Implementing Soft-Based Voice-Over Internet Protocol (VoIP) Technology for Emerging Tertiary Institution Sustainability

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***Abstract*—Voice-over-Internet Protocol (VoIP) framework, as an integral pa rt of smart computing, is increasingly changing the way telephony system is being used in our modern world. Telecom and Mobile Internet Service Providers (ISP) especially in Nigeria are constantly introducing flexible cost-friendly services for individuals and organizations. Moreover, it is challenging for emerging tertiary institution to manage and minimize cost of telephony and communication services. On the other hand, existing services are usually flooded with poor Quality of Service (QoS). We therefore, provide a soft-based open-end architectural communication framework to implementing a campus-wide communication system on toll-free basis. In addition, a variable length on-demand subnetting technique was adopted to provide for secured and flexible Internet Protocol (IP) address large pool. Call performance and evaluation of the campus toll-free system provided effective cost minimization and wider call coverage.**

Keywords: smart Computing, telephony, mobile, cost-friendly, toll-free.

I. 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, service area and high speed; which are in the forefront of technological innovation and constant stream of ideas [2]. The mobile data services, an intensive application area of Internet today, are made available via mobile networks with continual and rapid growing use of both Data and Voice services as Voice over Internet Protocol (VoIP) technology. Most Social Media Networks (SMN) are enjoying the combination of both voice and data services today.

Nigeria has witnessed over the last two decades; as evidenced by the rise of many existing and new telecommunication company (such as MTN Nigeria, Airtel, Globacom Nigeria, 9Mobile) providing services spanning provisions of VoIP; a tremendous growth of users' penetration in telecom. With users' experience, cost has been lowered in respect to more efficient way for voice communication than the traditional telephone system can offer using VoIP [3]. The cost benefits scenario of VoIP is feasible from 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 is growth and as a feature of socially defined people; everyone is in communication with others in the social context. Whatever the context 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 employees. One of the most common ways of communication aside letter and memo is verbal communication done through voice. Effective means of voice communication within and even outside the organization has been in the front-liner through the use of VoIP technology.

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 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. Meanwhile the public telephone service is associated with noise and high cost. To overcoming these challenges, VoIP is considered to provide a toll-free (a call setup that allow individuals, group of people or corporate organization to make call and/or use telephone services at no cost) communication system with the possibility of flexible reconfiguration of the networks where users are provided with call services at no cost.

We therefore, focus on providing a mixed-mode design framework (smartphone, computers, and Physical IP-phones) with interest in Internet Protocol (IP) address conflict resolution and security issues. The design would provide for both Quality of Service (QoS) and Quality of Experience (QoE) being the recent standardization activities in speech quality research study [4].

II. AIM OF RESEARCH

It is a clear fact that telephone communication is the most frequently used means of voice communication in an organization but it is associated with attenuations, disruptions and distortions in the case of either fixed lines from Public Switched Telephone Network (PSTN) or wireless lines from Global System for Mobile Communication (GSM). Moreso, the cost of making calls using these providers is high. Although service providers are constantly reviewing call and data cost with provision of customer-friendly services such as 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 [5]. Nearly all these tariffs do not provide flexible reconfigurable features provided in VoIP implementation through software-based approach that could cut across both staff and students need and personalized call features provisioning in educational environment. Aside cost associated with calls and data services, hardware implementation of VoIP does not allow flexibility in extending the network (IP Phones extensions or nodes) beyond the capacity of the hardware-Private Branch Exchange (PBX). Therefore, the need for soft-based PBX for VoIP implementation. Though, some drawbacks were highlighted in [3] for VOIP such as poor voice quality, identity & 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. These drawbacks are part of the focus to be addressed in this research work while implementing VOIP as smart computing campus initiatives. The design is implemented within the campuses of Federal Polytechnic, Ile-Oluji with the support of 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.

III. RELATED 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].

IV. THEORETICAL APPROACH

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 this is the adoption of IP technology to the analog system resulting into IP-PBX. The attendant effect allows for implementing telephony system as hardware-based. Software implementation of PBX allows for better management and reconfiguration.

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. We focused on the use of full IP-PBX techniques running soft-based IP-PBX with interconnections of fiber network connections, and a mapped allocation numbering plan of extensions. Routing, call setup configuration and scripting are further carried out on the connecting devices.

Call initiation principles were considered in the design to comply with the E-model standard (see Fig. 1). Fig. 2 provides the detailed framework and the core features of VoIP are presented in Table I.

V. METHODOLOGY

I. Step I: Design study

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 and providers' tariff effectives. For the purpose of our design, the call toll-free system is referred to as FedplelTalkOn.

II. Step II: :Network Mapping, Design and Extension Numbering

The FedpolelTalkOn system is designed using a 3-tiered network technique (see Fig. 3) running on fiber optics backbone. The soft IP-PBX server system was implemented using an open-source software-asterisk [11]. The software-based PBX provides the core functionality of a VoIP system configured to provide the following:

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

III. Step III: :Subnetting Techniques

A variable length On-demand IP-address subnetting concept is implemented at this stage of 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:

11000000.10101000.00000000.00000000

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 1.

$$2^{n-2} \tag{1}$$

For 100 users/systems, the question is, how many user bits will give us 100 users from the right to left.

11000000.10101000.00000000.00000000

The number of user bits (0s) to accommodate this is 7 bits as shown in Table II., which implies by converting binary to decimal with equation 1 for n=7, we have

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

TABLE II. AN OCTET BIT-DECIMAL CONVERSION

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

Usable IP-address

$$192.168.0.1 - 192.168.0.126$$

Broadcast address

$$192.168.0.127$$

Subnet Mask

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

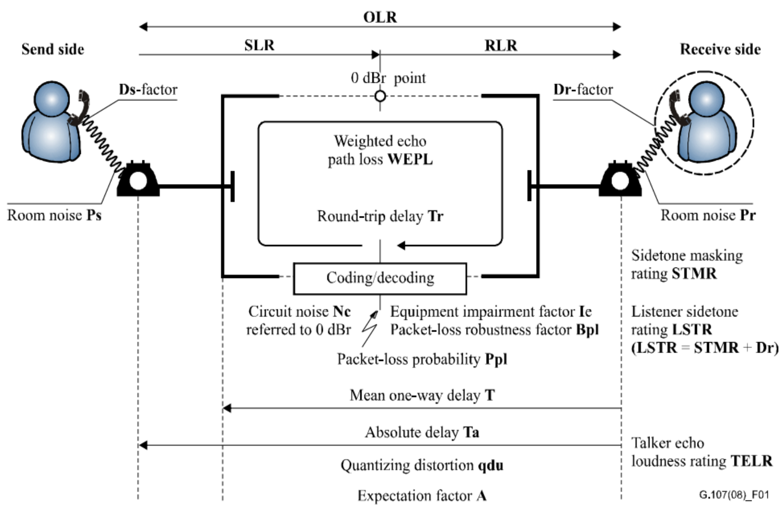

Fig. 1 E-Model Connection Framework (source: ITU, [12])

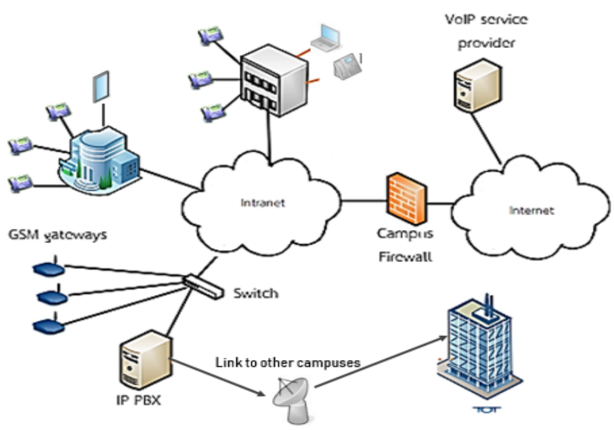

Fig. 2 Framework of VoIP as FedpolelTalkOn Toll Free System

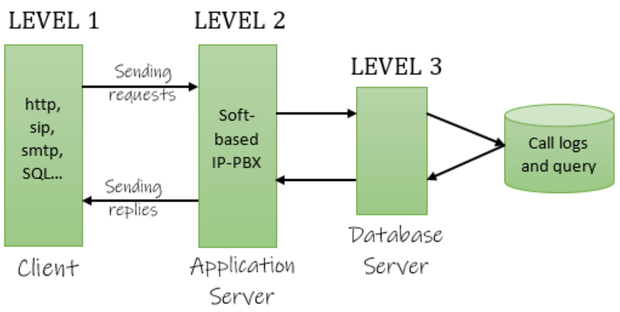

Fig. 3 The 3-tiered Architecture of FEDPOLEL Talk

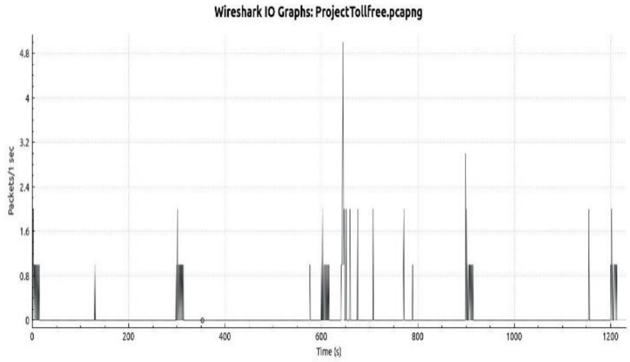

Fig. 4 Packet Transmission of FedpolelTalkOn System

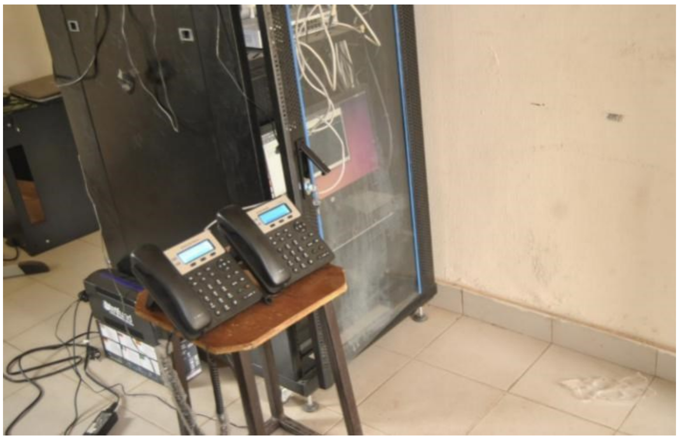

Fig. 5 A typical setup scenario of the Server and the IP-Phone extensions

TABLE I. 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)

VI. EXPERIMENT SETUP

The campus network is run on a fiber optics network as a backbone while the cat 6 cable is used for interlinks. The system installation and parameters configuration are implemented on Ubuntu Linux-based server virtual machine running asterisk and Wireshark [13]. The other client system setup includes smartphone, softphone and hard IP-Phones. (as detailed in Fig. 1 and 5 respectively).

IV.

VII. RESULTS AND DISCUSSION

A pilot test comprising twelve (12) IP-Phones extensions in mixed mode approach was 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:

- provision of larger IP-address pool to accommodate user growth on the network including IP Phones extensions
- IP-address resolution, thereby enabling each node to have sufficiently one unique IP address.

- better packet transmission was recorded even after the IP Phones extension for voice were added on the existing data network. (see Fig. 4).
- provision of better network security (for example IP-address spoofing and malicious act) because the different subnet composition on the network.
- Better voice quality was also experienced.

VIII. CONCLUSION

The paper has demonstrated the implementation of VoIP as an effective means of providing communication system for seamless communication services. A framework to implementing a call toll-free system using an open-source software with the intention of operating on the existing network infrastructure was also proposed. Having to merge the existing network facilities with the additional nodes requires efficient, secured and large pool of IP addresses. These issues have been practically addressed in the paper, thereby achieving greater Quality of Service (QoS). Therefore, smart computing in today's 21st century campus's administration and services cannot be over-emphasized. Other quantitative parameters and metrics to test network efficiency were not considered in this paper. We recommend that E-model as highlighted in the paper be further investigated to determining these metrics.

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