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Analysis of the Effectiveness of Voice over Internet Protocol on a LAN/MAN/WAN Network

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ABSTRACT

Voice over Internet Protocol (VoIP) is an advanced telecommunication technology that enables voice communication over Internet protocol hence the name "VoIP". VoIP can be implemented at a cheaper cost yet delivers great Quality of service (QOS). This paper will address and analyze the performance aspect of VoIP on digital communication platform. The main focus will be on how VoIP can handle Quality of service, packet loss, mean opinion score and jitter. Furthermore, will investigate the impact VoIP might have in terms of delay and distortions if traffic load was to be increased on the network. This will be achieved through designing a topology with extra models that will be added on the network to make it more realistic and that quality of service for the entire network can be evaluated then measure the impact. In addition, so much innovations surrounding VoIP Technology is taking place. This paper will further explore software technology that can allow VoIP Technology to be configured and implemented on the Desktop computers even mobile devices like Laptops, iPads, Tablets, smart phones only to mention a few. The software based technology will enable reduction in costs that come with procurement of expensive VoIP handset phones and ultimately cost associated with UTP cable LAN setup. The software once installed and configured will make it possible to achieve VoIP communication via a computer system within the defined LAN/MAN/WAN network. This paper again is aimed at exploring different innovations supporting and promoting VoIP Technology to enhance internal and external communication via a PBX.

Opnet IT guru will be used to simulate the VoIP performance on different levels of the Network.

Keywords: VoIP, Communication, Network, Cost Reduction, Quality of Service, Software & Computer Systems.

Acronyms used in this Paper

VoIP: Voice over Internet protocol

IP: Internet Protocol

TCP: Transport Control Protocol

UDP: User Datagram Protocol

RTP: Real Time Protocol

RTCP: Real Time Control Protocol

ETE: End-to-End Delay

MOS: Mean Opinion Score

PSTN: Public Switched Telephone Network

LAN: Local Area Network

WAN: Wide Area Network

MAN: Metropolitan Area Network

POE: Power over Ethernet

PBX: Private branch exchange

1.0 Introduction

1.1 Overview

VoIP communication is one of the latest technologies being implemented by many organizations today be it on the small, medium or corporate level. VoIP technology comes with the ability to send voice message over IP (Internet Protocol) based data networks with the assurance of getting quality of service (Qos). The VoIP concept is achieved through incorporating voice traffic in IP datagram using the same medium that will allow exchanging both voice and data. Furthermore, this will reduce the cost of running telephone cables separately while enjoying the cost benefits of using VoIP.

As much as VoIP implementation depends on TCP/IP to achieve communication, protocols for signaling and sending conversation data in the IP medium are incorporated to complete the implementation.

Real time protocol / Real time control protocol is used to send conversation data in the IP medium through UDP. To control voice packet and voice quality, real time protocol (RTP) is responsible while real time control protocol (RTCP) takes care of the message exchange between the sessions then takes note of the delay, lost RTP packet just to mention a few.

To monitor the call progress, call release and call setup, signaling protocols are required. Below are some of available protocols that can be used:

- International Telecom Union (ITU-T): H.323
- Internet Engineering Task Force (IETF): SIP and S/MGCP
- MEGACO/H.248 jointly developed by ITU and IETF

An acceptable level of Mean Opinion Score (MOS) for VoIP which is between 4 – 4.5 was put into consideration. Latency should also not exceed 150ms, while jitter should be within a tolerable level of approximately less than 20ms as recommended by the International Telecommunications Union (ITU).

1.2 VoIP and PSTN Comparison

As VoIP technology keep on advancing many people and organizations have appreciated the technology especially that recently so much innovations have been explored that VoIP setup has been extended even on different mobile devices running different operating systems provided they are configured within a defined network. Configurations are quite simple on the end user side that one does not necessarily need to be an ICT expert. With the phones that are dual-mode, users are able to switch between the cellular service and VoIP services on an internal wireless network. Due to this facility, carrying a cell phone and a Desktop phone to switch calls is a night mere now. In this regard, VoIP technology is able to provide mobility when configured on a VoIP mobile device within the Wi-Fi coverage. Unlike traditional PSTN phones that are labour intensive when laying permanent copper cables for every point.

VoIP is not only limited to internet but can also be implemented on the Ethernet LAN for both internal and external communication. A real life example were VoIP has been implemented on the LAN/MAN/WAN is Zambia Railway LTD, where they have even moved a step further employing VoIP GSM cellular gateways that makes external calls even cheaper. The VoIP GSM cellular gateway has provisions for SIM cards, were you can insert SIM cards of different network providers such that whenever a call is made, it will choose a SIM card for that particular network one is calling, hence avoiding cross network that can make a call expensive. ZRL has presence throughout the line of railway and VoIP technology have been implemented only in the major stations for now though still extending to smaller stations has well. CEC – Liquid is providing the Data circuit to all ZRL regional offices via fibre optic cable that reliable communication throughout regional offices situated along the line of rail can be attained. ZRL has considered communication has key to its operations, hence a redundancy link using digital microwave has also been installed as a secondary carrier in case fibre goes down due to vandalism it's prone to or accidental cuts. The incense of the redundancy link is to achieve constant communication as to meet the nature of ZRL business demands. Below is figure 1.1 illustrating VoIP implementation using Ethernet cables and Wi-Fi connection.

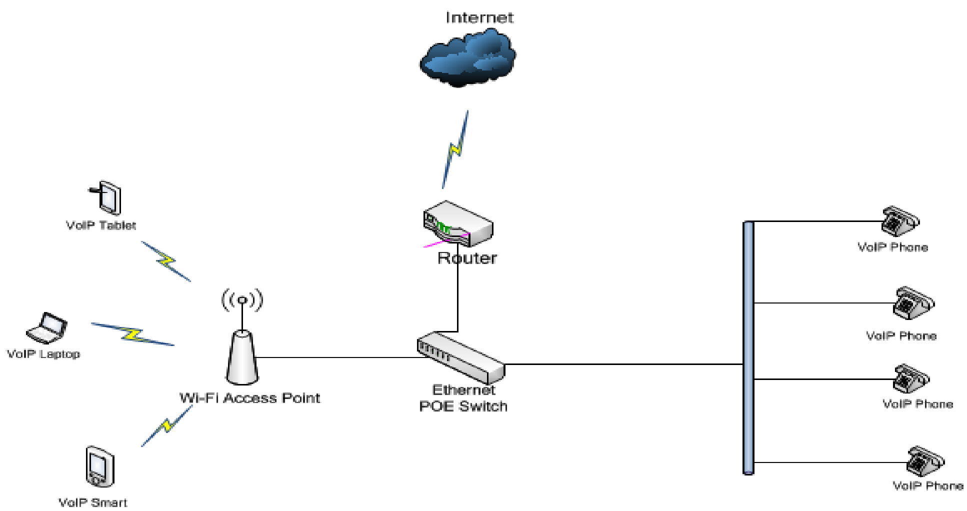

Figure 1.1 VoIP on Ethernet cable and wireless

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VoIP technology has received a positive response both in homes and corporate institutions today and it's being utilized worldwide. Figure 1.1 above illustrates a simple VoIP setup both on Wi-Fi and Ethernet cable platforms. Today different VoIP communication software's are available on the market and some can even be downloaded from the internet at no cost. E.g. Skype can be downloaded and installed on a Laptop, smart phone or a tablet. Other software's available are AIM and windows live messenger, though these purely depend on internet connectivity.

1.3 VoIP Concepts

1.3.1 Jitter

Many times VoIP suffers the problem with Jitter, which is defined as a delay in receiving a voice data packet. Packets are sent in a continuous stream at the sending side and evenly spaced apart. As a result of network congestion, configuration errors or improper queuing, this can course delay between each packet to vary instead of remaining constant. The quality and length of Ethernet cables can also course jitter. Should jitter be left uncontrolled or not minimized then will surely lead to a good number of packets being discarded by the jitter buffer in the receiving VoIP device. This means distortion in a call will be experienced, hence compromising the quality of a voice call. Therefore, in this project a network design that will reduce jitter to the acceptable minimum level will be implemented.

1.3.2 Packet Loss

This is where some of the voice packets are dropped by routers or switches on the network when congestion is experienced. With real time streaming technology like skype and live messenger, dropped and lost packets are easily noticeable. In every communication technology packet loss is experienced and acceptable but should be within manageable level. They are many factors that contribute to packet loss, among them are:

i. Corrupted packets can be dropped in transit.
ii. Poor networking infrastructure can also be the source of packet loss especially if most equipment and cables are sub-standard.
iii. Channel congestion can also lead to packet loss.

Although packet loss in a call can also assist in assessing the performance of the network, that prompt measures can be taken, either by changing the network design or reduce the number of packets to be transmitted.

1.3.3 End to End Delay

The total transit time for a packet in a data stream to reach the destination point is what is referred to as End to End Delay. Unfortunately the quality of a call is measured on how much time it has taken for the message to reach the intended recipient. Echoing has always been the major factor of end to end delay. A simulation to check if VoIP can operate within 200ms threshold was done in this project model in order to address the end to end delay problem.

1.4 Objectives of the Project

1.4.1 General Objective

The general objective of this project was to conduct several test experiments with VoIP through running different simulation scenarios using riverbed academic edition software. Thus determining the advantage VoIP has over the traditional PSTN.

1.4.2 Specific Objective

And the specific objects are as follows;

i. To determine if VoIP Network design can have a negative impact on the quality of service (Qos) delivered, knowing very well that QoS is a key factor when implementing VoIP technology. In this regard, the design should meet the standard level that will be able to give a quality of service to all the users on the network, bearing in mind of the unforeseen circumstances that would prevail at times, such as medium to high traffic loads on the network. Initially IP networks were purely designed to carry data traffic only and real time communication was not an issue by then. Now that VoIP technology has come on board with much support globally, a real time communication is required to support it.

ii. To determine how factors like; end to end delay, jitter, bandwidth, packet loss rate and utilization might influence the overall performance of VoIP.

iii. Explore how VoIP can be implemented on different computing devices using software based telephone, hence eliminating the cost that comes with procurement of expensive VoIP handsets.

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iv. To explore different levels VoIP technology can be implemented, in this study two (2) levels will be explored that is; Internal VoIP setup to achieve internal communication within the organization or home network, and secondly is an external VoIP setup that will allow calls outside internal network. As earlier mentioned, VoIP technology works on TCP/IP protocol; hence a well-defined physical network design will be ideal for VoIP implementation. Much focus should be on the type of cabling used, in this scenario UTP Cat6 cable was used for the Ethernet points.

1.5 Internal VoIP setup

This kind of setup purely depends on the LAN setup with a server computer configured to run PBX software. The common configurations on the server side may include; the numbering pattern to be used, creation of new users, assigning extension numbers to users, defining passwords for new users though can be changed at a later stage and many other functionalities. Under the live experiment I curried out, a server box running PBX software was assigned a static IP address of 192.168.0.3 being in the same range defined on the 192.168.0.0 Network. Below is the screenshot showing the sample of VoIP setup that was configured on the server side and figure 1.3 shows the screenshot of the client side configurations in the IP Phone.

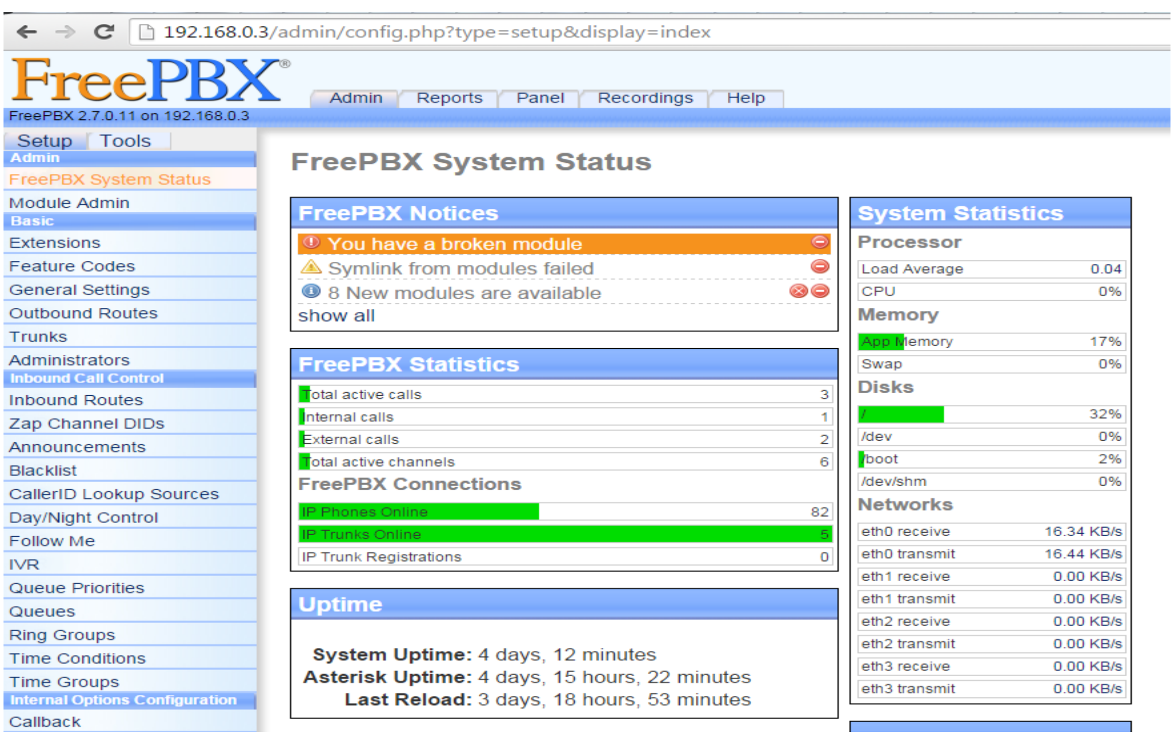

Figure 1.2: showing FreePBX System Status

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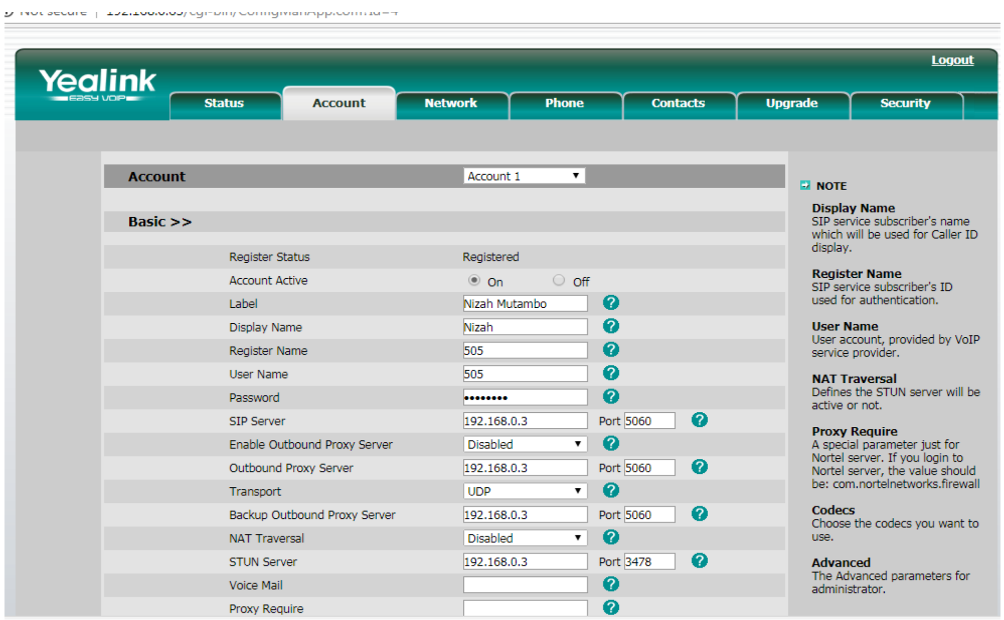

Figure 1.3: VoIP configuration in the client side (IP Phone)

Figure 1.4: VoIP softphone already configured

Figure 1.4 above shows a screenshot of a softphone software (3VX) already configured under my name. This is so powerful and easy to use on most operating systems like window, Mac, android and iOS.

3CX allows easy management of the phone calls, whether in the office using CTI and the desk phone, or on the road using a laptop / Tablet.

Unlike other PBXs, no additional licence fees are charged and because the clients are fully integrated they are easy to deploy and manage for the administrator, as well as easy to use for the employees in an institution.

2.0 Literature Review

2.1 Introduction

The literature review has been carried out to closely examine the pros and cons VoIP technology comes with. Based on the results gained, better solution of the system that is more secured, cheap and reliable is highlighted in this paper. Due to installation and maintenance costs that PSTN lines attract, VoIP technology was designed as an alternative since it has been proven to be more cost effective. In this regard, this literature review will provide the road map on how best we can transfer our knowledge and experience in coming up with desired results.

2.2 Voice over Internet Protocol (VoIP)

VoIP technology is the latest technology on the market that is able to allow voice transmission over packed switched network. Internet search defines VoIP as *a methodology and group of technologies for the delivery of voice communication and multimedia session over internet protocol (IP) networks, such as internet. Other terms commonly associated with VoIP are IP phone, Internet telephone, broadband telephone, and broadband phone service*. (En.m.wikipedia.org/wiki/Voice_over_IP). VoIP creates a platform were calls can be made from a computer to the other computer or to a mobile phone. Because of internet aspect that is associated with VoIP, other facilities are provided other than the services that a conversional PSTN landline phone is able to offer.

Since inception VoIP services have increased with time due to the low cost associated with the technology. Wiring cost aspect is eliminated because setting up VoIP will use the already existing

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Ethernet LAN for data or Wi-Fi connection can also be used for VoIP setup which is not the case with the traditional PSTN landline.

Voice over internet technology is relatively new in the communication industry. Although VoIP is the latest technology still the concept of converting the analogue singles into digital signals when sending and convert back to analogue signal at the destination. This is so because of internet being used as a medium. Packet switching is the digital mode of transmission while circuit switched is traditional analogue way of communication which is quiet costly to implement.

Voice over internet protocol works best with a high speed internet connection if quality of service is to be achieved. Voice transmission can work with DSL connection which is a broadband though it's in the process of being phased out completely and replaced with fibre optic which is noise free and can cover a longer distance without signal attenuation. Extra information is added to the voice data during the conversion from analogue to digital that internet protocol should know what the packet encloses. Many times a priority tag is added to the voice packet that makes internet protocol (IP) to identify that this is a voice packet and needs to be given first priority to data packet when transmitting. Voice packets receive a higher priority that quality of service (QoS) is not compromised.

With voice over internet protocol transmission, the speech signal is split into frames of about 20 to 30ms of length although it usually depends on the available bandwidth to transmit on. Each speech frame is processed first with the respective speech codec for compression that is when dealing with the data rate on the access and core network.

Transport control protocol (TCP) is no longer used in voice transmission but instead user datagram (UDP) is used. UDP is a real time protocol, meaning when the call is made no more waiting for acknowledgement from the receiving end for another voice packet to be sent like in TCP, hence transit time between the source and the destination is reduced. PSTN lines can only transmit at around 56kbps thus not ideal for multiple communications at the same time like with DSL which can transmit at 10mbps which is much better. If the speed is less then delay will be experienced frequently. However, fibre optic cable has come into the picture as a better option to DSL. Today many ISP's and corporate organization they are now migrating to using fibre optic cable as the currier because of its great bandwidth capacity.

However, the major drawback voice over IP has over PSTN line is that when power failure is experienced VoIP phones also go off. This challenge can be eliminated through provision of reliable backup power solution, for instance installation of the industrial power generator with capacity to handle the entire premises.

2.2.1 Technological requirements

When implementing VoIP, various aspects have to be put into consideration. Firstly, you have to determine the size of the network required, that is in terms of the number of points for the phones needed. Once the number is known then it will be far much easier to know how much capacity of bandwidth is needed to support VoIP. In addition, it has to be known weather physical IP phones will be used or computers / Smart devices through installation and configuration of the VoIP software.

For bandwidth to be properly managed, it is important that a router capable of doing intelligent routing is used. In case were the number of telephones are increased the router should be able to make a decision on the bandwidth allocation.

Zambia Railways recently did way with the tradition landlines phones that were being used for local communication and implemented voice over internet protocol phones. This has proven to be cheaper in terms of maintenance, setting up the phone and the quality of service has improved. Although for external communication a VoIP GSM also known as cellular gateway is connected and configured on the network and assigned a static IP address in the same range of the network that calling outside the company can be achieved at a minimum rate. The cellular gateway is able to accommodate SIM cards of different networks that when a caller calls any number it will pick a SIM for that particular network to go out hence reducing cross network charges.

2.2.2 Quality of Service

The most critical service when implementing voice over internet protocol (VoIP) is the quality of service (QoS). Therefore, this simply means the client should get the highest possible service from every call that is made. No time delay when calling is expected and that there should be no noise in the conversation.

Implementing VoIP on point to point radios at times might not meet the quality of service required though this highly depends on the type of radio being used. However, it's a well-known fact and proven that fibre optic link is able to deliver quality of service expected. In addition, if VoIP is implemented on fibre optic platform assurance of security, durability, reliability and continuity will always be achieved.

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Quality of service in VoIP will always again depend on the network structure starting with the LAN all the way to the WAN. It is cardinal that the network is perfectly designed and managed or else if not properly designed will have a direct impact on the voice quality. There are certain measure's that can be put into consideration to attain quality of service. Some of them are:

i. **Prioritization Techniques**

Through prioritizing voice against data that is less critical in the routers and switches, will reduce jitter and delay significantly without noticing the effect on the data traffic. Institute for electric and electronic engineering (IEEE) 802.1p/D and Diffserv are some of the available standards that can provide guidance when implementing differentiated services. Furthermore, to reserve end to end bandwidth for voice connection, resource reservation protocol (RSVP) can be used.

ii. **Capacity Management**

Mainly capacity management is incorporated on the wide-area network connections and other access links. Otherwise on the local area network setup its rear that problems concerning capacity management are experienced. However, if VoIP is implemented on the wide area network it's important to allocate the bandwidth size according to the demand of each office. That is the number of VoIP users. Capacity management can help reducing wastage of bandwidth were its not needed much, hence quality of service being attained.

iii. **Network Monitoring**

Network monitoring and management should continuously be done especially with the fact that network requirement are subject to change at any time. Challenges are faced in most cases if traffic is routed through unmanaged network.

2.2.3 Cost comparison

In every implementation of a new project it's very cardinal to compare the cost aspect between the proposed new system and the competing system. In this case some individuals and organization are still debating on which system is cheaper between PSTN and VoIP.

Well, PSTN is generally a very vast network of landlines connecting to a particular exchange which are normally big and occupies a lot of space in the exchange room. The circuits switch equipment is quite expensive and the rate of the call is relatively on the higher side. On the other side, voice over internet protocol is a high end network of packet switching that comes with additional quality control services. The initial cost might look to be on the expensive side if you are setting up from the scratch because of the network devices and accessories involved in building up a VoIP Network. But if there is a network already in place it becomes cheaper because there will be no need to do the UTP cabling and buying of other equipment needed. In short after VoIP implementation the cost of a call is relatively cheap as compared to PSTN call. Besides some companies are now manufacturing devices to support VoIP technology. One of the companies is CISCO who is now manufacturing equipment like routers and switches specifically for VoIP setup. This gesture will result in acquiring the initial devices needed at a cheaper cost.

2.2.4 Analysis

Voice over internet protocol does not require a separate network architecture if there is a packet based internet network already in existence. If VoIP is implemented on the corporate level, many limitations are there to come across. VoIP is not just about the analogue to digital and digital to analogue technology. Improvements on the higher level are required. Many are time when implementing VoIP network, IP MPLS technology is often preferred because it works between layer two (2) and layer three (3) of the ISO model and its swiftness.

2.2.5 Voice over internet protocol architecture

VoIP architecture is defined in four (4) layers as below:

I. Access layer: used for connections between the user and the service provider
II. Core Layer: this is the main network layer of the service provider that is generated to form a WAN. MPLS is a good example implemented on the core layer.
III. Connectivity Layer: this layer connects the core to other elements needed to process the basic calls. Various servers are connected with the core, which includes the billing server. Application machines are also found in the connection layer.
IV. Application Layer: various application machines with different functions are found under this layer.

2.2.5.1 How VoIP Works

Voice over internet protocol can be implemented on different levels. VoIP can be done through direct calling between computers using for example skype or through a private branch exchange (PBX). A private branch exchange basically allows exchange of calls between extensions within the organization. Setting up of an auto attendant, a call queue for a call Centre or a voicemail server can be achieved using the private branch exchange.

VoIP network originates calls from a computer or any other mobile garget with internet connectivity. The analogue to digital conversion is done when a call is made to a PSTN number while the internet medium is digital. The call is handed over to PSTN network for identification of the destination and same happens when calling a PSTN number to a VoIP number. The concept with voice is that a telephone number is mapped with the IP address such that when a call is made to a VoIP number, the IP address changes to a phone number that PSTN can handle it as an ordinary call.

2.2.5.2 Vulnerabilities

VoIP technology has its own vulnerabilities as well ranging from VoIP application to the operating system being used. However, many VoIP flaws are based on three features that every communication system security will depend on and these are confidentiality, availability and integrity. Below are some of vulnerabilities briefly discussed.

i. **Physical security**

   A major concern for every information system is the physical security. Unfortunately no monitoring too is available to distinguish flaws unless somebody physically visits the place. The fact that many exploits happen on the networking layer of the system remains challenging though some depends on the physical attach vector that exists in the VoIP physical equipment. Authentication is another feature that can be utilized to reduce chances of exploitation. Physical access to the VoIP equipment should only be for authorized users.

ii. **Quality of service**

   The principle behind quality of service is to install latency sensitive technology like VoIP that is given a higher priority over conventional data which is not based on real time platform. VoIP is only badly affected if internet medium is experiencing high congestion levels. This is usually the case if the medium is overwhelmed with other traffic and that bandwidth management is not

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applied. Although delay is always there in VoIP but difficult to detect that the user can't find any delay. However, quality of service is the only answer to reduce this kind of delay.

Low Latency queuing is the quality of service (QoS) technique that is recommended for VoIP set-up because of the combination of priority Queuing and class based weighted fair Queuing. Through using this method, the router takes all the packets meeting the requirement and forwards them to the serial port first before any other packet. This way delay in conversation is greatly reduced.

iii. Bandwidth

For VoIP to be implemented bandwidth is definitely required more especially were calling using skype as an example is involved. Therefore, direct calling using internet will require higher bandwidth capacity. Otherwise greater delay will be experienced in calls if bandwidth capacity required is not addressed.

iv. Connection Down Time

The major drawback of VoIP technology is internet dependence. This simply means if internet connectivity is cut down due to power failure or disconnected then no calls can be made. However, this can be mitigated through installation of back-up power and subscribe internet from a reliable internet service provider with a service agreement signed.

2.2.5.3 Voice compression

Compression techniques are applied for voice to be transmitted accordingly. These techniques are required to compress the high weighted packets into less weighted packets which will result in easier transmission of packets. Compression software basically encodes the voice packets from analogue to digital form, thus quality and efficiency of the encoding software resulting in attaining voice quality in conversations. A question is always being asked if compression occur, cant the data get lost on the way? The answer is no. the compression type called lossy compression which compresses parts of the voice which do not make sense to the human ear, like noise from the surroundings. This is discarded through the use of the compression technique. This simply means important voice information is kept intact. H 323 is the codec being used in VoIP compression

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2.2.5.4 Protocol H 323

H 323 is the protocol used in the VoIP. H 223 was the previous protocol which was very vulnerable especially to denial of service and the execution code. Though even now still problems do exist with the H 323 protocol, it has been discovered that other vulnerabilities are there. The software ports are left open sometimes because it does not work ok with the firewalls, hence prone to attacks.

2.2.5.5 Real Time Protocol

Real time protocol (RTP) is also among protocols that are used in VoIP implementation. This protocol helps in differentiating between real time and non-real time events. Vulnerabilities still exists in the real time protocol. User datagram protocol is the most preferred protocol for transmission in voice over internet protocol. Many ports are left open when UDP is used thus making it vulnerable to attacks such as denial of service that will make the service unavailable.

2.2.5.6 Session Initiation Protocol

Session Initiation protocol (SIP) is the highly desirable protocol that is used in voice over internet protocol for initiation of multi sessions. SIP is a signaling protocol which is widely used for starting and ending of the real time events.

2.2.5.7 SiVus

SiVus is a vulnerability scanner and known to be the first monitoring tool to be used in VoIP set-up. SiVus scans the vulnerability in three stages; firstly it starts by generating the SIP message, then generates SIP component discovery for target analysis and finally used for checking the security of the SIP phones.

Advantages of VoIP

VoIP being the latest technology on the market, the advantaged are quite many but only the critical ones are listed below:

I. For as long as broadband internet connection is available, VoIP can easily be used to call between computers with software installed that can support VoIP e.g. skype.
II. VoIP is portable and flexible such that a user can make a call anywhere in the world provided there is connectivity to broadband internet.

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III. Quality of service is guaranteed provided techniques to enhance the QoS are implemented correctly.

IV. VoIP can be implemented on the wireless Local Area Network hence making it mobile.

V. VoIP is relatively cheap to set-up. No need of installing new cables of there is an existing LAN in place.

VI. VoIP can be implemented on the Wide Area Network through the use of the PBX.

VII. Making a call via voice over internet protocol is easy and cheap more especially if the bandwidth is properly managed.

VIII. Since internet is at the core of this form of communication, no much hardware is required to set-up a VoIP communication.

IX. Voice over internet protocol is the latest technology for telephony and it transmits data through a packet switched method, hence increasing the effectiveness of data transmission.

Disadvantages of VoIP

I. VoIP set-up for calling between computers is internet dependent. In case of internet failure then no communication will be achieved.

II. A reliable back-up power has to be installed in case power failure or interruption is experienced. At least should be able to support continuity of communication until power is restored. Else VoIP will go down.

III. Since internet is not very secured, that also makes VoIP not to be very secured

IV. Emergency call feature is not yet incorporated in the VoIP set-up.

2.3 Key issues to consider in the design and implementation phases

Voice over internet protocol is relatively new on the market. Many companies are eager to switch from using the tradition PSTN to VoIP technology. Some companies have already implemented VoIP technology successfully; one of them is Zambia Railways LTD. Although many other companies would love to migrate the soonest, but unfortunately they do not know where to start from due to limited knowledge on how to get started through to implementation stage.

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The success of this project was supported by the following recommendations that were followed from design through to implementation stage.

I. Voice over internet protocol network should be implemented based upon common architecture for data communication and real time event communication. In return this will save a lot of money on the setup cost and management is so easy.
II. It's a well-known fact that VoIP is still a new technology and still advancing, so it's important always to leave room for future expansion.
III. Open standard protocol structures should be applied when deployment is taking place, that the network should not be vendor dependent.
IV. Always maximize the bandwidth capability, that quality of service can be attained at all times.
V. Session initiation protocol is the standard for VoIP and should be used as the only signaling protocol for the real time services.
VI. Single connection should be used to access the bandwidth rather than shared. In a case were shared connections are used then an intelligent router should be installed to reduce the cost.

2.4 Conclusion

Voice over internet protocol (VoIP) is the latest technology on the market for telecommunication. And it is widely being implemented both in homes and at the corporate level because of its cost effectiveness. In this report various technologies and techniques surrounding VoIP technology have been addressed. In the literature review, VoIP was discussed how it works, how it is implemented and the challenges which are currently there. The comparison with the competing technology was done were I highlighted that VoIP is relatively cheap to make a call compared to the tradition PSTN call. In addition, VoIP is cheap and faster to setup on an existing LAN unlike PSTN were cables are installed for every new connection. The report was going to be incomplete without mentioning the advantages and disadvantages that voice over internet protocol has. One of the greatest advantages is the quality of service (QoS) VoIP can deliver given a higher bandwidth and properly managed. On the other hand, VoIP is internet dependent as one of the disadvantage. Finally, VoIP has the potential to reach beyond the limitations. Let's all be innovative and get the best out of VoIP technology.

3.0 Systems Design

3.1 LAN and WAN Model

The overall network model was implemented using Riverbed Modeler academic Edition. The project focused mainly on the Local Area Network (LAN) and the Wide Area Network (WAN) setup, and then thereafter run experiments under different scenarios to determine the best setup for VoIP implementation.

3.2 Configuration and Design

The project had a total of Five (4) independent scenarios. Like eelier mentioned, Riverbed academic edition was used to measure VoIP performance and the comparison was done under different circumstances as below:

I. Local call vs long distance calls
II. Ethernet Hub LAN Network vs Ethernet Switch LAN Network
III. Increased Traffic Load on the Network
IV. WAN with FTP access while transmitting Voice.

3.3 Distributed Topology Infrastructure

The network was setup in (3) different locations namely; Head office in Lusaka and two (2) regional offices northern (Kitwe) and Southern regions (Livingstone) respectively. Two backbone links were used to measure one that can work best with VoIP. T1 link and fibre channel were used in this case.

3.4 LAN Topology with FTP

File Transfer protocol (FTP) was configured on the network just to increase traffic load then measure how the bandwidth will be affected and ultimately the effect on the VoIP quality of service should both services be accessed simultaneously by users on the network.

3.5 Topology with Delay in a Hub

VoIP LAN setup topology was designed and implemented using the Hub as the central connectivity point for all the IP phones on the LAN. The Hub was used to measure how voice gets affected in terms of Jitter, voice Packet delay and voice packet End-to-End delay. The comparison in terms of performance was done between a Hub and a Switch on two different Network setups. This experiment helped in understanding which networking devices is best for VoIP implementation.

4.0 System Development

4.1 LAN Switch and Hub Model

Three (3) major scenarios were analysed and investigated to determine how VoIP responds under different scenarios. The first scenario was implementing VoIP on a wired LAN using a Hub, the second one was a switch as a central connecting point and the third one involved connecting the three (3) regional offices that are geographically spaced and investigate how VoIP can be affected with distance using two (2) different link types for connectivity. Below are screenshots in figure 4.1.1 showing scenario 1, 2 and 3.

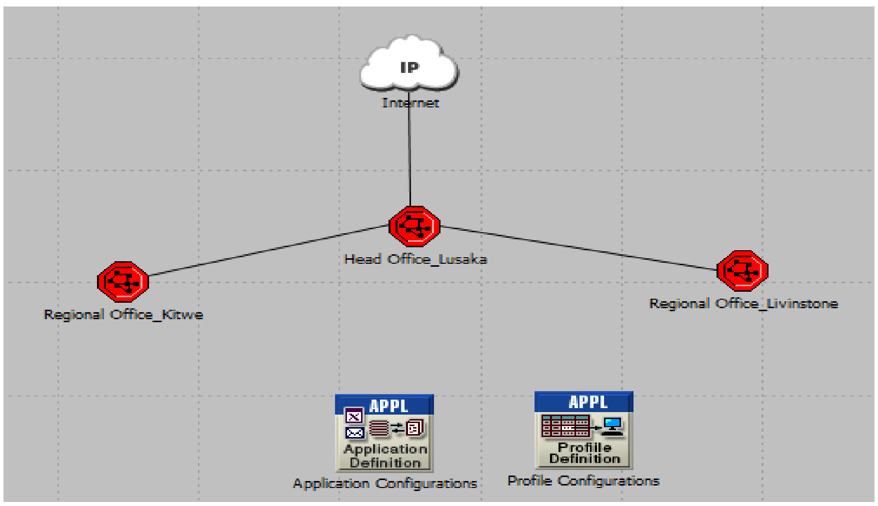

Figure 4.1.1: showing scenario 1 and 2 LAN setup

The Application and Profile configurations for the voice over internet protocol Network setup are shown below in figure 4.1.2 and 4.1.3 respectively. In the application configurations, two (2) applications where defined namely; voice and FTP services. Knowing very well that quality of Voice output is the key objective of this project, hence no need of defining many applications apart from FTP to be used investigating how voice transmission can be affected should FTP be incorporated on the Network using the same medium with voice. FTP application was set to high for all the three regional offices.

As for the profile configurations, three (3) profiles were created namely; Head Office_User, Northern Region_Users and Southern Region_users respectively. All the profiles will have access to the two applications already defined, that communication can be attained.

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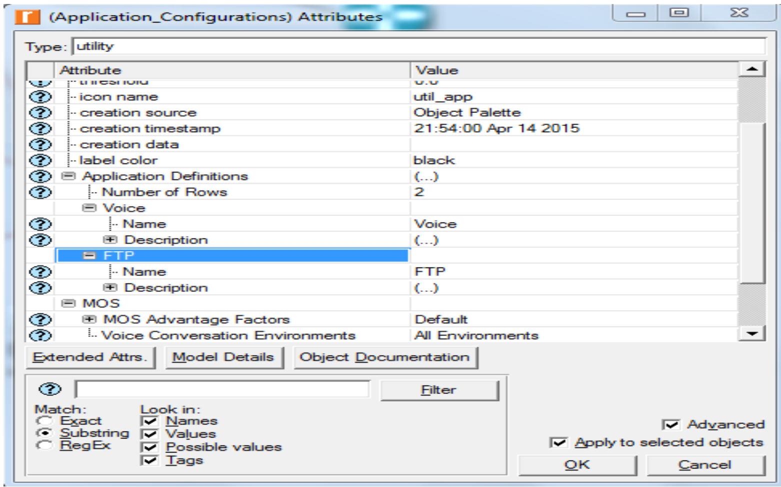

Figure 4.1.2: Application configurations

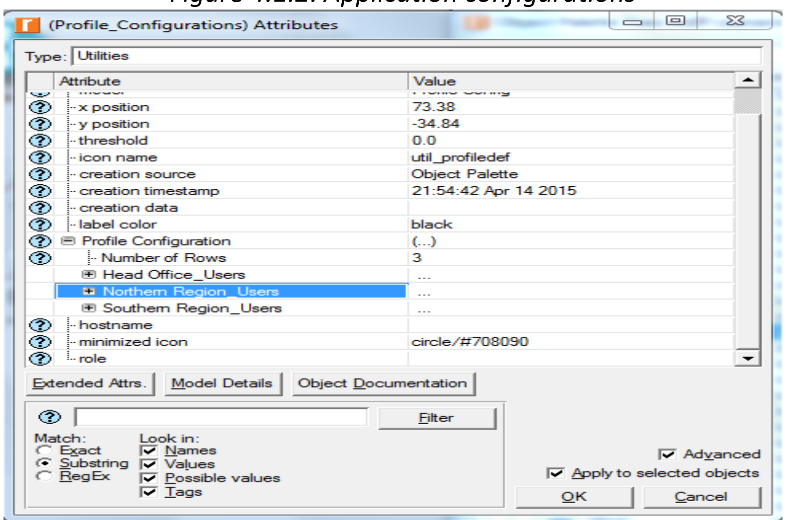

Figure 4.1.3: Profile Configurations

4.2 Local Area Network Topology using a Hub

This topology shows scenario 1 of the voice over internet protocol (VoIP) network setup, where a Hub is used as the central connectivity device for all the IP phones at Head office. Riverbed academic edition17.5 was used to simulate this scenario. The dimension of the office building is 100 x 100 meters with ground and first floors of office space. The building has about 150 Ethernet points for the IP Phones connectivity. 100 Base T cable was used for all the connections in the LAN. Below is figure 4.1.4 showing the topology for scenario 1.

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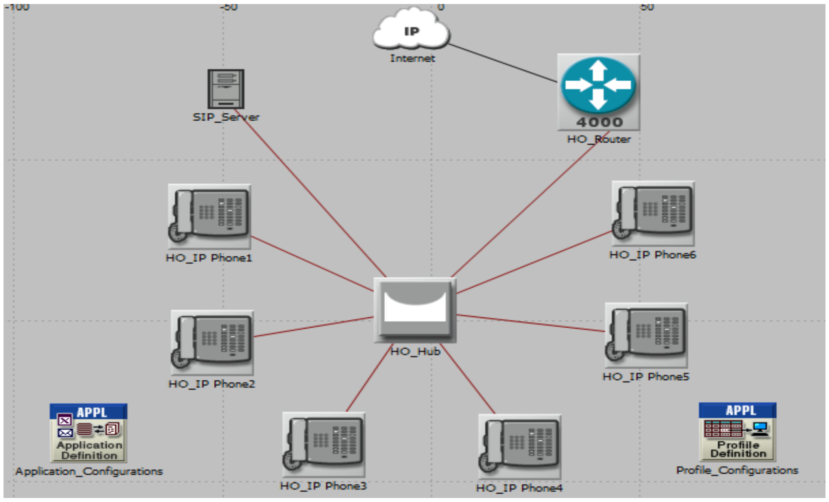

Figure 4.1.4: Scenario 1 VoIP LAN topology with 6 Nodes

The initial VoIP setup for scenario 1 had only six (6) IP phones connected on the network through the Hub and was able to make and receive calls internally. The SIP server was also part of the network to facilitate all the configurations for internal communications. The following figure 4.1.5 shows the topology of the VoIP network with increased number of VoIP workstations and IP Phones connecting to the internet through the Hub.

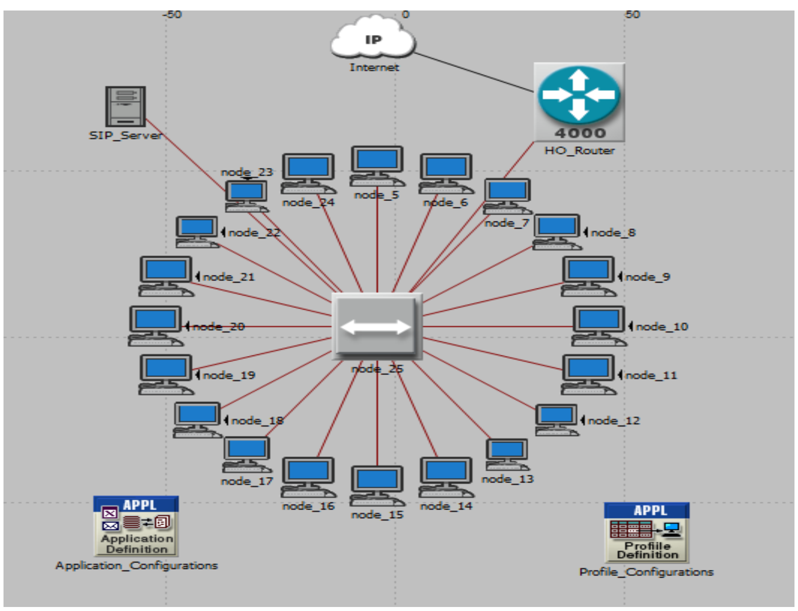

Figure 4.1.5: VoIP LAN Network with 20 stations using a Hub

After the simulation with the initial 6 nodes connected on the network was carried out, the number of connected VoIP machines was increased to 20 that we can measure how voice is affected in an event the number of users is increased while using a Hub. The results were obtained and documented.

4.3 Local Area Network using a Switch

A switch is defined as a computer networking device that connects devices together on a computer network, by using packet switching to receive, process and forward data to the destination device. Unlike less advanced network hubs, a network switch forwards data only to one or multiple devices that need to receive it, rather than broadcasting the same data on all its ports.

In a similar setup shown above in figure 4.1.5, the second scenario was designed and implemented but this time using a switch as a central connection point instead of using a Hub like in the first scenario. The initial setup had 6 IP phones connected on the LAN and the simulation for the voice network performance was done. Results were obtained and compared with results obtained after increasing the number of VoIP users on the network. The screenshot in figure 4.1.7 below shows the VoIP LAN setup with increased number of users to 100.

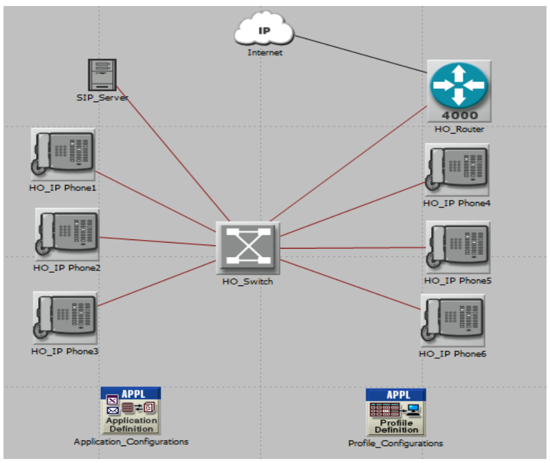

Figure 4.1.6: VoIP LAN using a switch with 6 Nodes

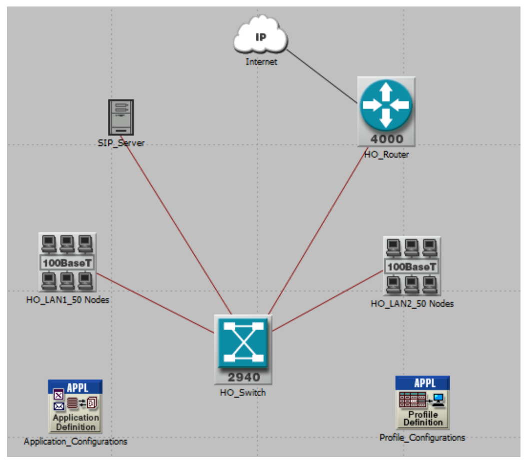

Figure 4.1.7: *VoIP LAN setup using the switch with 100 Nodes*

4.4 VoIP setup on the WAN

The setup for VoIP on the Wide Area Network involved three (3) towns; Lusaka being centrally located is the Head office for Zambia Railway with two regional offices, one in the south (Livingstone) and the other one in the North (Kitwe). The Head office has the main PBX server to allow internal communication for the organization. SIP phones are installed in the three (3) regional offices and running on both IP phone handsets and on the computer using the SIP 3cx software that enables making extension calls through the computer with the microphone and earphone provision. The internet service provider gives internet to the company through the Head office for proper management purposes and then distributes it to the two (2) regional offices. The Internet will allow VoIP communication through VoIP application software installed on the computer E.g. Skype. The following screenshots shows the main WAN setup and regional office LAN setup. The design and configurations were done using Riverbed Academic edition 17.5.

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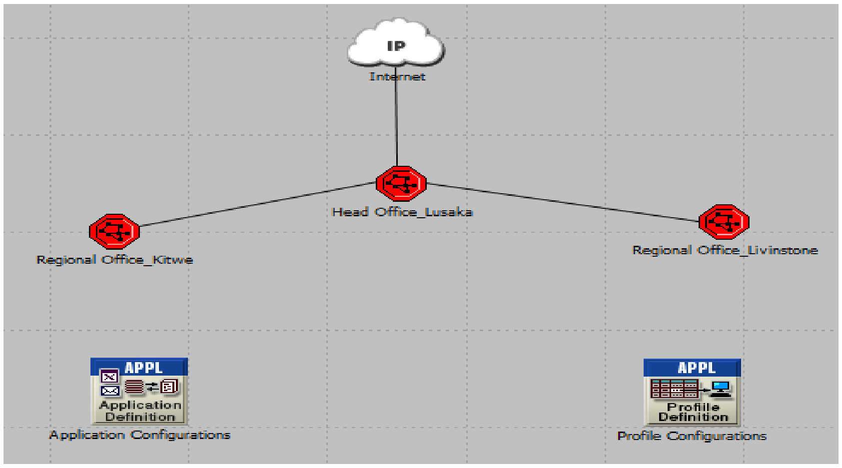

Figure 4.1.8: VoIP setup on the WAN

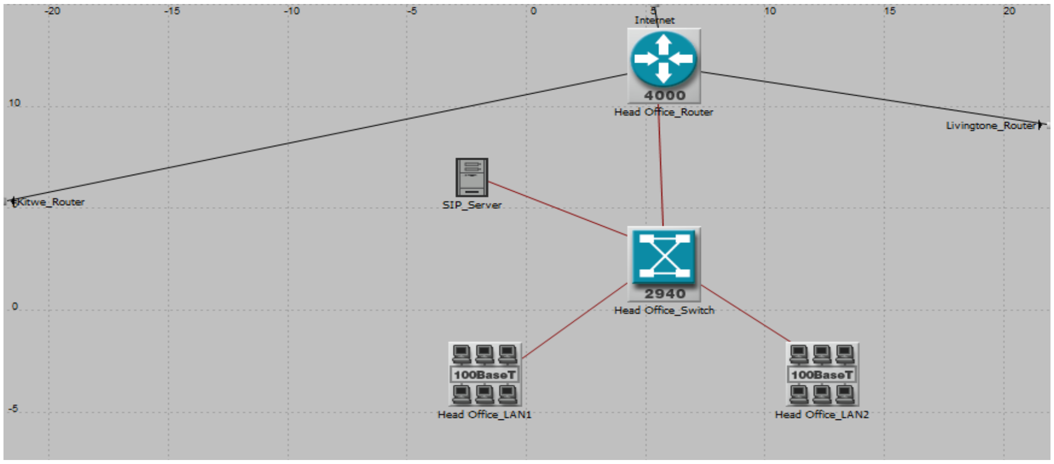

Figure 4.1.9: VoIP LAN Setup at Head Office – Lusaka

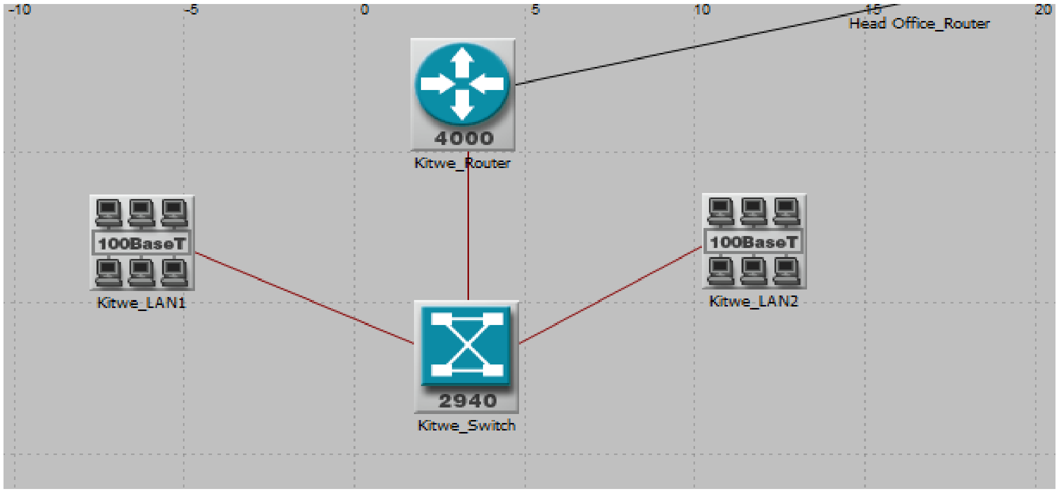

Figure 4.1.10: VoIP LAN setup in the Northern Region – Kitwe

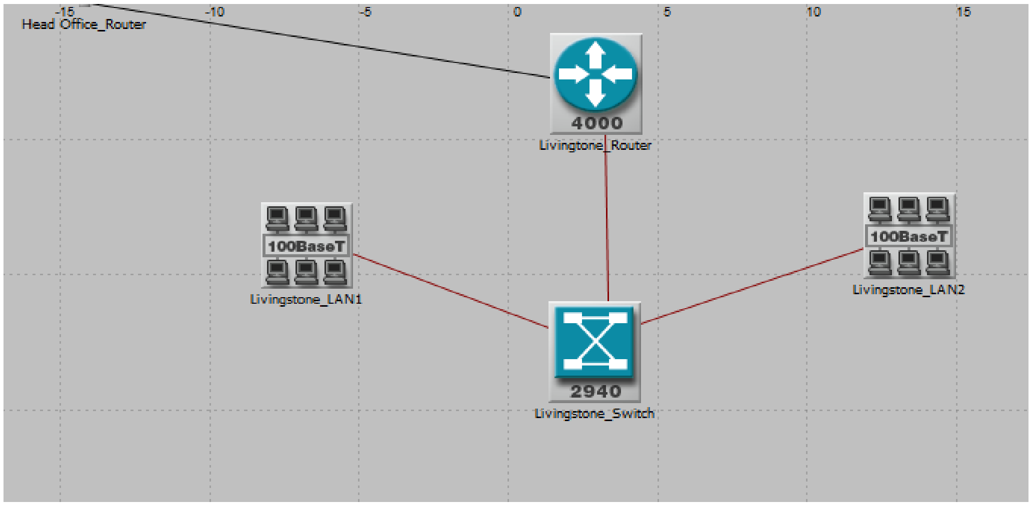

Figure 4.1.11: VoIP LAN setup in the Southern Region - Livingstone

Figure 4.1.6, 4.1.10 & 4.1.11 shows the LAN setup in the three regions. Head office with 100 VoIP users, northern region with 50 and Southern region 50 VoIP users as well.

4.5 VoIP Setup on the WAN with FTP Running

This scenario is exact the same with scenario 3 except that on this one FTP service have been added on the Network to measure how it can affect VoIP communication if they were to be accessed continuously at the same time. Will compare the results from the VoIP setup without FTP services running and with the one with FTP running. The following screenshot shows the VoIP LAN setup at Head office –Lusaka. The FTP server is defined at Head office then the two regional offices are accessing it through the WAN.

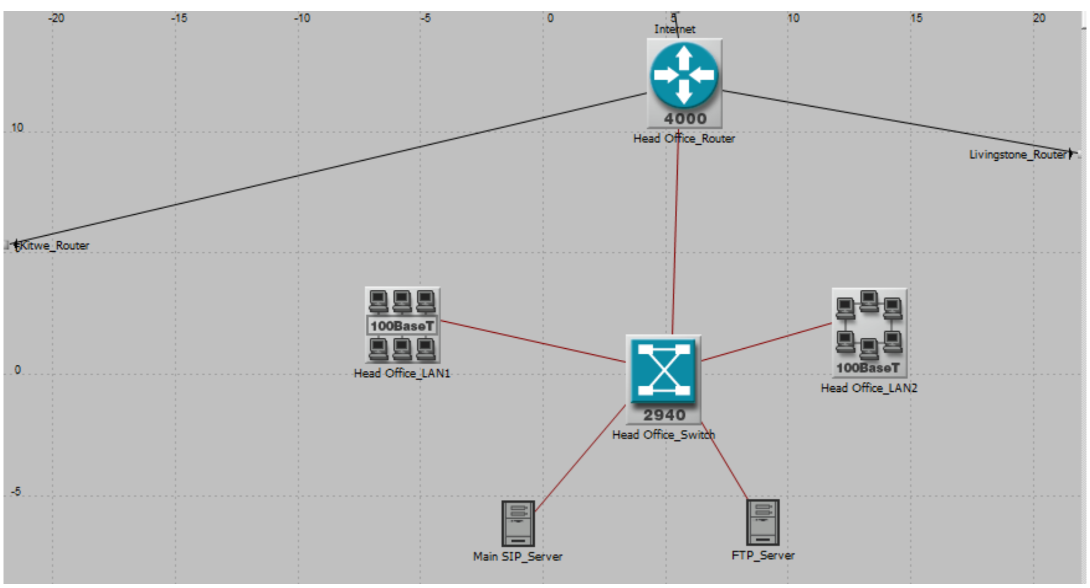

Figure 4.1.12: VoIP WAN setup with FTP services running

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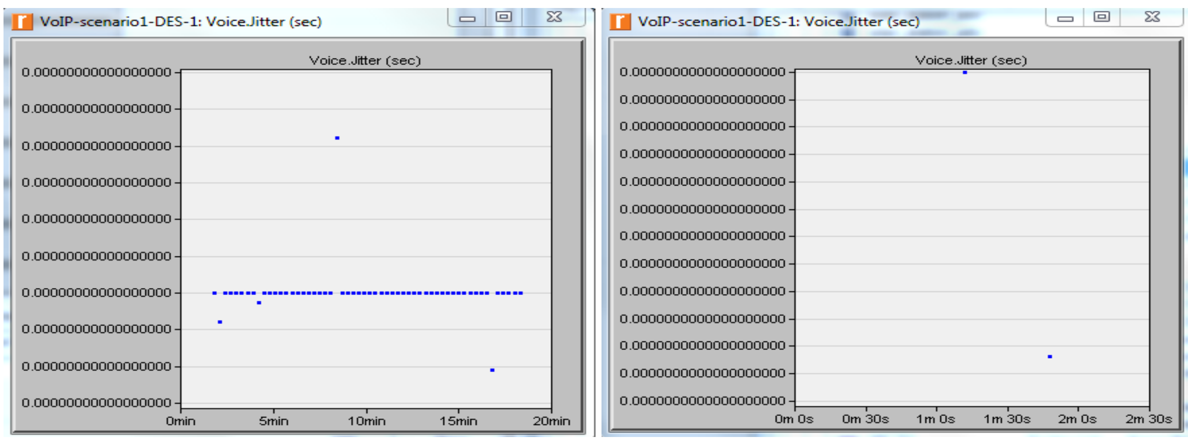
With the file transfer protocol added on the network through connecting it to the Head Office Switch. The traffic load is expected to go up that in return will have the negative impact on the bandwidth utilization.

5. Evaluation of Results for the Simulations

5.1 Scenario 1: VoIP Call on the LAN with a Hub

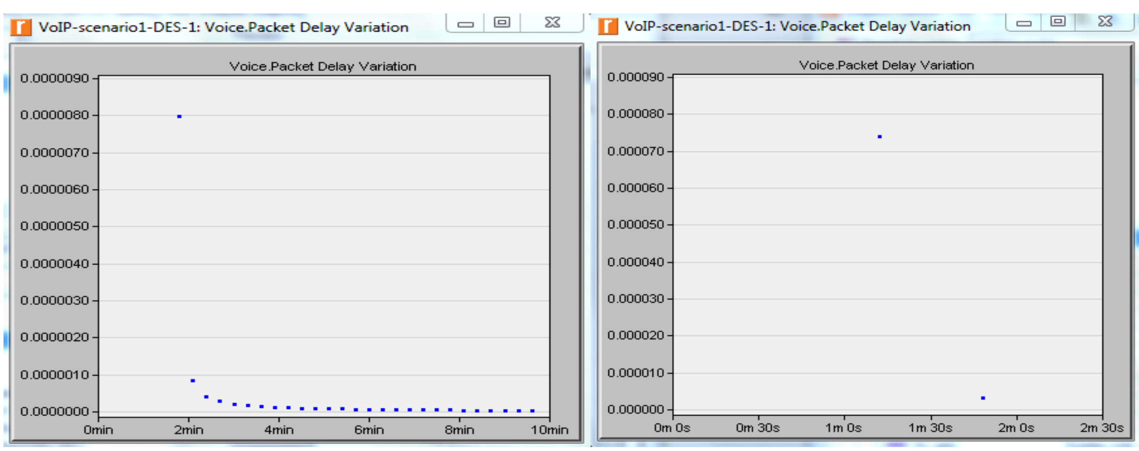
The all idea of coming up with scenario 1 was to test the performance of a VoIP call on the LAN of six (6) nodes connecting to the Hub. A comparison was done between the VoIP LAN setup with six (6) bodes and that of 20 nodes. Below are screenshots in figure 5.1.1 showing results after running the simulations.



Jitter on the LAN with 6 nodes　　Jitter on the LAN with 20 nodes

Figure 5.1.1: Simulation results for voice Jitter

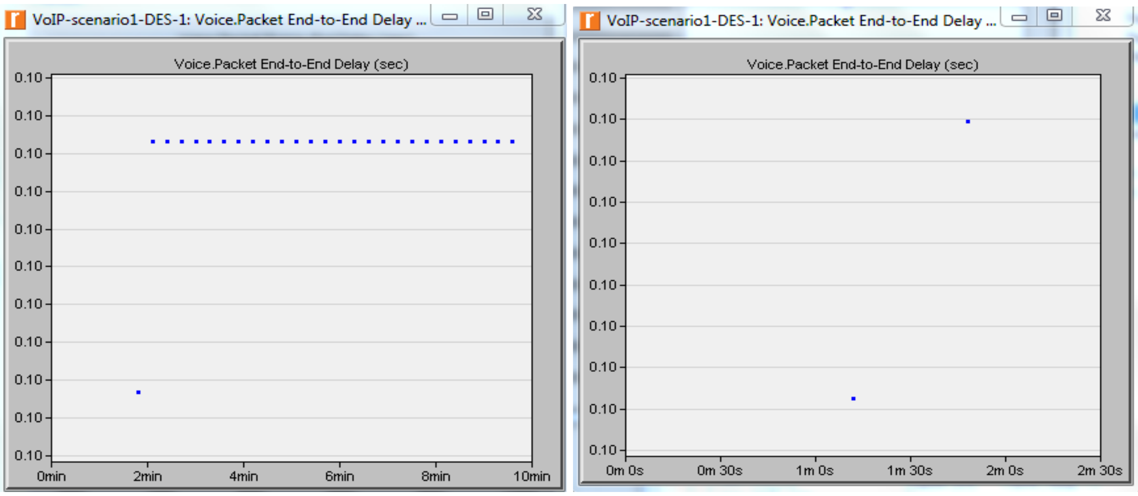
The two (2) screenshots above in figure 5.1.1 clearly shows the increase in the voice jitter after the number of nodes was increased from 6 to 20.



Voice packet Delay with 6 nodes　　Voice Packet Delay with 20 nodes

Figure 5.1.2: Simulation results for voice packet delay.

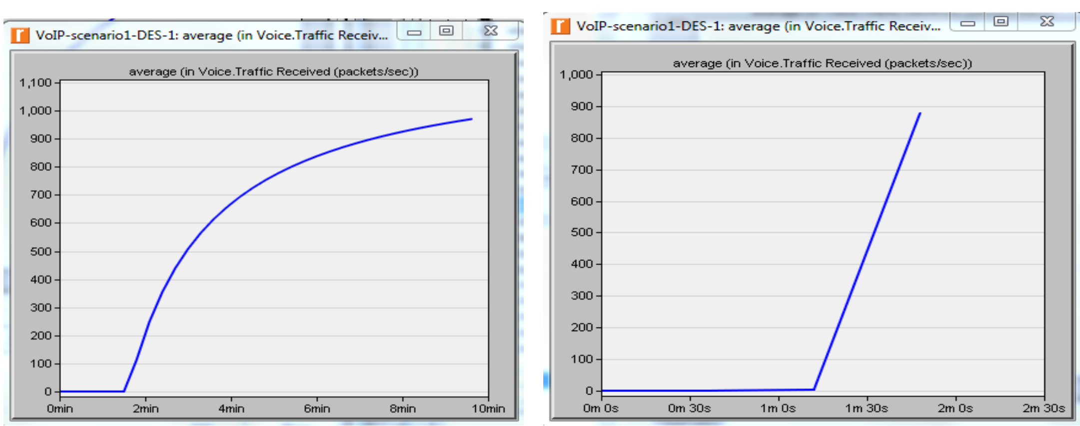
Figure 5.1.2 with two screenshots above shows the simulated results for voice packet delay. The packed delay variation was noticeable after the number of nodes was increased to 20.



Voice End-to-End Delay with 6 Nodes　　Voice End-to-End Delay with 20 Nodes

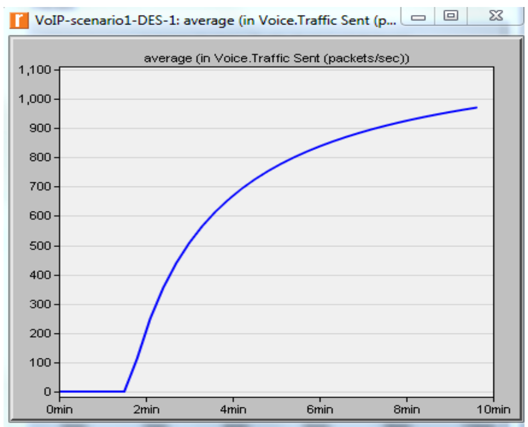
Figure 5.1.3: Simulation results for End-to-End delay

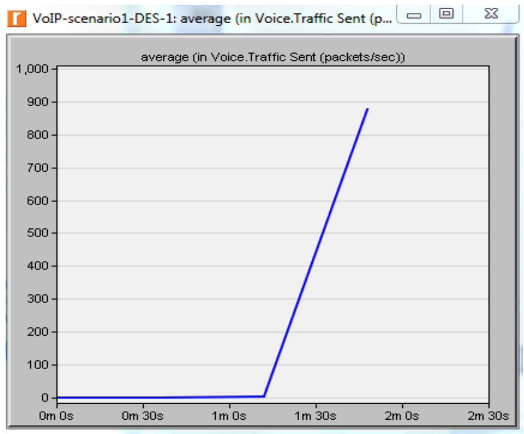
Figure 5.1.3 above shows the voice packet End to End delay difference under the setup with 6 nodes and 20 nodes respectively.



Average Voice traffic Received with 6 Nodes　　Average Voice traffic Received with 20 Nodes

Figure 5.1.4: Simulation results for voice Traffic Received





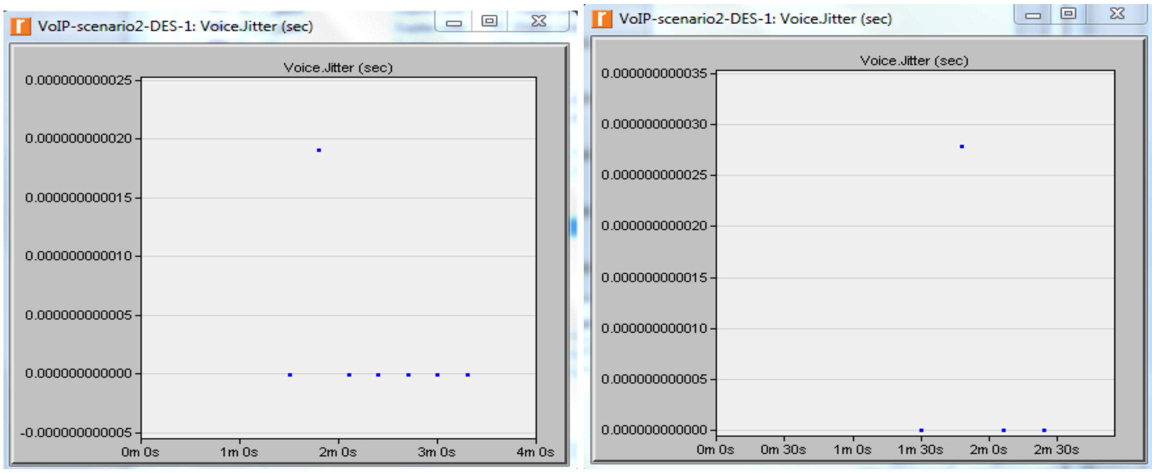
Average Voice Traffic Sent with 6 nodes   Average Voice Traffic sent with 20 nodes

Figure 5.1.5: Simulation results for voice Traffic Sent

The two screenshots in figure 5.1.5 shows the simulated results for the voice traffic sent on the network. In the instance were only 6 nodes were connected on the network; the traffic sent was 990 packets. This was so because of less collusion domains being experienced in the Hub. However, the second instance were more users where added to the network the graph dropped to somewhere 690 packets being sent per second. This was due to the fact that because of the increase in the number of users, the Hub formed a good number of collusion domains hence less packets being sent.

5.2 Scenario 2: VoIP Call on a LAN with a Switch

Scenario 2 is exact the same as scenario 1 except that this time around a switch was used as the central connection point for all the Ethernet nodes. In this scenario will evaluate how the quality of service for voice is affected with the increased traffic on the network. The initial VoIP LAN setup has 50 VoIP users connected and in the second instance, the number of users was increased to 100. The simulation was done for both instances and results compared. Voice Jitter, packet delay, End to End delay, traffic received and traffic sent was evaluated in this scenario. The following figure 5.2.1 shows the voice jitter results.

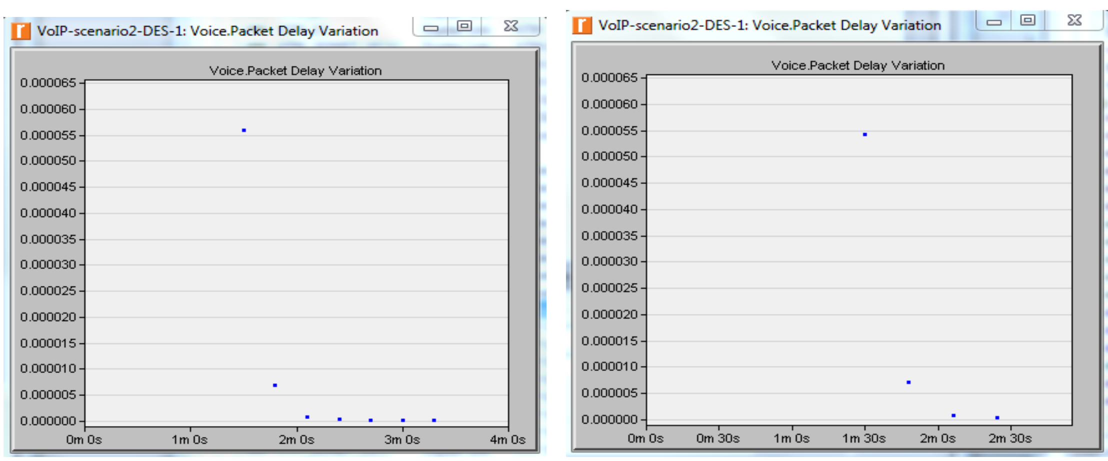


Jitter with 50 Nodes

Jitter with 100 Nodes

Figure 5.2.1: Simulation results for voice Jitter

The above figure 5.2.1 shows the results after the simulation for voice jitter was done. The first simulation was done with the total number of 50 VoIP users on the network, while the second simulation was conducted with doubled number of VoIP users to 100. As clearly shown from the two graphs, 50 VoIP users produced voice jitter of 0.000000000019 and with 100 users, voice jitter increased to 0.000000000028. This will surely have a negative impact on the quality of service.



Voice Packet Delay with 50 Nodes

Voice Packet Delay with 100 Nodes

Figure 5.2.2: Simulation results for packet delay

Figure 5.2.2 above shows voice packet delay variation. While the number of VoIP users were 50 and after the number of users was increased to 100.

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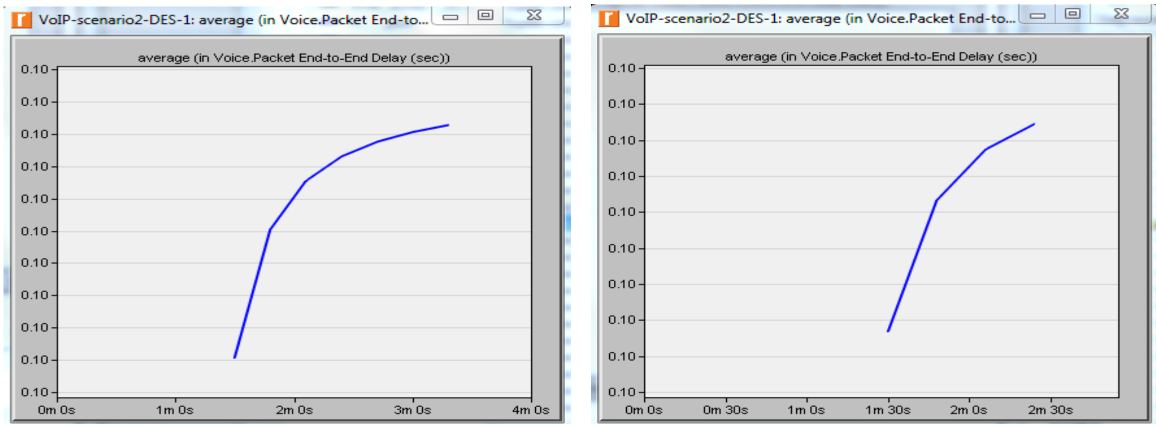
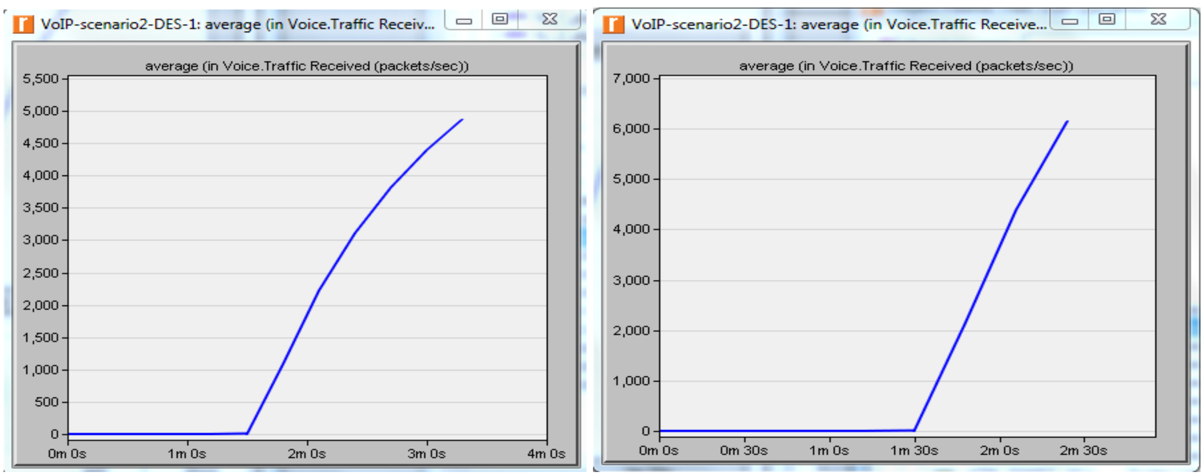


Average End-to-End Delay with 50 Nodes　　Average End-to-End Delay with 100 Nodes

Figure 5.2.3: Simulation results for End to End delay



Average Traffic Received with 50 Nodes　　Average Traffic Received with 100 Nodes

Figure 5.2.4: Simulation results for average voice traffic received

The graphs above in figure 5.2.4 clearly show that 50 VoIP users on a network received voice traffic of 4,900. After doubling the VoIP users to 100 on the network, traffic received increased to 6,200.

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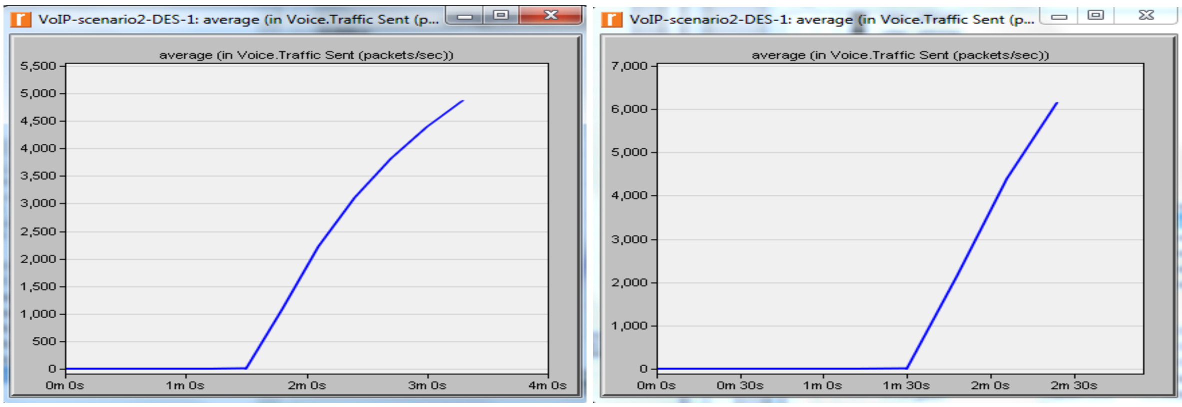


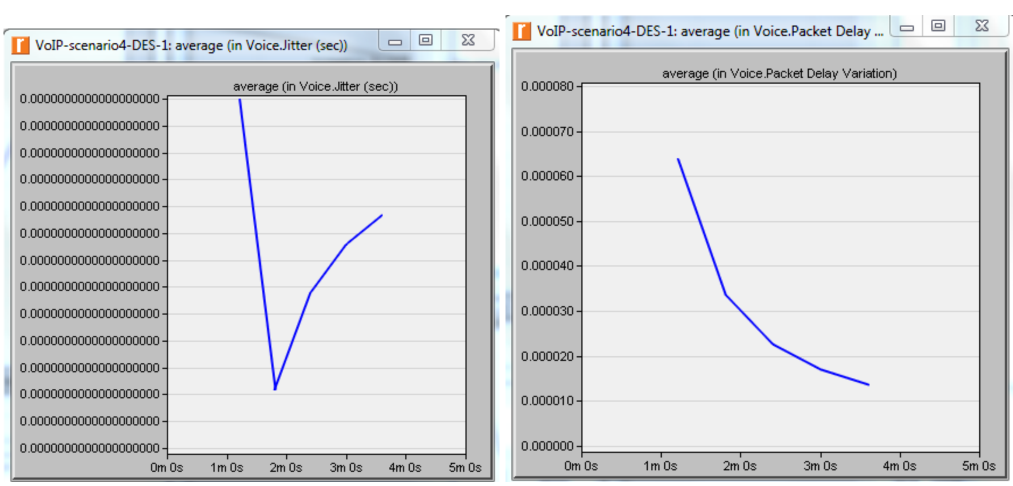
Average Traffic Sent with 50 Nodes   Average Traffic Sent with 100 Nodes

Figure 5.2.5: Simulation results for average voice traffic sent

The above figure 5.2.5 shows the simulated results for voice traffic sent for two instances. The simulation that was done in the first instance included 50 VoIP users on the network and the amount of voice traffic that was sent according to the graph was 4,800. While in the second simulation, the number of VoIP workstations was doubled to make it 100, which prompt the traffic sent increasing to 6,200 according to the graph. This scenario evidently proves that should more VoIP users be added on the network, then congestion will be experienced due to increased traffic on the network.

5.3 Scenario 3: VoIP Call on the WAN Setup

The following screenshots in figure 5.3.1 shows VoIP performance on a WAN environment using the T 1 link to connect the two regional offices. This scenario shows the average voice jitter, packet delay, packet End to End delay, traffic received and traffic sent between the three regional offices.



Average Voice Jitter   Average Packet Delay

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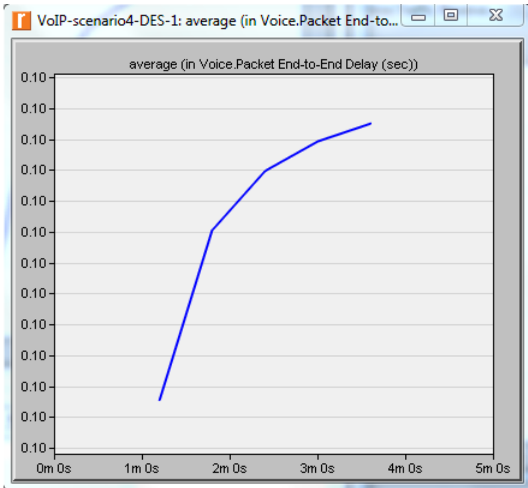
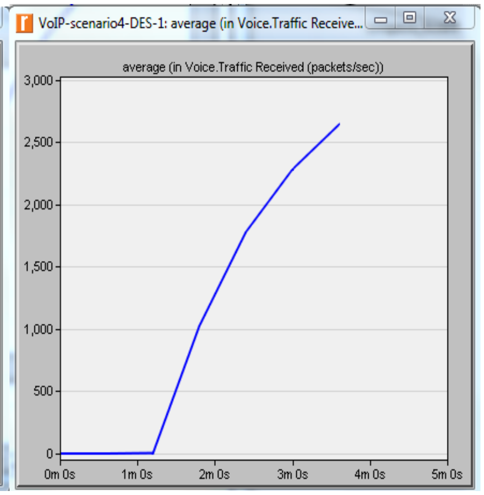
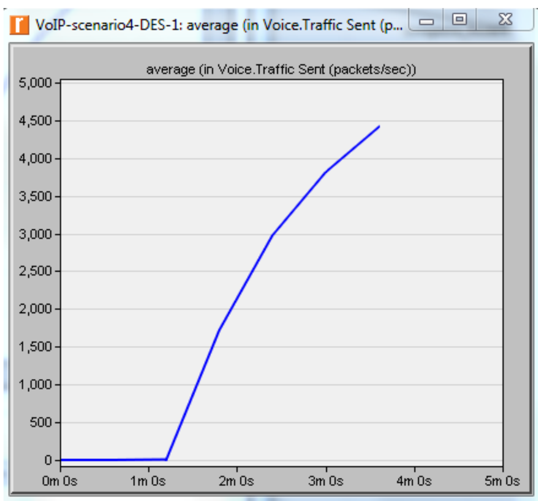


Average voice packet End-to-End



Average Traffic Received

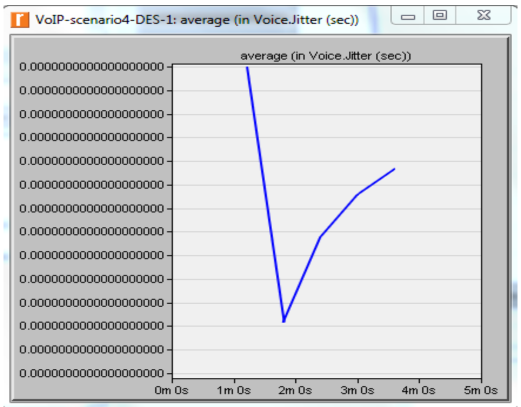


Average voice traffic sent

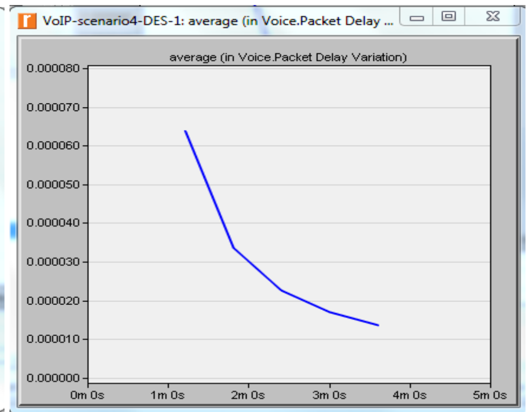
Figure 5.3.1: voice results after simulation

5.4 Scenario 4: VoIP Call on the WAN with FTP Running

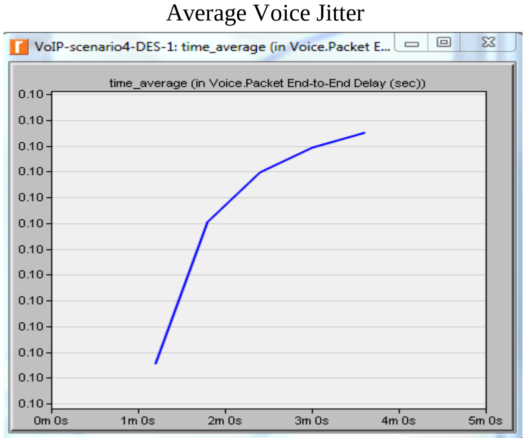
The following Screenshots in figure 5.4.1 below were taken after adding the FTP services on the network and did the simulation



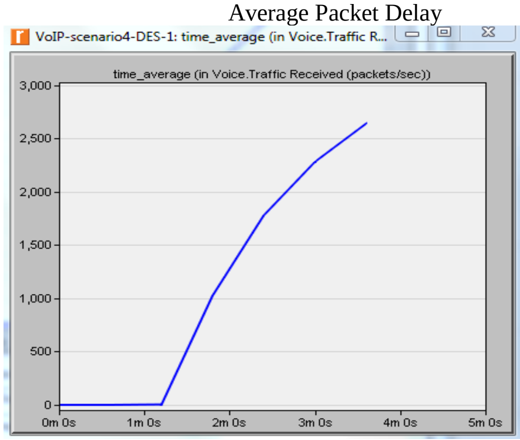
Average Voice Jitter



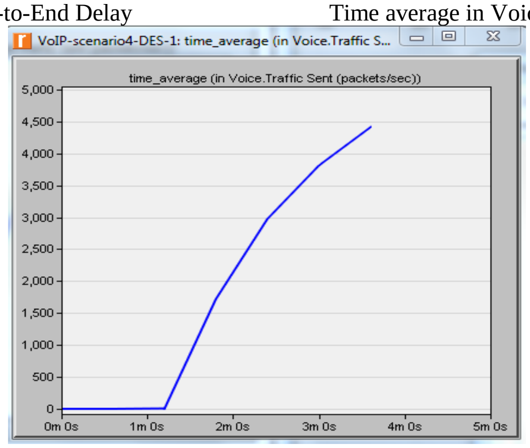
Average Packet Delay



Average Packet End-to-End Delay



Time average in Voice Traffic Received



Time average in Traffic Sent

Figure 5.4.1: Voice simulation results and adding FTP service on the network

The graphs in figure 5.4.1 shows the results after the simulation were done. In this scenario the VoIP WAN setup is exactly the same as of that in scenario 3. The only difference is that this scenario is running FTP services on the WAN. FTP was added on the WAN network to see what effective it may cause in trying to attain best VoIP quality of service on the WAN. The screenshots in figure 5.3.1 and figure 5.4.1 clearly show that no much impact was made on the network; this was due to the fact that when using an intelligent router and a switch on the network, voice will always receive high priority in packet transmission hence not being affected with the introduction of FTP.

6.0 Conclusion

Voice over internet protocol (VoIP) has made communication much more affordable while promoting efficiency and availability. The technological innovations have advanced in the recent years to the extent of cutting down many costs associated with communication equipment installations. In this paper I looked at how VoIP can be implemented on the LAN/MAN/WAN internal communication of an organization. I further demonstrated how VoIP can be implemented on mobile devices and allow communication internally and externally using 3CX software based phone. This innovation drastically reduces the costs associated with procurement of expensive VoIP handsets and ultimately the installation of the Local Area Network backbone to support VoIP using hand held IP Phones. Additionally, recommendations were made to ensure high standards are adhered too when designing the Network to support VoIP technology, that quality of Service (QoS) is achieved with minimum distortion.

Away from the hands on experiment, I further configured a software based Network setup for dipper understanding of how VoIP technology reacts under different conditions. Four (4) scenarios were covered under the experiment using riverbed simulation software. The first scenario was aimed at determining how voice can be affected if hubs were used to setup a VoIP network and it was proved that Hubs have a negative impact on the QoS due to the fact that Hubs share bandwidth on all the ports. Meaning Hubs are not ideal for multiple communications. The second scenario was aimed at determining if VoIP works well in a LAN setup implemented using a switch as a central connectivity device for all the points connecting IP Phones, especially with the known fact of the Switch intelligent decisions making. It's ideal also to install Switches with Power over Ethernet (POE) that you can also reduce the burden of running separate power cable to power the IP Phones. Further, the third scenario looked at how VoIP technology can be implemented on the wide area network involving three (3) towns that are geographically spaced and finally the fourth scenario was to determine if FTP service can disturb VoIP QoS in an instant it's added on the network.

Unfortunately, VoIP technology has a urge challenge with communication continuity in an event utility power supply is unavailable. To overcome this challenge, a reliable backup power supply should be considered when designing VoIP Network. This approach will guarantee continues communication.

VoIP Technology has indeed proved to be cost effect technology for many institutions. I too recommend VoIP exploration to those still using traditional copper wired telephones.

7.0 Acknowledgements

In a special way, I would like to thank the almighty God for the Wisdom, strength and knowledge have been blessed with and above all the great gift of life that today have this opportunity of Studying Master in Information & Communication Technology with Information & Communication University (ICU).

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To my fellow students, Special thanks for the tireless contribution and brilliant ideas you have been giving me that made me to have a positive outlook towards this project. Let’s keep up with the same spirit of togetherness.

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