internet connected computer anywhere in the world [2].

Many years ago it was discovered that sending a signal to a remote destination could have be done also in a digital fashion: before sending it we have to digitalize it with an ADC (analog to digital converter), transmit it, and at the end transform it again in analog format with DAC (digital to analog converter) to use it. "VOIP works like that, digitalizing voice in data packets, sending them and reconverting them in voice at destination". Digital format can be better controlled: we can compress it, route it, convert it to a new better format, and so on; also we saw that digital signal is more noise tolerant than the analog one (GSM vs. TACS) [3]. TCP/IP networks are made of IP packets containing a header (to control communication) and a payload to transport data: VOIP uses it to go across the network and come to destination.

In telecommunications (beyond the level of two cans and a piece of string) there are 2 bi-directional streams (possibly over the same channel. In fact, it was common for a number of years to have the control stream riding on the same medium as the voice stream. This is known as "in-band" control or signaling.

A new technology called Voice over Internet Protocol (VOIP), or Internet telephony, promises an evolutionary leap beyond the standard telephone service we have been accustomed to, as well as a host of benefits for consumers. The new technology transmits voice signals the same way email is sent, using the Internet's data-transfer protocols to break conversations into digital packets that can be sent on lower-cost, more efficient "packet-switched" networks. That innovation makes many other innovations possible, from eliminating the distinction between local and long distance calls, to easily maintaining several telephone numbers in a single account, to sorting and storing voice messages on your computer.

However with the advent of digital transport systems, most in-band signaling has been replaced by a separate control channel which is known, appropriately enough, as "out-of-band" signaling), a "voice" stream (which includes voice, music, fax squeals, modem shrieks, etc.) and a control stream. VOIP refers to the transport of a telecommunications voice stream over a data network using the data transport mechanisms associated with the Internet, called the Transmission Control Protocol/Internet Protocol (TCP/IP) suite.

A vendor presents the following ingenious and useful application as part of their VOIP strategy. It is a solution primarily for remote workers. The worker needs an IP connected computer and a Plain Old Telephone System (POTS) line. The worker uses a browser to connect to a web-site which, in turn, communicates with the IP PBX. The user then identifies and authenticates, giving the phone number of the POTS phone. The PBX connects to the telephone, and provides all the sophisticated features of the PBX like hold, conference calling, etc. by means of the phone line plus feature access through the computer interface) [4].

A private branch exchange (telephone switching system within an enterprise) that switches calls between VOIP (voice over Internet Protocol or IP) users on local lines while allowing all users to share a certain number of external phone lines. The typical IP PBX can also switch calls between a VOIP user and a traditional telephone user, or between two traditional telephone users in the same way that a conventional PBX does [4]. The abbreviation may appear in various texts as IP-PBX, IP/PBX, or IPPBX.

With a conventional PBX, separate networks are necessary for voice and data communications. One of the main advantages of an IP PBX is the fact that it employs converged data and voice networks. This means that Internet access, as well as VOIP communications and traditional telephone communications, are all possible using a single line to each user. This provides flexibility as an enterprise grows, and can also reduce long-term operation and maintenance costs. Like a traditional PBX, an IP PBX is owned by the enterprise.

LITERATURE REVIEW

Voice over Internet Protocol, VOIP, also known as Broadband phone or IP telephony service, is changing the telephony world. Traditional phone lines are slowly being phased out as businesses and households around the world embrace the benefits and features that VOIP technology has to offer. As this evolution accelerates, it is worthwhile to stop and take a look at the history of VOIP. One will find that as interesting as the history of VOIP may be, the future of VOIP is even more intriguing and exciting.

The history of VOIP shows that this technology started as far back as 1995, when a small company called Vocaltec released the first internet phone software. This software was designed to run on a home PC and much like the PC phones used today, it utilized sound cards, microphones and speakers. The software was called "Internet Phone" and used the H.323 protocol instead of the SIP protocol that is more prevalent today. Vocaltec had initial success with Internet Phone, and had a successful IPO in 1996. It was the Skype of the mid 90s [3].

By 1998, VOIP traffic had grown to represent approximately 1% of all voice traffic in the United States. Entrepreneurs were jumping on the bandwagon and were creating devices which enabled PC-to-phone and phone-to-phone communication. Networking manufacturers such as Cisco and Lucent introduced equipment that could route and switch the VOIP traffic and as a result by the year 2000, VOIP traffic accounted for more than 3% of all voice traffic.

Most organizations, companies, and institutions are switching to IP PBX compared to the conventional PBX. An IP PBX handles both voice and data, it is cheaper since it requires only one network to install and maintain instead of two, It reduces equipment costs (only IP based products; no voice products needed), reduces long distance charges for inter-branch office calls (by using the data network), is easier to provision (just plug in from wherever), supports services such as unified messaging, is more flexible, is more scalable, makes it easier to provide new services, such as data and video collaboration, allows remote configuration (over the Web), and supports modular software upgrades, new technologies (new CPUs, etc.) are easy to incorporate. Therefore this makes the future of this solution seamless and promising.

Aim of Research

Over the past couple of years, there has been much discussion in the industry about the convergence of voice and data and the pros and cons of transmitting voice over a data network. The purpose of this white paper is to address the question of what voice over IP telephony is using IP/PBX solution.

The aim of this research is to build the perfect IP PBX solution capable of integrating internet telephony with other communication exchange services like electronic mail, file transfer protocol and instant messaging all working seamlessly in the same exchange box or server which can be implemented in an environment or an area such as our institution using our department as a case study.

METHODOLOGY

VOIP telephony service using an IP-PBX solution can be implemented in several ways. In achieving the stated objective the following would have to be put into consideration:

- A review of literature leading up to this research work.
- Where is it going to be implemented and assembling of hardware devices needed for the IP PBX VOIP system?
- What type of network infrastructure is made available?
- What call solution is needed?
- Installation of the software components and configuration of the required servers.
- Writing and modification of scripts needed for the GUI interface.
- What kinds of equipment are going to be used to achieve this solution?
- What extra feature the customer/user might want to use this solution for?

Circuit- Switched Telephony

Before digital networking took off, everyone had to use the one and only Plain Old Telephone Services (POTS). POTS runs over a network called the Public Switched Telephone Network (PSTN). The PSTN has been around since Mr. Bell invented the telephone. That is why most companies today have POTS related systems in place [6]. These POTS telephone systems use the old tried-and-true (and more expensive) method of telephone service known as circuit-switched.

A good illustration of POTS and PSTN is the experiment where you had to take two tin cans and a length of wire to create an archaic telephone system [1]. As strange as it seems, this antiquated method of telephony is the principal means underlying the operation of POTS operating over the PSTN. “What changes in the real POTS-based telephony system are the number, length, diameter, and type of wire or cables used? These elements have grown immensely in variety and type”.

In addition, the types of telephone equipment have changed dramatically both at the customer end and at the carrier provider end. But POTS telephony continues to use “circuit switched” rules (or protocols) of operation [6].

Packet-Switched Telephony

From POTS to packets unlike circuit-switched POTS, which always require use of the Public Switched Telephone Network (PSTN), VOIP technology has enabled telephony and other new and novel features and services to run over dedicated and wireless networks including even your computer network [6].

These newer network types use packet-switched protocols. Packet-switched VOIP puts voice signals into packets. Along with the voice signals, VOIP packets include both the sender's and receiver's network addresses. VOIP packets can traverse any VOIP-compatible network. Along the way, they can choose alternate paths because the destination address is included in the packet [6]. The routing of the packets is not dependent on any particular network route. In a circuit-switched network, the destination address is not included in the signal; routing directions are determined physically by the actual POTS line. So the routing must follow a specific network line similar to how a train follows a designated set of railroad tracks. If the line is down, the call cannot go through.

In a packet-switched network, if one of the network lines is down, the packet can switch while in route between locations to another working route to keep the call up. Using VOIP, voice signals can be packetized like computer data packets [6]. This enables companies to consider using the same network infrastructure to support both data and voice applications. Companies can consolidate their physical networks (while maintaining redundancy in their routing patterns) and build an enterprise-class communications network with the latest advanced IP-based features.

IP Telephony

IP Telephony enables voice communication over Internet Protocol (IP) networks". It unites an organization's many locations—including mobile workers—into a single converged network. It promises cost savings by combining voice and data on one network that can be centrally maintained. But more importantly, it brings advanced features and applications that enhance productivity throughout the organization.

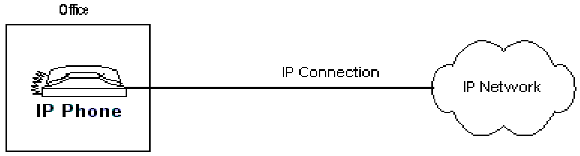

Figure 1: An IP Phone Connecting to an IP Network.

VOIP

VOIP is a solid technology available since some years that allows people to communicate via voice using the IP protocol instead of telephone lines [7]. VOIP is a technology for communicating using "Internet protocol" instead of traditional analog systems. Some VOIP services need only a regular phone connection, while others allow you to make telephone calls using an Internet connection instead [4]. Some VOIP services may allow you only to call other people using the same service, but others may allow you to call any telephone number - including local, long distance, wireless, and international numbers.

Voice over IP – the transmission of voice over packet-switched IP networks – is one of the most important emerging trends in telecommunications. As with many new technologies, VOIP introduces new security.

IP-PBX

Before VOIP, the PBX was the mainframe of corporate telephony. PBX stands for Private Branch Exchange or Premises Business Exchange. An IP PBX is a private branch exchange (telephone switching system within an enterprise) that switches calls between VOIP (voice over Internet Protocol or IP) users on local lines while allowing all users to share a certain number of external phone lines. The typical IP PBX can also switch calls between a VOIP user and a traditional telephone user, or between two traditional telephone users in the same way that a conventional PBX does. The abbreviation may appear in various texts as IP-PBX, IP/PBX, or IPPBX [4]. With a conventional PBX, separate networks are necessary for voice and data communications.

One of the main advantages of an IP PBX is the fact that it employs converged data and voice networks. This means that Internet access, as well as VOIP communications and traditional telephone communications, are all possible using a single line to each user. This provides flexibility as an enterprise grows, and can also reduce long-term operation and maintenance costs. Like a traditional PBX, an IP PBX is owned by the enterprise.

IP PBX system consists of one or more SIP phones / VOIP phones, an IP PBX server and optionally includes a VOIP Gateway. The IP PBX server is similar to a proxy server: SIP clients, being either soft phones or hardware based phones, register with the IP PBX server, and when they wish to make a call they ask the IP PBX to establish the connection. The IP PBX has a directory of all phones/users and their corresponding SIP address as shown in Figure 2. Thus is able to connect an internal call or route an external call via either a VOIP gateway [4].

However, it delivers the most value out of all four as well. Some key value points are:

- The PBX can use dedicated high-bandwidth lines out to the carrier or to other locations on the company's network.
- Interfaces can be used on the PBX to provide full motion videoconferencing.
- The PBX has extensive Call Management capabilities and the capacity for setting up and controlling multiple call centers.

By using their own system, companies reduce the total number of POTS lines required by a factor of one line for every six to eight employees. The phone system's circuitry integrates multiple users over fewer lines. With the PBX, videoconferencing and other high-bandwidth applications could be integrated.

Although companies could reduce the total number of lines required and therefore their total MRC, they still have to pay for local and toll usage. But with their own system, they are able to provide most of the traditional telephony call features at no extra cost.

The good news is that integrating IP Telephony and VOIP onto your computer network can be done while keeping your conventional POTS-PSTN telephony systems operational. Because the two are physically separate networks, they can operate simultaneously.

Fault-isolation can be more readily processed because you do not need to troubleshoot what network the problem may be on.

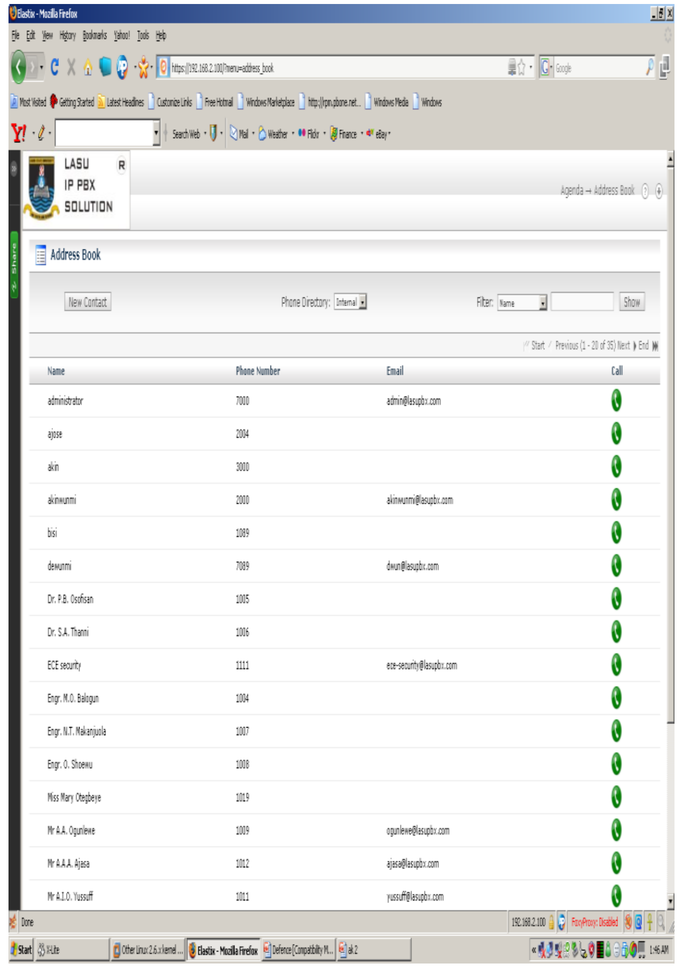

Figure 2: LASU IP PBX Directory.

There is only one VOIP network with one or more distinct LANs running IP Telephony. Because you will unify its support staff into one department, the ensuing cross-training and convergence experience to be gained by all in this department can result in a reduction of the your dependence on outside experts [5].

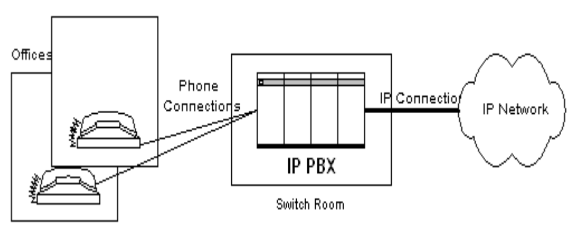

Figure 3: An IP PBX System.

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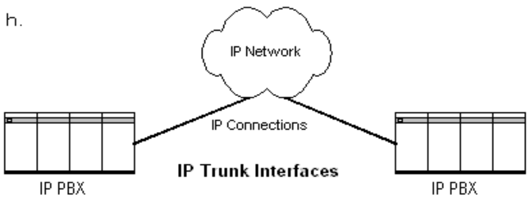

Figure 4: Interfacing two IP PBX Systems Together.

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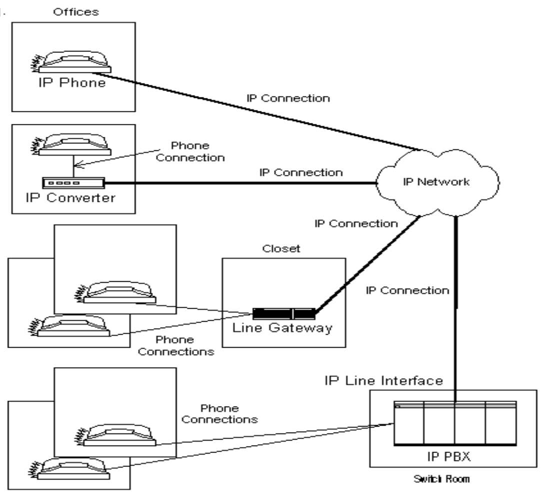

Figure 5: Interconnection of Phones in an IP PBX System.

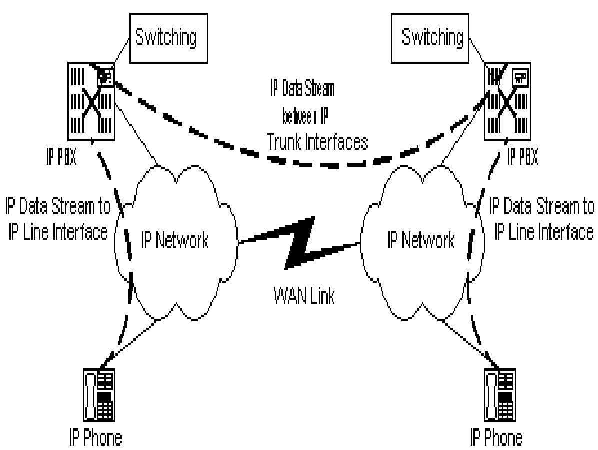

Figure 6: IP Phones Connecting to a Wide Area Network (WAN).

SIMULATION AND TESTING OF IP PBX SYSTEM

This project involves the design and simulation of a VOIP telephony system using IP PBX solution. To implement this task, the project has been divided into two segments:

1. Hardware components
2. Software components

Using the Electronics & Computer Engineering Department of LASU as a case study, the implementation can be done in all departments using a local area network patched in all offices, a switch Poe preferable, an IP PBX Linux based. This would be customized to be the VOIP box and the TFTP server for the SIP, registration of extension, IP protocol and packetization of data because VOIP is packet switch and not circuit switched so voice is converted to data in little bundles packets and travels via IP through any available route so when one route goes down it can still find another route without losing its data unlike PSTN which is in one route where the circuit is been formed. CAT 5 straight cables, network interface card, IP phones, heard set. It can also be used to make external calls by getting an analogue card and connecting it via a GSM module were the GSM SIM can be connected to make local calls or a digital card using T1/E1 lines to connect to the PSTN.

A VOIP Phone System / IP PBX system consists of one or more SIP phones / VOIP phones, an IP PBX server and optionally includes a VOIP Gateway. The IP PBX server is similar to a proxy server: SIP clients, being either soft phones or hardware based phones, register with the IP PBX server, and when they wish to make a call they ask the IP PBX to establish the connection. The IP PBX has a directory of all phones/users and their corresponding SIP address and thus is able to connect an internal call or route an external call via either a VOIP gateway or a VOIP service provider [10].

An IP PBX system is easy to configure and install but you should know the structure of IP PBX phone systems. An IP PBX phone system consists of:

- IP PBX server
- VOIP phones (SIP phones)

- VOIP gateway (optional)

The following components are needed to set up IP PBX:

- IP PBX itself
- Phones (VOIP soft phones, VOIP phones or ordinary phones with VOIP adapters)
- SIP Gateway (to call other people on the PSTN).
- Large storage hard drive depending on the capacity of the database but basically for an organization it's good to start with a large storage because database gets larger by the day.

The center of the IP PBX system is the IP PBX server that performs functions similar to those of a proxy server. All the clients (VOIP phones) are registered with the IP PBX server. When a client needs to make a call the IP PBX server should give the permission to establish the connection. Clients may be either VOIP phones (SIP phones) or VOIP soft phones. In case you want to use ordinary telephones you should fix ATA (VOIP adapters that convert voice into digital data and vice versa).All VOIP phones (clients) are listed at the IP PBX server along with their SIP addresses. The IP PBX server can establish the connection either through a VOIP gateway or a VOIP service provider [10].

To solve the problems about bandwidth, how to create a real time streaming of data, we enable a right real−time manager protocol in each router we cross, then we try to use a very high rate compression algorithms (like LPC10 which only consumes a 2.5 kbps bandwidth, about 313 bytes/s). Then we start to classify our packets, in TOS field, with the most high priority level, so every router helps us having urgently. Important: all that is not sufficient to guarantee our conversation would always be good, but without a great infrastructure managing shaping, bandwidth reservation and so on, it is not possible to do it, TCP/IP is not a real time protocol [2].

The H.323 family of standards under the aegis of the International Multimedia Teleconferencing Consortium (IMTC) and the International Telecommunications Union (ITU) is central to VOIP. This set of standards includes standards for voice (G.711, G.722, G.723, G.728, G.729) and video (H.261, H.263) compression which are accepted industry-wide [4]. On the other hand, the accompanying signaling/control standards (e.g., H.225) are less widely implemented. In fact, the IETF has a competing standard, called Session Initiation Protocol (SIP).

COMPONENTS OF AN IP PBX SERVER

Four main components needed to set up VOIP PBX:

- Elastix, the Asterisk powered IP PBX
- The phones (or soft phones)
- The VOIP gateway services that let you call other VOIP users and people on the PSTN.
- Have a network and broadband access with a router and hubs/switches if needed.

If you are restricting the usage to PSTN only, you will not need the broadband, example of such network with the interconnection of PBX system components are shown in Figure 7.

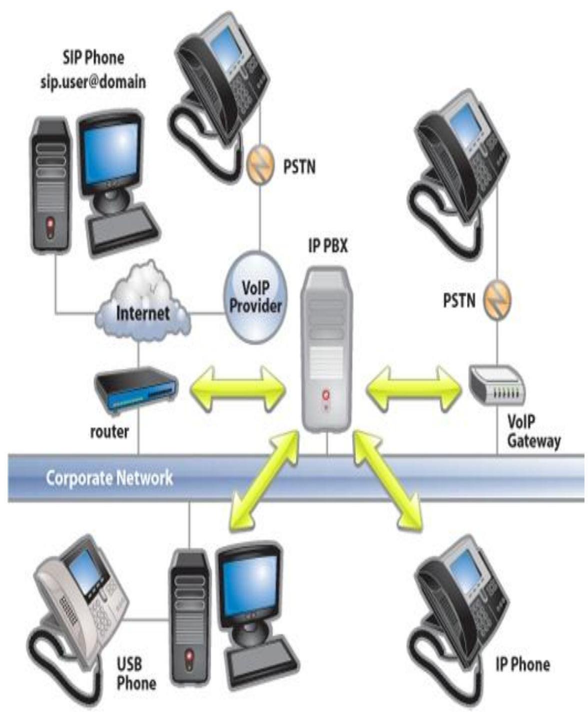

Figure 7: Interconnection of an IP PBX System with its Components.

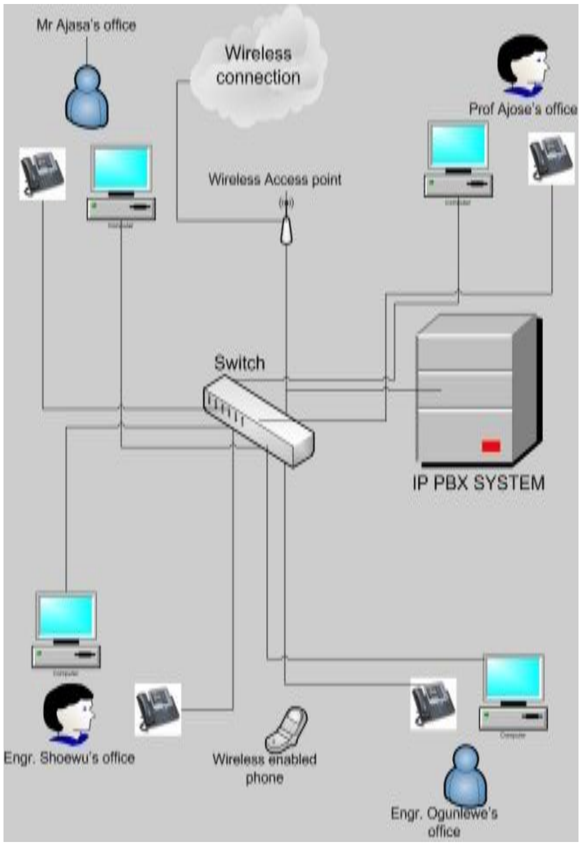

Figure 8: IP PBX System Schematic.

RESULT AND DISCUSSION

During our simulation and implementation in the department we discovered that PCs which were running firewall protection services affected the voice quality of the test and in some cases disallowed the communication of the VOIP soft phones between users.

There were no DHCP and DNS server readily available for use during the simulation so we configured a DHCP server in the PBX box to assign IP to users instead of assigning static IP to users which made the implementation faster and easier.

During implementation, multiple users were able to register on the PBX server at the department and communicate with themselves using the various services available like the instant messaging, electronic mail and VOIP calls.

We were also able to achieve a file transfer speed of 116mb/s wirelessly using a D-link wireless N router between users using the FTP services.

We were also able to achieve the follow me services as unavailable user received voice mails which were redirected to the email created for them and calls redirected to the follow me extensions configured.

It was also discovered that using multiple switches and routers resulted in degradation in the quality of the VoIP service during the implementation at the department.

A VoIP telephony system using an IP-PBX solution / system was achieved with optimum benefits and little limitations which supports other branch exchange services like instant messaging, electronic mail and file transfer protocols all working seamlessly in the main server to achieve the goal of eradicating the persistent communication problem which exist in the Department of Electronics & Computer Engineering, LASU which was used as a case study for one of the main objective of this project. Also a cost effective communication branch exchange system was achieved which makes this solution preferable to other private branch exchange services made available and can be easily deployed and maintained.

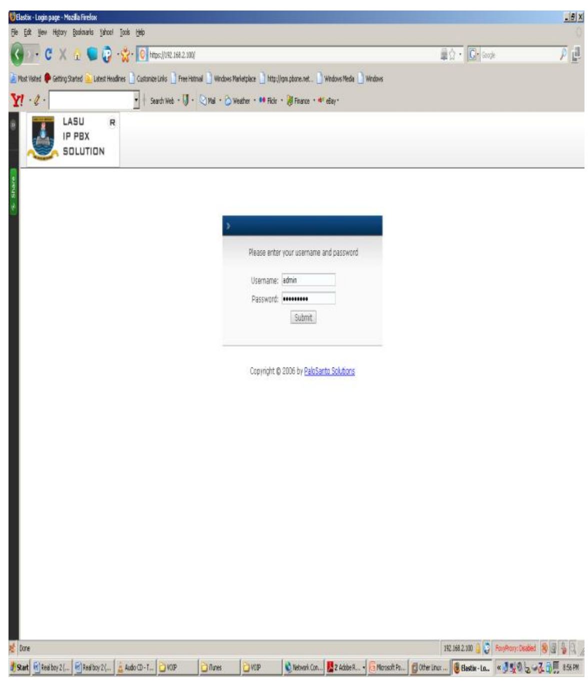

Figure 9: LASU IP PBX GUI Web Interface Page.

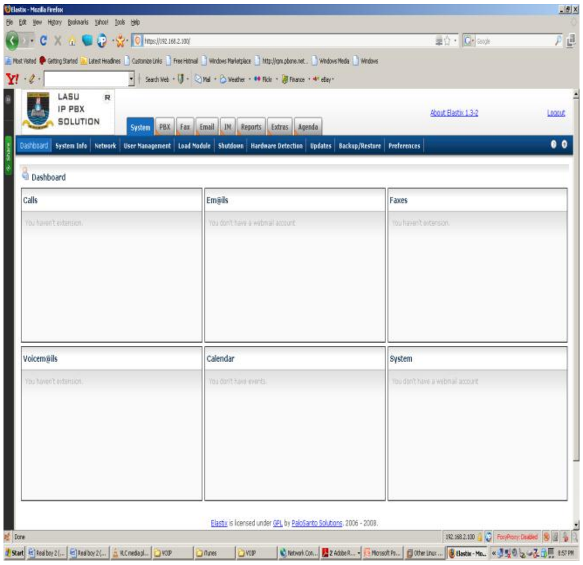

Figure 10: PBX Extension Configuration Page.

IP PBX RESULT INTERFACES SCREEN

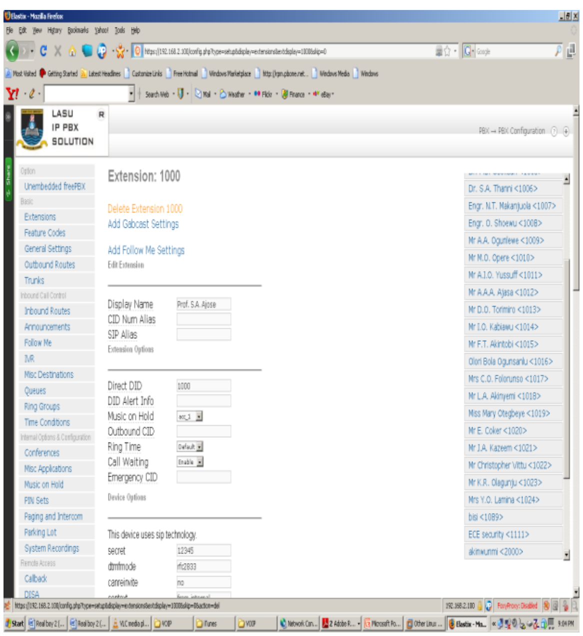

Figure 11: LASU IP PBX GUI Web Interface Page.

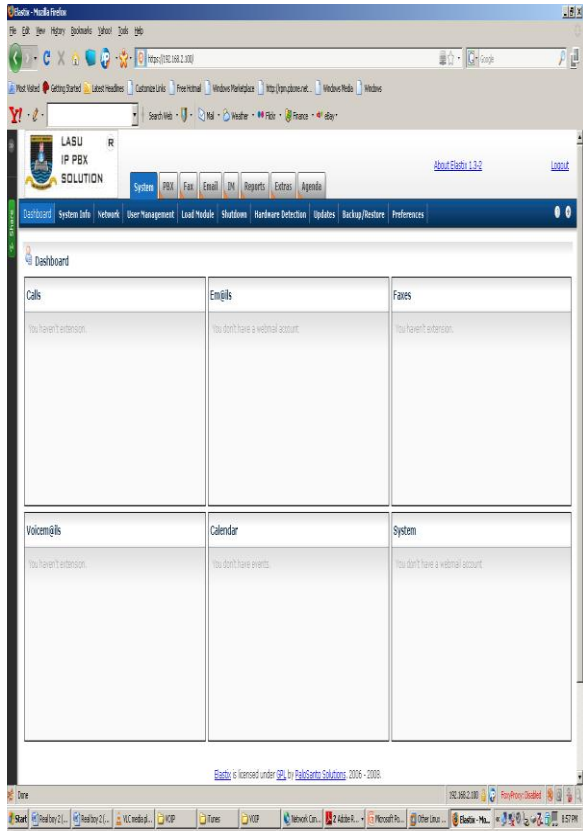

Figure 12: PBX Extension Configuration Page Showing a Configured User.

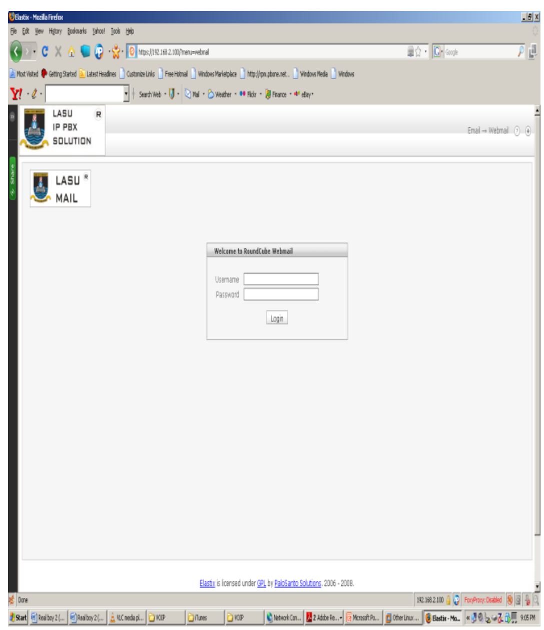

Figure 13: Webmail Login Interface Page.

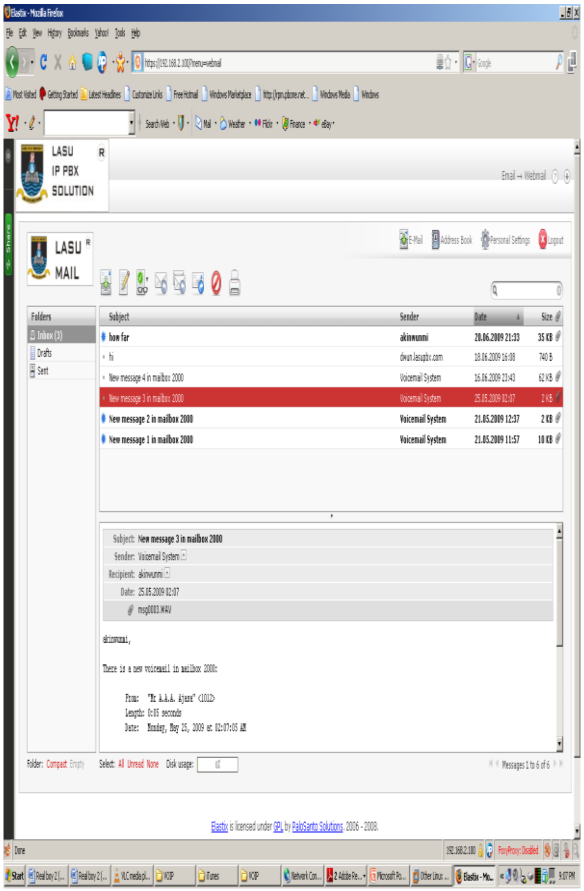

Figure 14: Webmail User Interface Page.

RECOMMENDATIONS

Like most other systems in the world, this project had some limitations and therefore has room for improvement. Part of the limitations and their respective recommendations are.

- Unlike the traditional phone system that supplies voltage along its voice data network path to power the phone. There RJ11 phone jack just need to be plugged into the network before the phones powers on or receives tone. But in the VOIP system using hard IP phones this is not done so the phones have to be powered on power packs built for the phones. With excessive wiring around in places where it's been implemented it could bring none conformity to such organizations that's why we would recommended using a power over internet switch (POE) for such implementation to power such phones through their RJ45 jacks only without the need for a power pack.
- When setting up the network infrastructure for such implementation excessive network components like using multiple switches, routers and firewalls should be minimized as they could cause delay in the voice data transmission which would affect the quality of service render (QOS).
- Power outage frequently could crash the IP PBX server so I would recommend an efficient UPS and backup supply should be made available to enable little or no down time.
- Since the server is meant to run all day, we would advise when choosing the area it's going to be installed the immediate surrounding environment should be cold to prevent hardware malfunction when it gets hot.
- We would recommend that VOIP implementation using an IP PBX solution should be used by organizations, institutions, company's, etc., particularly the Department of Electronics and Computer Engineering, LASU, Epe Campus, for their communication service as a result of its flexibility and almost limitless benefits with little limitations received using this communication technology.

CONCLUSION

If the late 1990s were characterized by irrational (or at least premature) exuberance for the innovative promise of the IT revolution, the last few years have been characterized by the opposite trend: an irrational pessimism about the long-term benefits of IT and VIOP in particular. As public attitudes settle on a more realistic middle ground and new innovations spur growth, one driver of continued progress will be VOIP. Many people have proclaimed that VOIP enables all kinds of new services that were never possible before. This is certainly true, though the hype far exceeds reality and what is practical. Even so, there are a number of new capabilities which are practical and will come forward as we continue to deploy VOIP telephony PBX systems.

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