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Design and simulation of an efficient VoIP network scenario for the University of Africa (Bulou-Orua campus), Bayelsa, 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. However, the application of this technology is still underutilized in many tertiary institutions in Nigeria. In light of this, an efficient VoIP network that tackles voice communication issues was designed and developed for the University of Africa, Bulou-Orua Campus. This study proposes a Voice over Internet Protocol (VoIP) system that can help users in the University to communicate freely by using voice communication devices such as the IP phone. Various sessions of the simulation were run and configured using the Cisco packet tracer for the development of efficient and flexible prototype VoIP network architecture. Our proposed system achieved results that are on a par with the baselines and the current systems employed in other universities.

Keywords: Bulou-Orua, Network, University of Africa, Voice over Internet Protocol

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1. INTRODUCTION

The development of the personal computer brought about major positive changes in business, industry, science, education, and virtually every sphere of life. For identification of IP address that matches with the phone number of the recipient, a set of rules is employed. Once the recipient answers a call, the voice is changed into a digital signal and split into a series of packets. The entire process begins by changing analog voice pointers to digital. Because digitized voice demands a vast number of bits, "a compression algorithm is used to reduce the volume of data to be transmitted" (Endler & Collier, 2006; Ayokunle, 2012; Mazalek et al., 2015; Antwi-Boasiako et al., 2016; Hasan & Hussain, 2017; Perigo et al., 2020). The need to access the Internet to download and upload information quickly and accurately and at any given time is on the increase.

An enterprise network is often understood as that portion of the computing infrastructure that provides access to whole network communication services and resources to end users and devices spread over a single geographic location. It might span a single floor, building, or even a large group of buildings spread over an extended geographical area. Some networks may have a single campus that also acts as the core or backbone of the network and provide interconnectivity between other portions of the overall network.

The invention of VoIP network has led to an adjustment of the average personal computer (PC) to create a softphone. A softphone is a collection of a headset, personal computer, software and a cheap connection service (Endler & Collier, 2006; Ayokunle, 2012; Mazalek et al., 2015; Antwi-Boasiako et al., 2016; Hasan & Hussain, 2017; Perigo et al., 2020). Voice over Internet Protocol is the infrastructure that helps to "dial telephone numbers and communicate with people on the other end of a connection who have a VoIP system" (Chong & Ang, 2010; Ibhaze et al., 2020; Shambour et al., 2020; Kolhar, 2021; Abualhaj et al., 2022; Al-Mimi et al., 2023). Because most Internet based phones are affordable and IP in most devices, is for their communication protocol, this makes VoIP taking the lead as service platform for future application (Kumar, 2006).

On the other hand, the Public Switched Transfer Network (PSTN) is a connection-oriented and circuit-switched network that employs for transmission, dedicated channels. To solving the problems related to its original analog transmission, the PSTN had switched to digital signals transmission using Pulse Code Modulation (PCM) for the conversion of all signals that are analog into digital transmissions at the caller end and the receiver end. However, PSTN has two notable important disadvantages: (1) PSTN is not cheap to employ, and this is due to the high-priced bandwidth, and (2) the network channels are underutilized.

One reason many institutions opt to use the VoIP system is that it is cheap compared to other modes of communication. The VoIP system does not use the switched telephone networks. Consequently, it is cheap to transmit voice messages over long distances. The VoIP system does not send voice traffic over the ordinary telecommunication line configurations (Jiang & Schulzrinne, 2000). Instead, the traffic travels over private data network lines or the Internet. The goal of this work is to deploy a telephony service that is based on VoIP using the existing network system in the University of Africa (UAT), and it covers connecting different offices together. By using simulation of the desk IP phone, the possibility was revealed on how cheaper calls can be realized.

In Hephibah et al. (2022), VoIP is rapidly replacing PSTN (a name given to networks created by phone companies) because of the established fact of Voice over Internet Protocol

(VoIP) technology evolving with various approving characteristics over the older standard Public Switched Telephone Network (PSTN). According to Rana et al. (2023), Next Generation Network (NGN) is the first IP-based telecommunication network developed by ITU-T in late 2003, and is considered a new telecommunication infrastructure to replace the old legacy Public Switch Telephone Network (PSTN). Simplification of the old complex network was the basis and important impact presented by the NGN network. In addition to simplicity, NGN is capable of supporting multimedia services. NGN provides a single platform for all services. It is the first step toward the conversion of non-IP-based networks into IP-based packet networks (Alo & Henry, 2013; Ifijeh et al., 2015; Chaudhari, 2018; Dubi & Muniyandi, 2019; Gurrapu et al., 2019).

2. MATERIALS AND METHODS

The mode of communication at the University of Africa (Bulou-Orua campus), Bayelsa, Nigeria 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 smart phones with the induced service cost. For any form of communication to occur between offices, departments or units, mobile phones or smart phones 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 are 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, 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, the university can benefit greatly from this technology that has been around for more than a decade.

2. 1. Materials Employed

Cisco Packet Tracer Version 8.2.0, publicly available software was employed in running the behavior of the system specification and simulation. With this version, support was provided for the several Application Layer protocols that were simulated. “The simulation devices are listed as follows:

a. Cisco Router 2811
b. Cisco 3650 Layer 3 Switch
c. Cisco 2960 Switch
d. Cisco IP phone 7960 with a large pixel based LCD display, full duplex speaker phone and an Internal Cisco Ethernet switch.
e. LAP-PT, WRT300N and a Home router
f. PCs and laptops”

For the configuration of the VoIP simulated system, the G.729 voice codec was employed; this is because it makes use of modest bandwidth and has high quality of voice streaming.

2. 2. Methods Employed

The steps involved in the simulation of VoIP Telephony system by Cisco Packet Tracer are as follows:

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

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

Step 3: IP addresses were assigned to the six (6) different endpoints on the campus via the DHCP (Dynamic Host Configuration Protocol) from the sever room.

Step 4: Static and dynamic routing protocols were configured on the routers.

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. Table 1 shows the Blocks and number of IP phones they contains. Table 2 shows the names of VLANS in the network. Table 3 shows the VLANS and their IPs.

Table 1. Blocks and number of IP phones they contains

BLOCKS	NUMBER OF IP PHONES
ADMIN BLOCK	4
ICT	3
ART AND EDUCATION	3
SCIENCE	3
E-LAB	1

Table 2. Showing names of VLANS in the network

VLANs	NAMES
VLAN 10	STAFF
VLAN 20	STUDENT
VLAN 30	IP PHONES

Table 3. VLANS and their Ips

VLANs	NAMES	IPs
VLAN 10	STAFF	172.168.16.0

VLAN 20	STUDENT	172.168.32.0
VLAN 30	IP PHONES	172.168.64.0

2. 3. Conceptual Framework of System Development

In this work, a conceptual VoIP framework was analyzed for the implementation of VoIP technology at the University of Africa, Bulou-Orua Campus. The implementation is done using Packet Tracer and the network is based majorly on the Star topology. The infrastructure of the campus was considered during the framework design.

2. 3. 1. Attributes considered during the implementation of the framework

i. Placement of VoIP for Internal use over the current network.
ii. VoIP control and management of calls would be done by using the Cisco Packet tracer.
iii. Users can receive/make calls by using IP Phones.

Figure 1 shows how the packets are sent out over the IP network to their destination where the reverse process of decoding and releasing of the received packets is carried out.

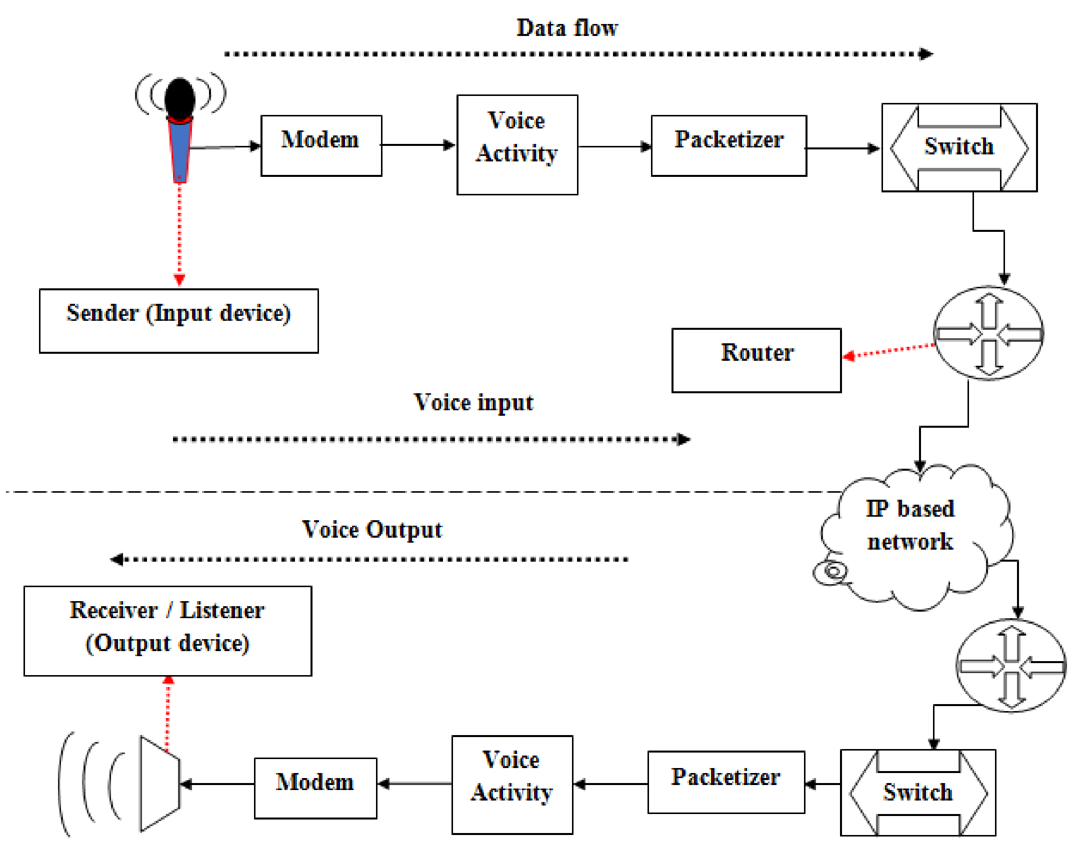

Figure 1. Signal flow process

2. 4. System Flowchart

The first telephony process represented by the flowchart shown in Figure 2 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.

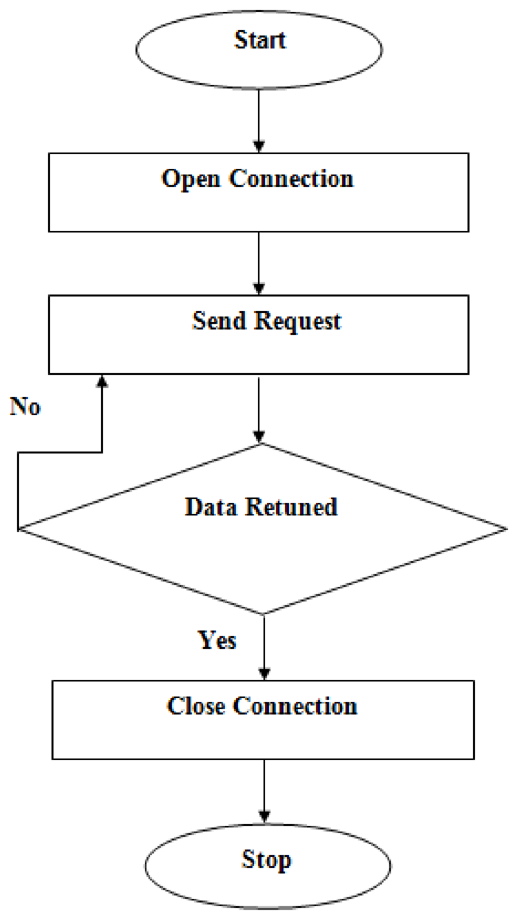

Figure 2. Flowchart for VoIP System

3. EXPERIMENTAL / RESULTS

Presented in this section are the analyses of the experiment conducted based on the conceptual framework as shown in Figure 1 in the previous section.

3. 1. Network Design Specification

The simulation of the VoIP system consisted of six network design model, and the sever room connecting all networks together; one for each of the three sites as shown in Figure 3.

a) The Sever Room Model: It was designed as a star topology with a central switch through which all other connections are made as shown in Figure 3.

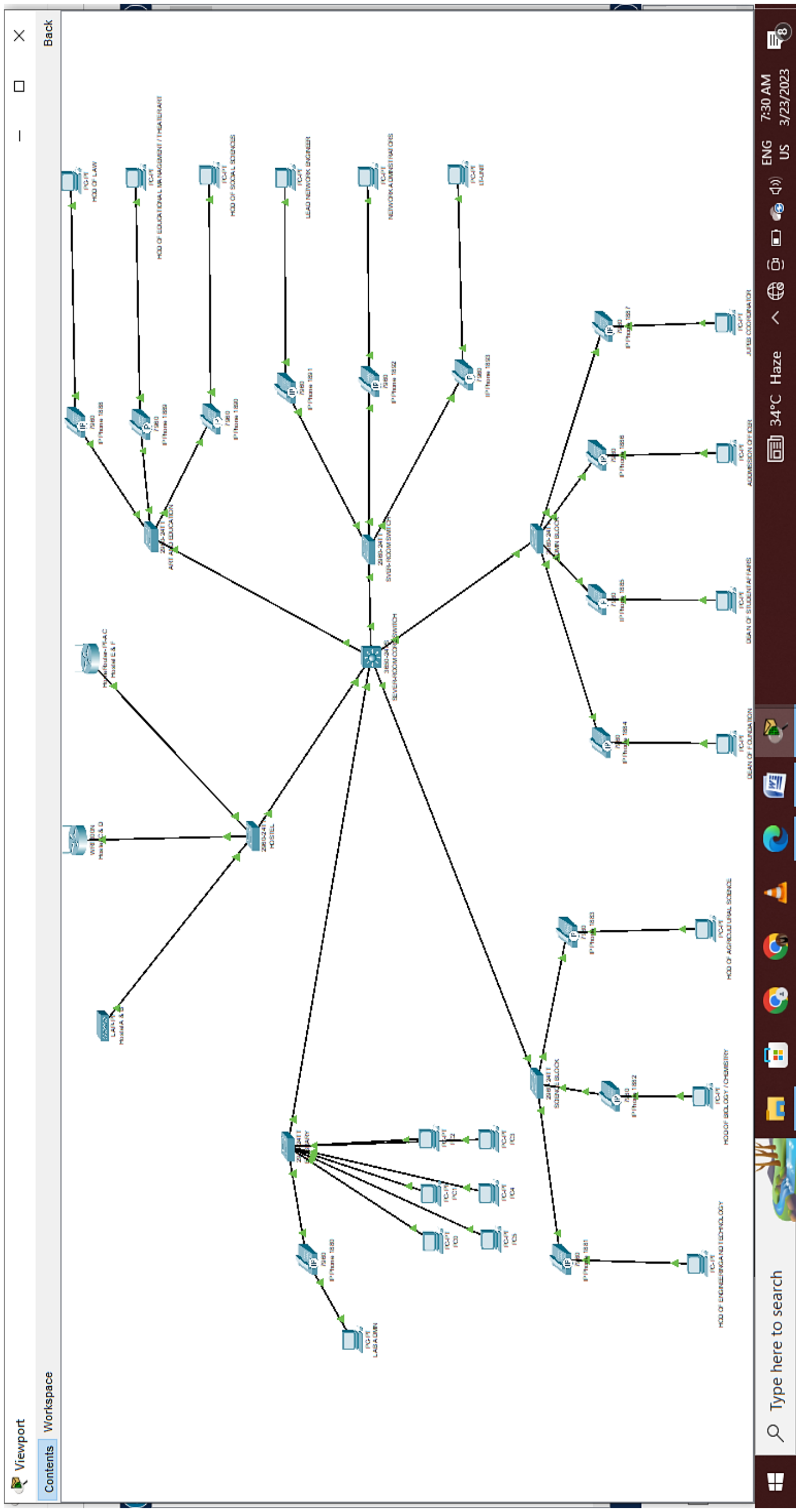

Figure 3. Star topology of the sever room

b) The Administrative Model: It was designed as a star topology with a central switch through which all other connections are made. The administrative block comprises four IP phones as shown in Figure 4.

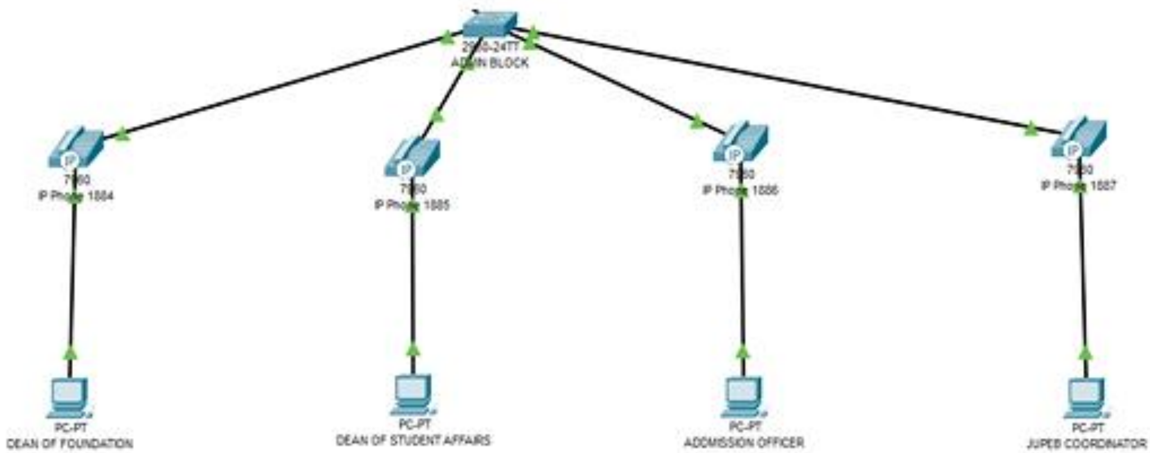

Figure 4. Star topology of the Administrative block

c) The E-library Model: It was also designed in a star topology but with more computers and IP phones than the administrative block as shown in Figure 5. The IP phone is been deployed only for the lab technician. This is due to the fact that there are more numbers of users (students) in the library site than the administrative block site.

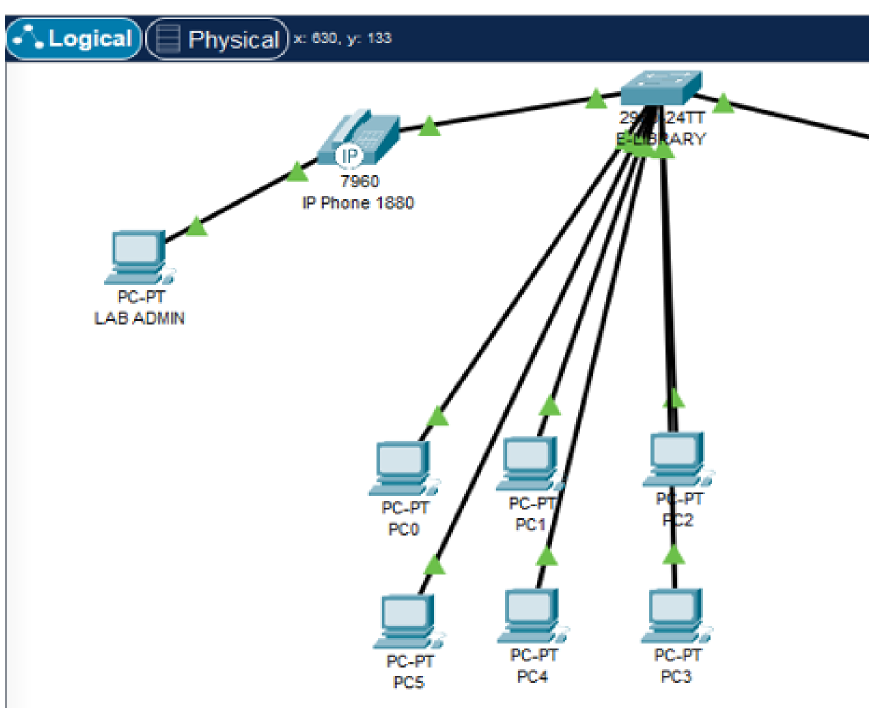

Figure 5. Star topology of the E-library

d) The Department Model: In order to simplify the simulation, the bus topology was chosen for the department because a prototype VoIP system was designed and simulated for one, just one department and this can be re-used for other departments as shown in Figure 6.

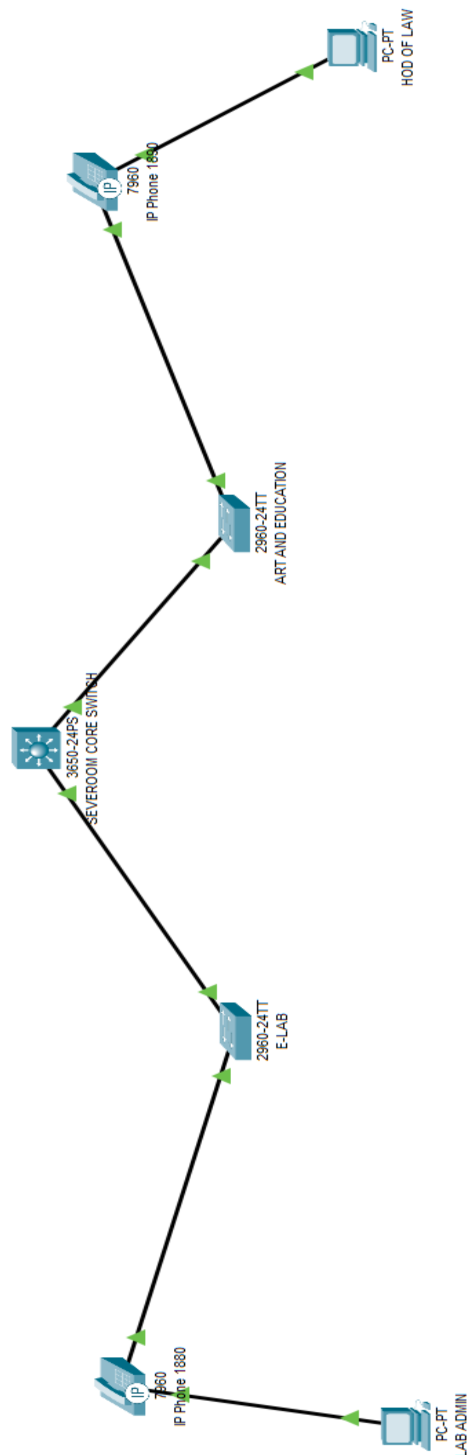

Figure 6. Bus topology of a single department

e) The Hostel Model: It was also designed in a star topology but with access points for wireless connections as shown in Figure 7. This is due to the fact that students can connect to the WiFi to do assignments, research work etc. after school period even at their various bunks. Students are to connect to the network using their personal device, i.e., smart phones, laptops, etc.

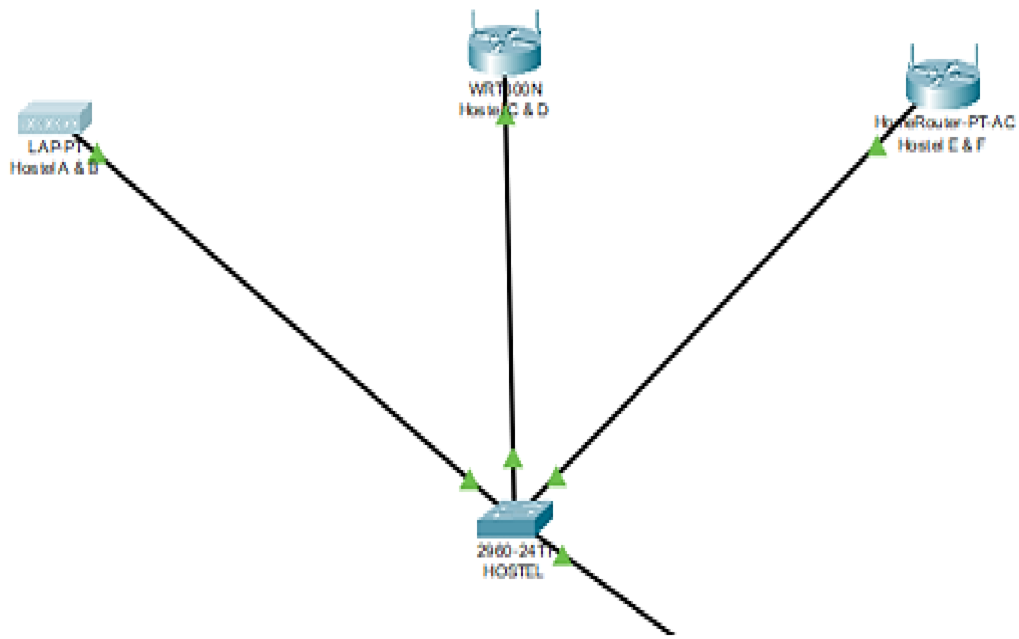

Figure 7. Star topology of the Hostel model

f) The Full Site Model: It comprises a router that is been fed by the Toru-Orua campus for Internet connection. This router is been configured to distribute IPs within IP phones located at various departments, switches and PCs for communication and data processing as shown in Figure 8.

4. DISCUSSIONS

It has been revealed in this work the benefits of VoIP over PSTN. The steps employed in Cisco Packet Tracer Simulation of the VoIP Telephony system added to the performance accuracy achieved in the work. The steps are: Step 1: Network devices were selected from the Cisco packet tracer device database and connected, Step 2: The VLANs and IP networks were created for different user segments and network traffic, Step 3: IP addresses were assigned to the six (6) different endpoints on the campus via the DHCP (Dynamic Host Configuration Protocol) from the sever room, Step 4: Static and dynamic routing protocols were configured on the routers, 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. Moreover, the results for star topology of the sever room, star topology of the Admin block, star topology of the E-library, bus topology of a single department, star topology of the Hostel model, and full site model as shown in Figures 3-8 are on a par with the work presented in Singh (2023).

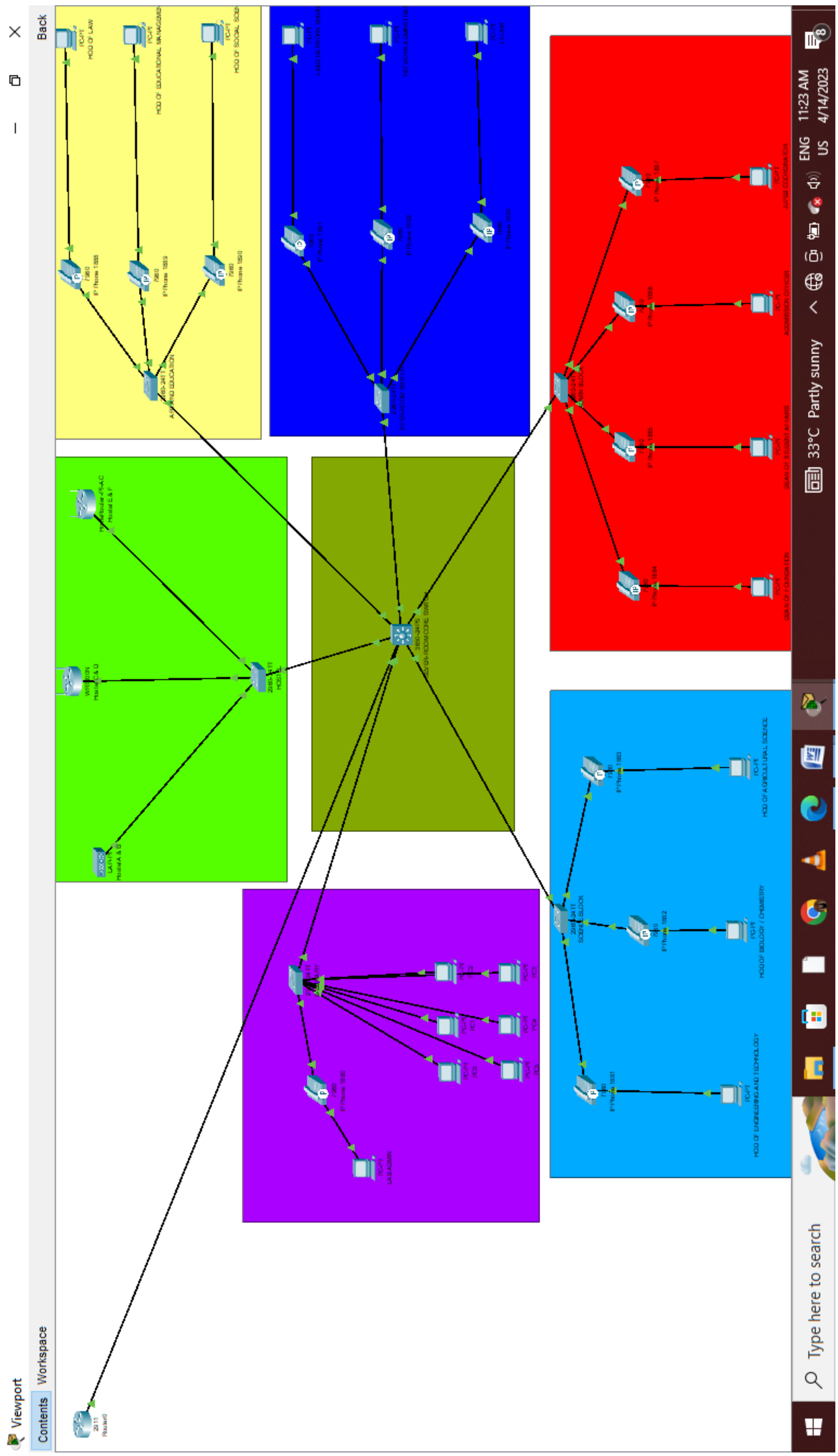

Figure 8. Full site model

We recommend the following; 1) 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, 2) Network security infrastructures such as VPNs, firewalls and so on optimize voice, and therefore should be implemented because they are capable of supporting the advanced security requirement of VoIP, and 3) VoIP comes with new complex threats; therefore, it is highly recommended that network security upgrades be carried out.

5. CONCLUSIONS

A prototype campus network has been designed and implemented in this work using the Cisco Packet Tracer software. The objective to design and simulate an efficient VoIP network scenario for the University of Africa (Bulou-Orua campus), Bayelsa, Nigeria and to also configure the virtual network devices of the simulation, evaluating point-to-point connections to ensure proper communication between various offices and departments has been attained. 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. Conclusively, this work has contributed to the understanding of individuals of the whole concept of VoIP and its ever-increasing demand in present times.

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