# IMPLEMENTING VOICE OVER INTERNET PROTOCOL ON A LOCAL AREA NETWORK.

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# DEDICATION

I dedicate this project first to the Almighty God who by His grace has sustained and seen me through to this time and also to my beloved Parents, Pastor and Mrs. Adeyeye, my siblings Phebe, Peace and Pearl who have always been there for me with their love, care and support.

# DECLARATION

I, Adeyeye Philip Oluwapelumi, declare that this work was done by me and has never been presented elsewhere for the award of a degree. I also relinquish the copyright to the Federal University of Technology, Minna.

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# ACKNOWLEDGMENT

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# ABSTRACT

Voice over Internet Protocol (VoIP) is a branch of Telecommunication Engineering which carries the whole populace along because of its importance to every person in terms of connecting to one another. Over time innovations came and communication via the internet became possible. However, an organisation still had to get a land line to communicate with the outside world. How possible is it to make a packet switched network become a telephone in a Local Area Network (LAN) without disrupting the flow of data was the challenge in hand. At the end of the project, it was discovered that with some modification of the existing configurations on the routers and switches in the network, and the incorporation of a Call Manger into the network, the existing LAN will become a VoIP as well as a normal packet switching LAN. The project was tested on a simulator and the results were positive showing that voice can be transmitted without the internet in a Local Area Network. Some recommendations were made that the project be taken further to a Metropolitan Area Network and Wide Area Network and also that the school provides the necessary equipment and laboratories for testing such projects.

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# CHAPTER ONE

# INTRODUCTION

Man has always wanted to be in touch with his surrounding neighbours, especially with another man. Different means like letter writing, emails, etc has evolved over time to make this possible where direct communication can not be made. One of the newer services that are attracting the interest of campuses and home users alike is the sending of voice signals over an IP-based network such as the internet. The practice of making telephone calls over the internet has a number of different names such as packet voice, voice over packet, voice over internet, Internet telephony, and Voice over IP (VoIP). But it is like the industry has settled for the 'Voice over IP' in reference to Internet Protocol (IP), which controls the transfer of data over the internet [1]. The question now is, what is Voice over IP?

Voice over Internet Protocol (VoIP) is a general term for a family of transmission technologies for delivery of voice communications over IP networks such as the internet or other packet-switched networks. Other terms used frequently and are synonymous with VoIP are IP telephony, Internet telephony, voice over broadband (VoBB), broadband telephony and broadband phone aside other terms earlier mentioned above. VoIP refers to communications services – voice, facsimile voice-messaging applications – that are transmitted via the packet switched network or computer network rather than the Public Switched Telephone Network (PSTN) [2]. VoIP is supported by many types of networks including Local Area Networks (LAN), public, cooperate, home, and even Wireless Local Area Networks (WLAN). One should not be deceived by the word Internet.

Implanting VoIP on a Local Area Network (LAN) refers to bringing the communication services down to the local area perhaps where no Internet service is in place, as small as a peer-to-peer. A local area network connection today is the most common type of network. It is found in almost every business, cooperate offices and academic environments – and even in many home. A local area network is a communications network that interconnects a variety of *data* communication *devices* within a small geographic area and broadcasts data at high data transfer rates [1]. Devices can be computers, printers, disk drives, modems and in our case, a telephone. Most users on a LAN expect to share information, data, software and hardware.

As technology continues to evolve, the realm of voice, which was traditionally kept completely separate from data, has now begun to merge with the data network. This brings together two different worlds of people: data technicians—historically accustomed to working with routers, switches, servers, and the like—and voice technicians, historically accustomed to working with PBX systems, digital handsets, and trunk lines [3].

## 1.1 Objectives

The aims and objectives of this project include

1. To evaluate and understand the principles of IP telephony technology and also be able to demonstrate a simple Voice over Internet Protocol (VoIP).
2. To show that VoIP can implemented on a local area network and make call within a local area without the intervention of NITEL NG and the likes.
3. To show that it is possible to incorporate IP telephony into an existing data local area network without having to set up a separate LAN for the voice.

## 1.2 Significance of the study

Voice over IP is our new technology for making calls but why do we need it? This study exposes to us the possibility of making calls from our offices to another office in a campus like ours and still carry out our internet service on the same network. It removes the need to setup a new system for the purpose of telephony thereby cutting down the overhead cost for Internet services and voice services. If implemented using a wireless local area network, with a laptop, calls can be received anywhere provided there is a broadcast of the network available. With Voice over Internet Protocol, there will be little or no need for the public switched telephone network generally served by our communication network providers in the country. They can only be used as a backup in case of failure of the packet-switched network.

## 1.3 Principles of Voice over IP

The basic steps involved in originating VoIP are; the conversion of the analogue voice to digital format. The digital format is then compressed/translated into the Internet Protocol (IP) packets for the transmission over the Internet. This process is reversed at the receiving end. It is important to note that the Internet protocol is protocol which operates on the *network layer* of the OSI Model.

## 1.4 Advantages and disadvantages.

As numerous as the advantages of VoIP may be, there are also some disadvantages.

### 1.4.1 Advantages of Voice over IP

There are many benefits to the use of VoIP otherwise there wouldn't be need for VoIP, it would have just been another option for communication. It`s not so because the advantages over the existing PSTN is clear. Listed below are advantages of VoIP.

1. The first thing to note is the substantial cost saving. This is because of the following reasons:

    - ✓ Greatly reduced number of access lines or reduced cost of cabling.
    - ✓ Reduced recurring carrier charges

- ✓ Reduced access line fees and surcharges
- ✓ Reduced access line taxes
- ✓ Elimination of call feature charges.

2. Users of interactive multimedia exchanges will benefit from services that do not exist or crudely exist without any extra cost like call conferencing, call forwarding.
3. Better customer services from organisations or campuses – LAN.
4. It puts communication at the core of business thereby enabling faster decisions, revitalised business processes and new business models especially in our country where the need for entrepreneurship is being emphasised today [4].
5. It brings about more productivity which in turn leads to loyalty from customers [5].
6. VoIP is very flexible; it can be on one's desk as a normal telephone, on a computer using an IP Softphone, mobile phone, and also PDA.
7. It offers some advance services like voicemail and web surfing.
8. Management of the Internet facilities and telephony is now done by a single body, no need for separate agent from NITEL/MTN/STARCOMS to come in for maintenance services while another manages the Internet. A person trained for the maintenance of packet-switched network can as well manage the VoIP with just little training.

### 1.4.2 Disadvantages of VoIP.

One major disadvantage of VoIP is the initial cast of setting up the network. This is because the technology is still relatively new and also the cost of the gadgets involved like routers, call managers, servers, etc

Also is the need for trained staff in the Voice over IP field to manage the network.

## 1.5 Methodology

This project as the title states, 'Implementing VoIP on a LAN' will be implemented on an existing local area network in order to see the possibility of using the same network for internet and voice. This project is intended to be carried out without the Internet. This means the local area network is setup but without an internet connection and the IP telephony is incorporated into the network.

First a local area network is simulated by configuring a perfectly working network, which can transfer a packet-switched data to and fro the network. All necessary connections between the hardware are done. When the network is tested and working, the VoIP configurations are now brought into the picture. First, all equipments needed for the working of the project are connected, the phones, routers, cablings are not left out also. Next is the configurations are then testing. At the end of the project, communication from one end of the LAN to the other is expected to be possible either by a ping or a direct call.

## 1.6 Scope

This project intends to probe into the whys, whats, wheres and Hows of VoIP in an existing local area network. What has to be done to make a Voice over IP possible in the LAN? What equipments have to be included in the network to make this possible? Would there be a need for PSTN services for the project to work?

At the end of this project, information needed for VoIP to be functional on an existing LAN will be obtained. The Figure 1.1 below shows a model of the expected LAN with VoIP implemented on it. It also shows the possible connection between the network devices and the hosts.

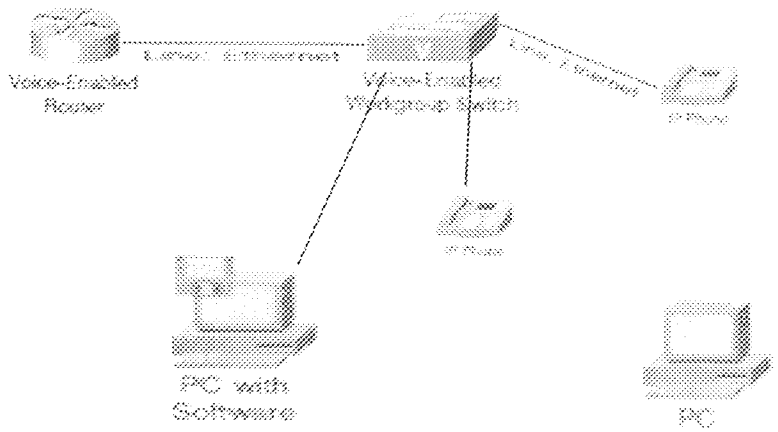


Figure 1.1 Different Types of Connection

# CHAPTER TWO

# LITERATURE REVIEW

## 2.1 History of VoIP in Brief.

There are two fundamental technologies that are necessary for the existence of VoIP. The first, and most widely used, is the telephone. The second technology is the Internet. The telephone was as direct result of the (independent) work of Alexander Gram Bell and Elisha Gray in the 1870s [7]. The first regular telephone exchange was established in New Haven in 1878. Early telephones were leased in pairs to subscribers. The subscribers were required to put up their own line to connect with another. In 1889, Almon B. Strowger, a Kansas City undertaker, invented a switch that could connect one line to any of 100 lines by using relays and sliders. This switch became known as "The Strowger Switch" and was still in use in some telephone offices well over 100 years later [8]. To make a call, the user needed to push a button on their phone the required number of times to dial the receiver's phone number. This button was replaced in 1896 with the rotary system. Philadelphia was the last city to give up the dual service (rotary and button) in 1943.[9]

About the same time the transistor was invented, a mathematician Dr. Claude Shannon published "A Mathematical Theory of Communication," which promoted the concept of communicating in binary code. Dr. Shannon's paper formed the basis of the entire digital communications revolution, from cell phones to the Internet. Fifteen years later, in 1963, AT&T used Dr. Shannon's ideas and created "TouchTone" dialling. This evolution of technology allowed calls to be switched digitally and, later, enabled all manner of automated menus and functionality that eliminated the need for human operators. In 1984 the US government broke apart AT&T – allowing home users to stop leasing their phones from AT&T and allowed them to purchase their own phones. These changes lead to a wave of new designs and functions for the home phone.

In 1968 the Internet was first developed by ARPANET (Advanced Research Projects Agency Network), founded by the U.S. Department of Defence in 1957. ARPANET was developed to provide a decentralized communications network that would not be disrupted by a potential global war

Developed in the 1970s, and in parallel to the Internet, were time-share computer networks owned by large companies who would rent out their large mainframe computers during the evening and weekends when they lay virtually unused. In 1979 CompuServe started a time-share computer service to consumers during these evening downtimes. As the PC became popular, online service companies (e.g. Prodigy and AOL) formed to provide proprietary information and email services. Subscribers would dial into the network with their telephone lines and would pay an hourly fee to receive the services offered. [10]

In 1989, Tim Berners-Lee and a group of researchers at CERN (an international scientific organization based in Geneva, Switzerland) created hypertext transfer protocol (HTTP) and a text format code called hypertext markup language. They also invented a universal resource identifier (later universal resource locator, or URL) to identify document locations. These inventions formed the foundation of the World Wide Web.

Although the telephone and Internet were vital to the existence of VoIP, there is another technology that is closely related, and just as important. In 1972 Dr. Vint Cerf was the man who invented Transmission Control Protocol / Internet Protocol (TCP/IP) – the technical protocol that defines the form of network data packets and how they travel to their destinations.

Now that the groundwork has been documented, we can examine the short brief of VoIP. From most accounts, VoIP started in February of 1995 by a small company in Israel called Vocaltec, Inc. Their product, *InternetPhone*, allowed one user to call another user via their computers, a microphone and a set of speakers. Additionally, this application/product only worked if both the caller and the receiver had the same software setup. By 1998 some entrepreneurs started to market PC-to-phone and phone-to-phone VoIP solutions. The phone calls were marketed as "Free" nation-wide long distance calls. When the caller would start the call he/she had to listen to advertisements before the call was connected. Another development in 1998 was the hardware's foray into the market. There were three IP Switch manufactures that introduced VoIP switching software as a standard in their routing equipment. By the end of 1998 VoIP calls had yet to total 1% of all voice calls. By 2000, VoIP calls accounted for 3% and by 2003 that number had jumped up to 25%. [9]

## 2.2 Theoretical Background of VoIP

Before digital networking took off, everyone had to use the one and only *POTS*, which stands for *plain old telephone service*. POTS runs over a network called the *PSTN*, or *public switched telephone network*. These POTS telephone systems use the tried-and-true method of telephone service known as *circuit-switched*. Voice over IP represents a significant change from the traditional way that telephone calls have been handled until recently. Even so, the genesis of VoIP is rooted in the history of networks, specifically, the history of the circuit-switched phone network.

The roots of VoIP go all the way back to the 1870s. In 1879, Alexander Graham Bell forgot his Internet password and, knowing that his assistant had stashed it away, uttered the famous words "Watson! Are you there?" He never got on the Internet, but he did prove that the human voice could be carried electronically over a pair of wires. He also demonstrated that the endpoints for these wires had to be connected to the right equipment —hardware that he invented. Mr. Bell's inventions ushered in an age of communication that made the world much smaller than it had ever been before.

When Mr. Bell invented the telephone and thereby gave birth to the telephone network, VoIP was not even a consideration. As a matter of fact, the idea of a network was not yet a consideration either. Other inventions would be required before VoIP could become a reality. The first telephone equipment was analogue. Historians and technicians alike have labelled the first phone service *POTS*, or *plain old telephone service*. VoIP won't function very well over a POTS system; it requires a digital network.

Digital networking for telephones was invented in the 1920s, but the first digital networks would not leave the laboratory until much later, in 1964. Today, most phone companies have updated their equipment to include digital service. Over time, the POTS network gave way to the *PSTN, or public switched telephone network.*

Although it occurred in what seems like the ancient past, Alexander Graham Bell's work is important in understanding VoIP. The POTS network that began with his invention has grown into the largest circuit-switched network in the world. It has also become an expensive network, with individuals and companies spending hundreds of billions of dollars each year for communication services.

VoIP, which was developed in 1995, is gradually replacing the PSTN. Some view the PSTN as the antithesis of VoIP, but it still remains the standard of quality by which VoIP is measured. For instance, people often ask whether VoIP provides voice quality as good as what is delivered through the PSTN. Most of the factors used to evaluate the quality of VoIP are based in some way on the PSTN, so understanding a bit about the older networks is important [5].

### 2.2.1 Analogue Telephone Circuits

As mentioned, phone technology originally was analogue, from start to finish. *Analogue modulation* is the technique used to convert sounds (such as our voice) into an electromagnetic form. The analogue circuitry of the POTS telephone transmitter converts the voice patterns coming from the caller's mouth into continuous electromagnetic signal patterns. These patterns are carried on a telephone line circuit, sometimes called a *trunk line,* where they are carried to the terminating end of the circuit. There, analogue circuitry converts the signal back into audible sounds so they can be understood by humans.

A good basic illustration of a POTS circuit can be found in an old primary school science experiment where we are told to make a hole in two tin cans, connect them with a long string and speak into it. The other person at the end hears even a whisper.

This simplistic experiment taught the basics of the POTS network: Sound was converted to an analogue signal (vibrations) that was carried over the taut string to the receiving can. The string, in turn, vibrated the can and converted the analogue signal back into audible sounds [5].

### 2.2.2 Telephony Goes Digital

Scientists, never content with two tin cans and a string, looked for different ways to transmit sounds over long distances. The pioneering work of Harry Nyquist in the 1920s gave us the basics of sampling theorem. In the 1940s, Claude Shannon would mathematically prove Nyquist's sampling theorem. Their work is the foundation for what we now call digital networking. Basically, they proved that you could take the analogue signals of any POTS call and convert them to digital form. This meant that POTS calls could originate in analogue form, be converted to digital form, and be transmitted on the PSTN using the now familiar ones and zeroes of computers. Digital networking had arrived, setting the stage for the beginning of VoIP.

The work of Nyquist and Shannon led to many telephone and computer network inventions. For example, Nyquist is credited with the patent that led to the first coder-decoder, or *codec*, device. Codecs can come in many sizes and shapes and are often found in the electronic circuitry of large networking devices. Codecs basically convert analogue signals to digital form and vice versa. Nyquist's work led to the design of many other networking devices such as dial-up modems, high-speed broadband modems, IP routers, and VoIP gateway servers. The ability to convert analogue signals to digital form also led to the development of several types of computer networks.

From the early 1960s to the present day, several types of digital networks, including fibre-optic-based networks and wireless networks, have emerged in support of computers and telephone systems. Today's digital networks, regardless of the form they take, are capable of supporting VoIP telephony [5].

## 2.3 Analogue and Digital Signalling

Everything one hears, including human speech, is in analogue form. Until several decades ago, the telephony network was based on an analogue infrastructure as well. Although analogue communication is ideal for human interaction, it is neither robust nor efficient at recovering from line noise. (Line noise is normally caused by the introduction of static into a voice network.) In the early telephony network, analogue transmission was passed through amplifiers to boost the signal. But, this practice amplified not just the voice, but the line noise as well. This line noise resulted in an often unusable connection.

Analogue communication is a mix of time and amplitude. Figure 2.1, which takes a high-level view of an analogue waveform, shows what voice looks like through an oscilloscope. The farther a person lives from the end office switch (which provides the physical cable to home users), an amplifier might be required to boost the analogue transmission (voice). Analogue signals that receive line noise can distort the analogue waveform and cause garbled reception. This is more obvious to the listener if many amplifiers are located between the home and the end office switch. Figure 2.2 shows that an amplifier does not clean the signal as it amplifies, but simply amplifies the distorted signal. This process of going through several amplifiers with one voice signal is called *accumulated noise* [6].

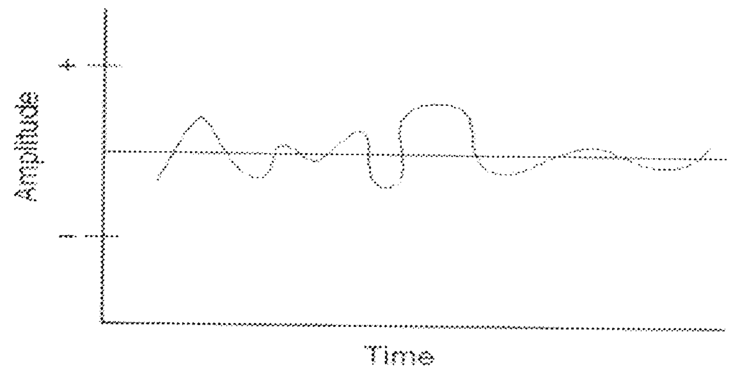


Figure 2.1 Analogue Waveform

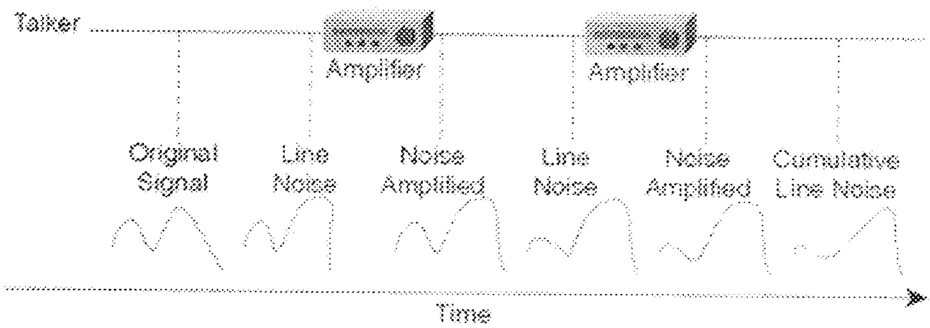


Figure 2.2 Analogue Line Distortion

In digital networks, line noise is less of an issue because repeaters not only amplify the signal, but clean it to its original condition. This is possible with digital communication because such communication is based on 1s and 0s. So, as shown in

Figure 2.3, the repeater (a digital amplifier) only has to decide whether to regenerate a 1 or a 0.

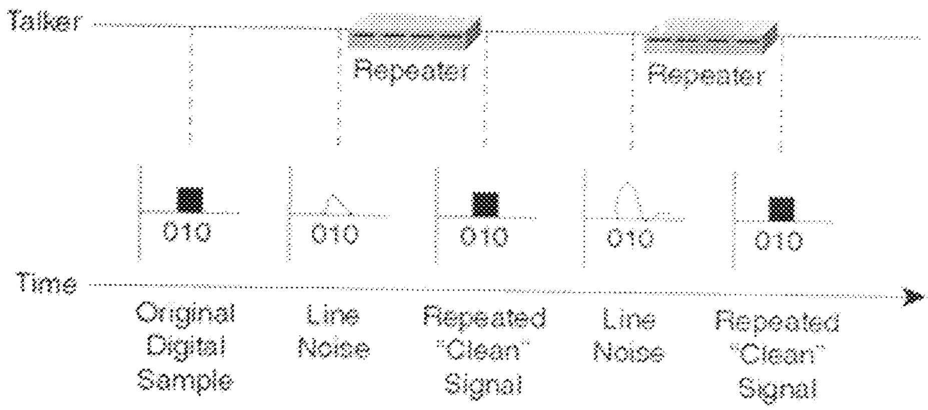


Figure 2.3 Digital Line Distortion

Therefore, when signals are repeated, a clean sound is maintained. When the benefits of this digital representation became evident, the telephony network migrated to pulse code modulation (PCM).

## 2.4 Local Loops, Trunks, and Interswitch Communication

The telephone infrastructure starts with a simple pair of copper wires running to your home. This physical cabling is known as a *local loop*. The local loop physically connects your home telephone to the central office switch (also known as a Class 5 switch or end office switch). The communication path between the central office switch and your home is known as the phone line, and it normally runs over the local loop. The communication path between several central office switches is known as a trunk [6].

## 2.5 VoIP Protocols

As of this writing, a number of voice protocols are in use. According to [2] Voice over IP has been implemented in various ways using both proprietary and open protocols and standards. The most common protocols in use are:

1. H.323
2. IP Multimedia Subsystem (IMS)
3. Session Initiation Protocol (SIP)
4. Real-time Transport Protocol (RTP)

## 2.6 Integration of Voice and Data Networks

Many users have the Data network. The problem has always been setting up a telephone system aside the existing data network. Laying of cables could take several months, after the laying of the cables comes the setting up of the system, all these could take a long time. With Voice over IP now, the existing network will be integrated with voice. There will only be need for purchase of the voice equipment to be added. The voice will be brought to the desktop of every user in the campus who needs it. Many proprietors like Cisco, Nortel, Avaya, Skype have their equipments for such purposes although some of them have their limitations. But the OSI reference model was designed to create standard methods of connecting and communicating across the data network. Cisco is the world's leading internetwork organisation. Figure 2.4 [3] below shows the breakdown of this equipments need for Voice over IP integration into the data network.

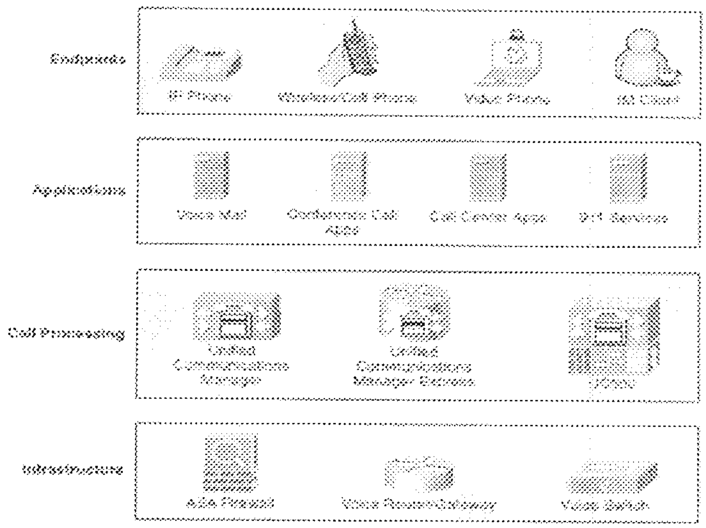

Figure 2.4 Voice Communication Layers

### 2.6.1 Infrastructure Layer

The infrastructure layer represents the devices that build the infrastructure of the data network. These are devices such as routers, switches, and voice gateways. The infrastructure layer is logically located at the bottom of the model because it represents the foundation that supports the voice network.

When it comes to the infrastructure layer, redundancy and Quality of Service (QoS) are key. The uptime of traditional Private Branch eXchange (PBX) systems is 99.999 percent (approximately 5 minutes of downtime per year). In order to achieve this same level of performance in a converged network, the network infrastructure should be tuned for lightning-fast failover should any device or physical link fail. In the same sense, QoS should carry the voice traffic from source to destination using priority bandwidth that is untouched by data traffic. Voice phone calls should never compete for bandwidth from any data application.

### 2.6.2 Call Processing Layer

The call processing layer is responsible for just that: processing calls and all the functionality that goes with them. When a user picks up their phone handset, the call processing layer gets involved to generate a dial tone. Each digit that is dialled is then analyzed and processed. The ring signal is then generated at the remote device and the call is connected. As the users converse, if one of them presses the Transfer button, for example, the call processing layer steps in to process this request. Just about any time someone touches a phone, the call processing layer gets involved.

### 2.6.3 Applications Layer

At the next layer of the VoIP structure which is the Application Layer, we encounter the applications that expand the functionality of the voice network in some way. Many applications have already been developed for the VoIP solution, each of them adding its own special features to the voice network. Three of these application servers stand out as "essential applications" for many VoIP networks: Cisco Unity (voice mail), Interactive Voice Response (IVR)/Auto Attendant, and Unified Contact Centre

**Cisco Unity Products:** Cisco has designed the Cisco Unity product line to encompass everything dealing with messaging. Whereas traditional phone systems are geared to deliver messages to telephone handsets, Cisco Unity allows you to deliver messages to a variety of clients. This allows VoIP network users to unify (thus the name) all messaging into a single point of access. For example, fax messages, voice mail, and e-mail can all be delivered to a single inbox [3]. Other platforms have their own equivalent systems for messaging and voicemail but they all carry out the same function of message unification.

**Interactive Voice Response/Auto Attendant:** IVR provides prompt and collect features, meaning it can play a recorded message to a user and request that the user press a key in response. These types of systems can be used to provide a variety of information to callers. For example, our school could use an IVR system to provide information for callers through an automated system. The most popular use of IVR is an auto-attendant application. An auto attendant can allow callers to direct themselves to the correct person or department in a campus or organisation without requiring a dedicated receptionist.

**Unified Contact Centre:** The Contact Centre software runs on a dedicated server and provides an automatic call distributor (ACD) service, which distributes calls to different groups in the organization, in addition to IVR capabilities. In addition, the Contact Centre uses computer telephony integration (CTI) that can automatically pop up a window on the agent's screen with information about a caller (based on the caller ID), which could include contact information and previous case numbers. In staying true to the VoIP network integration vision, the Contact Centre software also supports chat and web collaboration and e-mail integration. This allows what used to be solely a voice call centre to become a call centre of many communications types. The Unified Contact Centre product line is often called a "call centre in a box" because it provides robust call centre features in a single-server platform.

### 2.6.4 Endpoints Layer

The final layer of the Cisco VoIP structure contains the endpoints of the system. This is, most likely, the only layer that the end users will interact with directly. Bearing in mind that VoIP takes audio and converts it to IP-based packets, it means that *any* device connecting to a network could potentially be a VoIP endpoint. For example, an instant messenger client or even a website could be considered a VoIP endpoint.[3]One major endpoint device is the IP phone.

Figure 2.5 IP Phone

The Figure 2.5 shows an IP phone, the IP phone is the endpoint for most users who wants to make call. There are different types of IP phone and they vary in their capabilities. It is also worth noting that the IP phone is a digital phone and they have their own unique IP Address which distinguishes them from the analogue phones used on the traditional PSTN system. Also worth knowing is that the already existing PSTN can be merged with new IP network and used for VoIP. Those analogue phones would be connected via an FXS port. This is a port that connects to any analogue device and provides dialling tone to the attached device.

The IP phone can be connected to the networks in three different ways. It can be connected directly to the switch and stand alone. It could either be powered via the switch through the Power over Ethernet (PoE) technology or using an external adapter. Using this system of connection, the phone will be assigned its own IP address and the systems on the network will have their own IP addresses.

The second method is connecting the computer system to the IP phone and the IP phone in turn is connected to the switch. With this connection, the computer system and the telephone may decide to use the same IP address and in a situation where its needed to be separated, it can also be done through the use of Virtual LAN on the networks. With a VLAN in place, the phone and the connected system will use different addresses.

Finally, the last method is the use of IP Softphone. This refers to a software version of the IP Phone. This will work just like the phone. All functions on phone can be found on the softphone and the system and the softphone uses the same IP address. Figure 1.1 shows the connections explained above.

Figure 2.6 below shows the LAN without voice capability. At the end of the project, the LAN is going to be as shown in Figure 1.1.

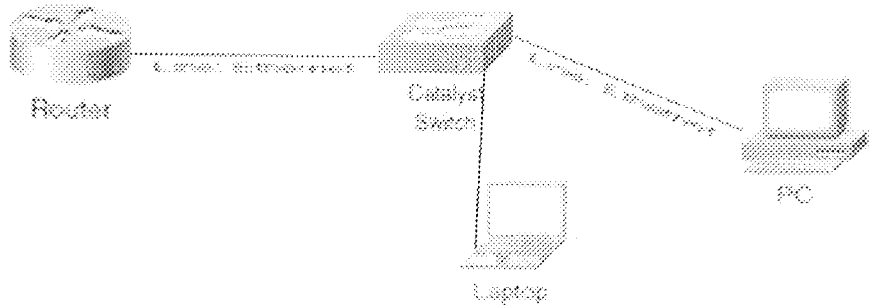


Figure 2.6 A Local Area Network

# CHAPTER THREE

# DESIGN AND IMPLEMENTATION

Figure 3.1 shown below is a model of Voice over IP. The model will be divided into two major parts. The first part is the *physical connections* involved and secondly, the *configurations*. This will be considered in details.

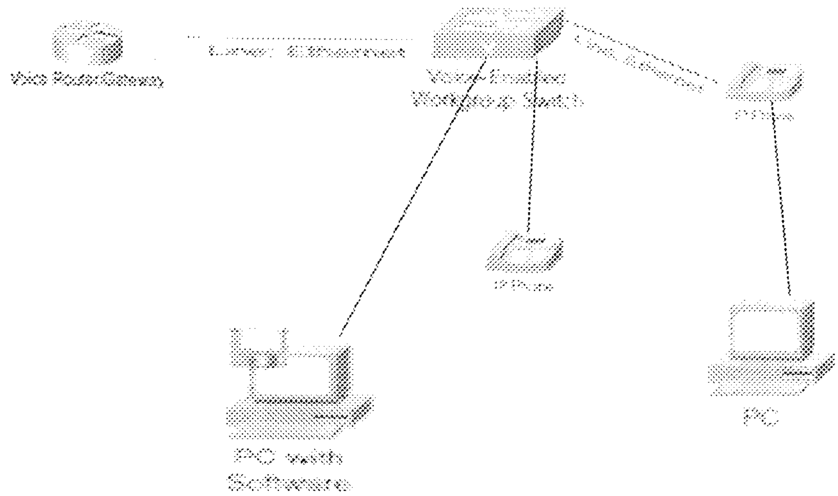


Figure 3.1 Voice over Internet Protocol

## 3.1 Physical Connection

The devices to be connected as shown in the model above are the voice-enabled router, voice-enabled switch, IP Phones and the computers.

First on the list is the power connection. The routers, switches and computer systems are powered from the wall with the normal 220V. The IP Phones are powered from the wall with an adapter or via the switch. Power of IP Phone from the switch is referred to as Power over Ethernet (PoE) or power inline. This will be further explained later.

These devices are going to be linked up with the other and they will communicate via the links put in place. There are different kinds of links/cables. The one in consideration here is the Ethernet cabling. This cable is of three types and they are based on the pin configuration:

- Straight-through cable
- Cross-over cable
- Rolled cable.

The **straight-through** cable is used to connect the following: *host to switch/hub, Router to switch/hub*. A host can be a computer, phones and other end user devices. This cable is made up of four pairs of wires which make eight pins. Four of those cables are used for data transfer while the remaining four had been left dormant until recently when its now been used as Power over Ethernet (PoE). PoE is aimed at removing the need for external powering of devices. The pins used for the configuration of the eight pins are pins 1, 2, 3 and 6 which are connected to pins 1, 2, 3, and 6 respectively. Due to the invention of PoE, pins 4, 5, 7 and 8 are now used for power. This now implies that an Ethernet cable now transmit both Data/Voice and power.

The **cross-over** cable is another type of cable used to connect any of the following *Switch to Switch, Hub to Hub, Hub to Host, Hub to Switch, Router direct to host.* Pins 1, 2, 3 and 6 connect to pins 3, 6, 1 and 2 respectively.

**Rolled cable** is used to connect the *router/switch* console serial communication port and with the HyperTerminal of the computer, the *router/switch* is then configured from the port. Pins 1, 2, 3, 4, 5, 6, 7, 8 connect to pins 8, 7, 6, 5, 4, 3, 2, and 1 respectively.

Pins 4, 5, 7, and 8 are used for power online to power device like the IP phone. The PoE supplies maximum of 15.4Watts of power. An end of the cable is connected and to a RJ45 connector and the second end is connected based on what kind of cable is needed for the devices. The table 3.1 below summarises the different types of cables and their configurations.

Table 3.1 Different Types of Ethernet PIN Configuration

| First End | Second End | | |
|---|---|---|---|
| | Straight-Through cable | Cross-over cable | Rolled cable |
| 1 | 1 | 3 | 8 |
| 2 | 2 | 6 | 7 |
| 3 | 3 | 1 | 6 |
| 4 | 4 | 4 | 5 |
| 5 | 5 | 5 | 4 |
| 6 | 6 | 2 | 3 |
| 7 | 7 | 7 | 2 |
| 8 | 8 | 8 | 1 |

When the configuration of the cable is done, the next thing is to connect the cables to the devices. The straight-through cable is connected between the router and the switch and also between the switch and the hosts. To open the configuration window, known as the console, the Rolled cable is connected from the com/serial communication port of the host/computer system and the RJ45 end to the port labelled console on the router. The next thing is to open HyperTerminal window on the computer, a name is entered for the connection and the Bits Per Seconds (BPS) is set to 9600. At this point the configuration is set to begin. The router is turned on and the decompression of the Internetwork Operating System, IOS begins.

## 3.2 Configuration

The Figure 2.6 in chapter two above, is an existing data LAN. The LAN has no voice capability in it, the voice-enabled or voice capable router is therefore brought into the picture. To configure the model shown earlier in the chapter (i.e Figure 3.1), we start with the router.

### 3.2.1 Voice-enabled or Capable Router (Call Manager)

Before talking of Voice over Internet Protocol (VoIP) on a LAN, the Internet Protocol is a protocol that resides in the network layer of the Open Systems

Interconnection reference (OSI) model. The device which operates in the network layer of the OSI is the router. The router carries all the major configurations needed for the VoIP to function. The router also internetwork devices in a local area network. A router serves as a gateway to a LAN so it carries all information coming from outside. using the IP address, it locates the right device to send the packet being routed.

Before a router can function as a call manger, the call manger software has to be loaded and installed on the IOS. The call manger contains the information needed by the IP Phones on the network, these information that assist us in making our calls- caller ID configurations. Voice mail, and other Phone related configuration like the phone button configurations, the ringing tones, speed dial, monitor lines and others depending on the quality of the IP phone in use.

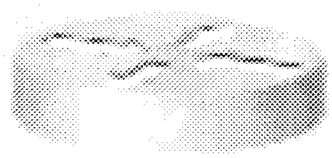

Figure 3. 2 Voice Router/Gateway

To achieve the model in Figure 3.1, the router which becomes a unified call manager will be enabled to route both voice and data network but with priority to voice. This is to avoid voice delay.

When the router is switched on. the first thing shown on the screen is:

Continue with configuration dialog? [yes/no]:

'no' is typed and entered.

It then prompts to press RETURN to continue.

The enter key is pressed.

It brings us to a command line to continue...

Router>

From this point, the following configurations are entered into the router.

Below is an extract directly from the console port of the router.

```
        --- System Configuration Dialog ---

Would you like to enter the initial configuration dialog? [yes/no]: n


Press RETURN to get started!

*Mar  1 00:00:12.735: %LINEPROTO-5-UPDOWN: Line protocol on Interface VoIP-Null0
, changed state to up
*Mar  1 00:00:12.739: %LINEPROTO-5-UPDOWN: Line protocol on Interface IPv6-mpls,
 changed state to up
*Mar  1 00:00:14.731: %LINK-5-CHANGED: Interface FastEthernet0/1, changed state
to administratively down
*Mar  1 00:00:14.735: %LINK-5-CHANGED: Interface FastEthernet0/0, changed state
to administratively down
*Mar  1 00:00:15.107: %SYS-5-RESTART: System restarted --
Cisco IOS Software, 2600 Software (C2691-ADVENTERPRISEK9_SNA-M), Version 12.4(13
b), RELEASE SOFTWARE (fc3)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2007 by Cisco Systems, Inc.
Compiled Tue 24-Apr-07 15:33 by prod_rel_team
*Mar  1 00:00:15.119: %SNMP-5-COLDSTART: SNMP agent on host Router is undergoing
 a cold start
*Mar  1 00:00:15.731: %LINEPROTO-5-UPDOWN: Line protocol on Interface FastEthernet0/1,
changed state to down
*Mar  1 00:00:15.735: %LINEPROTO-5-UPDOWN: Line protocol on
Interface FastEthernet0/0, changed state to down

Router>enable
Router#configure terminal
Enter configuration commands, one per line.  End with CNTL/Z.
Router(config)#line console 0
Router(config-line)#password cisco
Router(config-line)#login
Router(config-line)#line vty 0 4
Router(config-line)#password cisco
Router(config-line)#login
Router(config-line)#exit
Router(config)#interface fastEthernet 0/0
Router(config-if)#no ip address
Router(config-if)#exit
Router(config)#interface fastEthernet 0/0.10
Router(config-subif)#encapsulation dot1Q 10
Router(config-subif)#ip address 172.16.1.1 255.255.255.0
```

```
Router(config-subif)#description ROUTE INTERFACE FOR VOICE VLAN
Router(config-subif)#interface fastEthernet 0/0.50
Router(config-subif)#description ROUTE INTERFACE FOR DATA VLAN
Router(config-subif)#encapsulation dot1Q 50
Router(config-subif)#ip address 172.16.2.1 255.255.255.0
Router(config-subif)#exit
Router(config)#Router rip
Router(config-router)#network 172.16.0.0
Router(config-router)#exit
Router(config)#ip dhcp excluded-address 172.16.1.1 255.255.255.0
Router(config)#ip dhcp excluded-address 172.16.2.1 255.255.255.0
Router(config)#ip dhcp pool VOICE_SCOPE
Router(dhcp-config)#network 172.16.1.0 255.255.255.0
Router(dhcp-config)#default-router 172.16.1.1
Router(dhcp-config)#option 150 ip 172.16.1.1
Router(dhcp-config)#dns-server 4.2.2.2
Router(dhcp-config)#ip dhcp pool DATA_SCOPE
Router(dhcp-config)#network 172.16.2.0 255.255.255.0
Router(dhcp-config)#default-router 172.16.2.1
Router(dhcp-config)#dns-server 4.2.2.2
Router(dhcp-config)#^Z
%SYS-5-CONFIG_I: Configured from console by consoleRouter#
Router#copy run start
Destination filename [startup-config]?
Building configuration...
[OK]
```

The configurations above setup the following: the password to login to the console port of the router and the password to login to the telnet ports. It also setup the Fast Ethernet port of the router which is the link to the LAN with two sub-interfaces for the data and voice transmission. This sub-interfaces are two logical interfaces on a single physical Fast Ethernet port. This configuration also set the pool or address range to be used both by the Phone and Computer Systems connected to the LAN. This is done by creating two Virtual Local Area Network (VLAN) which were named VOICE and DATA. Also the addresses to be excluded were also configured.

This router is doubling as normal router and Call Manager. The configurations below are now added to the router. It enables the router to communicate with the IP Phones. It is assumed that this router is carrying both the Call Manager software which consists of every information the IP Phone need to operate.

```
Router#config t
Enter configuration commands, one per line.  End with CNTL/Z.
Router(config)#telephony
Router(config)#telephony-service
Router(config-telephony)#max-ephones 24
Router(config-telephony)#max-dn 48
Router(config-telephony)#load ?
  12SP      Select the firmware load file for 12SP+ and 30VIP phones
  7902      Select the firmware load file for 7902
  7905      Select the firmware load file for 7905
  7910      Select the firmware load file for Telecaster 7910 phones
  7912      Select the firmware load file for 7912
  7914      Select the firmware load file for sidecar 7914
  7920      Select the firmware load file for 7920
  7935      Select the firmware load file for 7935 Conference Station
  7936      Select the firmware load file for 7936
  7960-7940  Select the firmware load file for Telecaster 7960 & 7940 phones
  7970      Select the firmware load file for 7970
  7971      Select the firmware load file for 7971
  ATA       Select the firmware load file for ATA

Router(config-telephony)#load 7960 p00308000500
Router(config-telephony)#create cnf-files
Creating CNF files
Router(config-telephony)#exit
Router(config)#ephone?
Router(config)#ephone-dn 1
Router(config-ephone-dn)#
*Mar  1 00:09:03.787: %LINK-3-UPDOWN: Interface ephone_dsp DN 1.1, changed state
 to up
Router(config-ephone-dn)#?
Ephone DN configuration commands - configure phone lines for ephone

...

Router(config-ephone-dn)#number 1000
Router(config-ephone-dn)#exit
Router(config)#ephone-dn 2
Router(config-ephone-dn)#ephone-dn 2
*Mar  1 00:09:38.371: %LINK-3-UPDOWN: Interface ephone_dsp DN 2.1, changed state
Router(config-ephone-dn)#ephone-dn 2 dual-line
Router(config-ephone-dn)#number 1001
Router(config-ephone-dn)#EXIT
Router(config)#ephone 1
Router(config-ephone)#mac-address 0014.1c48.e71a
Router(config-ephone)#exit
Router(config)#telephony-service
Router(config-telephony)#^Z
Router#sh
*Mar  1 00:13:37.479: %SYS-5-CONFIG_I: Configured from console by consol
% Type "show ?" for a list of subcommands
Router#copy running-config startup-config
Destination filename [startup-config]?
```

```
Building configuration...
[OK]
```

These configurations are made on the Call Manager to carry the configuration for the IP Phone. The phone numbers are configured and the mac-addresses of the phones are assigned to each number. The maximum number of phones to be connected and the phone directory is also setup.

The next device to setup is the switch so as to enable the vlans.

### 3.2.2 Voice-enabled Switch

The switch operates on the second layer of the OSI model which is the Data Link Layer and it is divided into two parts which are: Media Access Control (MAC) and Logic Link Control (LLC). The switches use the MAC address to locate devices on the LAN. Whenever a frame comes into the network, it checks through it's filter table and forwards the frame to it's rightful owner. It transfers the frame to it's respective VLAN which will be created and identified as shown in the configurations below.

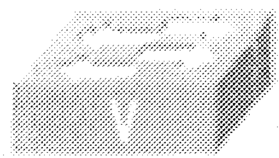

Figure 3.3 Voice Switch

This is the configuration from the console port of the switch.

```
Press RETURN to get started!

Switch>enable
Switch#configure terminal
Enter configuration commands, one per line.  End with CNTL/Z.
Switch(config)#vlan 10
Switch(config-vlan)#name VOICE
Switch(config-vlan)#vlan 50
Switch(config-vlan)#name DATA
Switch(config-vlan)#exit
```

```
Switch(config)#mls qos
Switch(config)#interface fastEthernet 0/24
Switch(config-if)#description CONNECTION TO ROUTER CALL MANAGER
Switch(config-if)#switchport mode trunk
Switch(config-if)#switchport trunk encapsulation dot1q
Switch(config-if)#switchport priority extend trust cos
Switch(config-if)#switchport voice vlan dot1p
Switch(config-if)#switchport mode access
Switch(config-if)#switchport acess vlan 50
Switch(config-if)#switchport voice vlan 10
Switch(config-if)#exit
Switch(config)#interface range fastEthernet 0/1 - 4
Switch(config-if-range)#switchport mode access
Switch(config-if-range)#switchport voice vlan 10
Switch(config-if-range)#switchport access vlan 50
Switch(config-if-range)#^Z
Switch#
%SYS-5-CONFIG_I: Configured from console by console
Switch#copy running-config startup-config
Destination filename [startup-config]?
Building configuration...
[OK]
Switch#
```

After this configuration, the host are now connected and in turn the switch begins to dynamically configure the hosts. But some little configuration still has to be made on the host – IP Phones/Softphones and Computer sytem for an effective communication.

### 3.2.3 Hosts

Hosts refer to the end user equipments to be used on the network like computer, phones, mobile phones, etc. On the computer system, the Ethernet communication also known as the Local Area Connection is configured at the network connections window. From there, the *properties* of the *Local Area Connection* are accessed. The Internet Protocol TCP/IPv4 properties are then set to *Obtain an IP address automatically*.

The phone is automatically powered via the Power-over-Ethernet cable switch. The phone automatically goes to the router to find its own configurations.

For a softphone, it uses the systems IP address. But the following must be provided for the softphone: the IP address, Port number and server name. All this being set up, the VoIP is set for service.

# CHAPTER FOUR

# TEST, RESULT AND DISCUSSION

## 4.1 Test

Each stage of the project was simulated and tested using internationally recommended simulators for such projects. At least two major simulators where used which are Cisco Packet Tracer and GNS3. The test was first carried out on the simulators before carrying them our on the real equipments.

After setting up the network design as shown earlier in this project write-up, the test was made by sending in some protocols that was able to show details of the transmission of voice and data protocols without interfering with the other. Below is a caption of the configurations as shown in the running router and switch.

On the router and switch, the configurations are viewed using the command 'show running-config'. This shows the current running configurations of the device and can be used for troubleshooting if there is no connection in the network.

**ROUTER**

```
Current configuration : 1104 bytes
!
version 12.2
no service timestamps log datetime msec
no service timestamps debug datetime msec
no service password-encryption
!
hostname Router
!
!
!
enable secret 5 $1$mERr$enz27brmFHiKOfquF.qeO.
enable password cisco
!
!
!
ip name-server 0.0.0.0
!
!
```

```
!
!
!
!
interface FastEthernet0/0
 no ip address
 duplex auto
 speed auto
!
interface FastEthernet0/0.10
 description INTERFACE VLAN LAPTOP
 encapsulation dot1Q 10
 ip address 172.16.1.1 255.255.255.0
!
interface FastEthernet0/0.50
 description INTEFACE VLAN PC
 encapsulation dot1Q 50
 ip address 172.16.2.1 255.255.255.0
!
interface FastEthernet0/1
 no ip address
 duplex auto
 speed auto
 shutdown
!
router rip
 network 172.16.0.0
!
ip classless
!
!
!
ip dhcp excluded-address 172.16.1.1 172.16.1.9
ip dhcp excluded-address 172.16.2.1 172.16.2.9
!
ip dhcp pool LAPTOP
 network 172.16.1.0 255.255.255.0
 default-router 172.16.1.1
 dns-server 4.2.2.2
ip dhcp pool PC
 network 172.16.2.0 255.255.255.0
 default-router 172.16.2.1
 dns-server 4.2.2.2
!
!
!
line con 0
 password cisco
 login
line vty 0 4
 password cisco
 login
!
```

```
!
!
end
```

**SWITCH**

```
Switch con0 is now available

Press RETURN to get started.

Switch>enable
Switch#show running-config
Building configuration...

Current configuration : 1318 bytes
!
version 12.2
no service timestamps log datetime msec
no service timestamps debug datetime msec
no service password-encryption
!
hostname Switch
!
!
interface FastEthernet0/1
 switchport voice vlan 10
 switchport access vlan 50
 switchport mode access
!
interface FastEthernet0/2
 switchport voice vlan 10
 switchport access vlan 50
 switchport mode access
!
interface FastEthernet0/3
 switchport voice vlan 10
 switchport access vlan 50
 switchport mode access
!
interface FastEthernet0/4
 switchport voice vlan 10
 switchport access vlan 50
 switchport mode access
!
interface FastEthernet0/5
!

!
!
interface FastEthernet0/22
```

```
!
interface FastEthernet0/23
!
interface FastEthernet0/24
 description CONNECTION TO ROUTER CALL MANAGER
 switchport mode access
!
interface GigabitEthernet0/1
!
interface GigabitEthernet0/2
!
interface Vlan1
 no ip address
 shutdown
!
ip classless
!
!
line con 0
line vty 0 4
 login
!
!
End
```

Each layer in the model uses Protocol Data Units (PDU) to communicate and exchange information [11]. A PDU is sent from one system to another in form of an ICMP- Internet Control Message Protocol. The message is delivered to its destination and a reply sent back to the sender of it's delivery. This shows that the connection is okay. Figure 4.1 show the model on the simulator.

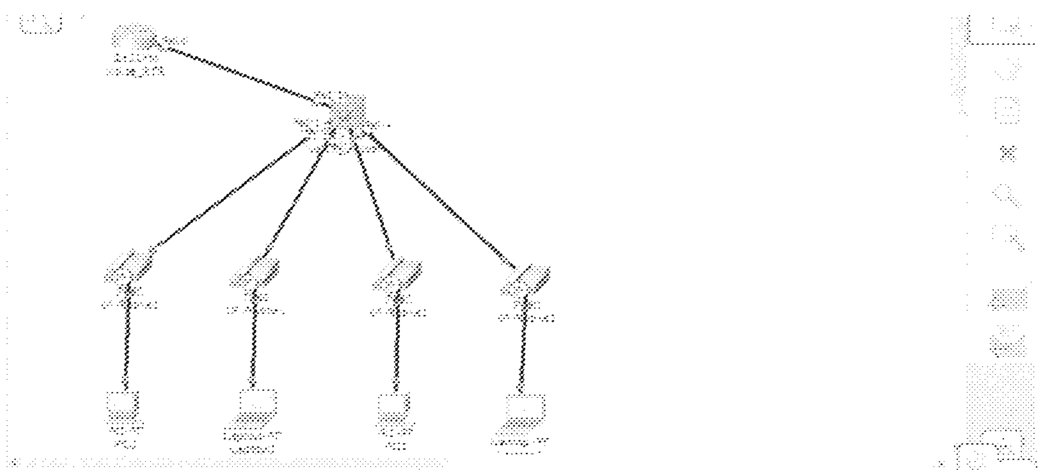

Figure 4.1 Model

The figure above shows a network model of a campus with Voice over IP implemented in the existing LAN. The next set of Figures show a test of the network.

## 4.2 Results and Discussion

The PDU packet being sent can be referred to as a ping. The test carried out gave the following result. The result is detailed based on the layers of the OSI model. The OSI layer 1, 2 and 3 are majorly operated on the LAN. The figures below show that there can be communication without interference between the voice and data signals. A PDU is sent from PC2 to Laptop 2 in this model below.

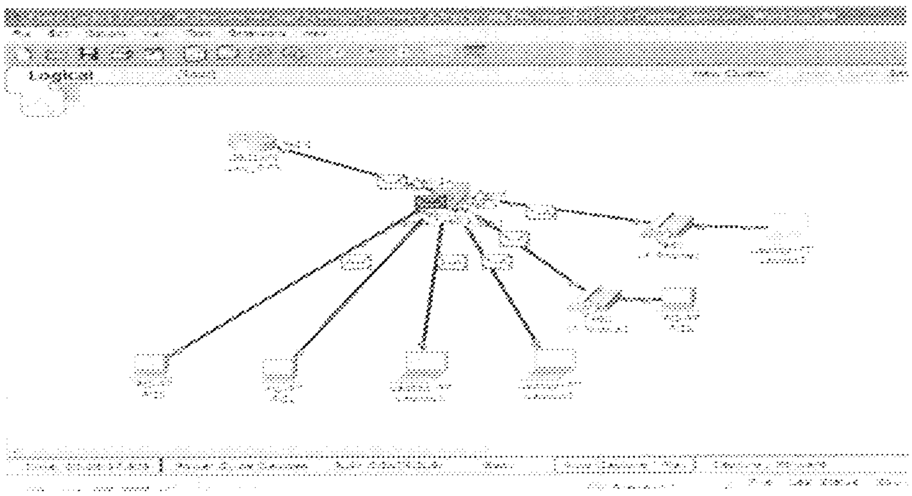

Figure 4.2 A sent request being broadcasted

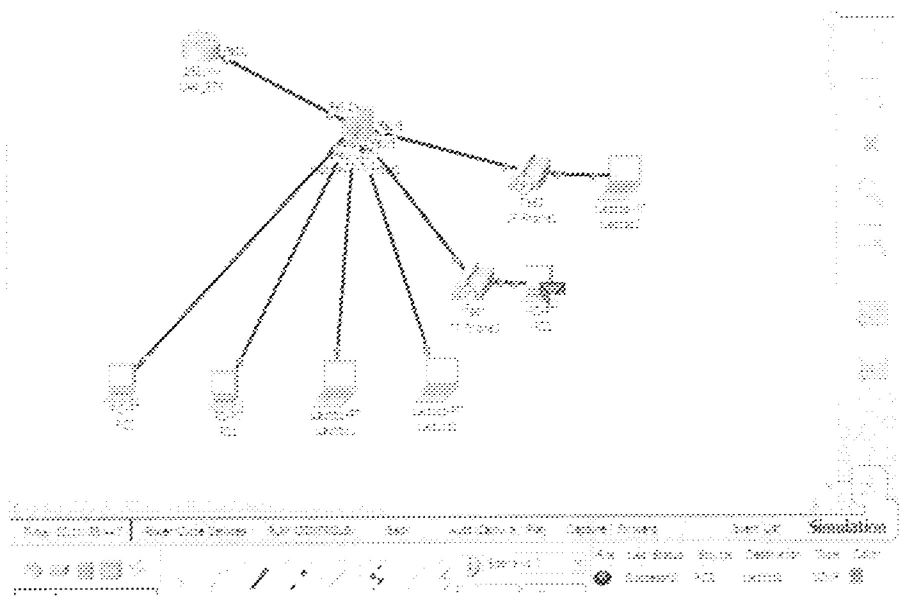

Figure 4.3 Successful Test

The result of the above test is detailed for every device and according to the layers in the OSI reference model. The router is a device for the layer 3, switch for the layer 2 and Ethernet for layer 1. This is shown where they are involved in the following result.

**At PC2**

Layer 2

1. The next-hop IP address is a unicast. The ARP process looks it up in the ARP table.

2. The next-hop IP address is not in the ARP table. The ARP process tries to send an ARP request for that IP address and buffers this packet.

Layer 3

1. The Ping process starts the next ping request.

2. The Ping process creates an ICMP Echo Request message and sends it to the lower process.

3. The source IP address is not specified. The device sets it to the port's IP address.

4. The device sets TTL in the packet header.

5. The destination IP address is in the same subnet. The device sets the next-hop to destination.

*Outbound*

Layer 2

1. The ARP process takes out this packet from the buffer and resends it.

2. The device encapsulates the PDU into an Ethernet frame.

Layer 1

1. FastEthernet sends out the frame.

**At the IP Phone 0**

*Inbound*

Layer 1

1. Port PC receives the frame.

Layer 2

1. The IP Phone sends a valid LACP/PAgP frame to the higher process.

2. The frame source MAC address was found in the MAC table of IP Phone.

3. This is a unicast frame. IP Phone looks in its MAC table for the destination MAC address.

*Outbound*

Layer 2

1. The outgoing port is an access port. IP Phone sends the frame out that port.

Layer 1

1. Switch port sends out the frame.

The IP Phone 0 has seen that the frame received on its PC port does not belong to it and sends it to the switch to locate the MAC address of the destination host.

**At Multilayer Switch**

*Inbound*

Layer 1

1. FastEthernet port linking PC2 receives the frame.

Layer 2

1. Sending a valid LACP/PAgP frame to the higher process.

2. The frame source MAC address was found in the MAC table of Multilayer Switch.

3. This is a unicast frame. Multilayer Switch looks in its MAC table for the destination MAC address.

*Outbound*

Layer 1

1. The outgoing port is an access port. Multilayer Switch sends the frame out that port.

Layer 2

1. FastEthernet port linking Laptop2 sends out the frame

**At IP Phone1**

*Inbound*

Layer 2

1. Sending a valid LACP/PAgP frame to the higher process.

2. The frame source MAC address was found in the MAC table of IP Phone.

3. This is a unicast frame. IP Phone looks in its MAC table for the destination MAC address.

Layer 1

1. Switch port of the IP Phone receives the frame.

*Outbound*

Layer 2

1. The outgoing port is an access port. IP Phone sends the frame out that port.

Layer 1

1. Port PC sends out the frame.

The Port PC of the IP Phone has sent out the frame to the Laptop.

**At Laptop 2**

Layer3

1. The packet's destination IP address matches the device's IP address or the broadcast address. The device de-encapsulates the packet.

2. The packet is an ICMP packet. The ICMP process processes it.

3. The ICMP process received an Echo Request message.

Layer 2

1. The frame's destination MAC address matches the receiving port's MAC address, the broadcast address, or a multicast address.

2. The device decapsulates the PDU from the Ethernet frame.

Layer1

1. FastEthernet receives the frame.

At this stage, the request gets to Laptop 2 and it in turn sends its reply which the PC1 also receives on its FastEthernet port.

A success in this ping shows that a data Packet will go through to the system even though an IP Phone is connected and a voice packet will just as well go to the Phone. This ping represents a call for a voice network and a data sent for Data network.

## 4.3 Limitations

In this project, some limitations were encountered. They are as follows:

- The VoIP is Call Manager dependent, if the call manger is down, placing of calls will not be possible.
- Packet-switched network can be unreliable compared to the traditional PSTN hence the need for backup.
- The voice-enable router in this project doubles as a Call Manager this also makes the VoIP network prone to external attacks.
- There was also the problem of insufficient time to thoroughly execute the project.

# CHAPTER FIVE

# CONCLUSION AND RECOMMENDATIONS

## 5.1 Conclusion

In conclusion, this project which involves the implementation of Voice over Internet Protocol on a Local Area Network was successfully implemented. This project is intended to be an exposure to VoIP and to show the possibility of VoIP being implemented on an existing packet switched network which was made for data initially. The project also was intended to implement Voice over IP in a cost effective way and this was achieved by making the call manager to double as a router linking to the outside network. With the proper implementation, their will be no need for intercom and a dedicated Public Switch Telephone Network (PSTN) once the internet LAN is existing.

This project can be adopted for our immediate campuses, that is the Federal University of Technology, Gidan Kwano and Bosso campuses which are already connected to the internet, and also in other organisation like banks and other co-operate offices and companies.

## 5.3 Recommendations

I would recommend that:

1. The department provides networking equipment like routers, switches, Call Managers and IP Phones for the implementation of internetwork projects like this.
2. This project should be implemented on a Metropolitan Area Network (MAN) and subsequently, a Wide Area Network (WAN).

3. The project should be attempted at linking up two or more Local Area Networks (LANs) for instance, the two campuses of our university being linked up to a single VoIP.

# APPENDIX

**802.3af Power over Ethernet (PoE)** Industry-standard method of supplying power over an Ethernet cable to attached devices.

**Call Manager**: In the Cisco Call Manager, profiles are set up for each individual phone based upon the static Media Access Control (MAC) address of the IP phone.

**Ethernet:** The oldest and most popular protocol used for establishing data networks. Ethernet is used in more than 98 percent of corporate America for LAN networking. Ethernet is increasingly being used as a MAN backbone standard. The fundamentals of Ethernet are modified slightly to support WiFi and WiMax, popular forms of wireless Ethernet.

**Foreign Exchange Station (FXS)** Analogue interface type that connects to a legacy, analogue device (station). FXS ports provide dial tone to the attached device.

**Foreign Exchange Station (FXS) ports** Analogue interfaces that allow you to connect a legacy analogue telephony device to a VoIP network.

**H.323** Protocol suite created by the ITU-T to allow multimedia communication over network-based environments.

**ICMP** Internet Control Message Protocol: Documented in RFC 792, it is a Network layer Internet protocol for the purpose of reporting errors and providing information pertinent to IP packet procedures.

**IEEE 802.3:** The standard defining early forms of the Ethernet networking protocol.

**IEEE:** Institute for Electrical and Electronic Engineers. The main standards certifying body for protocols such as Ethernet (IEEE 802.3), WiFi (IEEE 802.11), and WiMax (IEEE 802.16).

**IP address:** An address comprised of four numbers, each ranging from 0 to 255, and normally expressed with each number separated by a period (such as 192.168.2.100). IP addresses are used to route network traffic from sender to receiver. The IP address is a major component field of a VoIP packet and is used to map the VoIP telephone call to a specific telephone number. In a VoIP telephony call, both source and destination (caller and receiver) addresses are used to establish and maintain the VoIP call.

**IP soft phone:** Software that enables a computer to function as a VoIP telephone, including an on-screen dialling pad for point-and-click dialling.

**IP telephony:** IPT. A technology that allows traditional voice calls to be carried as data over a local area network. IPT is technically VoIP on a LAN (and VoIP is IPT outside the LAN).

**IP:** Internet Protocol. One of two major protocols used in the TCP/IP family of protocols. The IP protocol is one of the protocols used to implement the Internet.

**LAN:** Local area network. A data network limited to a small geographic area. A LAN can be as small as a couple of devices connected on the same network or as large as a campuswide installation with numerous buildings and thousands of addressable devices on the same network.

**MAC address:** An address that uniquely identifies a network device. The MAC address is typically represented in hexadecimal notation, as in 00-04-23-58-90-6E.

**MAC frame:** The format specified for encapsulating bit signals in the IEEE 802.3, IEEE 802.11, and IEEE 802.16 standards.

**MAC:** (1) In network terminology, an acronym for media access control. The lower sublayer in the Data Link layer, it is responsible for hardware addressing, media access, and error detection of frames. The part of the network interface that controls physical access to the LAN through the MAC address. (2) In telephone system administration terminology, an acronym for moves, adds, and changes. MAC describes the most common type of maintenance necessary in traditional telephone systems.

**MAN:** Metropolitan area network. A type of network designed to cover a large geographical area, such as a city.

**OSI** Open Systems Interconnection: International standardization program designed by ISO and ITU-T for the development of data networking standards that make multivendor equipment interoperability a reality.

**OSI reference model** Open Systems Interconnection reference model: A conceptual model defined by the International Organization for Standardization (ISO), describing how any combination of devices can be connected for the purpose of communication. The OSI model divides the task into seven functional layers, forming a hierarchy with the applications at the top and the physical medium at the bottom, and it defines the functions each layer must provide.

**Packet-switched:** Packet-switched networks such as a VoIP network use the addressing information contained in the packet to determine the route the packet takes to its destination.

**PDU** Protocol Data Unit: The processes at each layer of the OSI model. PDUs at the Transport layer are called segments; PDUs at the Network layer are called packets or datagrams; and PDUs at the Data Link layer are called frames. The Physical layer uses bits.

**Ping** Packet Internet Groper: A Unix-based Internet diagnostic tool consisting of a message sent to test the accessibility of a particular device on the IP network. The term`s acronym reflects the underlying metaphor of submarine sonar. Just as the sonar operator sends out a signal and waits to hear it echo ("ping") back from a submerged object, the network user can ping another node on the network and wait to see if it responds.

**POTS line:** The physical line that supports plain old telephone service (POTS)

**POTS telephone:** The telephone that supports plain old telephone service (POTS).

**POTS:** Plain old telephone service. The most basic form of circuit-switched telephone service.

**Private Branch eXchange (PBX)** A system that allows a company to run an internal, private voice network; PBX systems are usually used in larger companies and provide unique extensions to all devices.

**PSTN:** Public switched telephone network. The oldest and largest communications network in the world.

**Pulse-Amplitude Modulation (PAM)** The process of sampling an analogue waveform many times to determine numeric electric amplitude values for digital conversion; PAM is typically combined with pulse-code modulation (PCM).

**Pulse-Code Modulation (PCM)** The process of converting pulse-amplitude modulation (PAM) values into binary number equivalents that voice equipment can transmit over digital circuits.

**QoS** Quality of service: A set of metrics used to measure the quality of transmission and service availability of any given transmission system.

**RTP:** Real-time transport protocol. Operates at the application layer of the TCP/IP model to provide end-to-end network transport functions for digital voice signals encapsulated in the VoIP packet.

**SIP:** Session initiation protocol. An interoperable protocol in the TCP/IP family of protocols. SIP uses text formatting to set up and maintain communication sessions with various endpoints. These endpoints can include cell phones, desk phones, PC clients, and PDAs. SIP permits these various endpoints to operate as a single system.

**Softphone:** Software that enables a computer to function as a VoIP telephone, including an on-screen dialling pad for point-and-click dialling.

**Spanning Tree Protocol (STP)** A method designed to prevent loops in switched networks due to redundant inter-switch connections.

**TCP/IP:** Transmission control protocol/Internet protocol. The family of interoperable protocols consisting of more than one-hundred-twenty protocols, each of which performs one or more services to support various network applications. The early developers of the Internet agreed upon the name TCP/IP because, at the time, TCP and IP were considered the two most important protocols for any network connection.

**TCP:** Transmission control protocol. One of two major protocols used in the TCP/IP family of protocols. In VoIP telephony and videoconferencing calls, the TCP protocol is replaced by its sister protocol, UDP.

**Virtual LAN (VLAN)**: A group of devices on one or more logically segmented LANs (configured by use of management software), enabling devices to communicate as if attached to the same physical medium, when they are actually located on numerous different LAN segments. VLANs are based on logical instead of physical connections and thus are tremendously flexible. It's a configuration used to break a switch into multiple broadcast domains and IP subnets.

**VoIP:** Voice over Internet protocol. A network service that supports carrying telephone calls over packetized networks. VoIP reduces substantially or eliminates the need for a separate, circuit-switched telephone network to carry telephone calls.

**WAN:** Wide area network. A larger network that connects two or more LANs using dedicated transport lines.

**WLAN:** Wireless Local Area Network. A network configuration that uses radio waves for intercommunication.

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