COLLEGE OF ENGINEERING

DEPARTMENT OF COMPUTER ENGINEERING

A Proposed Design for VoIP Integration with Legacy Communication Systems on Kwame Nkrumah University of Science and Technology Campus

Project submitted in partial fulfillment for a Degree of Bachelor Science (BSc.) in Computer Engineering

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DECLARATION

We hereby declare that except for specific references which have been properly acknowledged, this work is the result of our own research and it has not been submitted in part or whole for any other degree elsewhere.

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ABSTRACT

Voice over Internet Protocol, otherwise known as VoIP, has seen a lot of interest and patronage in the 20^{th} century. This is due to the development of its underlying network infrastructure and internet as a whole. Also, it brings to board a communication system that is much cheaper to use and maintain than the traditional legacy communication system.

KNUST boasts of an efficiently functional VoIP deployment over its local network, allowing for users, mainly administrators and other faculty and staff members to make calls over the VoIP system. Though this system is reliable and serves its purpose well, it has failed to fully exploit other capabilities a VoIP system can employ. There is no integration of the Public Switched Telephone Network (PSTN) system, hence users can reach other users when they are away from their IP phones through the VoIP system. The reason for its absence is history of users misusing the PSTN service for long, personal calls. Also, calls to external networks contend for bandwidth on the router, delivering low quality call experience during peak hours.

This project addresses all the above issues, with recommendations for future improvements. A proposed design is made for integration of PSTN with the existing VoIP system. This design incorporates control mechanisms to reduce misuse of the service by users. The design also tackles externally made calls, especially to satellite campuses and other universities. The new design passes all simulation tests for ITU-T recommended metrics. It is also low-cost and highly feasible, making it a suitable improvement to be implemented.

DEDICATION

We dedicate this project to our parents, who have supported and seen us through school. We also appreciate the efforts of our ever-sturdy supervisor, Dr. Eric Tutu Tchao. May God Almighty reward him.

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Our first thanksgiving is to the Almighty GOD, by whose grace we could never have come this far. His careful hand has indeed guided us through.

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We also say a special thank you to our classmates and friends, you were always there to hold us up when we were down. Our moments together will forever be cherished.

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LIST OF ABBREVIATIONS

AVVID	Architecture for Voice, Video and Integrated Data
DoS	Denial of Service
FTP	File Transfer Protocol
GARNET	Ghana Academic Research Network
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
IP	Internet Protocol
ISP	Internet Service Provider
ITU	International Telecommunication Union
ITU-T	ITU Telecommunication Standardization Sector
JMF	Java Media Framework
KNUST	Kwame Nkrumah University of Science and Technology
LAN	Local Area Network
MOS	Mean Opinion Score
NOID	Network Operating and Infrastructure Department
NPRU	Nakhon Pathon Rajabhat University
OPNET	Optimized Network Engineering Tool
OTT	Over-The-Top
PBX	Private Branch Exchange
POTS	Plain Old Telephone System

PSTN	Public Switched Telephone Network
PBX	Private Branch Exchange
SIP	Session Initiation Protocol
QoS	Quality of Service
VoIP	Voice over Internet Protocol
V2oIP	Video voice over Internet Protocol
WAN	Wide Area Network

CHAPTER ONE: INTRODUCTION

1.1 BACKGROUND OF THE STUDY

Voice over Internet Protocol (VoIP) is a system of communication where calls are placed on an IP network or over the internet [1]. It allows for voice signals, which traditionally would have had to be sent via traditional phone systems, to now be sent over a network the internet connection. VoIP is quickly becoming a preferred mode of communication as the internet becomes more and more available. Also, VoIP is easy to maintain and its cheap nature is attractive for many big organizations and businesses as compared to Public Switched Telephone Network (PSTN) alternatives [2]. Much has already been done in the way of research into usability and functionality of VoIP [3][4][5].

This study seeks to build off previous work to design a comprehensive VoIP – PSTN integration for the Kwame Nkrumah University of Science and Technology (KNUST), explore how to extend these services to other campuses and analyze the functionality and reliability of the proposed implementation.

With the rise in internet-enabled devices and interconnected systems, the need and modes of communication have also taken a massive turn. Messages that took weeks through the post office began to take minutes via fax, then took seconds via email and are now almost instantaneous in many messaging apps today. The needs and style of communication have changed for many customers [6]. In large academic institutions where the vital resource of information is exchanged in immense volumes, it is necessary that a communication system that is both reliable and cheap is provided. Today, many campuses are equipped with traditional Private Branch Exchange (PBX) systems [7]. These allow for point to point communication but are heavily limited to perform more complex communication functions that are so needed in today's world. In the past two decades, great strides have been made to allow the internet to not only carry data but also voice and video. This has led to a much faster internet which serves as the backbone for voice over the internet, better known as voice over IP [8].

Currently, the Kwame Nkrumah University of Science and Technology has a VoIP system built over the university's Ethernet Local Area Network. It is the focus of this study to assess the extent and functionality of this voice system thoroughly. The next thing would be to design a retrofitted system which incorporates the ability to make calls to traditional telephones via the PSTN. Also, the study is ended with notes and specification on how to extend the VoIP system to other satellite campuses across the nation. Finally, diagnostics are run to determine QoS, network reliability and availability of the proposed implementations [9].

1.2 PROBLEM STATEMENT

There is a current worldwide move to employing the power of internet connected devices to empower our day-to-day communication [10]. The PSTN communication system has stood the test of time but it is very limited in the face of modern call client needs [7]. This leads to a lot of money spent on much less its worth. For the same number of connected devices and maintenance, VoIP is found to be much cheaper than legacy systems. Monthly telephony bills are not an issue anymore when using VoIP. Repair and maintenance works on local telephony are both complex and expensive. Also, PSTN requires that every user has a physical phone. With VoIP, applications called softphones can be installed on desktops or local devices that are already existent. This minimizes the number of physical devices that need to be purchased. Lastly, it is a one-time purchase with the only cost being the already existing internet infrastructure saves the university the exorbitant monthly pricing PSTN demands. According to Fayyaz et al., comparing pricing of VoIP providers like Vonage to traditional PBX systems proved VoIP to be much cheaper yet a wider range of services [6].

The communication system in KNUST should be reliable and efficient, opening users up to an array of communication services like instant messaging and conference calls. Currently, KNUST has a reliable VoIP communication system which is used by various faculty and administrative staff. The VoIP system, however, lacks a PSTN integration due to former misuse of resource by users, leading to high telephony bills. Also, the deployment is only on KNUST Main Campus so other campuses do not communicate with KNUST via VoIP. These factors show that the potential of VoIP has not been fully exploited to an appreciable extent given the network infrastructure in KNUST.

The solution this paper seeks to realize is a VoIP-PSTN integrated system that seeks to empower the current telephony system with reliable and optimized VoIP capability, which is a much cheaper and plausible option than a full overhaul, and also extend these services to satellite campuses.

1.3 OBJECTIVES OF THE STUDY

The general objective of this study is to design a VoIP system integrated with the PSTN which employs the existing IP networks and infrastructure of the main Kumasi campus of Kwame Nkrumah University of Science and Technology (KNUST) to reduce the cost of voice communication and upgrade the quality of voice and video in communication. The study would use the communication system in KNUST as a case study.

The specific research objectives are as follows:

1. To use the existing IP network in KNUST to design a voice communications system that incorporates the Public Switched Telephone Network (PSTN) to ensures quality voice services and improve communication efficiency.

2. To propose an internal IP communications system that enables branches (Accra City campus and Guesthouse) to communicate with each other over the IP network and communicate with the PSTN.

3. To use the reliability mechanisms provided by the IP network to improve service availability and network reliability of the VoIP system for external calls.

1.4 SIGNIFICANCE OF THE STUDY

The adoption of Voice over IP (VoIP), and largely, IP telephony has remained slow and steady for a variety of reasons. The main factors have been the reliability of legacy phone systems and VoIP's extreme sensitivity to delay, packet loss and security concerns on the other hand [11]. However, the debate on PSTN vs. IP telephony is still ongoing, and an increasing number of institutions are opting to replace their PSTNs with VoIP alternatives [12] [13]. IP telephony is also threatening traditional circuit-switched voice calls of mobile telecommunication.

In Ghana, the massive growth in data usage over the last years has caused a wholesale use of innovative means of communication, including Over-The-Top (OTT) services. OTT service providers allow users to transmit media, including voice and video over the internet, bypassing traditional distribution. According to [14], VoIP has changed the game for Telecommunication industries in Ghana, especially the voice business.

VoIP systems generally offer reduced telecommunication costs. The cost of a VoIP service mostly depends on the service provider and the calling features which the system integrates. Also, the setup cost is usually minimal since a number of institutions that switch to IP telephony already have a functional IP infrastructure on which the service is built and this is good especially for businesses and government institutions like Kwame Nkrumah University of

Science and Technology, which run on cautious budgets. Typically, computer-based VoIP systems allow free PC to PC calls, aside from the initial software cost, and a relatively marginal cost for PC to phone and phone to phone calls. This is very attractive, given the relatively high cost of phone bills and maintenance from using the Plain Old Telephone Service (POTS).

Also, with sufficient bandwidth, VoIP offers improved digital call quality. Such systems also have the ability to integrate advanced phone features such as caller ID, free call waiting and forwarding, digital voicemail forwarding, follow me call routing, auto attendant and many more. [15]

Many institutions maintain the use of telephone services provided by PSTN and are disinclined to switch to the emerging VoIP technology. This is mainly due to the leg up of the old technology on VoIP in areas such as its unmatched inherent reliability, security, and emergency location services. To such institutions, an old reliable technology is preferred to a relatively unreliable forward-looking technology.

Despite the attractive advantages of VoIP systems, there are still a number of drawbacks including low voice quality, high dependency on bandwidth and security. The system is highly susceptible to network security threats such as call tampering, denial of service (DoS) etc. [16]

Juxtaposing the benefits offered by both communication technologies, an integration of the two will offer a groundbreaking communication system in KNUST. VoIP users can easily make calls to traditional telephones on the PSTN and vice versa.

Taking a cue from the afore-discussion, we illustrate how a VoIP system can be designed as an integration of VoIP with the legacy communication systems on KNUST campus in order to provide an overall improved communication system that is capable of being scaled up to provide communication services between other branches i.e. Accra City Campus, Guest House Main Kumasi Campus.

1.5 ORGANIZATION OF THE STUDY

The rest of this study is organized as follows:

Chapter two describes the motivation for this work. Specifically, it discusses the current technologies used for voice communication in KNUST and the reason why new ways must be sought to better them. Also, existing literature on the design and integration of VoIP in a campus environment will be analyzed, identifying the downsides of these implementations.

Chapter three describes the research methodology used for the proposed design of the KNUST VoIP – PSTN integration to solve.

Next, in chapter four, the results of this design are discussed in detail and are used to perform a voice quality evaluation of the proposed design.

Our work is concluded in chapter five, where notable challenges encountered during the study are identified and propositions for a better future study are made.

CHAPTER TWO: LITERATURE REVIEW

2.0 INTRODUCTION

This section reviews related studies and relevant literature in the area of the deployment of VoIP technology on a campus environment. The discussion highlights the methodology, benefits and the drawbacks of the deployed systems. This underlines the necessity of a proposed design for VoIP integration with legacy communication systems. A discussion is made on existing communication system, IP Network, Public Switched Telephone Network (PSTN), and protocols used in VoIP.

2.1 RELATED WORKS

PSTN is a circuit switched technology and guarantees dedicated Quality of Service once call lines have been set up. VoIP is a packet switched technology, and depends on a reliable network to function effectively with undesirable amounts of packet loss, delays or jitter. In this chapter we take a look at VoIP implementations in campuses and how various designs that deal with inherent issues of VoIP.

In 2001, the Westminster Information Technology (IT) Department embarked on a project to replace the aging PBX communication system with a modern VoIP Cisco AVVID system. The former had reached its capacity of line extensions, requiring frequent repairs and suffering heat damage. A cost analysis was done and VoIP turned out to be more profitable in the long run. As earlier stated, VoIP is dependent on network performance. The 100-megabit backbone was upgraded to a 1-gigabit backbone to support the new technology. Packet loss where significant causes pieces of information to be lost during communication. Redundancy was introduced at core routers and switches to deal with packet loss. Switches were placed close to hard phones to directly provide power instead of having to depend on auxiliary power sources.

In Westminster, the voicemail server was built from scratch. A web interface served as the platform to program all phones, gateways, phone numbers and monitor user usage. Installing the IP-PBX required re-training of the staff. In the implementation, the old cabling for the PBX was not removed as this meant additional overhead. A few of the phones were retrofitted with a digital card to give them VoIP capabilities. The total cost of the installation was $750,000. As at the writing of the paper, the campus runs over 1000 phones on the IP-PBX. A post-evaluation showed that the new technology brought about huge cost savings, and an upgrade in the data network meant a boost in performance of the VoIP system also. Its expandable nature allows for a growing campus and the system continues to perform stably. [17]

In NPRU, a project was undertaken to integrate VoIP into the existing analogue telephone communication system. The existing PBX provided for a total of 300 users. It provided call functions like auto-attendant, call waiting and call identification. It was connected to a PSTN service that allows for internal calls (within the confines of the campus) and "external" calls (outside the campus but within Thailand). Despite its reliability it suffered two major drawbacks: (1) Electrical surges frequently damaged the analog telephone cards and (2) the limit on the number of phone lines prevented additional phones from being added. VoIP is built to run on fast networks thus making fiber-optic backbones ideal in implementation. An existing underground fiber wiring meant NPRU could leverage on the network. Fiber optic wiring is immune to electrical surges and hence inherently deals with the issue of damaged analog cards. Also, making the move to VoIP would allow for expandability for more phone lines to be added on. A cost-analysis on whether to remove and install a new PBX system completely or integrate VoIP functionality to the existing analog system favored the latter. The traffic usage was analyzed and used to determine trunk lines running between the analog and the IP-PBX enabling communication between both PBXs. This considered both internal and external calls, and the average call duration.

The expression [18] is used in estimating traffic flow.

$$A = C \, x \, T$$

where A = traffic flow

C = calls in a one-hour duration

T = call duration

The bandwidth required to support the determined traffic flow was computed. This bandwidth requirement influenced the choice of codec used to handle calls for optimal voice quality. A final decision was made on the popular G.711. This codec has a rate of 1Gbps and handles approximately 6250 simultaneous calls [19]. The influencing factor for this choice was because call quality was preferred over bandwidth consumption. To handle external calls, two types of trunks are available: VoIP gateways or VoIP service providers. These trunks are to connect VoIP users with the PSTN network, and have their merits and demerits. VoIP service provides provide an all set software system and removes the need for hardware installation. However, the administrator has little control over some features. Furthermore, it is also dependent on the internet and consumes the bandwidth. VoIP gateways give the administrator full control over all features. There is however an added cost in hardware installation and maintenance. The deciding factor fell on security; hence gateways were chosen to make the system less vulnerable. *Elastix* functioned as the underlying platform that the IP-PBX was built on. IP phones were used as terminal endpoint for placing calls. Analog telephone cards connected to the PBX phones allowed them to make calls via the VoIP gateways. A set number of digits with varying leading digits identified which phone system a call was placed from.

This study [21] focuses on the design and implementation of a campus-wide VoIP infrastructure using an Asterisk server. The voice quality was a big issue coming in, and great strides were taken to ensure that the quality of calls were top notch. Fedora, a Linux based operating system, served as the base OS for the Asterisk server. Linux servers are known to be superior when it comes to network configurations. This may be because they are lightweight and generally more secure. Tests were run in OPNET to test performance of asterisk. The system was loaded with calls so as to test the different codecs. The tests helped select the best codec for the Asterisk PBX. End to end delay is also tested using call loads. The existing system supports both LAN (100mbps) and Wi-Fi (54mbps). The network topology selected was centralized, with Asterisk at its epicenter. Users could register over both networks. Registration however remained limited for 20 users who have ID, username, passkey and extension. Static IP given for connection. The resources were allocated as per the extension given.

Many codecs with respective data rates were taken into consideration. Below is a table of them all.

Table 2.1: Codec Specification with data rates

Codec	Data Bitrate(Kbps)	License requirement
G.711	64 Kbps	No
G.726	16,24,32, or 40Kbps	No
G.729A	8 Kbps	Yes (no for pass through)
GSM	13 Kbps	No
iLBC	13.3 Kbps (30-ms frames) or 15.2 Kbps (20-ms frames)	No
Speex	Variable(between 2.15 and 22.4 Kbps)	No
ULAW	14 Kbps	No
ALAW	13 Kbps	No

To implement the server certain tools were employed: a pc compatible with the fedora platform, all asterisk packages downloaded and compiled. Next the firewall was configured to allow access to incoming and outgoing server connections. The SIP is also configured as a signaling protocol. Softphones are then installed on user devices and connected to the Asterisk server.

The softphone used is the X-Lite application. Using the network simulator, the topology was modeled on the computer. Now different call loads and codecs were used to test performance metrics. End-to-end delay is used to determine the codec having the minimum delay. Users were to rate the call quality by their personal experience from a range of 10. One finding was that the voice quality decreases as the number of calls increased, but the severity differed among the varying codecs.

S. Dhanalakshmi *et al* [22], explored how the system performance of an Ethernet LAN based VOIP in a campus network is affected by the traffic arrival distributions, voice codec schemes and increasing number of VOIP clients. In a comparative assessment, the study also investigated the performance of VOIP over wireless Ethernet (IEEE 802.11). With WLANs, users do not require a physical connection to the network, allow mobile users to connect from remote locations. This is very useful when network cables may not be available, or its use is not recommended. An IP network's services are usually shared by a number of utilities, services and devices. This will mean that integrating VOIP, which is a real time service with an existing IP network will require that measures to provide a quality of service (QoS) to users are ensured [23]. Such competition, when not checked, could result in delays and packets lost. Several tests cases on different simulation scenarios were considered to identify the optimum network conditions for implementing VOIP on the campus network. Two scenarios were generally used; a two-floor office and two different locations across Canada. The former had a local LAN call, WLAN local call, WLAN with interference and FTP under LAN as test cases whiles the latter was tested via a long-distance Ethernet. Through these simulations using the OPNET 14 software, they considered real time communication parameters such as packet end-to-end delay, delay variation for each call, and other Ethernet performance parameters which showed relatively worse VOIP performance of wireless networks than Ethernet.

S. Dhanalakshmi *et al*, identified that to achieve a reliable VOIP communication over a wireless network, a maximum number of simultaneous voice connections should be allowed. They continued by noting that findings indicated that for both Ethernet and wireless LAN, increasing the number of VOIP clients has a significant impact on the performance of VOIP calls. They established that, as compared to wireless LAN, Ethernet has a more stable and less delay connections which makes it preferable for VOIP implementation and scalability. According to the study, VOIP performance over WLAN can be improved by implementing effective queue management schemes, choice of voice codec, and payout buffer algorithms. Their work, however, was limited to performance analysis of only VOIP traffic, ignoring other traffics which may affect faithful voice transmission. Also, the test cases were all peer-to-peer voice calls. That is, advanced VOIP features such as conferencing and text messaging were not considered.

In this study [24], the researchers examined an existing Wi-Fi infrastructure that did not cover the whole campus. They sought to solve the call drop problem in dead zones. They proposed a dual mode approach to switch calls from Wi-Fi to GSM in dead zones. Another problem arose, that is uncomfortable call discontinuity as the exchange was being made. Finally, they employed cross-layered signal processing algorithms to mitigate the call discontinuity issue. Further tests showed stellar performance and good prospects for future studies on their findings. In this paper, the university campus (National Taiwan University) has SIP-based VoIP service replete with the necessary infrastructure that allows users to place calls over the WLAN, provided they are in coverage zones. The study seeks to find solutions to user discomfort in using the service caused by dropped calls and calls not going through at all.

A solution proposed is the dual-mode communication. This seeks to employ the capabilities of cellular coverage to extend the reach of calls over WLAN. Basically, it works this way. A user makes a call over WLAN. As he moves along he enters into a place out of the WLAN coverage.

His call is automatically transferred to cellular GSM to make sure the call continues. Should he move back into a WLAN coverage area, the call is transferred back to the WLAN. A scaling factor of 1.25 was chosen to extend an audio portion of 1.2s to mitigate a 300ms delay. To check the forthcoming performance, 30 listeners were surveyed. Each was made to listen to both the audio with the delay and the audio that had the time-scaling algorithm applied. The listeners were to identify when the audio delay occurred. Then the listeners were to rate their experience on a scale of 1-5. The audio gap stream scored a MOS of 2.1 whereas the time-scaled stream scored a higher MOS value of 3.7. This proves that the time-scaling algorithm gives a better user experience and deals with the audio gap problem.

George Asante [25], aimed at designing a 3-tier architecture VOIP system to allow communication over the data network at Kumasi Technical Institute (KTI) at virtually no cost. Leveraging on the Local Area Network on the campus connecting the school's four departments with the administration block, he proposed a cost-effective communication system with voice and video call, text messaging and file sharing capabilities.

Asante's 3-tier architecture was made up of a main and database server hosted on the same computer and a Java-based user application. The main server contains the address-to-location mappings for each user whilst the database server contains information about users. All the users would be registered on the server so the server can broadcast all users. To start, a user creates a network peer by establishing network links with their friends. This is the network from which a user can accepts calls from. A client software is installed on the communicating computer or android mobile phone to provide graphical interface for initiating calls and file transfers. Also, to handle the transmission of media, the Java Media Framework (JMF) API was used. The design provides users with services such as text chat, audio chat (voice calls), video chats and file transfer.

However, the proposed system had no capabilities for voice and video conferencing. Also, the client application for mobile devices was made to only run on devices with Android operating system.

In this paper [26], the authors sought to integrate a video voice over IP in the existing network and test its performance. The problems experienced were call quality degradation during peak hours. Their objectives were to develop V2oIP services on campus and to make a comparative analysis of the performance of V2oIP over USB video phone and softphone. To monitor performance Network Management Systems were used. They took records off softphones to determine peak periods.

The development stages of the V2oIP implementation were:

i. Planning and research, where extensive study was done on hardware and software requirements.
ii. Development of the network design.
iii. Implementation of the software and hardware installations over the network.
iv. Testing the USB Phone and Soft phone for performance. Performance metrics that were tested include delay, jitter, CPU usage and packet loss.

The software used for analysis and measurement are VQnet and Colasoft Capsa.

The experiment was carried out on a dual basis: Peak periods and Non-peak periods. To capture as closely as possible traffic was measured every two minutes.

Through simulation methodology, Abdul-Bary *et al* [27] sought to integrate VOIP service with the existing IP network infrastructure at Mosul University to enhance the communication experience of the staff. The aim of the study was to model and analyze the performance of the campus network, first without VOIP and then after the deployment of VOIP. Their methodology encompassed four development stages; planning and research, development, simulation and, testing. Like most institutions, the conventional communication system at Mosul University was a Panasonic phone system using the traditional analog telephone system. The best service this system could provide was only calls between instructors on the campus. According to the study, motivation for the integration of VOIP came from the robust network infrastructure provided by the fiber optic backbone cables covering the campus, with VLAN enabled Cisco Switch 2950 distributed on the campus. The various departments and colleges of the University connects to the IP-network in a star topology via fiber optic cables. Their proposed VoIP architecture consists of the user agents (IP phones or softphones), a SIP server and Gateway. The Panasonic PBX (KX-TDE600) which serves the traditional telephone system is also able to connect to the campus LAN through a virtual Gateway card. This is shown on the figure below. [18]

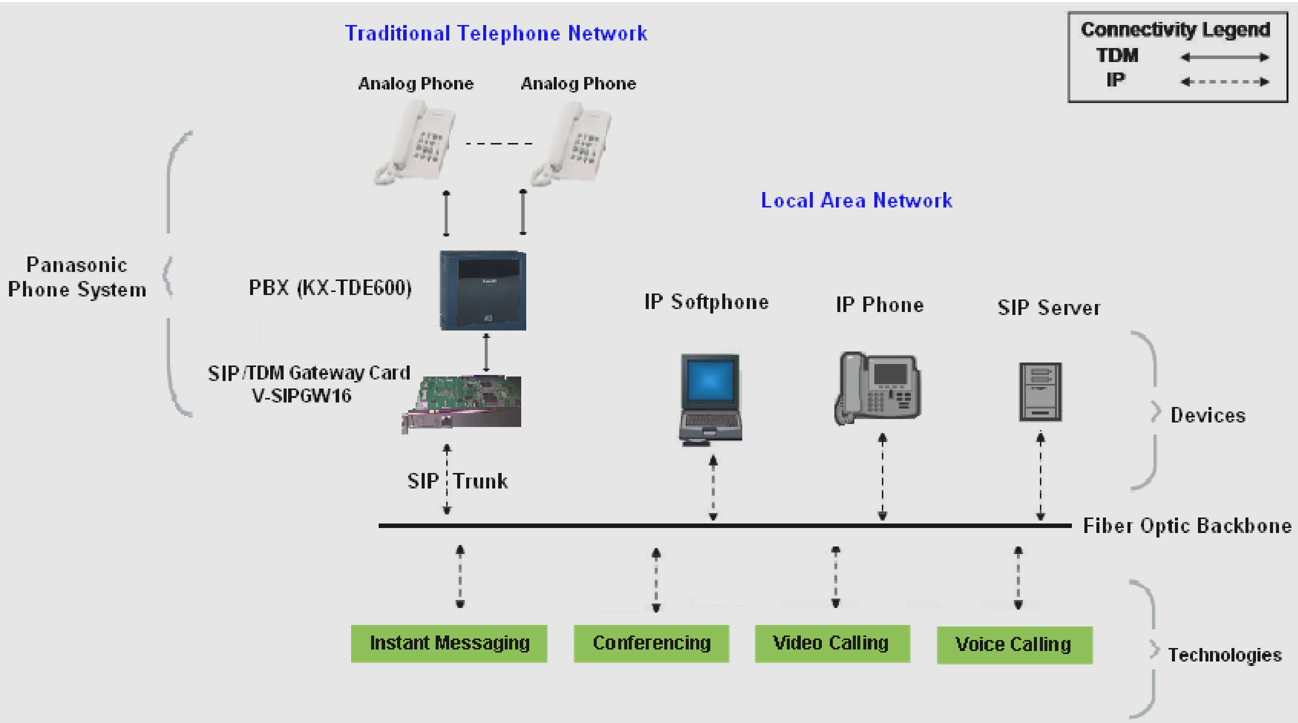

Figure 2.1: VoIP Architecture for Campus Environment.

Using the OPNET Modeler 14.0 [28], various network performance simulations were carried out. Three different VOIP scenarios were also undertaken to identify the best codec to use for the deployment. IP phones were added to the subnets to generate VOIP traffic was then added to the network traffic to test the network performance. With the average duration of a call set to 30s, results of the tests indicated that the simulation time increases with an increase in the number of SIP calls. Studies has shown that voice communication can only tolerate delays which are less than 150ms [29]. The G.732 codec recorded a delay of 120ms and the G.711 codec, a delay of 22ms. On voice quality, the G.711 had a MOS factor greater than 4.0 [30] which indicated a good voice quality performance of the proposed system. The Mean Opinion Score (MOS) is a qualitative voice quality measure, expressed as a number between 1 and 5, with 1 being Bad performance and a score of 5 indicating Excellent voice quality.

Tests performed to determine delay variations or jitters also revealed the value 0.5ns recorded for the duration of 5mins increased sharply when the number of simultaneous calls was raised. The higher the variation in delays during a VOIP call, the greater the degradation in voice quality. Bandwidth tests also showed that G.711 expense larger bandwidth as compared to G.723 and G.729 codec. However, the G.711 was chosen because of its better value of jitter, delay and MOS factors.

2.2 EXIST1NG NETWORK SYSTEM IN KNUST

2.2.1 KNUST WAN Infrastructure

The Kwame Nkrumah University of Science and Technology (KNUST) main campus is located at Ayeduase, about eight miles (13 km) to the east of Kumasi, the Ashanti Regional capital [31] of Ghana. The campus hosts a student population of nearly 50,000 [32].

The KNUST main campus has an established Wide Area Network (WAN), with the network backbone riding on a single link to the outside world with a bandwidth of 144 Mbps as of June, 2015. There is a cascade of firewalls, gateways and switches routes the link from the Network Operating and Infrastructure Department (NOID), from which the link enters the KNUST network. Connectivity to the 28 faculty areas, including the main administration block, main school library, central classroom block and Dean of Students office are established by fiber optic and Ethernet cables from distribution switches. Additional services provided by Vodafone Ghana, the Internet Service Provider (ISP) include support for VoIP [33].

The network structure of the KNUST WAN is as illustrated in the [33, Fig 1].

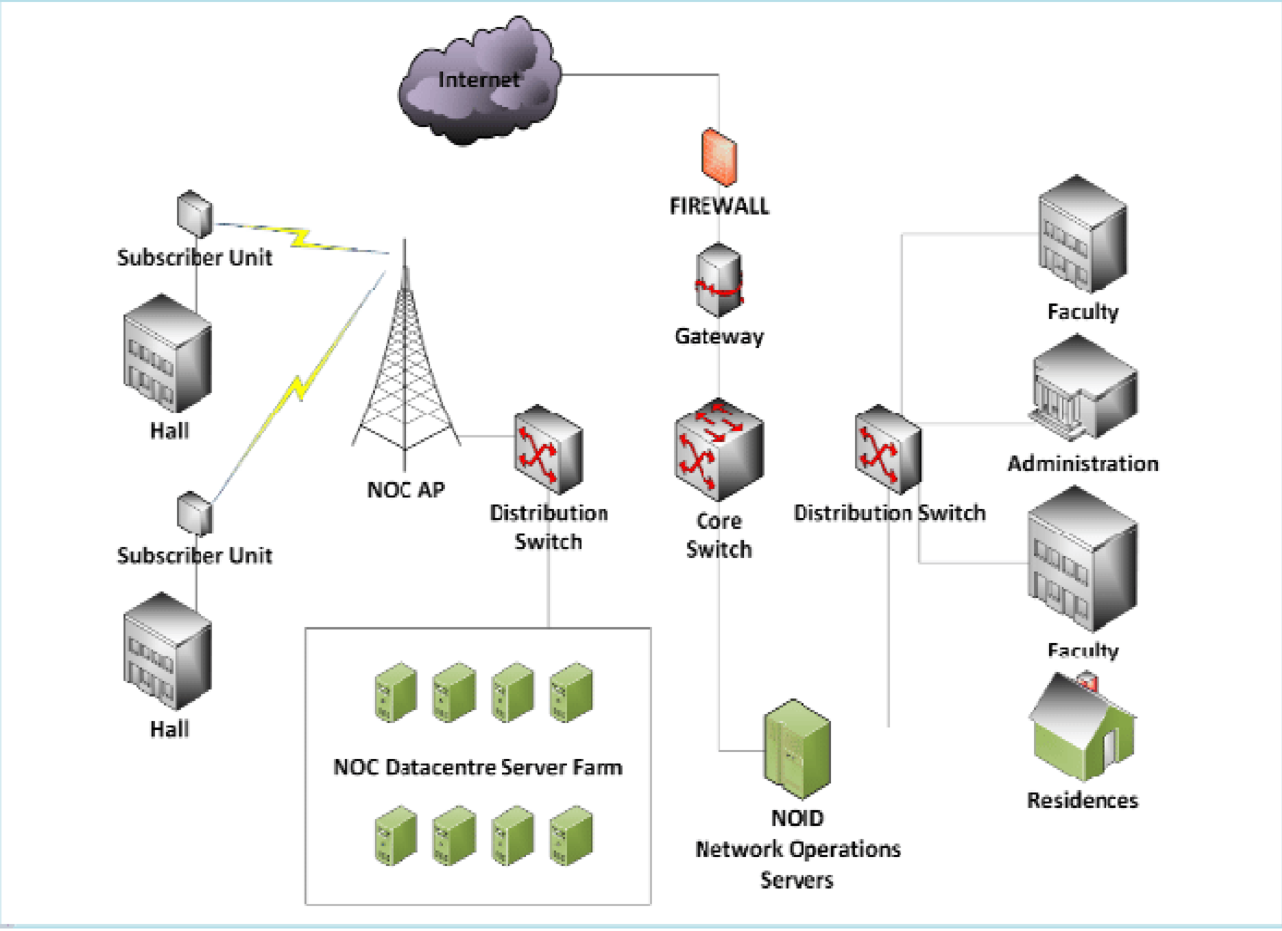

Figure 2.2: Simplified Network diagram of KNUST WAN [33]

2.3 PUBLIC SWITCHED TELEPHONE NETWORK (PSTN)

The Traditional telephony, developed in 1878 has evolved over the years. Alexander Graham Bell, considered as the inventor of the telephone, accomplished the first voice transmission in 1876, which set of the development of the telecommunications industry.

PSTN, generally used to represent the traditional telecommunication network. It was developed from the automated telephone switching technology referred to as the Plain Old Telephone System (POTS) and uses the circuit-switching technology for making calls [34]. In Circuit switching, resources such as telephone lines and hubs are used to establish a virtual circuit between the two endpoints when a voice call is initiated. This denies access to the channel by other calls during the entire call duration. This reserved circuit for calls ensure guaranteed delivery and a uniform guaranteed quality of voice throughout the call duration.

2.4 VOIP TECHNOLOGIES AND TERMINOLOGIES

2.4.1 Session Initiation Protocol (SIP)

The session initiation protocol is a protocol used to control multimedia sessions by establishment, maintenance and termination. SIP is an application layer protocol that has been standardized by the Internet Engineering Task Force (IETF). It functions as the basic framework for the text-based protocol RFC 2543. SIP works well in conjunction with other protocols like H.323. [35] [36]

A SIP URL identifies a SIP host, and addressing can be individual or for a group of users. A client can then locate the host via a SIP server using its corresponding IP address and port number. Once connected the client can then send requests and receive responses. This exchange can be done over a connection-oriented TCP or a connectionless UDP [37]

2.4.1 Codecs

Codec is a short term for coder-decoder. Its function is to digitize the audio inputs as packets and compress them for transmission over IP. One of the most used today is G.711. It provides an audio quality of 64 kb/s. G.711 is an integral part of the H320 and H323 protocols.

2.4.2 Latency

Latency is the delay or the amount of time it takes to send information from one point to another. It is more graphically explained as the taken from when speech is released till when it is heard by the listener. Latency is measured in milliseconds (ms). The ITU-T recommendation for latency on VoIP is 0ms – 150ms.

2.4.3 Packet Loss

Packets are the constructs that carry data from one point to another. Packet loss is when a data-carrying packet or packets traversing a network is unable to reach its destination. By ITU-T standards an upper limit of 1% loss is acceptable in voice over IP communication systems. Packet loss is expressed as a percentage.

2.4.4 Jitter

During communication, there are sometimes variations in latency or delay of packets carrying voice or video data over a communications channel. Jitter is the name given to this phenomenon. Jitter is measured in milliseconds (ms). ITU-T recommends a maximum of 100ms when operating with real-time data transmission like VoIP.

CHAPTER THREE: RESEARCH METHODOLOGY

3.0 INTRODUCTION

In this chapter, we present and justify the methodology adopted for this research. Section 3.1 discusses the Simulation methodology and introduces the OPNET Modeler 14.5, which is the simulation tool selected for the design. Section 3.2 covers the analysis of the existing network and VoIP system in KNUST. Section 3.3 then discusses the design stages undertaken in this research.

3.1 SIMULATION METHODOLODY

3.1.1 Simulation

Simulation methodology was adopted in this study for the modelling and analysis of VoIP over Ethernet LAN. The availability of various sophisticated and powerful simulation software packages has contributed greatly to the increasing popularity of the simulation methodology among computer network researchers [38] [39].

Simulation, as defined here [40], "is the process of creating an abstract representation of an existing or system in order to predict the behavior of the system." With simulation, you can predict the performance, in terms of strength and weaknesses of a system before the actual implementation of the model. The high degree of flexibility in model construction and the ability to easily control the scale of the network also makes simulation an ideal tool for studies on modeling and network performance quality. In most cases, a network may contain a large number of network nodes and that will be very costly and time-consuming to establish the physical network for studies.

However, current network simulation tools require a high learning curve on the side of the user due to the growing complexity of the simulators [41]. The user, therefore requires extensive knowledge to ensure that the simulation results are meaningful.

This study is going to design a proposed VoIP system integrated with PSTN using the OPNET simulator. Some simulation-based studies for VoIP network system design embarked on in recent years include [39], [42], and [43].

In their study, Zubairi & Zuber [39] developed a simulation-based model of a campus network with OPNET. This network simulation model was used to obtain the Ethernet delay, traffic statistics and other interesting data. To determine the capacity of the model to handle the demanding voice applications under different traffic load conditions, they also ran an interactive voice across the demanding voice across the network. Their results show very good performance under typical load conditions whilst the delays and jitters increase under loaded conditions.

Rabassa [42] presents a study involving the creation of a tool aimed at supporting the planning and design phases of IP networks carrying voice traffic while considering the network conditions and simulation features. Specifically, he develops a model to integrate real speech to the VoIP simulation offered. His study also involved the implementation of common speech quality assessment methodologies.

Capelle, et al. [43] also designed a campus network in OPNET and tested the VoIP traffic on a shared Ethernet. They investigated the network performance in case the university network offers VoIP services for each student room. Their reported results include voice end-to-end delay, delay variation for each call and Ethernet parameters.

This study will use a methodology similar to one adopted in the studies identified above. They can help to deploy a VoIP network by giving insights such as "how to generate VoIP calls." The studies also give a fair idea of what results are useful when analysing a proposed VoIP system.

3.1.2 The OPNET Modeler 14.5

A number of simulators exists for network modeling and performance analysis. Popular among them are the Optimized Network Engineering Tool (OPNET), QualNet Developer, NetSim, Shunra Virtual Enterprise 5.0, Ns-2, GloMosim, OMNeT++, P2PRealm, The Georgia Tech Network Simulator (GTNetS), and AKOROA.

OPNET Modeler 14.5 is adopted as the simulation tool for this study. OPNET is an object-oriented network simulator which provides a comprehensive development environment for the specification, simulation and performance analysis of computer and data communication networks [38].

OPNET uses a three-tiered hierarchical architecture for topology, data flow and control flow. Each tier describes a different aspect of the complete model being simulated. It has an easy to use GUI structure which enables users to design, simulate and view the results with on the go.

OPNET Modeler has been selected for its GUI and ease of use.

3.2 ANALYSIS OF EXISTING SYSTEM

3.2.1 Description

As expressed in the proceeding chapters, KNUST has a VoIP system deployed on the university's Ethernet LAN infrastructure. This makes the VoIP service accessible on the whole network across the university.

The system uses the Linux-based Asterisk software Internet Protocol Private Branch Exchange (IP PBX). Specifically, the open source Asterisk based *Issabel* software, which integrates a PBX, mailing and a database server is used.

Currently, it serves about two hundred and fifty (250) IP phones spread across faculty and administrative buildings on campus. The design exploits the VLAN technology implementation to separate the VoIP traffic from other traffics on the network. This is done to allow sufficient resource (bandwidth) allocation which will ensure quality voice and video calls. Network users can also access the service via softphones installed on their devices. VoIP calls are usually not concurrent.

3.2.2 Limitations

i. VoIP service is not available on the university Wireless Local Area Network, which is mostly used by students via access points in the faculties and halls of residence.

ii. There is no integration with the PSTN system. This means IP Phones are not able to make calls to the traditional telephones.

iii. The usage of the IP Phone is limited to administrators. Lecturers would have to use a softphone on their devices or like students, have to make GSM calls to their colleagues even when on the network.

iv. The Accra City and Kwabenya campuses of the university do not have VoIP deployment.

3.3 SYSTEM DESIGN AND DEVELOPMENT

3.3.1 Campus Network Modeling with OPNET

Here, we describe our effort to model the KNUST Kumasi campus network with the proposed VoIP integration and to simulate voice traffic between clients while measuring various voice quality parameters such as jitters, packets loss, total bytes sent and delays.

The network infrastructure on campus makes extensive use of Cisco product. For that reason, almost all the network devices used in our design consists of Cisco switches and routers.

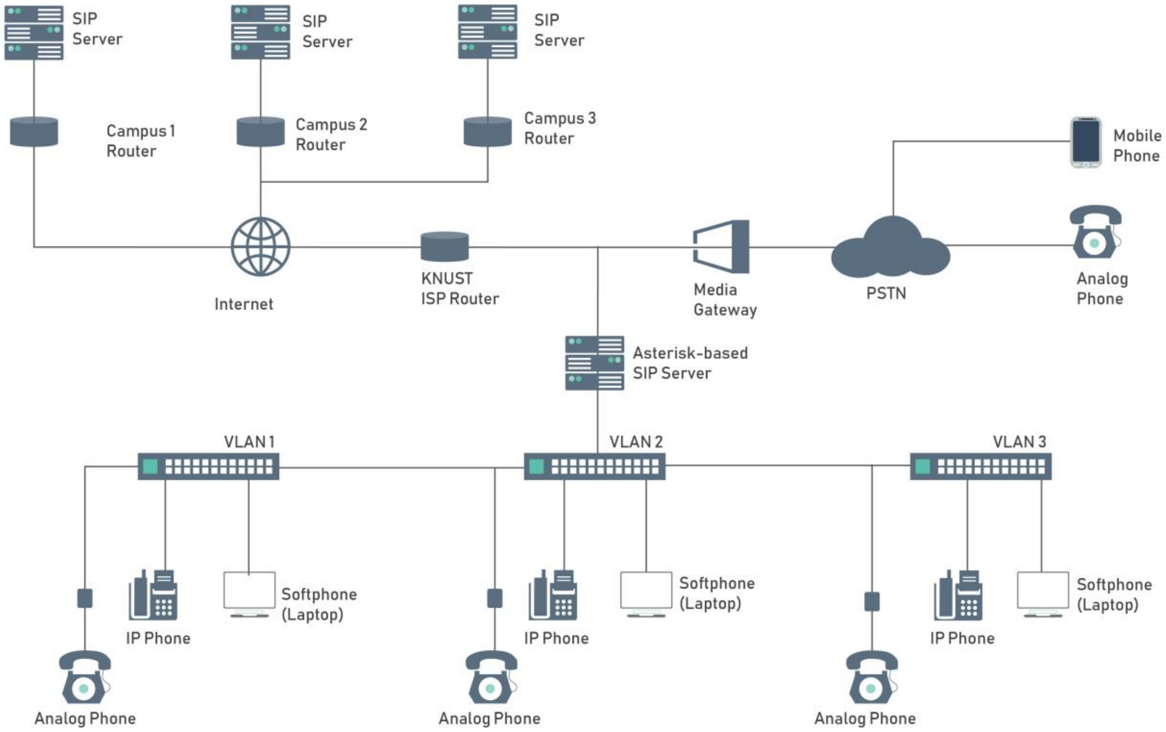

Figure 3.1 Proposed VoIP network diagram

3.3.2 Routers

A router is used to make the decisions regarding the best path for packet transmission on a network. The Cisco 2600 router will be used in order to make the study more demonstrative.

The default value parameters in OPNET are used with a forwarding rate of 25,000packets/second. The router attributes configuration is shown in Figure 3.2

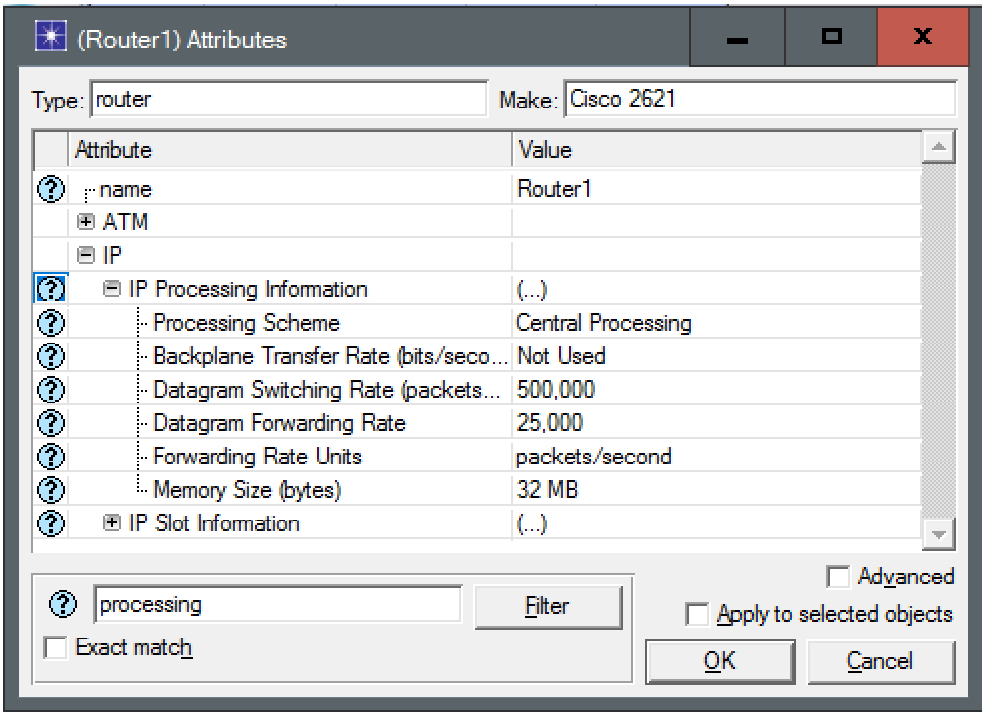

Figure 3.2 Router attributes configuration

3.3.3 Switches

A Cisco 6509 switch is used as the core switch to serve as the core of the computer network backbone. To increase the number of ports connected to the router, a Cisco 2948 switch is used. Similar to the routers, all parameters are configured in default values. The Cisco 6509 switch has a switching speed of 150,000,000 packets/second while the Cisco 2948 switch has a switching speed of 46,875,000 packets/second. Figure 3.2 and 3.3 shows the switch attributes configuration.

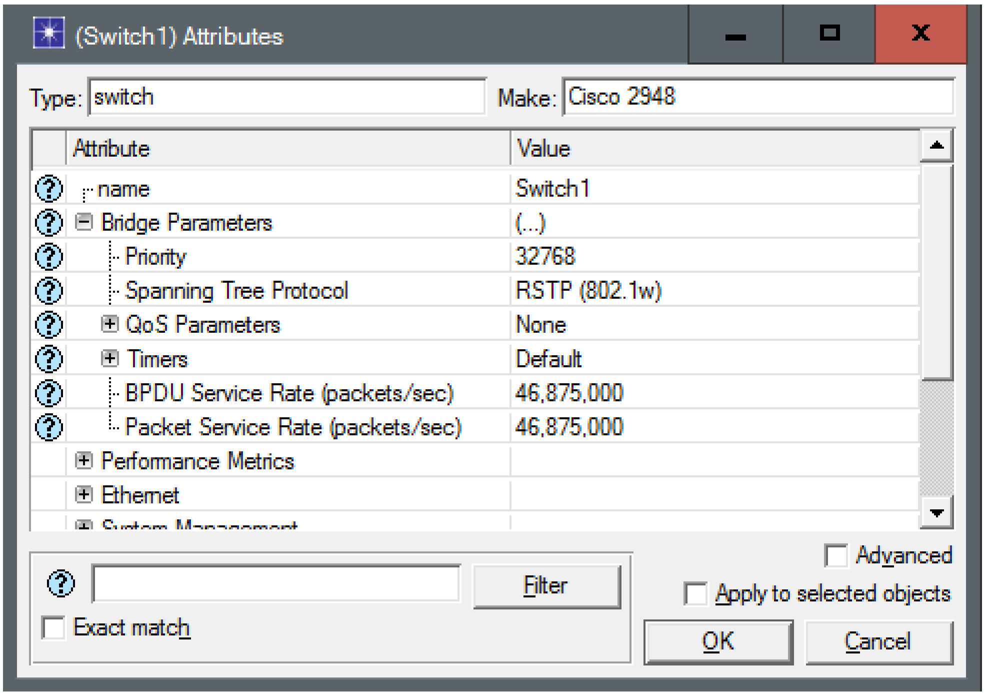

Figure 3.3 Core 6509 Switch attributes configuration

3.3.4 Cabling

The wiring consists of 10Gbps fiber optic cables, connecting VLAN switches to the core switch and T-1s are used for internet.

3.3.5 Server

In a VoIP network, VoIP server is set up to communicate with the IP phones, softphones or traditional phones to facilitate the users [44]. They are computers with specialized software which facilitate calling in a VoIP network. A Linux-based AMD ASUS system is used.

3.3.6 Media Gateway

A media gateway is a translation device to provide transformation and the interconnection of different telecommunication standards, protocols codecs and other technologies. This will allow voice calls to be made between the VoIP network the Public Switched Telephone Network (PSTN).

The OPNET model of the proposed design is shown in the Figure 3.5.

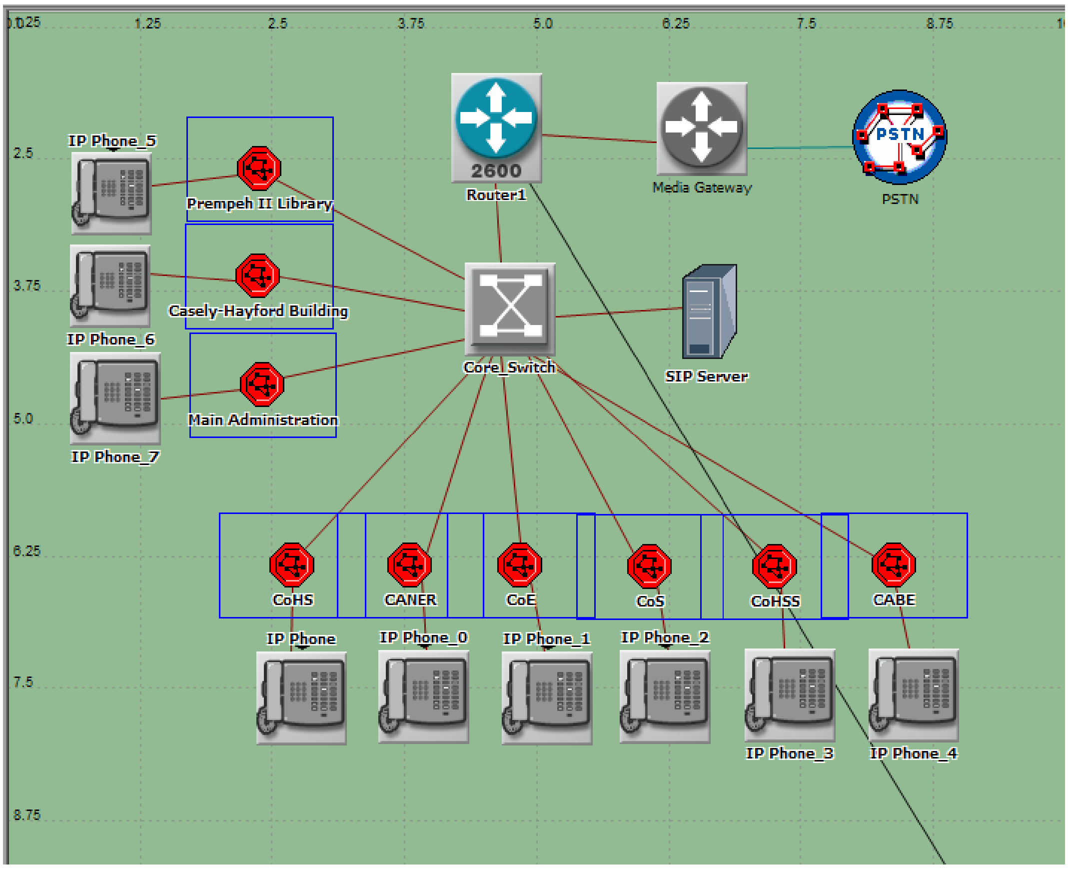

Figure 3.4 OPNET model of the proposed VoIP architecture.

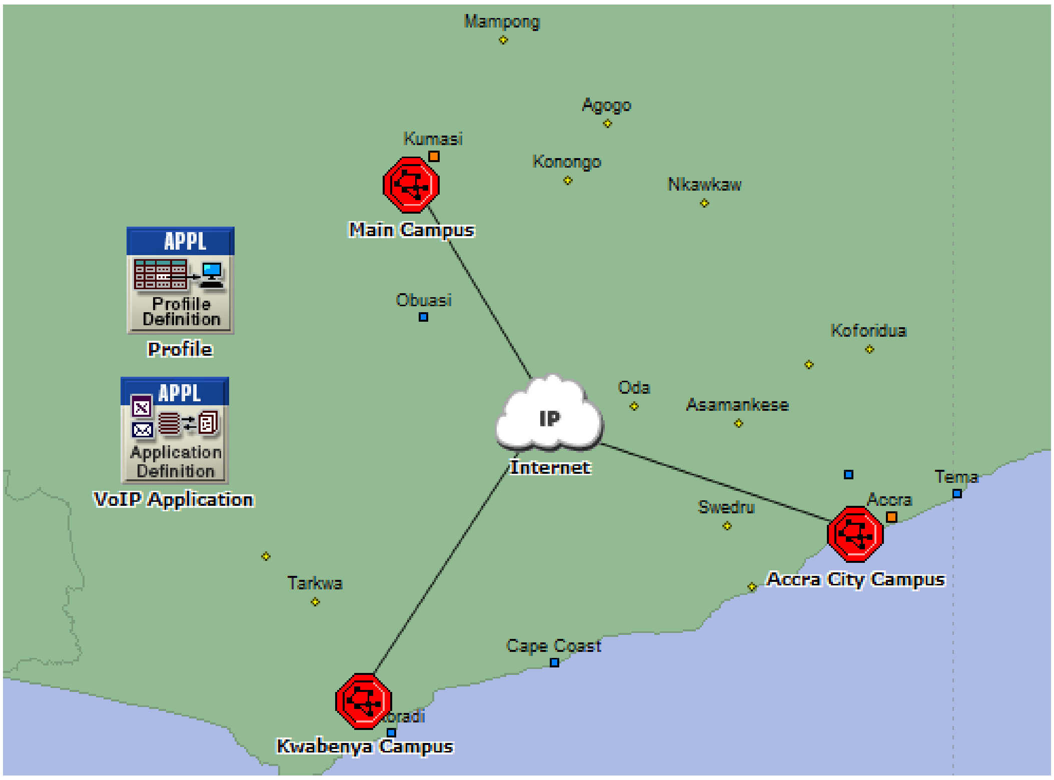

Figure 3.5 Map view of proposed design

3.3.6 VoIP Traffic Settings

The Application definition and Profile definition are used to model an application in OPNET. Therefore, a VoIP application has to be set up. The configured parameters of the VoIP application are shown in Figure 3.6.

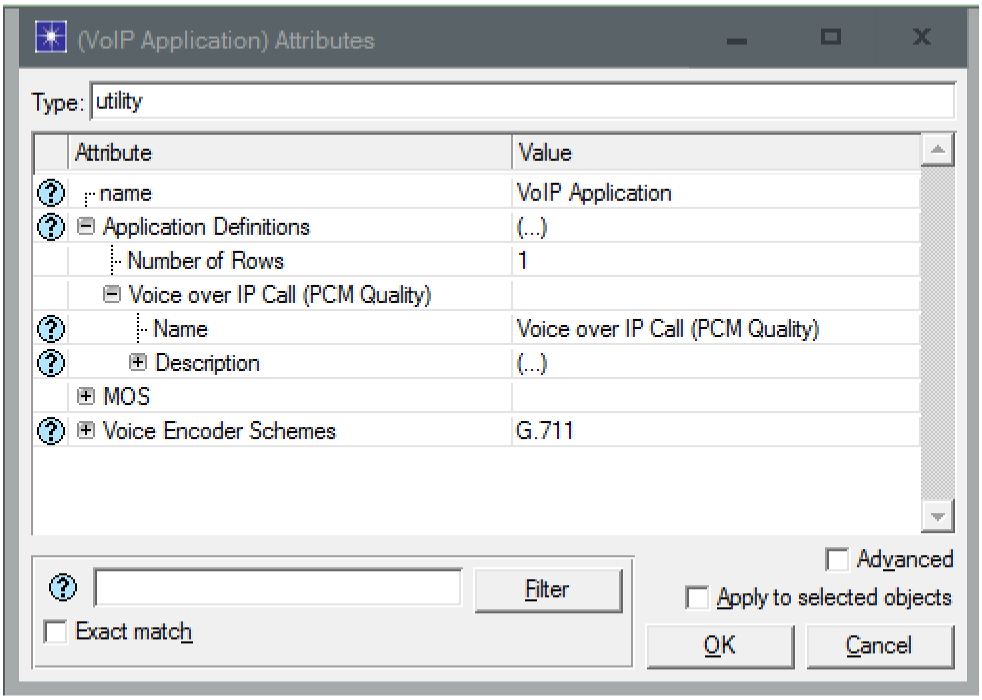

Figure 3.6 VoIP application configuration.

The "Voice Encoder Scheme" is set G.711 which has been adapted for this design. The G.711 codec offers a worst-case LAN bandwidth requirement since it consumes the most bandwidth. However, it provides the best voice quality as compared to other codecs.

Next, to allow the nodes to implement this VoIP application we have created, we need to specify the profile definitions. A profile is used to define the behavior of a network workstation. Here, only one profile would be used, called 'Profile'. The profile is configured to make workstations to generate VoIP calls.

Workstations would generate VoIP calls at a constant rate. The Profile is configured as shown in Figure 3.8

i. The attribute "Start Time Offset" of the VoIP traffic is set to 60 seconds.

ii. Attribute "Start Time Offset" is also set to 60 seconds.

iii. “Repeatability” is set to “Unlimited” and “Interrepetition time” to 5 seconds. This will make the workstations to generate calls every five seconds. This configuration is shown in Figure 3.8

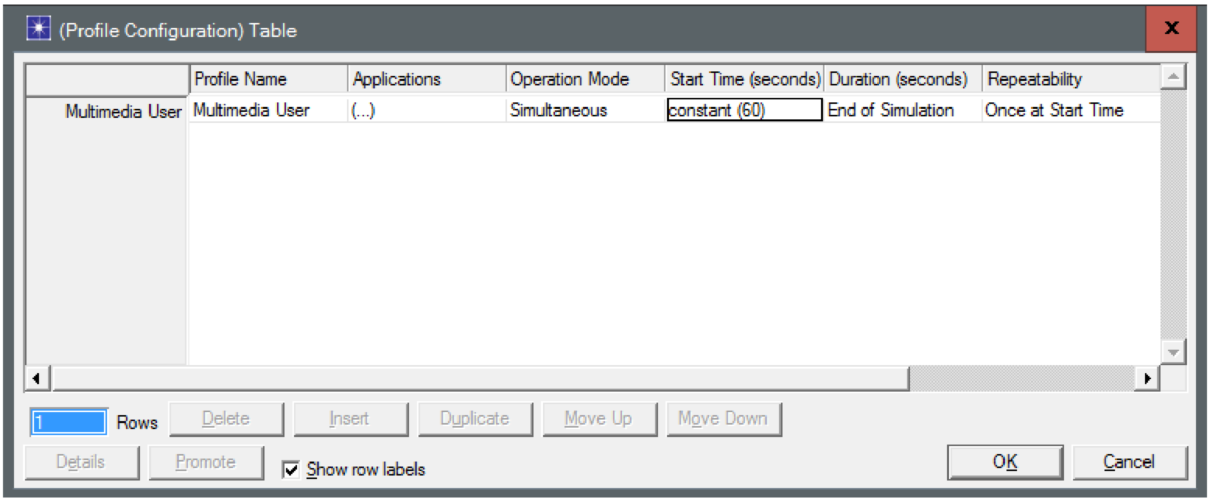

Figure 3.7 Profile Configuration

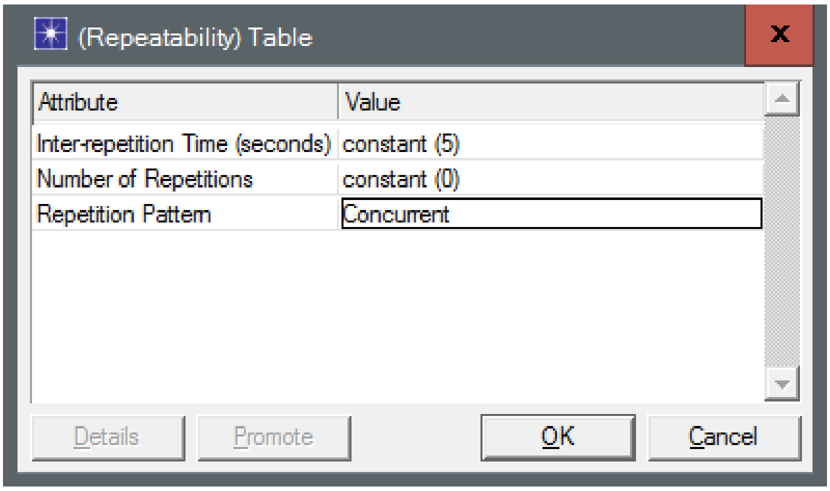

Figure 3.8 Inter-repetition Time

3.3.7 Simulation

The simulation is configured to run for eight minutes to determine the network functionality. The results on traffic, jitters, end-to-delays and packet loss are studied to determine the performance of the proposed design.

3.4 SUMMARY

In this chapter, we have discussed the methodology used for this study, a highlight of the state of the current VoIP infrastructure in KNUST and a detailed discussion of the development states of the proposed design. The OPNET Modeler 14.5 was used to model and simulate the new network.

CHAPTER FOUR: RESULTS AND DISCUSSSIONS

4.0 INTRODUCTION

In this chapter we take a look at the proposed design metrics, how the PSTN is integrated with it, and making calls to external campuses. In Section 4.1 we present the results from our simulation of the proposed design in the previous chapter. Using the OPNET network simulator, we run tests on the design to check ITU-T recommended parameters for Voice over Internet Protocol networks. These are outlined as graphs and further discussions made on the results. Section 4.2 is on overview on the integration of the Public Switched Telephone network with the model in Section 4.1, along with some control implementations. Section 4.3 deals with calls made to external campuses.

4.1 RESULTS OF SIMULATED NETWORK

4.1.1 Jitter

Jitters, or variation in packet arrival times, leads to unpredictable performance. To avoid this phenomenon, real-time data like voice must be kept below 100ms for quality call experiences. The result of the jitter in simulated network is shown below.

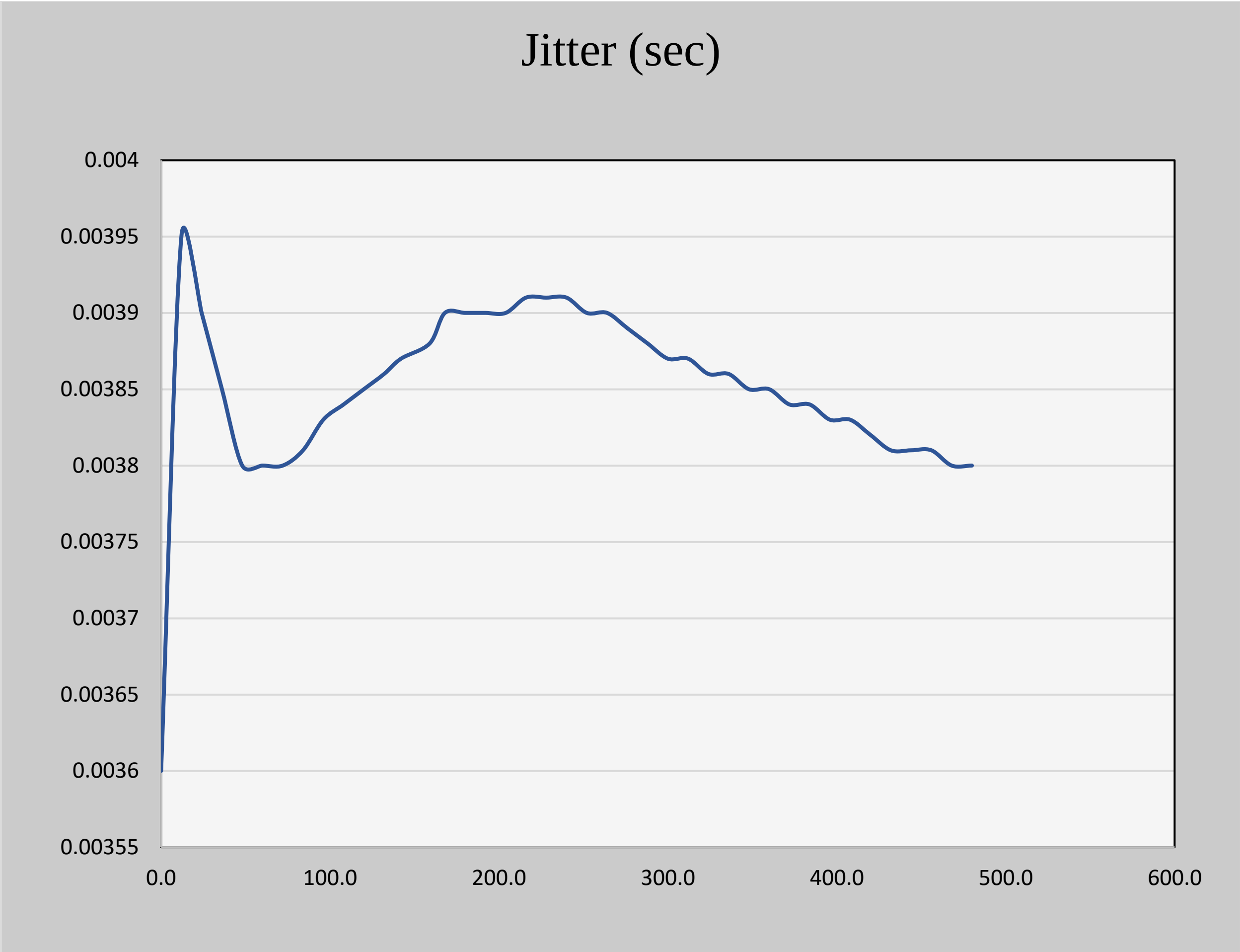

Figure 4.1 Results of jitter in simulated network

The simulation was run for a time of eight (8) minutes, which is a total of 480 seconds in the graph. Over this time there is a spike at the beginning of transmission to a peak of 3.97ms, which then drops to 3.8ms. It rises up slowly to a little above 3.9ms and comes to a relatively settled value of 3.8ms. Our average value of jitter in the simulation rests at a value of 3.8ms. The ITU-T recommendation is 100ms, and hence this value is very desirable in the network.

4.1.2. Latency

Latency, or end-to-end delay, is the total time from when speech is released by the speaker till when it is heard by the listener. Long delays in transmission of voice packets are unacceptable and degrade the performance of the VoIP system considerably. ITU-T recommends a range of 0ms – 150ms for latency in voice transmission systems. In the simulation, we chose the G.711 codec. This choice was made due to the codec's ability to offer very low end-to-end delay, thus producing a call experience in similitude to PSTN telephony networks. This is because there is little to no buffering needed. The results of end-to-end delay in the shown in the figure below.

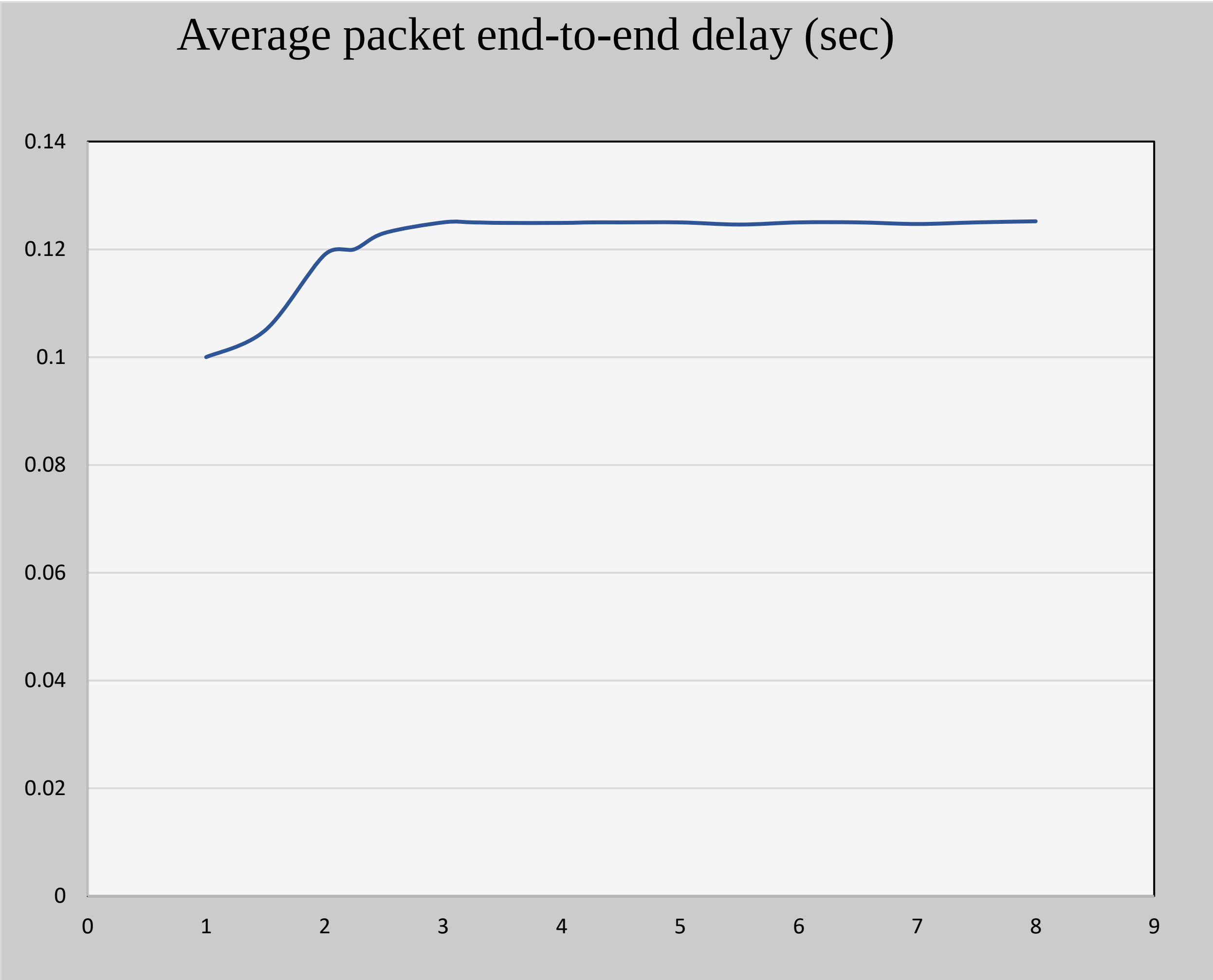

Figure 4.2 Results of end-to-end delay in simulated network

The simulation was run for a total time of eight (8) minutes, which is a total of 480 seconds in the graph. Over this time, the latency starts out at a value of 100ms and grows to a peak of 124ms. Here we realize a stabilization of the value around the peak value. An average value of 115ms is calculated as the end-to-end delay of the simulated design. This is lower than the recommended standard of 150ms and hence would deliver quality voice service.

4.1.3 Mean Opinion Score

The Mean Opinion Score, or MOS, is a methodology used to access the human-judged quality of voice in a network. It uses a range from 1, which signifies bad performance to 5, which represents excellent. Acceptable voice calls are above 3.5. The figure below shows the MOS of the simulated network.

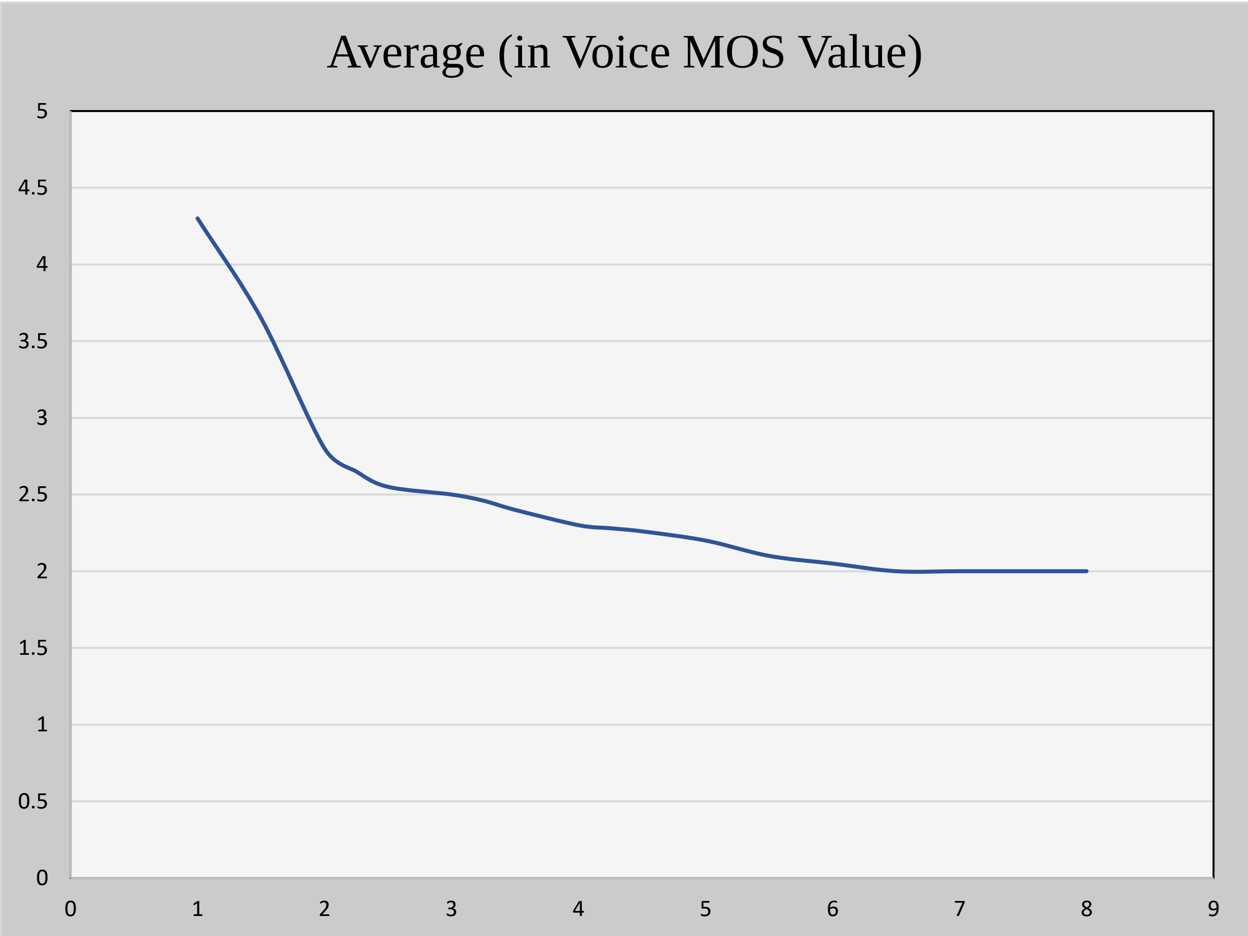

Figure 4.3 Results of MOS reading in simulated network

The MOS of the simulated design lies within a range of 2 – 4.3. This is common is acceptable to make voice calls.

4.2 PUBLIC SWITCHED TELEPHONE NETWORK INTEGRATION

Public Switched Telephone Network (PSTN) refers to the global network of interconnected telephone lines across local, national and international telephone network providers. Research shows that telephone bills for campuses that make use of the traditional legacy communication systems are high. In KNUST, there used to be an available PSTN network. However, this was discontinued due the high costs incurred by the school in its usage over time. One major contributing factor was the misuse of the service by users. The use of the service for personal calls and also misuse for long calls contributed to incurring high telephone bills. Many measures to bring this to a low were inadequate as these measures only addressed the issue after the harm had already been done. We seek to address to address this issue using proactive measures as outlined in the following sections.

4.2.1 Controlled Access

Access control is limiting the number of users that are privy to a particular resource, and stating who is to have access to what. This can be employed as a way of curbing the misuse of the PSTN integration. In this implementation, users are registered as extensions and their access levels declared. Users who have need of access to the PSTN service are identified by their extension numbers and initialized with PSTN access. All other users, whether on IP phones, analog phones or softphones that have not been initialized in this manner cannot make calls through the PSTN network. A code snippet of controlled access initialization in the Asterisk-based Issabel PBX included in the appendix.

4.2.2 Call Time Regulation

A control factor that could be implemented is regulating the time given to each authorized user to make calls through the media gateway, that is PSTN calls. For explanation purposes only, we assume every user is given 200 minutes' worth of PSTN calls per month. This limit is reduced with every call made. Should any user reach that limit in a month, the user's extension is automatically prevented from making any more calls through the PSTN for that month. This helps to curb the use of office resource of PSTN calls for personal purposes and also avoid wastage. This implementation is developed as a script of code embedded in the IP-PBX server. A flowchart of the algorithm is included in the appendix.

4.2.3 Extensions

In addition to all the control factors aforementioned and discussed, another way by which the misuse of the PSTN resource on the VoIP system is the provision of extensions for PSTN calls. All calls to be made through the media gateway must be prefixed by a specific extension number. This number should not be made public but only known by administrators and members of staff who have authorized access to make calls through the PSTN. This way, even when unscrupulous personnel try to make PSTN calls with authorized administrator phones they will be automatically barred.

4.3 CONNECTING KNUST TO SATELLITE CAMPUSES AND OTHER UNIVERSITIES

In the confines of KNUST campus, calls made are locally routed and therefore do not make use of the internet. VLAN logically separate physical buildings and allow for easier setup and configuration. However, making calls to the other satellite campuses and universities needs to be routed externally. This means that it would have to go through KNUST's ISP router to be directed to the router of the destination campus over the internet. During a voice call to other campuses, the call contends with all other internet requests being made on the router. VoIP calls are UDP based and hence require a reasonable amount of uninterrupted bandwidth for good performance. Hence, during peak-use hours of the day, the quality of externally made calls may greatly degrade. In this section we present a means of overcoming this phenomenon.

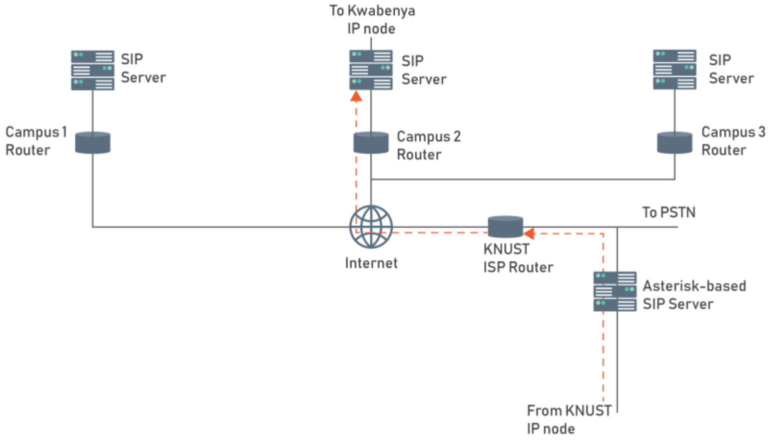

Figure 4.4 Graphic display of call to external campus

4.3.1 KNUST Bandwidth

By voice over internet protocol standards a voice call using the G.711 codec requires an average bandwidth of 100kbps. Below are the internet service providers for KNUST and the bandwidth the provide:

Table 4.1 ISP providers and the bandwidth provided to KNUST

ISP Network	Bandwidth Provision
MTN	1 Gbps
Vodafone	1 Gbps
Ghana Academic Research Network (GARNET)	1 Gbps

The bandwidth provision by all three providers makes up for a total of 3Gbps. However, a voice call can only be routed through only one of them at a time. In the next couple of sections, we discuss measures to ensure peak call quality for externally routed calls.

4.3.1.1 Static Bandwidth Allocation

One solution to prevent contention between voice calls and other internet requests is to statically allocate a fixed bandwidth purposely for voice calls. This ensures that whenever an external call is made, whether from or to KNUST, there is sufficient bandwidth to support it. There is one downside to that approach. Many VoIP calls are not concurrent, and also calls typically do not last very long. There are many hours, especially during the night, where users make or receive few or no voice calls at all.

Declaring a bandwidth allocation for voice calls alone at all times will lead to low utilization rates of the allocated bandwidth resource.

4.3.1.2 Dynamic Bandwidth Allocation

The proposed approach in this study is to dynamically allocate bandwidth as at when it is needed. Routers have the inherent capability to handle bandwidth allocation, and we seek to leverage that in the implementation of this method. When a user makes a voice call that is over the internet, the router recognizes this through the description in the packet headers. It then proceeds to allocate an amount of bandwidth that is sufficient to effectively manage the call during its session. The call is placed on this channel and send through the internet. When the call is terminated by either user the allocated bandwidth is returned to the rest of the channel stream. The same goes for incoming calls. When the router recognizes an incoming VoIP from any location, a bandwidth allocation is made to handle the call for its duration. This method is handling calls ensures calls have sufficient resource for good quality. Also, there is a very high utilization rate, and allocated bandwidth is promptly returned to the rest of the bandwidth channel stream when the call is terminated.

CHAPTER FIVE: CONCLUSION AND RECOMMENDATIONS

5.0 CONCLUSION

In this project, we have worked on analyzing the current underlying network infrastructure KNUST. We have also taken a look at VoIP system deployed on it, its performance metrics and what could be done by way of improvement. In this study we proposed an integration of PSTN on the current IP telephone network. Also, control factors were designed to order to effectively deal with the issue of misuse of PSTN services. In the later part of the study we take a look at how to effectively communicate with KNUST's satellite campuses and other universities. Our final design turns out to be one that is both easy to implement and low cost.

5.1 CHALLENGES DURING THE STUDY

The current VoIP system is being managed by the UITS department. Due to the sensitive nature of the network, access was highly guarded and very limited. This led to a lot of delays, affecting the project timeline.

Also, it was difficult finding offices to run tests in. Due to the security consciousness of both security personnel and staff members it was not so easy finding a place.

Lastly, the learning curve of the OPNET simulator was brutal. Very few tutorial videos exist and even with those we still found considerable difficulty learning the ropes.

5.3 FUTURE PROSPECTS

Ghana Academic Research Network (GARNET) is currently being deployed across universities in Ghana. When this is done, we recommend that KNUST liaises with these universities and deploy the VoIP system on GARNET. This acts as a local switch between all the networked campuses thereby enhancing quality of inter-campus calls and reducing costs, as the calls are logically viewed as local calls.

In KNUST, we are projecting a future where the VoIP access is deployed over the wireless network (KNUST WIFI). Students could use their existing school numbers as their softphone extensions. A softphone app for mobile devices could be developed in similitude to the current AIM app where users are allowed access only on merit of being students – thereby using student credentials. We envision a more intact communication system should this be enabled. However, we recommend a thorough network analysis to determine whether the network can support the load of students that would now have access to use the VoIP service.

5.4 RECOMMENDATIONS

The proposed design is highly feasible and low-cost due to the nature of the existing infrastructure. Also, we recommend that the design be implemented on other satellite campuses of KNUST to enhance their communication system.

It would also be expedient to run a needs analysis to determine how many minutes to allocate to authorized users of the PSTN service.

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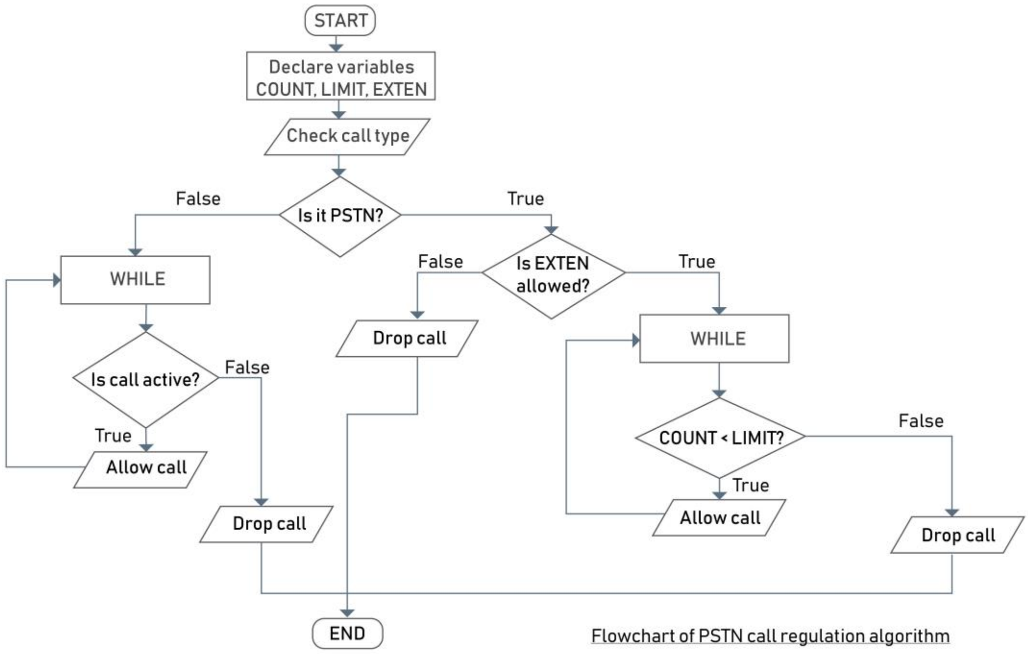
APPENDIX A: CODE SNIPPET TO INITIALIZE IP PHONE EXTENSIONS AND GRANT ACCESS

```
[macro-init-access]

PSTN=VF/g2

exten => 701,1,Dial(SIP/701|30)

exten => 701,102,Voicemail(b701)

exten => 702,1,Dial(SIP/702|30)

exten => 702,2, Dial(PSTN/702|30)

exten => 702,102,Voicemail(b702)

exten => s,2,Dial(${PSTN}/9w${dial_no},${time_v})
```

APPENDIX B: FLOWCHART TO DISPLAY ALGORITHM TO LIMIT USER BY CALL MINUTES



Flowchart of PSTN call regulation algorithm