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Implementing Toll-Free Communication Framework as Smart Campus Initiatives for Tertiary Institution Sustainable Development

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Abstract--- The era of smart computing is initiating innovative ideas and making life more comfortable with its diverse implementations. Voice over Internet Protocol (VoIP) framework, as an integral part of smart computing, is an increasingly used technology for modern telephoning system. Telecom and Mobile Internet Service Providers especially in Nigeria are constantly introducing flexible cost user-friendly services for individuals and organizations to lower cost on communication. In contrast, it is challenging for emerging tertiary institutions to meet the enormous recurrent cost on telephony and communication services. In this paper, an open-source software was implemented to provide a free billing (toll-free) communication system. The concept of subnetting was further implemented to solve the problem of Internet Address (IP) conflict associated data-voice network. Effective communication among staff and students was achieved with the system and recurrent communication cost incurred through private telecom providers would be lower at the long-run.

Index Terms— Framework, Internet, Performance, Smart Computing, Soft-based, Technology, Voice-over-IP

1 Introduction

THE era of smart technology is changing the paradigm of human activities in recent times. Smart concept continues to grow with no restrictions to a single entity, but it also covers the aspects of human life and activities such as smart city, smart campus, smart grid, and even smart agriculture [1]. The "Smart" in technology has leaned on the growth and broad acceptability enjoyed by the Internet in virtually every human life today. There is a constant and continuous place for its expansion and coverage area [2]. Mobile data, an intensive application area of telecom service today, is made available via mobile networks with continual and rapid growing use of both data and voice services as VoIP technology. Most Social Media Networks (SMNs) such as Facebook, WhatsApp etc employ the two services in their applications.

Nigeria has witnessed over the last two decades, as evidenced by the rise in telecommunication companies, varied telecom services. Tremendous growth of users' penetration in telecom had also been recorded. From users' experience, communication cost has been lowered in respect to more efficient way for voice communication than what the traditional telephone system can offer with the use of VoIP [3]. The communication Cost-Benefits-Analysis is dependent on other advanced call features such as video call, voice-mail, e-mail and call conference, which certainly lead to the improvement of the organization's productivity as a whole.

Meanwhile, the primary goal of an organization, and as a feature of socially defined people, is growth. For achieving optimal growth, everyone has to be in constant communication with others in the social context. Whatever the content is, individuals should exchange and share thoughts, news, in other words they should communicate. In this regard, especially in organizations where formal and informal communication exists such as the educational environment under study, "communication within organization" plays a vital role in structuring the organizational activities, objectives, policies and procedures. Communication within an organization should be of benefit to the Faculty members and students. One of the most common ways of communication aside letter and memo is verbal communication. One fastest means of communication is through telephone system. Since public telephone system is liable to distruption, distortions and very expensive, individual organizations provide telephone steup to allow for certain users, numbers or time of the day at no call charges (toll-free).

A toll-free call communication system therefore, is a call setup that allow individuals, group of people or corporate organization to make call and/or use telephone services at no cost. Although, some drawbacks were highlighted in [3] for VOIP, core technology for implementing toll-free system, such as poor voice quality, identity and service theft, viruses, malware and denial of service (DoS). Call tampering is also one of the notable threats considered destructive to constant non-breaking operation of the service being provided by VoIP.

The question therefore, is how do we lower communication

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cost while still maintaining good call quality for an emerging institution like Federal Polytechnic, Ile-Oluji? We therefore, focus on providing a mixed-mode design framework (smartphone, computers, and Physical IP-phones) using open-source software with further interest in Internet Protocol (IP) address conflict and security resolution. The concept, as intended, would address the aforementioned drawbacks for implementing VoIP and a basis for further investigating the design for both Quality of Service (QoS) and Quality of Experience (QoE) being the recent standardization activities in speech quality research study [4].

2 TELEPHONY SYSTEM

Telephony, in which VoIP is a subset, is defined as the technology associated with the electronic transmission of voice, fax or other information between distant parties using system historically associated with telephone, a handheld device containing both a speaker or transmitter and a receiver. The first telephones were connected directly in pairs. Each user had a separate telephone wired to the locations he might wish to reach. This quickly became inconvenient and unmanageable when people wanted to communicate with more than a few people. The inventions of the telephone exchange provided the solution for establishing telephone connections with any other telephone in service in the local area. Each telephone was connected to the exchange via one wire pair, the local loop. Nearby exchanges in other service areas were connected with trunk lines and long distance service could be established by relaying the calls through multiple exchanges.

It is a clear fact that telephone communication is the most frequently used means of voice communication in an organization using fixed lines, Public Switched Telephone Network (PSTN) and wireless lines from Global System for Mobile Communication (GSM). Various telecommunication companies in developing countries like Nigeria provide extensive services with a target to satisfy low-end to high-end users. Some of their customer-friendly services are TruTalk, AWUF 4 U, Ofala, Smart Life, On-net, Night Call, Campus Zone or booster, Talk Special, Same or intra Network, Time Variant, Closed User Group (CUG), Post-Paid, Int'l etc (see [5]). Nearly all these tariffs do provide flexible cost and reconfigurable.

3 REVIEW OF REALTED WORKS

There has been a wide range application of VoIP as presented in research works. [6] presented a parametric prediction model for perceived voice quality in secure VoIP. Emphasis was laid on prediction methods such as regression and neural network while system testing for voice quality was carried out with two client computers rather than a mixed mode approach of softphone-hardphone. A VoIP system based on soft-switch technology was presented in [7]. The soft-switch architecture uses SIP technique as the control protocol over an IP network consisting of call server and media server. Although the work provided a C++ API for its interfacing, the real-life implementation was not carried out. Other related works are study of Skype over IEEE 802.16 [8], toll-free automation in agriculture using a dual-tone multi-frequency beep technique [9] and advertisement service using improved VoIP [10].

4 THEORETICAL FRAMEWORKS

As a way of cutting the cost of telephone communication even while telephone lines are still used, leads to the invention of Private Branch Exchange (PBX) technology. A PBX is a telephone exchange or switching system that serves a private organization and performs concentration of central office lines or trunks and provides intercommunication between a large numbers of telephone stations in the organization. The central office lines provide connections to the public switched telephone network and the concentration aspect of a PBX permits the shared use of these lines between all stations in the organization. The intercommunication aspect allows two or more stations to establish telephone or conferencing calls between them without using the central office equipment. This has been so successful with analog telephone device. Meanwhile the era of analog telephone system is fading away from our operating environment such as the school campus with the proliferation of smart devices enabling both call and Internet services on-the-go. Arising from the drawback is the adoption of IP technology to the analog system resulting into IP-PBX. The attendant effect from this is that rather than implementing telephony system as hardware-based, it could be implemented as a software system alone (IP-PBX) or a mixture of the methods (PBX + IP-PBX).

Each PBX/IP-PBX-connected station, such as a telephone set, a fax machine, or a computer system, often referred to as an "extension or end-terminal" must have a designated extension telephone number that may or may not be mapped automatically to the numbering plan of the central office and the telephone number block allocated to the PBX/IP-PBX. Our design made use of full IP-PBX techniques running soft-based IP-PBX, inter-building fiber network connections, and a mapped allocation numbering plan of extensions. Routing, call setup configuration & scripting are further carried out on the connecting deices and the extensions. Fig. 1 provides the detailed network design and architectural framework and the core features of VoIP is presented in table 1.

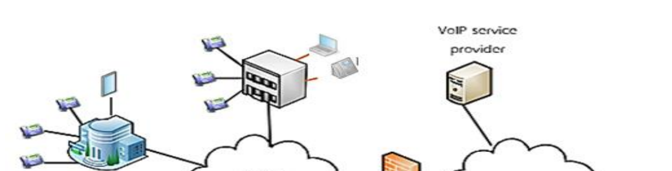

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TABLE 1
FEATURES OF VOICE OVER INTERNET PROTOCOL TECHNOLOGY

S/N	VoIP features	S/N	VoIP features
i.	Conference calling	ii.	Busy override,
iii.	call forwarding, and programmable caller ID	iv.	Call blocking
v.	Auto attendant	vi.	Call forwarding on busy or absence
vii.	Auto dialing	viii.	Call logging, Call pick-up
ix.	Automated directory services,	x.	Call transfer
xi.	Automatic call distributor	xii.	Camp-on
xiii.	Automatic ring back	xiv.	Conference call
xv.	Shared message boxes, Voice mail	xvi.	Voice message broadcasting
xvii.	Welcome message	xviii.	Call waiting
xix.	Public address voice paging	xx.	Do not disturb (DND)

5 MATERIAL AND METHODS

Our design, implemented within the campuses of Federal Polytechnic, Ile-Oluji was based on the existing network infrastructure. The existing infrastructure consist of fiber-optic backbone and a point-to-point network adequately supported by a range of distributed access points. An open-source software-Asterisk PBX [11] was fully impmented in the design.

5.1 System Review and Planning

The initial evaluation of phone system usage and VoIP awareness among staff and students of the institution was carried out to further provide answers to the quality of call, cost, providers' tariff effectives. For the purpose of our design, the call toll-free system is referred to as FedplelTalkOn.

5.2 System Design and Setup

The FedpolelTalkOn system is designed using a 3-tiered network technique (see Fig. 2) running on fiber optics backbone. The soft IP-PBX server system was implemented using an open-source software-asterisk. The software-based PBX provides the core functionality of a VoIP system configured to provide the following:

i. extension numbering, setup and management
ii. IP address setup and management
iii. Firewall setup and management for extensions- inbound and outbound information exchange.
iv. mixed mode extensions integration.
v. call functions and management among users (extension call, priority call, conference call, call transfer, call override)
vi. voicemail and call recoding

5.3 Network Design

Avariable length On-demand IP-address subnetting concept is implemented at this stage during the design. A variable length subnetting is a networking concept that make the network to have and use different subnet masks. The On-demand approach allows for flexibility in number of hosts-to-subnet design. A few hosts need a subnet mask that accommodates only these few hosts and vice versa. For instance, considering an IP address 192.168.0.0 in the Class C. The binary bit representation is:

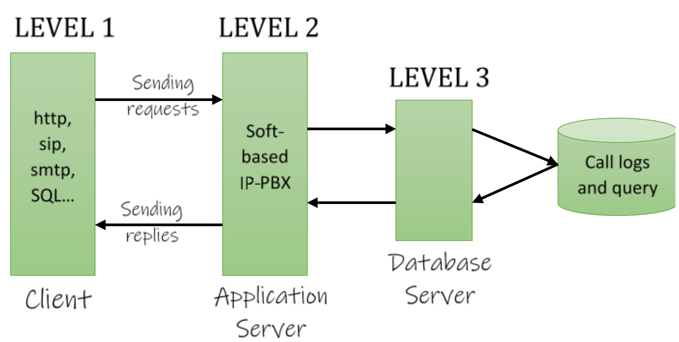

Fig. 2 The 3-tiered Architecture of FEDPOLEL Talk-On System

$$11000000.10101000.00000000.00000000 \quad (1)$$

1s are network bit and 0s are user bits. The number of 1s to turn to 0s is dependent on the number of users or systems to be connected to the network.

Let say, we need 100 nodes (e.g IP-Phones, Computer, printer etc.) on the network using class C: 192.168.0.0. To resolve the subnet, we need to determine the network bit (1) to turn to user bit (0) by using equation (2):

$$2^{n-2} \quad (2)$$

For 100 users/systems, the question is, how many user bits will give us 100 users from right to left of equation? (1).

i. 11000000.10101000.00000000.00000000

ii.

0	0	0	0	0	0	0	0
128	64	32	16	8	4	2	1

The number of user bits (0s) to accommodate this is 7 bits, which implies by converting binary to decimal with equation (2) for n=7, we have

$$2^{7-2} = 128 - 2 = 126 users$$

Usable IP-address

$$192.168.0.1 - 192.168.0.126$$

Broadcast address

$$192.168.0.127$$

Subnet Mask

$$256 - 128 = 128$$
$$255.255.255.128$$

6 Experiment Setups

The campus network runs on a fiber optics network as a backbone while the cat 6 cables were used for interlinks. The system installation and parameters configuration were implemented on Ubuntu Linux-based server virtual machine running *asterisk* and Wireshark. The other client system setup includes smartphone, softphone and hard IP-Phones. (as detailed in Fig. 1 and 4).

7 Results and Discussion

The design was carried out within the Academic Take-off Site (ATS) of Federal Polytechnic, Ile-Oluji as a pilot test. It comprising twelve (12) IP-Phones extensions in mixed mode approach setup on the existing network with over 500 initial users with an expected growth of over 1000 users. This means we need $2^{10-2} = 1022 users$. Applying the concept of On-Demand variable length subnetting concept, we have the following IP-address configuration:

Usable IP-address

$$192.168.0.1 - 192.168.0.254$$
$$192.168.1.1 - 192.168.1.254$$
$$192.168.2.1 - 192.168.2.254$$
$$192.168.3.1 - 192.168.3.254$$
$$192.168.4.1 - 192.168.4.6$$

Broadcast address

$$192.168.252.0$$

Subnet Mask

$$256 - 252 = 4$$
$$255.255.255.252$$

With this concept adopted, the followings were achieved as against the aforementioned drawbacks:

i. Provision larger IP-address pool to accommodate user growth on the network including IP Phones extensions

ii. IP-address resolution, thereby enabling each node to have sufficiently one unique IP address.

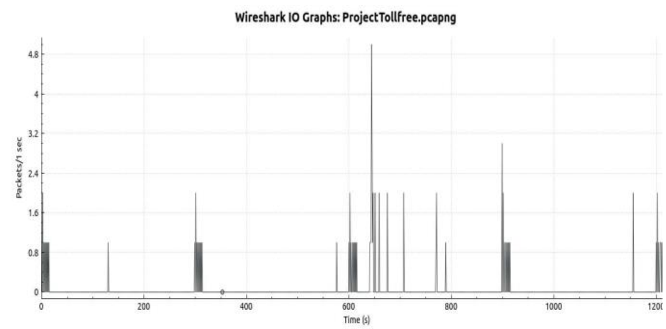

Fig. 3 Packet Transmission of FedpolelTalkOn System

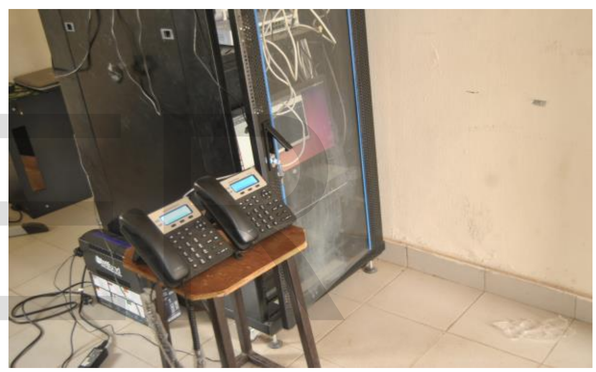

Fig. 4 A typical setup scenario of the Server and the IP-Phone extensions of System

iii. Better packet transmission was recorded even after the IP Phones extension for voice were added on the existing data network. (see fig. 3).

iv. Provision of better network security (e.g IP-address spoofing and malicious act) because the different subnet composition on the network.

v. Better voice quality was also experienced.

8 Conclusions

Initiating smart computing in today's 21st century campus's administration and services cannot be over-emphasized. The derivable benefits (such as efficiency, better performance, cost reduction and better collaboration) worth the capital investment.

We have been able to provide a framework for a call toll-free system and implementied same using an open-source software with the intention of operating on the existing network infrastructure. Our implementation further provides the concept of generating large IP addresses on the campus network to accommodate both data and voice, thereby overcoming the challenges associated with VoIP [3]. Meanwhile other quantitative parameters and metrics to test network efficiency were not considered in this paper. We recommend that E-model and other metrics be used to validate the efficiency and the performance of the campus network fo Quality of Service (QoS).

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