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Design and Implementation of Voice over Wireless Local Area Network

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Author's contribution

Author AOA designed the study, wrote the first draft of the manuscript, managed the literature searches, read and approved the final manuscript.

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Original Research Article

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ABSTRACT

Aims: The main aim is to design and implement voice over wireless local area network.
Study Design: Building of network system using cabled and wireless media.
Place and Duration of Study: College of Physical Science (COLPHYS) of Federal University of Agriculture, Abeokuta, Ogun State, Nigeria, between March 2014 and July 2014.
Methodology: The techniques of the Structured Systems Analysis and Design Methodology (SSADM) and Prototyping were adopted in this project.
Results: IP was observed to be ubiquitous and cost-effective and thus the school at large under which college of physical science exists can deploy new voice/data services which removes the need to manage separate voice and data networks in all the colleges. Also, the school can reap the benefits of a standard, highly flexible network, giving competitive market to equipment vendors, and encompassing a wide range of equipment for different market niches.
Conclusion: Utilization of cheaper IP-based backbone equipment to carry voice data can be conducted. While this work had demonstrated the feasibility of leveraging on the affordances of IP-enabled telephones and developing a VOIP-based campus-wide telephony, it can be used as a prototype for the full implementation in all departments, colleges and the university at large.

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1. INTRODUCTION

"Voice over IP (VOIP) uses the Internet Protocol (IP) to transmit voice as packets over an IP network. So VOIP can be achieved on any data network that uses IP, like Internet, Intranets and Local Area Networks (LAN). Here the voice signal is digitized, compressed and converted to IP packets and then transmitted over the IP network. Signaling protocols are used to set up and tear down calls, carry information required to locate users and negotiate capabilities. One of the main motivations for Internet telephony is the very low cost involved" [1]. "Demand for multimedia communication and demand for integration of voice and data networks are also been considered. In the world of computers, networking is the practice of linking two or more computing devices together for the purpose of sharing data. Networks are built with a mix of computer hardware and computer software" [2].

"Voice over Wireless Local Area Network (VoWLAN) is a combination of two technologies which are Wireless Local Area Network (WLAN) also known as Wi-Fi and Voice over Internet Protocol also known as (VoIP). With the rapid rate of use of internet as a means of networking devices around the globe, various technology organizations are evolving to use this means to reach their audience irrespective of where they might be as long as there is internet connection. In ancient times where communication was springing up from oral to digital means, there were development of technologies to aid this. The invention of telephone by Alexandre Graham Bell was a step ahead oral communication as sounds were sent via electrical strings" [3].

"A network consists of two or more computers that are linked in order to share resources (such as printers and CD-ROMs), files, or allow electronic communications. Networks are divided into two categories: Local Area network and Wide area network. They have different functionalities and characteristics. A Local Area Network (LAN) is a network that connects a relatively small number of machines in a relatively close geographical area. A local area network can be used in a school" [4]. Wide-area network (WAN) is a network that connects two or more local-area networks over a potentially large geographic distance. Wireless LAN is one of the mainly organized wireless technologies all over the world and is likely to play a major function in the next-generation wireless voice call networks.

"The architecture of this type of network is the same as Local Area Network (LAN) except that the transmission happens via radio frequency (RF) or infrared (IR) and not through physical wires/cables, and at the MAC sub-layer, uses different standard protocol. This technology provides people with a ubiquitous communication in offices, hospitals, campuses and factories due to its mobility, simplicity, scalability, edibility and cost effectiveness" [2]. "Also, wireless network allows nodes to communicate with each other wirelessly and it can be configured in two ways, infrastructure-less mode and infrastructure mode. In infrastructure-less mode it is called Ad hoc or peer to peer (P2P) network, there is no fixed point and each node can directly communicate with all other nodes. On the other hand, infrastructure mode is a method of wireless where the transmission between two or more nodes goes through a third node called Access Point (AP)" [5].

"VoIP is a form of communication that allows users to make phone calls over a broadband internet connection instead of typical analog telephone lines. Basic VoIP access usually allows users to call others who are also receiving calls over the internet for free" [1]. "VoIP also known as IP telephony or Internet telephony, is a set of protocols to transport voice traffic over IP-based packet-switched networks with acceptable quality of service (QoS) and reasonable cost. In addition, VoIP can efficiently provide compelling features and services, such as voice mail, voice conferencing by allowing the integrated transmission of voice and data over the same network" [3]. In simple terms Voice over Wireless Local Area Network (VoWLAN) is a combination of two technologies which are Wireless Local Area Network (WLAN) also known as Wi-Fi and Voice over Internet Protocol also known as (VoIP).

"The primary purpose of Wi-Fi when it was formed was to access the internet, or use with certain network applications like printing as technology progresses" [6]. "Voice over Internet Protocol (VoIP) was introduced in WLAN thereby forming a technology called VoWLAN. VoWLAN is a method of sending voice information in digital form over a wireless broadband network, similar to VoIP but differing from it in that VoWLAN delivers voice over wireless devices, not wired phones. Based on the same IEEE 802.11 specifications, VoWLAN transports data over wireless local area networks or the Internet. It is also called “VoWi-Fi” or “Wi-Fi VoIP”. VoWLAN uses voice-enabled wireless devices such as PDAs, Wi-Fi phones and laptops to communicate. When a call is made, the voice data is transmitted across to the destination within the local network or out onto the internet or Public Switched Telephone Network (PSTN). Software-based ‘Skype’ application or softphone may be used to communicate through devices such as a laptop or desktop computer” [7].

“VoIP consists of three essential components: CODEC (Coder/Decoder), packetizer and playout buffer” [6]. “At the sender side, the analog voice signals are converted to digital signals, compressed and then encoded by voice codecs” [8]. “There are various voice codecs developed and standardized by International Telecommunication Union-Telecommunication (ITU-T) such as G.711, G.729, etc. Packetisation process is performed which fragment encoded voice into equal size of packets. Furthermore, in each packet, some protocol headers from different layers are attached to the encoded voice. Protocols headers added to voice packets are of Real-time Transport Protocol (RTP), User Datagram Protocol (UDP), and Internet Protocol (IP) as well as data link layer header. In addition, RTP and Real-Time Control Protocol (RTCP) were designed at the application layer to support real-time applications. Although TCP transport protocol is commonly used in the internet, UDP protocol is preferred in VoIP and other delay sensitive real-time applications” [9].

“TCP protocol is suitable for less delay-sensitive data packets and not for delay-sensitive packets due to the acknowledgement (ACK) scheme that TCP applies. This scheme introduces delay as receiver has to notify the sender for each received packet by sending ACK [10]. The packets are sent out over IP network to its destination, where the reverse process of decoding and de-packetising of the received packets is carried out. Additionally, there are signaling protocols of VoIP namely Session Initiation Protocol (SIP) and H.323. These protocols establish VoIP calls. H.323 was standardized by ITU-T specifically to smoothly work together with PSTN. On the other hand, SIP was standardized by Internet Engineering Task Force (IETF) to support Internet applications, such as telephony” [11]. Fig. 1 illustrates VoIP protocol stack with respect to its TCP/IP protocol.

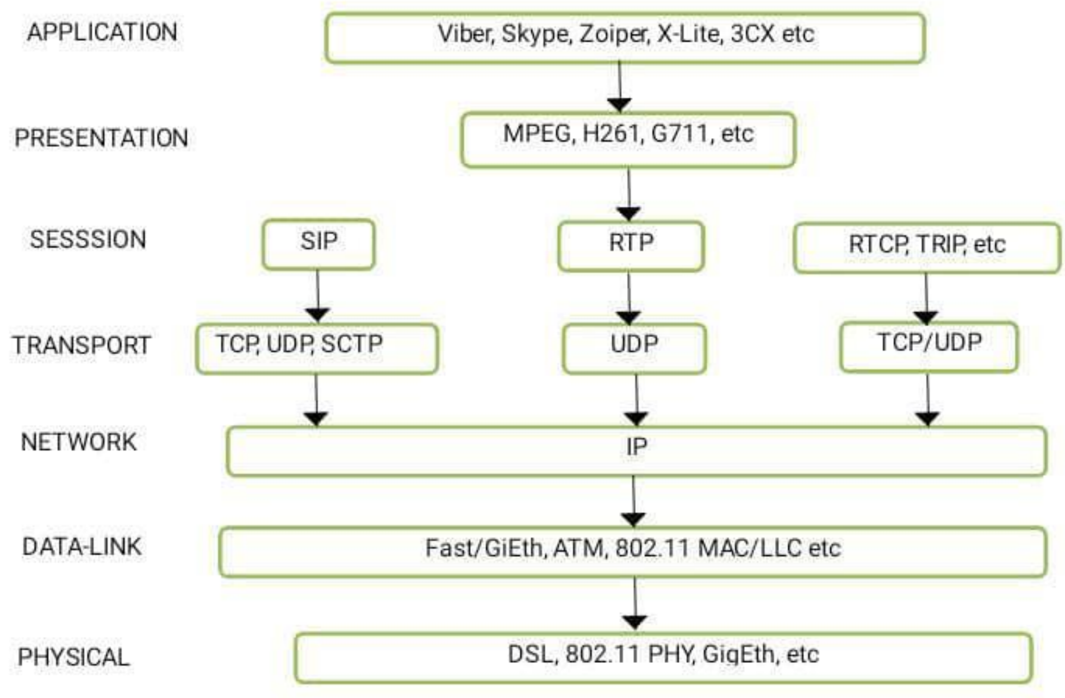

Fig. 1. OSI Layer Protocol Stack for VoIP [12]

The need for urgent Information, unreliable information dissemination, unreliable service providers and expensive means of communication in the College of Physical Science (COLPHYS) of Federal University of Agriculture, Abeokuta, Ogun State, Nigeria was the basis upon which this study was conducted. The specific objectives was to design, implement and test voice over wireless local area network (VoWLAN) communication between end devices that are suitable for use in COLPHYS.

2. METHODOLOGY

The techniques of the Structured Systems Analysis and Design Methodology (SSADM) and Prototyping were adopted in this project (Fig. 2).

Upon the examination of an applicable means of communication within the domain in study the local private automatic exchange box is suitably substituted with the VoIP system using a Wireless Local Area Network. The Analysis phases of the SSADM were adopted in carrying out a detailed study of the present telephony system in our institution while a prototype VoIP system was implemented live. In addition to the research methodologies, four physical connectivity options were adopted in this research:

i. In the first option, we used the several IP Softphone option. In this case, a Xlite application which was installed on a computer and smartphone was the phone used (Fig. 3). This helped to reduce the need for separate handsets.
ii. In the second option we connected the SEANET IP Phone to the VoIP network created for the project through wireless means (Fig. 4). In this way, we were able to use our VoIP network to provide connectivity to all devices both soft IP phones and hard IP phones.
iii. In the third option we connected the VoIP server to the network for the soft IP phones hard IP phones to receive configuration updates from the VoIP server (Fig. 5).
iv. In the Forth option the VoIP network was connected to two TP-Link routers In other to broadcast the network as Access Point (AP) through wireless (Fig. 6).

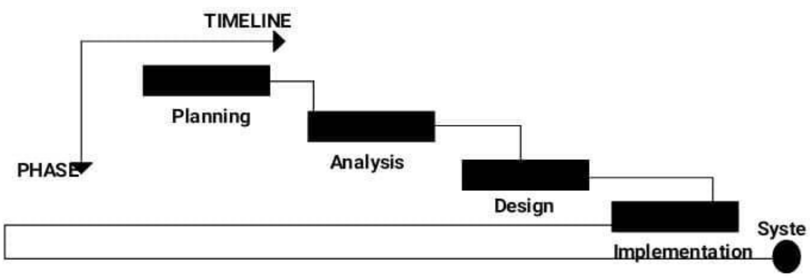

Fig. 2. Structured design methodology

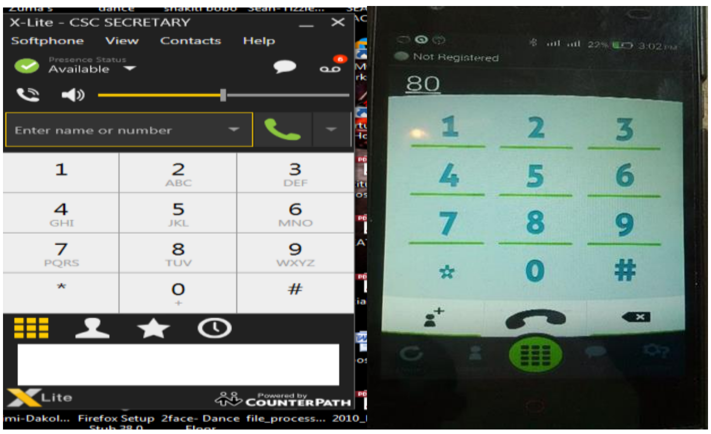

Fig. 3. (a) Xlite Softphone and (b) Tecno smartphone

Fig. 4. SEANET IP Phone

Fig. 5. VoIP server

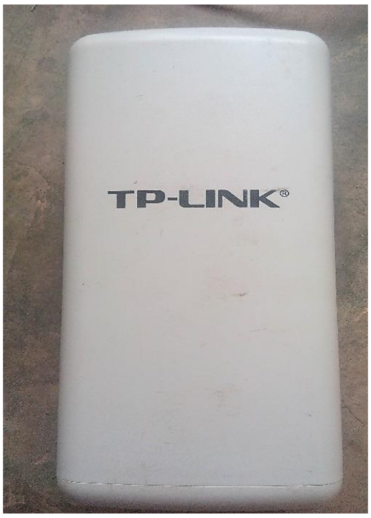

Fig. 6. Access Point (AP)

2.1 DOMAIN STUDY

The domain to be tested with this system is the College Of Physical Sciences of the Federal University of Agriculture Abeokuta. The radius of the domain is 4.0km with the height of the building equating to 5.0ft which will serve as a basis of choice of network devices to be used to propagate the wireless signal across the domain. Land Area Covered By the Domain

- Length:

384 (m), 419.95 (yd)

- Area:

9217 (sq/m), 14286378.57 (sq/in), 0.01 (sq/km), 0 (sq/mi), 99210.96 (sq/ft), 2.28 (acres), 0.92 (ha)

2.2 Building the Network System

The network system supporting the VoIP service is based on the 802.11b standard which transmits on a 5 GHz channel with data transfer rate of 54Mbps compared to the native 802.11a standard transmitting on 2.4 GHz channel with data rate 11Mbps. This higher frequency compared to 802.11b shortens the range of 802.11a networks. The higher frequency also means 802.11a signals have more difficulty penetrating walls and other obstructions.

Careful study of the domain and positions of the hard-phones shows that the network system has an open area to propagate the signals to every position of the hard-phones within the domain. Little obstructions will be encountered and this will have the least minimal effect on the quality of service on the signal of the network provided for the voice over wireless LAN system to be implemented.

Two means of connectivity were considered and the more suitable of the two was chosen as the type of connection to be used to implement the VoIP system. The two media are;

The Cabled Media:

This involves the use of network cables which in this case is a Category 5 unshielded twisted pair cable. This type of connectivity is usually considered because a cabled media is known to be a guided transmission media because the data sent over this kind of media is protected from diffraction and interference from other waves in transmission unlike the wireless.

The Wireless Media:

This involves the use of radio waves, infrared, electromagnetic waves to transmit signal in a network. This type of connectivity is usually considered because a wireless media is known to be an unguided transmission media because the data sent over this kind of media is unprotected from diffraction and interference from other waves in transmission unlike the cabled media but provides a neat and easily installation and set considering the topology of the domain.

2.3 Domain Existing Network

Federal University of Agriculture, Abeokuta is a large domain area with wide range of wireless network but for the purpose of this project the network is not usable to avoid network availability problems within the school, therefore the network CSC VOIP is set-up to attend to our network needs. Figs. 7, 8 and 9 illustrate FUNAAB, extracted COLPHYS and the proposed CSC VOIP Network respectively.

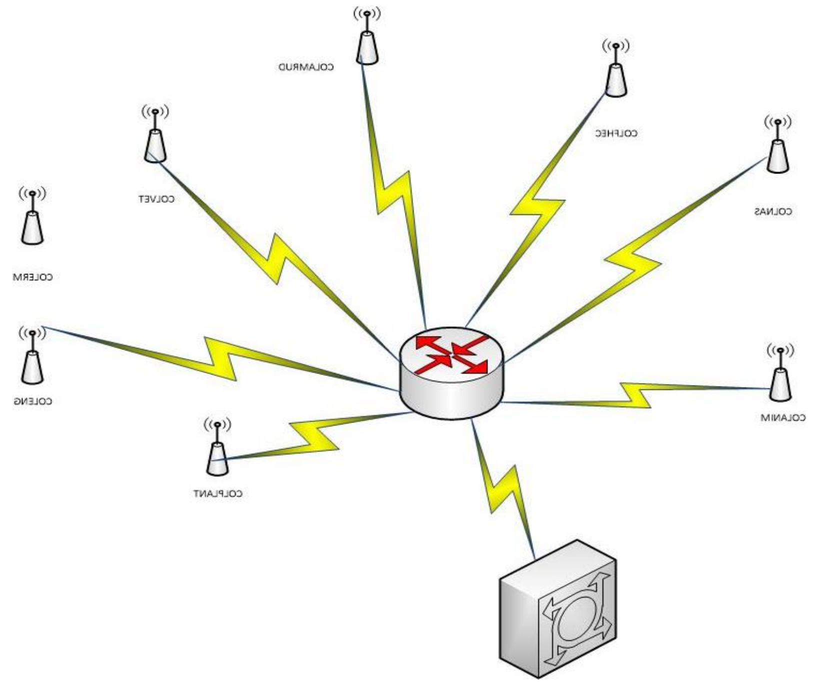

Fig. 7. FUNAAB Network

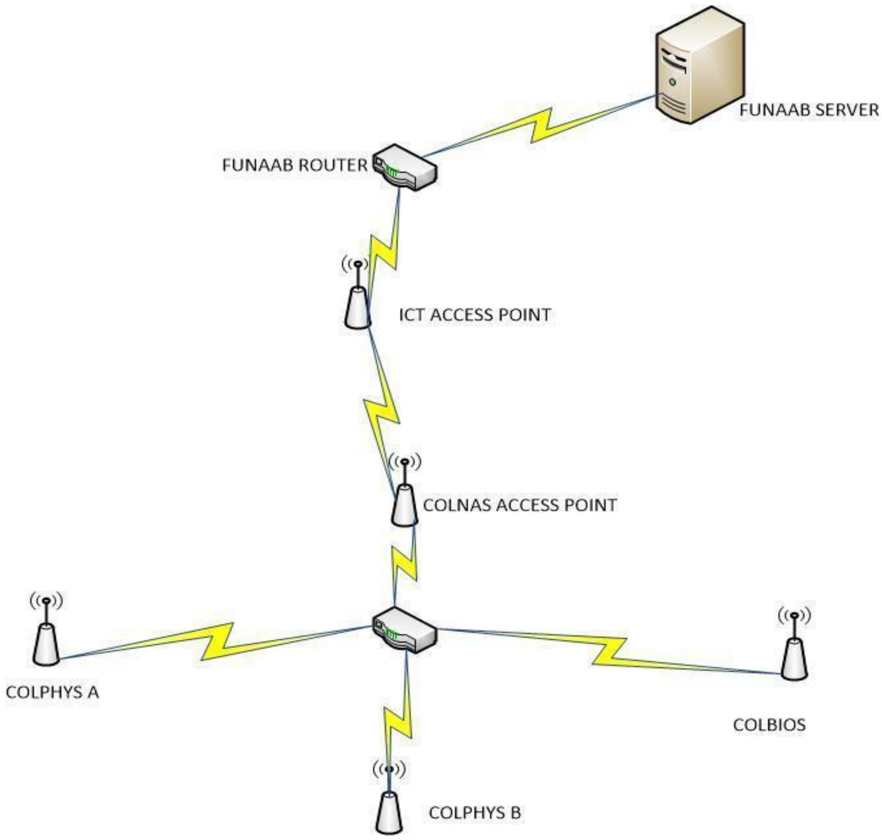

Fig. 8. Extracted COLPHYS Network

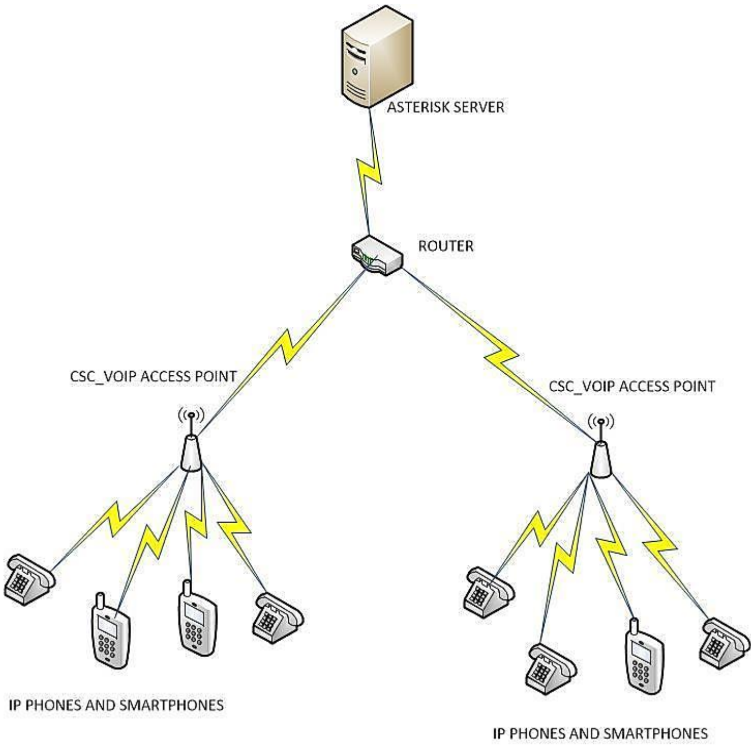

Fig. 9. CSC VOIP Network

2.4 Material Deployed for the Project

A 300Mbps wireless N Gigabit TP-Link Router: This is a combined wired/wireless network connection device integrated with internet-sharing router and a 4-port gigabit router. It creates wireless network with high speed of up to 300Mbps, which ensures simultaneous enjoyment of multiple high bandwidth consuming and interruption sensitive applications such as streaming HD video, making VoIP calls, sharing large files and playing online games. This device is used in the project to set-up a network with a SSID called CSCVOIP network.

TP-Link wireless Access Point: This a wireless network device that can be used to perform multiple function such acting as an access point to network by transforming a wired network into a wireless network, as a repeater to regenerate and retransmit a weak wireless signal into a strong signal, as a bridge to join two or more different network as make them appear as one or distinct depending on the choice of the user. The device is used in the project as an access point to cover both COLPHYS and COLBOIS to facilitate strong wireless network as it covers up to 15kilometers without obstruction.

AsteriskNow: This is a VoIP platform also called Private Branch Exchange (PBX) that is built on Linux for voice over internet protocol options, it is a free operating system with a command line interface (CLI) and a remote view graphical user interface (GUI). This supports several VoIP protocol, customizations, unlimited functions and account unlike other closed source PBX such as brekeke PBX, Elastix PBX, 3CX and others

SEANET IP Phones: This is a voice over internet protocol phone that supports SIP configuration for performing VoIP services

Softphones: This are voice over internet protocol telephone application that runs on both personal computers and several version of mobile operating system. For the purpose of this project several of it will be used to emphasize the flexibility of our VoIP system.

2.5 Designing the VoWLAN System

The design procedure of a VoIP Telephony system adopting a Voice over Wireless Local Area Network (VoWLAN) is shown as Fig. 10:

Step 1: The network architecture and layout of the college is studied in order to facilitate proper VoIP network Set-Up.

Step 2: A router with a SSID called CSC VOIP is set-up with a class B range of network Address and configure to suit VoIP services

Step 3: Two Wireless Access Points (AP) are configure and installed to give a signal boost to existing CSC VoIP network by retransmitting the signals in a reasonable scope within the two colleges.

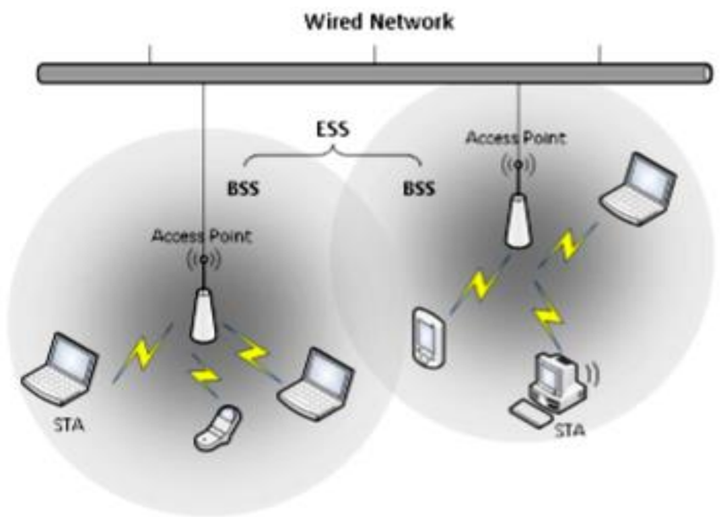

Fig. 10. System Architecture Diagram for A WLAN Phone System

Step 4: An AsteriskNow server version 13.1 is installed as a new operating system to the VoIP Server, to allow both session initiation protocols (PJSIP and CHAN SIP) version 13.1 is used considering the fact that PJSIP only exists in version 12.0 and above. The purpose of the AsteriskNow is to perform the work of our Private Branch Exchange (PBX) and a free PBX module has been integrated with it to enable session layer registration, extensions registrations, trunk configuration and many other configurations and also to serve as communication platform between the VoIP clients

Step 5: The telephony-service was then implemented by configuring the SEANET IP phones and the VoIP dial-peers

2.6 System Diagram

The overall IP telephony process is represented below using a flowchart (Fig. 11). The first activity is the opening of the data connection, followed by sending a call request. A returned data shows the completion of the call and t30he connection is closed, otherwise, the call request is re-sent until a data connection is returned. Fig. 12 gives the state diagram of the IP telephony system.

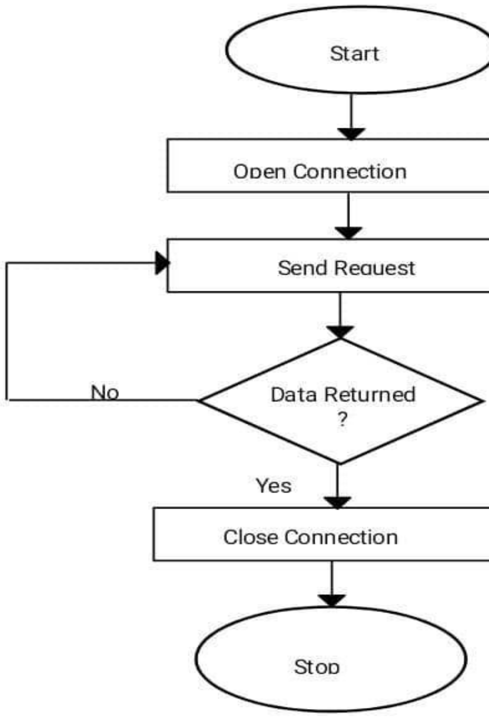

Fig. 11. Flowchart for the VoIP system

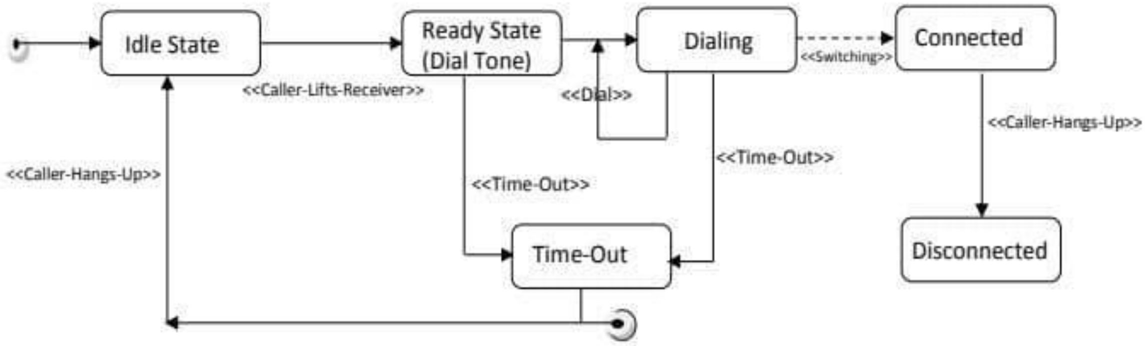

Fig. 12. State Diagram for the design of Both the Soft and Hard IP Phones

3. NETWORK DESIGN

The design and implementation of the VoIP system consisted of four network design model; one for each of the four sites and the multiple site central model.

i. The COLPHYS B Lower Block Model was designed as a star topology with 2 SEANET IP Phones and several softphones connected to a central wireless router through which all other connections are made (Fig. 13).

ii. The COLPHYS B Upper Block Model was also designed in a star topology but with less computers, 1 SEANET IP phone and several softphones than the COLPHYS lower block (Fig. 14). This is due to the fact that there are less numbers of targeted users in this site than the COLPHYS lower block site.

iii. The COLPHYS A Block Model was designed as a star topology with 2 SEANET IP Phones and several softphones connected to a wireless access point through which all other connections are made (Fig. 15).

iv The Multiple Site Central model comprises of softphones located at remote sites registered with the call manager cluster located at a central site. The star topology is deployed in this mode as other sites are connected to the central location (Fig. 16).

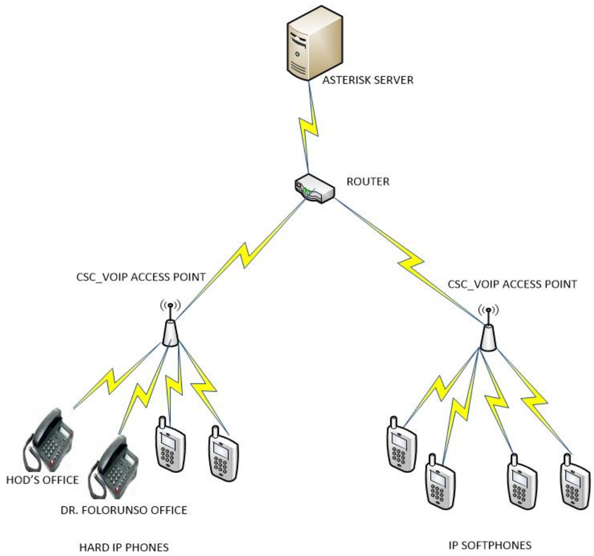

Fig. 13. COLPHYS B Lower Block Layout

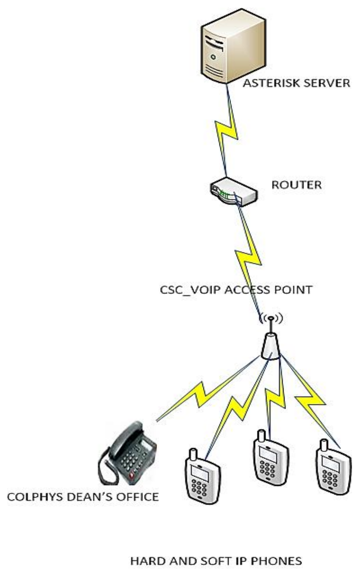

Fig. 14. COLPHYS B upper block layout

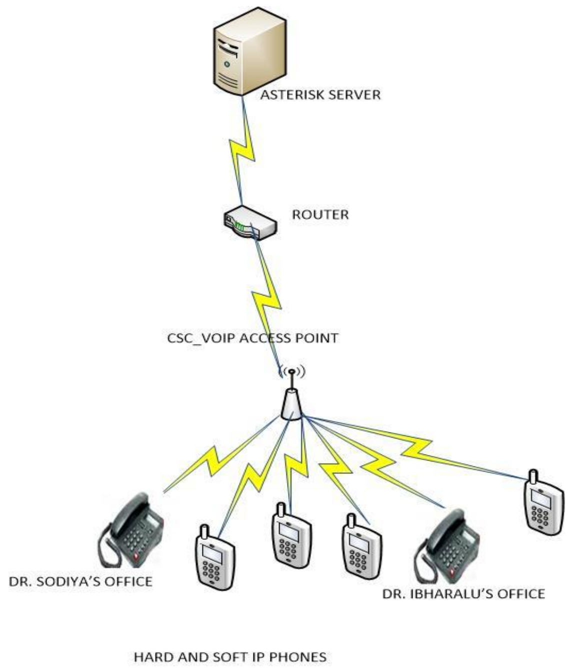

Fig. 15. COLPHYS A block layout

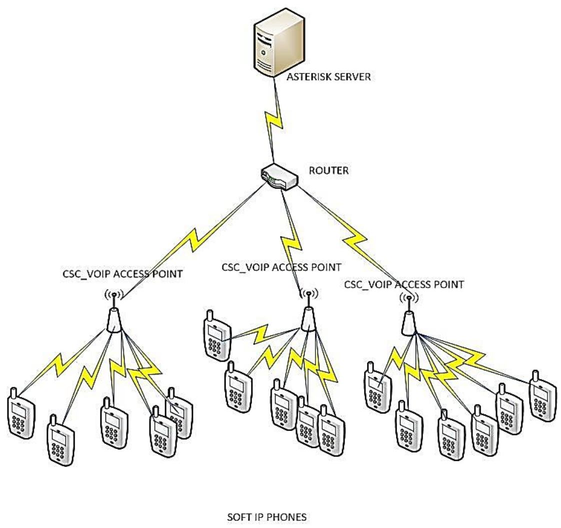

Fig. 16. Multiple Site central model layouts

3.1 Project Requirement

Voice over Wireless Local Area Network (VoWLAN) requires both hardware and software. The requirement varies with expected performance and they can be functional or non-functional requirements.

3.1.1 Software Requirements

a) Equipment or Development Tools

- Voiper, X-Lite, linphone, Skype and any other softphone (VoIP function)
- Microsoft Project 2010
- Microsoft Visio 2010

b) Operating System or Server

- AsteriskNow Server (A Linux based free PBX)

3.1.2 Hardware Requirements

- Wireless Switch (Access Point)
- Core i3 or Core i5 6gb RAM, 500Gb HDD
- Laptops with Wireless - Personal Computers (Pcs)

- Wireless USB adapter (for Client PC)
- Category 5 (CATS) Unshielded Twisted Pair (UTP) Cable
- Wireless Router

4. CONCLUSION

IP is ubiquitous and cost-effective. Thus, moving towards an IP-based network, the school can (i) deploy new voice/data services, removing need to manage separate voice and data networks in all the colleges, (ii) reap the benefits of a standard, highly flexible network, giving competitive market to equipment vendors, and encompassing a wide range of equipment for different market niches and (iii) utilize cheaper IP-based backbone equipment to carry voice data. While this work had demonstrated the feasibility of leveraging on the affordances of IP-enabled telephones and developing a VOIP-based campus-wide telephony, it can be used as a prototype for the full implementation in all departments, colleges and the university at large. This project provides various functions which includes free secure communication between two users, voice conferencing between multiple users, it can also be used as help desk for the department in which it provides information about the users, working hours and the department and it fosters free communication between the staffs of the department

COMPETING INTERESTS

Author has declared that no competing interests exist.

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