Campus-wide Internet Telephony Design and Simulation using Voice over Internet Protocol: A Case Study of Adamawa State University, Mubi, Nigeria

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Abstract - *Voice over Internet protocol (VoIP) is a technology also known as Internet protocol (IP) telephony because it uses Internet protocols to make enhanced voice communications possible. Internet protocols (IP) form the root of IP networking, providing support to the public, corporate, private, cable, and even wireless networks. VoIP unites an organization's many locations into a single converged communication network. In light of the objective, an efficient VoIP network which tackles Voice communication issues was designed and developed for Adamawa State University, Mubi. This study proposes a Voice over Internet Protocol (VoIP) system that can help users at the Adamawa State University campus to freely communicate by using voice communication devices such as the IP phone. Various sessions of the simulation were run and configured using the Cisco packet tracer. It was used to develop a prototype network architecture, giving flexibility and ease to implement an efficient VoIP system.*

Key Words: VoIP, QoS, Telephony, Codec, Cisco packet tracer

1. INTRODUCTION

Nowadays, in light of the current developments in the field of communication networks and due to the crucial need to send data or information in a short period and at the lowest possible cost, the VoIP technology was found. It is the technology that transmits analog voice over a digital network such as the Internet. VoIP increases functionality and reduces costs because telephone calls pass through a data network instead of the telecommunication network of a company.

This technology is a successor to the public Switched Transfer Network (PSTN) a connection-oriented, circuit switched network that uses dedicated channels for transmission. The PSTN runs over landlines or wires on poles and underground. The inception of VoIP brings a turnaround in communication technology in which we no longer need to connect overhead wires over long geographical location or distances to have access to telephony service instead all we need now is just the internet or our major network connection that can be in a form of LAN, WAN, etc. Today VoIP is one among the dominant technology in the communication world and many organizations around the world are implementing the technology while some are re-engineering the traditional PSTN they have been using for years into VoIP.

VoIP is built on open infrastructure allowing various vendors to, provide applications and access, unlike the public switched network, whose infrastructure is a closed system. The PSTN technology involves vendors only building applications specific for their equipment and its framework hasn't made it possible for vendors to develop new applications for it; VoIP allows the development and design of more creative applications as well as the convergence of data, voice, and video in one channel. Consequently, it is expected that the VoIP may completely replace the circuit switched PSTN system in the future.

Because of the efficient bandwidth and minimum cost that VoIP technology offers, in which both voice and data communication can be run on a single network, several organizations ranging from small businesses to large enterprises not excluding universities or colleges, are adapting and deploying the technology and it has proved to be useful in enhancing communication and distribution of information. Currently Adamawa State University, Mubi do not operate the VOIP technology and the benefits of an implemented campus-wide internet telephony are numerous. And since the university has a good number of departments and offices, VOIP can be used to promote inter-departmental communication and likewise that of the offices.

This study aims at designing and simulating a VOIP-based telephony network using the Cisco packet tracer network simulator, for the university communication system. I believe that if we can successfully implement the simulation, given the advantages of VOIP, a modern telephony system can be implemented on campus.

2. LITERATURE REVIEW

VoIP is often referred to as *IP telephony* (IPT) because it uses Internet protocols to make enhanced voice communications possible. Internet protocols are the basis of IP networking, which supports corporate, private, public, cable, and even wireless networks [1].

VOIP uses packet switching because circuit switching is not the most preferred mechanism that is chosen by data networks. Another reason is that the speed of the internet connection would decrease by a great amount if it had to maintain a continuous connection to the web page that is being viewed at any given time. So as an alternative, data networks simply send and recover data as needed. Also, instead of choosing to route the data over a dedicated channel, the packets of data flow through a hectic network that consists of various possible routes [2]. In the past, the circuit switching process was the most widely used to build a communications network it was used for ordinary telephone calls and allowed the sharing of communication equipment and circuits among users. The connection was established first between source and destination and after the transfer of information, it was terminated [2]

Public switched telephone networks (PSTN) is a traditional phone network system, using a circuit switching mechanism for voice transmissions. Basically, in circuit switching, resources are reserved along the entire communication channel for the duration of the call, whereas in packet switching, information is digitally transmitted into one or more packets [2].

2.1 VoIP REAL-TIME PROTOCOLS

The protocols used to send real-time streams of data across a network are called real-time protocols. Real-time protocols deliver audio and video over IP networks. RTP (Real Time Protocol) and RTCP (Real Time Control Protocol) comes under VoIP real-time protocols that run on the top of the User Datagram Protocol [3].

2.1.1 Real-Time Protocol (RTP)

Real-Time Protocol (RTP) a network protocol for delivering audio and video over IP networks. RTP is used extensively in communication and entertainment systems that involve media, such as telephony, video teleconference. RTP is used in alliance with a signaling protocol that assists in build-up connections across the network [4].

2.1.2 Real-Time Control Protocol (RTCP)

Real-Time Control Protocol (RTCP) is a control protocol and works in combination with RTP. RTCP provides Quality of Service (QoS) feedback and session information [5]. RTCP can monitor the fraction lost, jitter, packet loss and one-way delay [4].

RTCP allows participants to indicate that they are leaving a session with the use of the BYE packet. It partners with RTP in the delivery and packaging of Multimedia data but does not transport any media data itself [3].

2.2 Related Work

[6] Has demonstrated a survey on VOIP over WLAN, its advantages and challenges and also VOIP capacity over WLAN and the number of calls for different voice codecs and intervals based on IEEE 802.11b standard.

[7] Evaluates the quality of service of video transmission on Differentiated Services (Diff-Serv) with Multiprotocol Label Switching (MPLS) network is being simulated. The objective of this work is to study the influence of the QOS mechanism via DiffServ-MPLS on network parameters such as packet loss, delay and throughput for different video resolutions.

[8] Proposes how the implementation of voice over internet protocol (VoIP) system in UUM campus can help users to freely communicate by using the VoIP technique. According to him the proposed system also helps to increase the effectiveness of using the Internet bandwidth; since the users can communicate with each other without the need to have an Internet access.

[9] Proposed the optimization techniques that can be used to analyze and optimize the performance of wired and wireless networks of a campus area. Cisco packet tracer was used for the simulation

[10] Attempts to identify some of the network performance parameters that service providers will focus on to develop a VOIP over WIMAX communication tool that will serve as a voice communication broadband replacement technology to old circuit switch voice communication.

[11] Provides the quick and technical overview of concept, standard, technology and architecture for IEEE 802.16 WiMAX.

[12] Presented a Media Access Control Protocol that provides the quality of service for VoIP over wlan. In this, the characteristics of our proposed protocol are No hardware modification of VOIP STA. Backward compatibility in order to minimize the cost of development no modification of access points.

[13] Provides focusing on quality of service scheduling services and performance related metrics such as jitter, packet end to end delay and MOS (mean opinion score).

[14] Evaluated the performance measures such as delay variation, delay, page response time, throughput and packet drop for different types of traffic such as voice,

video, data in their movement in a congested network for both MPLS-TE and Conventional IP Network.

[15] Studies VOIP to a level that allows discussion of security purposes and concerns. In this work, VOIP components will include network components, gateway, end user equipment, call processors and two of common architectures.

[16] Work on the network performance analysis to evaluate the effects of the application of different voice encoder schemes on quality of service of VOIP system which is deployed with the UMTS network.

2.3 Conceptual Framework

In this research work, a conceptual VoIP model/framework was designed for Adamawa State University, Mubi. The implementation was carried out using Packet Tracer and the network is based on the Hierarchical Network Design Model and mesh topology. The infrastructure of the campus was considered during the framework design.

The following attributes were also considered during the implementation of the framework.

i. Placement of VoIP for internal use over the current network.
ii. VoIP signaling, control, and management of calls would be done by using the Cisco 2811 router's telephony service.
iii. Users can receive/make calls by using IP Phones.

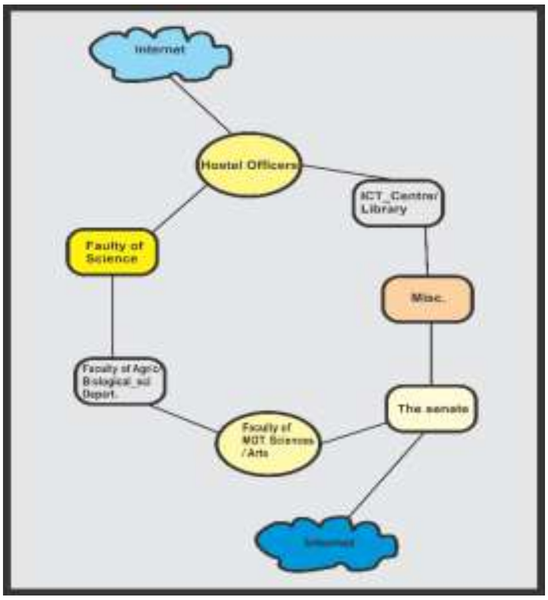

Fig -1: Conceptual framework of the system

3. METHODOLOGY

The mode of communication in Adamawa State University campus does not involve any communication system other than the current call services provided by Internet Service Providers (ISP) which involves the use of mobile Phones or smartphones with the induced service cost. For any form of communication to occur between offices or departments or units, mobile phones or smartphones are the major alternative means to communicate. Consequently, without service charges or costs, it will be difficult to communicate. Network Systems such as the VoIP system is being utilized by numerous organizations to save the cost charges induced by ISP for their call services. Therefore for effective and efficient voice communication to take place on campus between various offices, departments, and units, etc. a telephony system needs to be implemented. Deployment of this system will make communication a lot easier since no cost charges will be attached to voice communication, instead, the voice calls will be cost-free. With such an advantage, Adamawa state university can benefit greatly from this technology that has been around for more than a decade. The prototype network developed in this study can be used to implement this technology in Adamawa state university. Apart from that, the prototype will also address major VOIP voice quality issue that usually affects VOIP systems.

3.1 System Development

The Cisco Packet Tracer Simulation of the VoIP Telephony system is given below:

Step 1: Network devices were selected from the Cisco packet tracer device database and connected.

Step 2: The VLAN's and IP networks were created for different user segments and network traffic.

Step 3: IP addresses were assigned to the seven different endpoints on the campus via the DHCP (Dynamic Host Configuration Protocol).

Step 4: Static and dynamic routing protocols were configured on the routers with dual Internet access connection.

Step 5: Telephony-service was then implemented by configuring the cisco 2811 routers to support the Cisco IP phones, through directory assignments and call connections using VoIP dial-peers.

3.1.1 System Flowchart

The first telephony process represented in the flowchart below involves the opening of a data connection, followed by sending a telephone call request. When data is returned, it means a call is completed and the connection will be closed or else the call request will be sent again until a connection is returned.

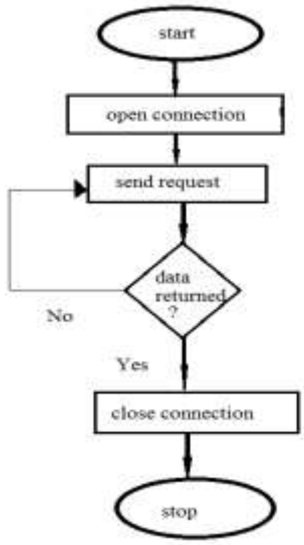

Fig -2: Flowchart for VoIP System

3.1.2 System's Use Case Diagram

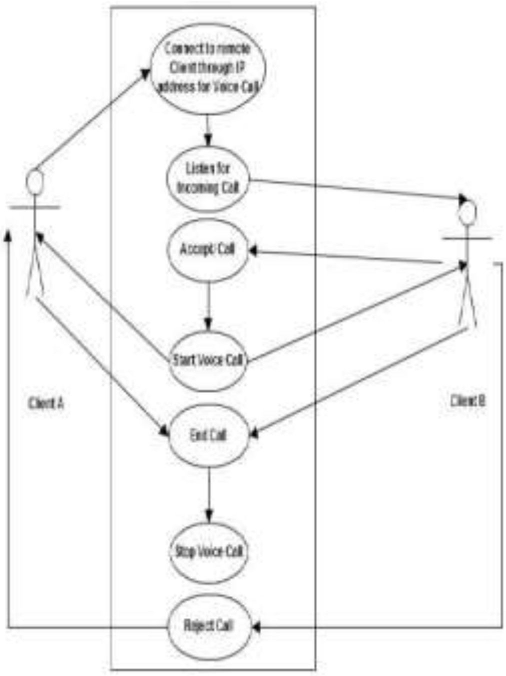

Fig-3: Use case diagram of the VoIP System

3.1.3 System's State Diagram

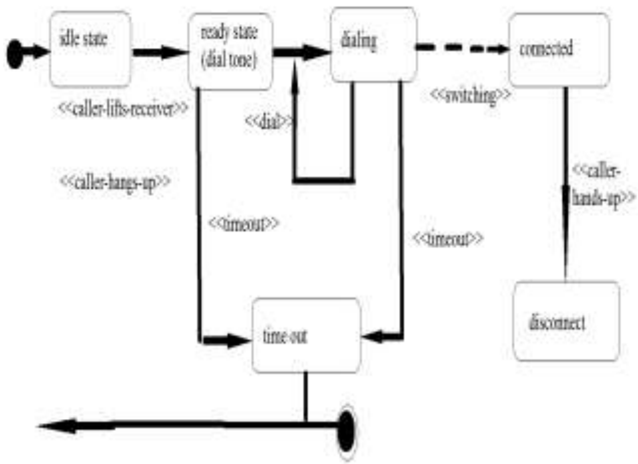

Fig-4: State Diagram for the Simulation of the Cisco IP phone calling process

4. RESULTS

In this research work, the results of the study are presented and discussed concerning the aim of the study, which is to determine the influence of using quality of service (QoS) Configurations to improve Voice communication efficiency for the VoIP network simulation of Adamawa state university, Mubi.

Table-1: The IP address and VLAN's

LOCATION/ROUTER		IP ADDRESS | VLAN
(A) Faculty of social & Management Sciences/arts	Data	10.10.12.0/23 || 12
	Voice	10.10.14.0/23 || 13
(B) The Senate	Data	10.10.16.0/23 || 16
	Voice	10.10.18.0/23 || 17
(C) ICT_Centre/ Library	Data	192.168.4.0/23 || 40
	Voice	192.168.6.0/23 || 50
(D) Resident Halls	Data	192.168.8.0/23 || 8
	Voice	192.168.12.0/23 || 12
(E) Faculty of science	Data	192.168.10.0/23 || 10
	Voice	192.168.20.0/23 || 20
(F) Faculty of Agric	Data	10.10.10.0/23 || 16
	Voice	10.10.8.0/23 || 17
(G) Miscellaneous	Data	192.168.10.0/23 || 11
	Voice	192.168.12.0/23 || 13

Table 1. Above gives the list of designated network addresses and VLAN's of faculty buildings and other sections of the University. Each one of them has a designated IP address for both Voice and data communication in the system. For example, the faculty of the science IP address is 192.168.10.0/23 for data communication and 192.168.20.0/23 for voice communication.

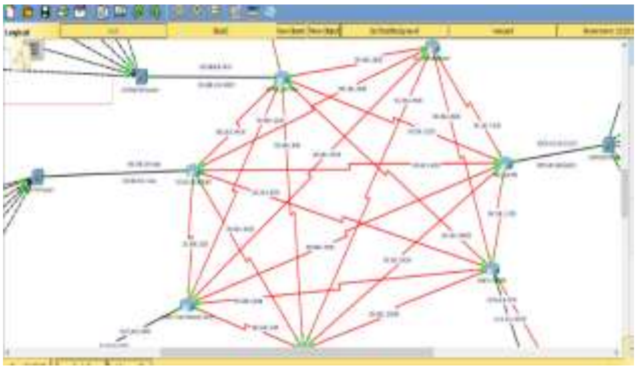

Fig-5: Connected routers of the network simulation with their designation.

Fig-5. Above shows, the designated routers of each faculty and various sections of the University connected in a mesh topology.

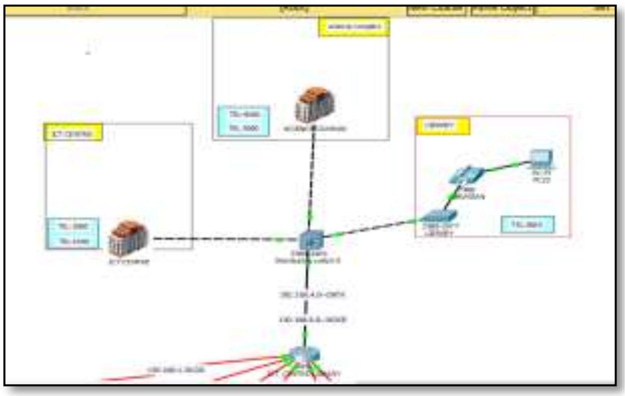

Fig -6: The simulation of ICT and Science complex VoIP clusters with the library VoIP network.

Fig-6. Above is a portion of the larger network, showing the clustered form of ICT Centre and science complex connected on a multilayer switch, together with the library Unit.

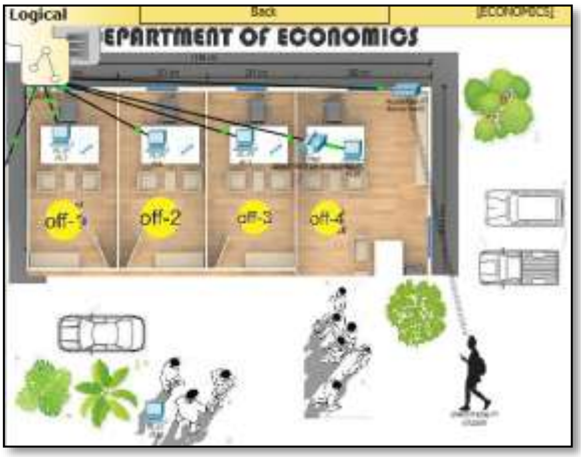

Fig-7: view of Economics department VoIP network

Fig 7. Above shows a view of the Economics department network and the various connected devices.

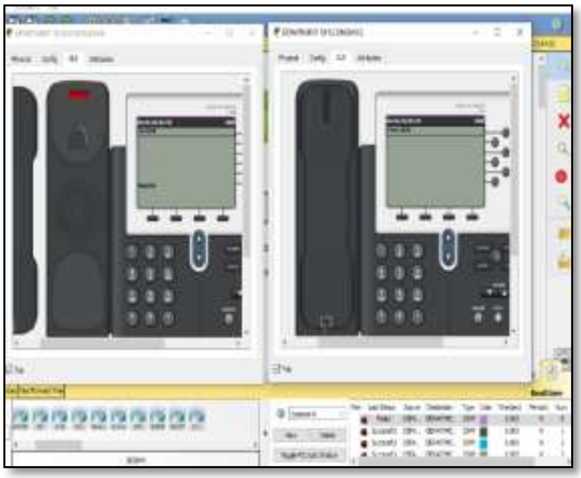

Fig-8: Calling from one IP phone to another

In Fig-8. Above the process of making a call between two destinations was depicted using a Cisco IP phone, in which the sender dials the destination phone number while the receiver sees an incoming call.

5. CONCLUSION

In conclusion, a prototype campus network was designed and implemented in this research work using the Cisco Packet Tracer software. Our objective was to design and simulate an efficient VoIP network scenario for the case study Adamawa State University and also to configure the virtual network devices of the simulation, evaluating point-to-point connections to ensure proper communication between various offices and departments. To implement this topology, we had to study the whole VoIP scenario, VoIP background, its features, benefits, drawbacks and its future in the networking world. Overall, this study improved our understanding of the whole concept of VoIP and its ever-increasing demand in present times.

5.1 Recommendations

i. The access list should be implemented in the network to provide more security to control which packets or routing updates are permitted or denied in or out of the network.

ii. Network security infrastructure such as VPNs, firewalls, etc. optimize voice and therefore should be implemented because they are capable of supporting the advance security requirement of VOIP.

iii. VoIP comes with new complex threats, therefore, it is highly recommended that network security upgrades be carried out.

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