A Virtual Private Branch Exchange Model For Improvement Of Voice Call Quality In Intranet And Internet Communication Systems

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***Abstract*—It has been observed that VoIP calls, such as on WhatsApp, Skype and Viber, have call quality issues, such as echo, jitter, noise, packet loss and delay. Various researchers have attempted to resolve these issues using packet loss concealment, machine language, codec selection and mathematical methods. However, the problem of call quality still persists. This study therefore developed a model for improving call quality in a VoIP communication system known as Virtual Private Branch Exchange (VPBX). The VPBX client-server application (app) model was designed using JAVA programming language in a NetBean environment. For the evaluation of the model, the client software application (app), and server application were installed on three computers. Two of the computers served as clients with a client app installed on them for VPBX system calls, while the third served as a server with a server app. The outcome of the study shows that one can leverage on the current broadband technology offered by the Internet to make calls both within and outside the organization. In conclusion, it has been demonstrated that VPBX offered a very good voice call quality by incorporating existing Internet facilities, without running cables. It is therefore recommended that organizations should use VPBX to make and receive calls within and outside organizations to save cost.**

Keywords—Local Area Network, Private Branch Exchange, Voice over Internet Protocol, Virtual Private Branch Exchange, Wide Area Network

I Introduction

The penetration rate of mobile broadband is driving the shift to Internet Protocol (IP) services. Over-The-Top services are services offered over the Internet without going through telecommunication operators to satisfy customers. Voice calls over the Internet are normally referred to "VoIP" which means Voice over Internet Protocol. Calls though cheap are associated with call quality issues such as echo, jitter, noise, packet loss and delay. However, the issue of call clarity in telecommunication, when receiving or making calls has remained a concern to both telecommunication operators and users.

According to [1] Efiong and Aranuwa (2017), mobile devices have gained 94% penetration in Nigeria. This means that we have high percentage of people who make calls, because most mobile devices are used in making and receiving calls, which is one of the real time processes of communication. With this high penetration rate of mobile devices, there is the need to solve the problem of call quality for users to enjoy the readily available utilities provided by network operators. With increasing globalization, communication has become a medium of empowerment because information is power. [2] Worldometer (2020) has it that Nigeria's current population as at Thursday, March 19, 2020 is about 204,632,055; which is 2.64% of the world's population. Imagine how difficult it would have been trying to communicate with each other using face to face communication, hence the need for an improved and cheap means of telecommunication. Various social media have been used in order to accomplish the quest to communicate with one another. These social media platforms include Facebook, Twitter, blogging, LinkedIn, Snapchat, Instagram, telegram and WhatsApp [3] (Gikas & Grant, 2013). However, this cannot be substituted to real time online voice communication which is the fastest means of communication worldwide.

Telecommunication users want to settle for the best communication quality, hence the need for improving service delivery by telecommunication operators. VoIP was one of the ways in which the cost of calling over long distances was reduced, particularly for international calls [4] (Jung, Mo & Park, 2017). The issue of voice clarity in communication, when receiving or making calls, has remained a challenge and a concern to phone users. Call quality is affected by challenges like echo, jitter, noise, packet loss and delay.

The objective of this paper is to utilize the broadband technology offered by Internet Service Provider (ISP) to utilize the services is the Internet to make voice quality calls within and outside organization/institution at no cost to the users. The paper provided a voice quality architecture that considered call quality characteristics such as listening qualities, conversational and transmission characteristics. Software applications which consist of both client and server apps were developed using JAVA programming language in a NetBean environment. This developed model application is known as Virtual Private Branch Exchange (VPBX), which is a model that is used in communicating over the Internet. The VPBX can be used on Local Area Network as well as Wide Area Network.

The remaining sections of this paper is sub-divided the following: section two is about review of related works on VoIP and voice quality. The third section is on methodology used in achieving the set objectives, the fourth section is result and discussions, finally conclusion and recommendation.

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II Literature Review

A. Architecture of a VoIP Model

The typical VoIP model architecture is illustrated in Figure 1. The voice source is an analog signal which is passed to a playback buffer with the function to collect and store data through the Internet Protocol (IP) network. These packets are then sent to the receiver end where they are decoded and the voice is received as an analogue signal.

Figure 1: Architecture of Voice over Internet Protocol. (Source: Jalendry & Verma, 2015).

Block Diagram of conventional VoIP Architecture
The current VoIP Architectural block diagram as depicted in [6] Figure 1 is presented in a block diagram form in Figure 2. This figure highlights some of the characteristics of current VoIP models.

The existing VoIP architectural communication system is made using a Public exchange system (PBX) and an Internet connectivity or either one of these connections using VoIP phones for connection as end devices.

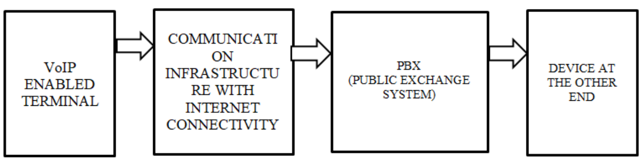

Figure 2: Block Diagram of VoIP Models
The existing system structure consists of groups of Internet phones attached to the Internet enabled devices that use IP telephony technology to route phone calls through the internet from one caller to another, as shown in Figure 2. The Internet infrastructure delivers the phone call to a PBX which is a telephone switching system that connects to the caller who in turn links the end user with whom the caller intended to communicate. The virtual switching server as software that minimizes delay, suppress noise, echo and jitter. The phone devices include computers and connection could be through intranet or internet or even both.

2.3 Review of Related Works
The tabular representation is a summary of closely related works done by other researchers all in the bid to seek for ways to improve voice call quality.
Perwej and Parwej (2012) noted that the new methods (Perceived Evaluation Speech Quality (PESQ) and E-model algorithms) have a superior cost benefit and suitable quality compared to the normal Adaptive Buffer Algorithm. Experiments were carried out to compare fresh models and Adaptive Buffer Algorithms with elevated and low network delay performances under distinct network circumstances. Only delay variation (jitter) and no other parameter such as signal loss and reverberation can be used with this technique. Newton and Arockiam (2013) proposed a method to store the QoS data transfer requirements and data delay performance evaluation. The work adopted the analysis of sample data to identify if the QoS problem occurred when the Service Data Unit (SDU) size was 128 byte or when it was 1024 byte. They went further to investigate if delay occurred during mean delay or 95 percentile delay. The outcome of the analysis was theoretical. A practical analysis of this evaluation would have been a bench mark on which other researchers would have ridden on to solve QoS problem related to delay in packet radio services. Slavata and Holub (2015) researched on the effect of codec and QoS technique used on speech transmission quality in IP networks. Ten samples were measured for the various methods of controlling and preventing congestion. The intervals of confidence were also calculated by a mathematical technique. In comparison to high-bit rate codecs, low-bit rate codecs yielded worse outcomes, however low-bitrates could be helpful in low bandwidth. In this work, the techniques were not addressed in detail however the results were shown graphically. Dahivadkar and Limkar (2017) used sampling method in their work. The proposed system was based on theoretically estimating instantaneous residual echo variation, nonlinear acoustic echo and delay. The proposed system took into account more factors that bring about undesired echoes. The proposed system efficiently helped reduce acoustic echo by the use of factors that were not considered in other systems showing that improvement can be made on the existing Acoustic Echo Cancellation (AEC) method. The work however did not consider other factors that could affect VoIP call apart from echo. Le, An Thanh, (2017) wrote on the title “Controlling and Monitoring Voice Quality in Internet Communication”, Markov Mathematical model of calculating Jitter without time stamp method was deployed. The output of the research gave a better result on how network could be controlled and monitored compared to E-Model and R-factor methods that were used by former researchers. Hidavat & Saputra (2019) discussed how VoIP was implemented as a way of reducing cost of communication in a University. The work used SWORT analysis to justify the importance to carrying out the work. No discussion has been given of the research method of Projecting, Planning, Designing, Implementing, Operating and Optimizing (PPDIOO).
Gap Analysis

[7] Perwej and Parwej, (2012) compared PESQ to Adaptive Buffer Algorithm to know which gave a better performance. [8]Newton and Arockiam, (2013) compared Service Data Unit (SDU) of 128 byte with that of 1024 to ascertain which gave a better QoS result. [9] Slavata and Holub (2015) compared Low bitrate codecs and high bitrate codecs, to ascertain which gave a higher voice quality in IP network. [10] Dahivadkar and Limkar (2017) was basically on acoustic echo cancellation and not considering any other parameter. [11] Le, An Thanh, (2017) approach was using a new mathematical model apart from e-model and R-factor to compare which gave a better monitoring of voice quality output in internet communication.[12] Hidayat & Saputra (2019) implemented an open source VoIP in a campus environment and used a survey method to get customers response on its advantage.

In all these researches and more, the researchers considered one or two VoIP parameters. In addition, none built a software to combat the challenge of poor voice quality in a VoIP system. Hence this research is bridging the gap by considering five VoIP parameters, namely: echo, jitter, noise, packet loss and delay. These parameters affect VoIP voice quality in one way or the other. The research has the added advantage of making free voice call at a clearer voice quality.

III Methodology

Call quality is affected by challenges like echo, jitter, noise, packet loss and delay. Figure 3 shows a block diagram of elements that affect call quality.

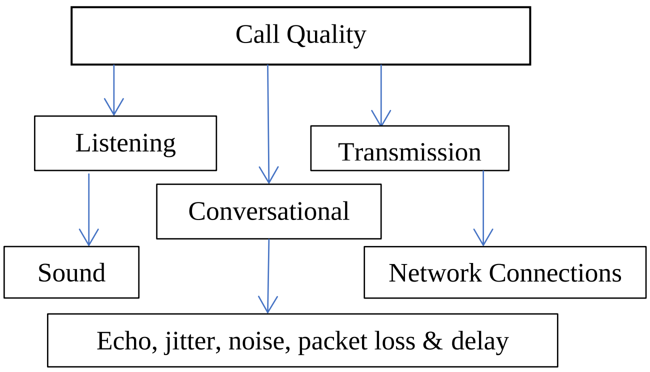

Figure 3 Conceptual block diagram of Call Quality

Call quality can be divided into three: listening qualities, conversational and transmission characteristics. Quality of hearing is related to quality of sound, conversational quality refers to both conversation and listening qualities which may include echo or delay issues that can affect conversation. Transmission qualities include network connection that is used to transmit the voice, in the area of network service quality.

This is the method the researcher used in developing the VPBX software app model to actualize voice call quality. To develop the software model for improving audio call quality in a VoIP communication system, a Virtual Private Branch Exchange (VPBX) Switching model was developed using JAVA programming language in a Netbean environment to ensure clarity of call. The system was based on client-server architecture. The VPBX server serves as a point of connection to all nodes on the network. The VPBX server was designed to use five commination channel ports. These are ports 8001, 8002, 8003, 8005, 8006. Port 8001was dedicated for call request from a client to the server and to another client. Port 8002 was used for call response in the server in order to prevent communication conjunction. When a node is communicating with the server and another node places a request to the server, the server opens another port, which is port 8003 for communication. Port 8005 and 8006 are used for speaking and hearing respectively, in other to allow free communication flow.

An enhance database was used to store the full name, username, password, current IP address and a short phone number (four digit number) in order to identify the client on the network.

The flow chart of methodology framework is as shown in Figure 4

This a framework showing the general overview of the processes followed to obtain the desired result.

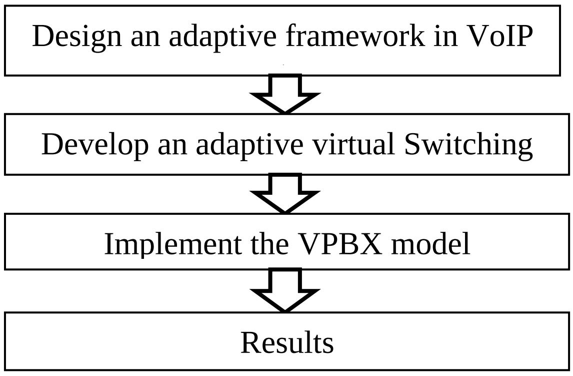

Figure 4 Framework of Virtual Private Branch Exchange (VPBX)

3.1 Software Used for VPBX development

An implementation Platform was developed in a network. The coding was done using JAVA in a NetBean environment.

3.2 Hardware Used

A Personal computer (PC) was used with 2.40 GHz Intel core i3 processor, Microsoft Windows 10 Operating System (64 bit), 8.0 GB of RAM system served as server. Other computers used as clients had 4GB RAM each.

3.3 VPBX Client-Server Architecture

The VPBX is a client-server architecture that has the client app and the server app. The server app can be installed on computers with RAM of about 8GB or more. This server end can run on organization/Institutions network to enable clients communicate with each other through Intranet, Internet, wired or wireless network services. Using GSM, individuals insert sim with registered number to send and receive calls. In other VoIP software's, one download the app and register mobile number to enable sending and receiving of calls. VPBX model app, enable administrator to records particulars of the clients on the server. The administrator has the right to insert, delete, populate the server to suite the organization/institutions need. The username and password to send and receive calls is given to individual by the administrator. The server is developed to respond to changes in the environment as the caller or receiver moves from one location to another, hence the adaptive nature of the VPBX model app.

3.4 Flowchart showing the Virtual Private Branch Exchange (VPBX) Server

Figure 5 shows the flow chart of the VPBX**.** The processes were divided into three (3) sections which include: input, process and decision making stages. The input stage includes: declaration of server address, declaration of server listening port and declaration of server parameters. The process stage comprises of "start listening for request", "get the header request", "test connection to the request in the header", "recipient not available", "recipient not ready" and "connect the two clients on a new port for calling". The decision stages are, "is request received", "if exit then" and "is client ready".

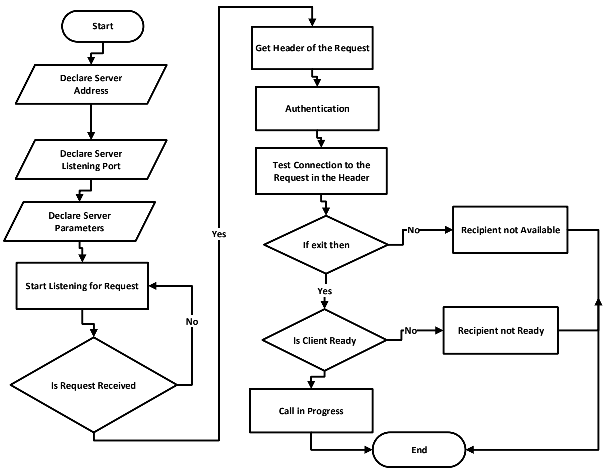

Figure 5 Virtual Private Branch Exchange (VPBX) Server Flow chart

Declare Server: The server is being setup in order to identify the node or client on the network. Declare Parameters: Username, password, IP address are all declared. In listening to request, the request coming from the client is listened to by the server. The request headers contain vital information about the resource to be fetched, or about the client requesting for the resource. The request header also allows the server to process as-well-as identify the client on the network.

3.5 Other information on each step of the VPBX flowchart explained as follows:

Declare server address:
This is an input point where the server is being setup in order to identify the node or client on the network.
Declare server listening ports:
During the development of the model, port 8001 was assigned as authentication port. This port allows the server to make request and also listening to request made by the clients on the network. Port 8002, allows server to respond to request made by users who have authenticated. Port 8003, allows for direct connection between the caller and the receiver. Ports 8005 and 8006, for sending and receiving of calls.
Declare server parameters:
This is an input point where the IP address and the ports of the server are declared, to allow client to connect to the server through the IP address.
Start listening for request.
This is a process whereby the server listens to request, coming from the client.
Is request received?
This is a decision process to check if the request received from the client is valid.
Get the header request.
This is a process where the client makes request to the server and the server gets the IP address, username and the password of the client. This process allows the server to identify the client on the network.
Authentication of client
This is a process where all users or client on the network authenticates in order to allow communication. No client on the network is able to communicate without authentication.
Test connection to request header.
This is a process whereby the server connects both clients, to allow free communication flow.
"If exit then"
This is a decision process to check if the client exist on the network, if YES it then proceed to "is client ready" and if NO the process terminate.
Is client Ready
This is a decision process to check if the client is ready for communication, if YES then communication continue, otherwise (if no) the process is terminated.
The processing algorithm of the VPBX is as displayed in the diagram in Figure 6.
An expanded VPBX architecture is as shown in Figure 6. The algorithm used works as streams; hence the challenges in the traditional PBX have been eliminated to enhance the voice quality of the system.

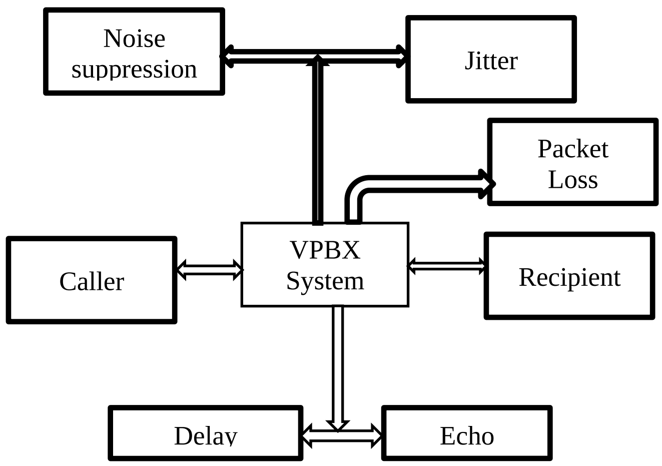

Figure 6 Expanded VPBX System (Source: Authors Field work)

As shown in Figure 6 the VPBX system which is the processing box, removes jitter, noise, minimizes delay and suppresses noise and thereby facilitates sound quality of the system. This is due to null values assigned when the model is being designed.

Some extracts from the code explaining what happens in the VPBX system. Please see appendix 11 for more details.

/* The system implements calling and receiving capability through thread classes

* There are two major thread classes in this research project. Namely

*Thread 1 and Thread2

*Thread 1 handles calling while

*Thread 2 handles receiving

Thread is a light weight packet code which programmers use to improve network performance through parallelism

*Two major development measures adopted to remove system generated echo, jitter, noise, packet loss and delay in delivery.

*1. The packet size of the audio message was increased. Regular size should be 256 from our experiment *we found that 512 and 1024 could work and this will help to transfer large signals.

*2. Dedicated port channels for communication were used by making sure that port assigned to the *microphone of sender is the same attached to the speaker of the receiver. And the port assigned to the *speaker of sender is also attached to the microphone of receiver making a direct exchange of audio *and avoiding collision that could result in using single channel. This is responsible for removing echo, jitter, noise, packet loss and delay in delivery.

*Dedicated channels for audio transmission only were ports 8005 and 8006

*other channels for communication and requests were

1. 8001. Used for login request to server or request form server

2. 8002. Used for call response in the server in order to prevent communication conjunction.

3. 8003. Used in establishing a fresh connection to a new user

*/

3.6 Mathematical Representation of Caller-Receiver

$u = f(x_1, x_2)$ Equation (1)

Where u –> Means, that call went through and it is a function of x_1 (caller) and $x_2(receiver)$

If there is no caller and no receiver, the call will not get through zero (0)

If there is a ready receiver and no one calling, we still have zero (0)

$x_1 = g(t_5, t_6)$ Equation (2)

Where x_1 is a function (g) of t_5 and $t_{6,}$,

t_5 ⟹port 8005 and t_6 implies (⟹) port 8006.

The caller x_1 uses t_5 to send and $t_{6,}$ to receive

$x_2 = h(t_6, t_5)$ Equation (3)

Where x_2 is the function of t_6 $and t_5$, t_6 implies for port 8006 and t_5 implies for port 8005

x_1 uses t_5 to send and t_6 to receive respectively

x_2 $uses$ t_6 to send and t_5 to receive respectively.

$u = f\big(g(t_5, t_6), h(t_6, t_5)\big)$ Equation (4)

$= (\alpha t_5 + \beta t_6 \cup \rho t_6 + \sigma t_5)$

Where α, β, ρ and σ are constants

The VoIP Architecture of Figure 1 is used to Benchmark the VPBX model in Figure 7. The feature of this model is compared with existing VoIP model and how they differ in functionality. The feature in table 1 gives a clear difference between how the developed VPBX model app is compared to the existing VoIP model in terms of functionality and equipment's.

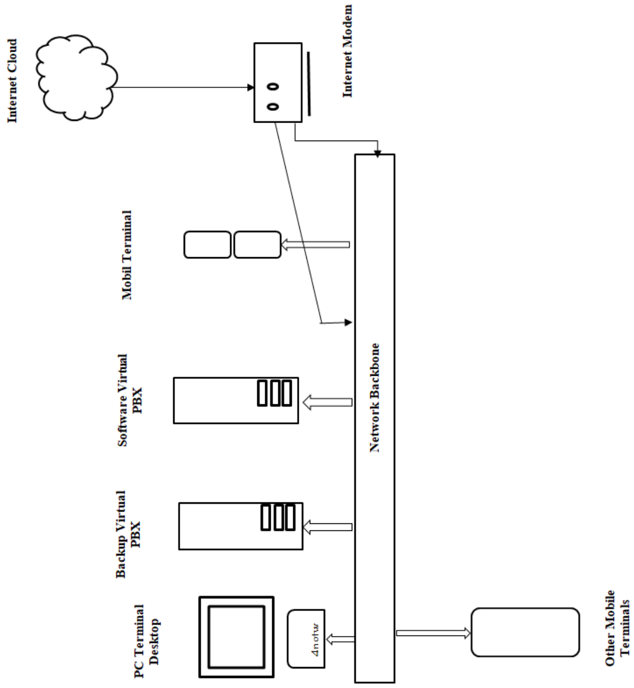

Figure 7 Design of VPBX Model for call clarity

3.7 Difference between benchmarked model and VPBX model

The benchmarked work is an architecture of Voice over Internet Protocol (VoIP) by [6] Jalendry and Verma (2015). This is achieved through tabular representation.

Table 1 Difference between benchmarked model and VPBX model

S/N	Features	Benchmarked Model	VPBX Model
1.	VoIP Provider	The model depended on external entity to provide VoIP	This model is a Server of its own, providing services to clients.
2.	Public Switch Telephone Network (PSTN)	This is required.	This is not required.
3.	VoIP Gateway	This is required.	This is not required.
4.	Session Initiation Protocol (SIP) Phone	This is required.	This is not required.
5.	Message Delivery	Messages are delivered in packets.	Messages are delivered in streams.

This research is aimed at allowing users to use organizations Internet infrastructures to make call with clear voice quality. Its call log information does not reside on the cloud but within the organization and can function like other PBX but using the organizations Local Area Network (LAN) or Wide Area Network (WAN) to function. Customized numbers of three or four digits are used instead of the long digits as in the case of global system for mobile communication (GSM). One will still be able to make calls within the organization using the Intranet system even when the internet is not functioning. The parameters used in developing the VPBX makes the model to evolve with the real world situations. The VPBX model is configured to integrate the services of Intranet, Internet, wired or wireless connections to communicate within and outside the organization.

IV Result and Discussion

This section of the research gives us the result of the VPBX and a discussion of how the result was obtained.

4.1 Virtual Private Branch Exchange (VPBX)

The virtual Private branch exchange (VPBX) is a model that does a similar function as the traditional Private branch exchange (PBX), without the normal running of cables but connects virtually in such a manner that it adapts to the environment without the manual switching process. In the case of the traditional PBX, cables are run from central junction PBX box to the various locations were the connections are needed. However in the VPBX, the clients communicate with the server by making a request to the server.

The network server caller interface is as shown in Figure 8. This enables the administrator to login to the server. The server enables the clients from both ends to communicate with each other. Once the server begins to run, it generates an IP address which enables the caller and the receiver to communicate with each other using the server IP address.

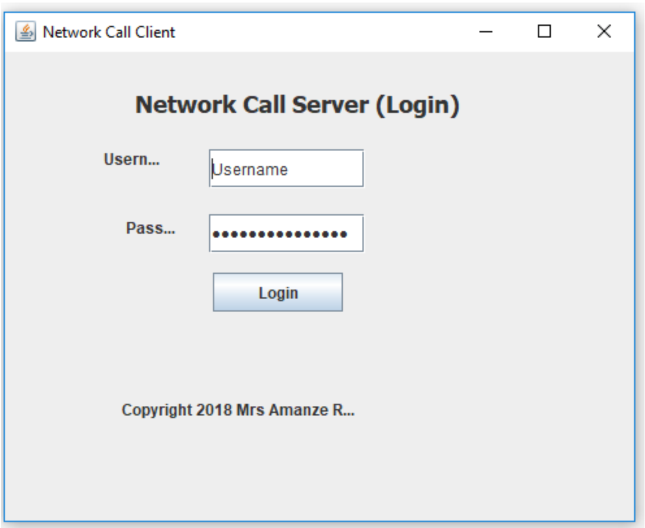

Figure 8 Network Call Server Interface

This interface is only for the administrator. The username and password of the server are not to be exposed to the general Public for security reasons. This is the power house of the system.

Once the request is made, a port in the server is opened which acknowledges the request from the server by sending a message.

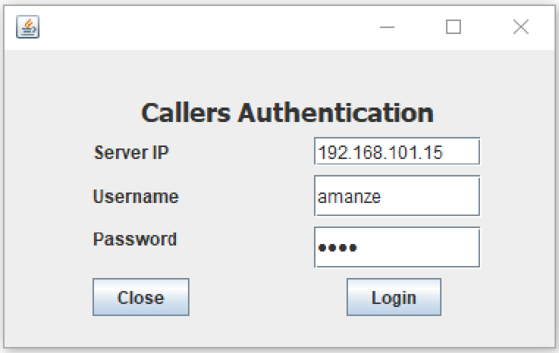

Figure 9 Caller authentication interface

Figure 9 shows the caller authentication interface, where the caller authenticates using server IP address, username and password. The authentication process is to ensure that the right person is authorized to use the VPBX system.

Once the caller logs in to the server, the sever sends a message as displayed in Figure 10

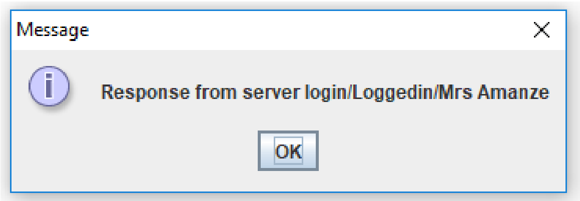

Figure 10 Response from server

This message signifies that the server has responded to the request of the caller. On the other hand if the server is not connected, the message will not pop-up, signifying that there is no connection at the server end.

To proceed in making the call, the caller (client), clicks ok, once this is done, the caller will receive a new interface as shown in Figure 11

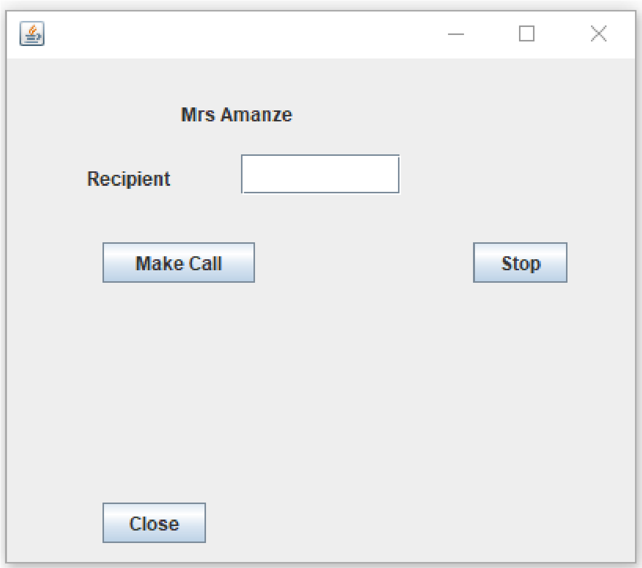

Figure 11 Call making interface

This interface enables the caller to type the number of the recipient, followed by clicking on the make call box. Once this is done (as long as the recipient has also logged in to the server), the caller hears the ringing tone signifying that the call is reaching the recipient end.
On the side of the recipient, the individual receives a customized ringing tone. Once the recipient accepts the call, by clicking the “accept” button, the two individuals, (the caller and the called (recipient)) are ready to communicate.

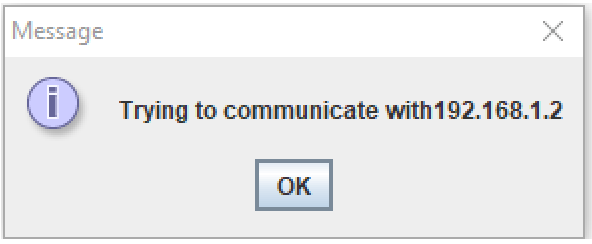

Figure 12 Connection interface

This message show that the caller is trying to communicate with the server. This message signifies that the server is up and ready for communication to take place.
The message in Figure 13 signifies that the recipient is expecting a message from a caller.

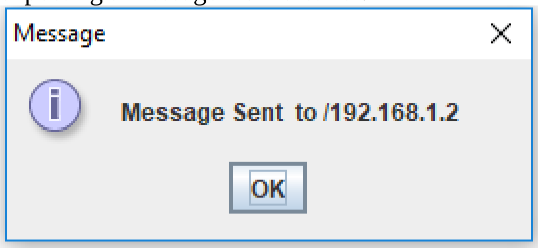

Figure 13 Message sent to the recipient

On the part of the person receiving the call, the message in Figure 14 displays the name of the person the call is coming from. It is now the responsibility of the recipient to accept or reject the call.

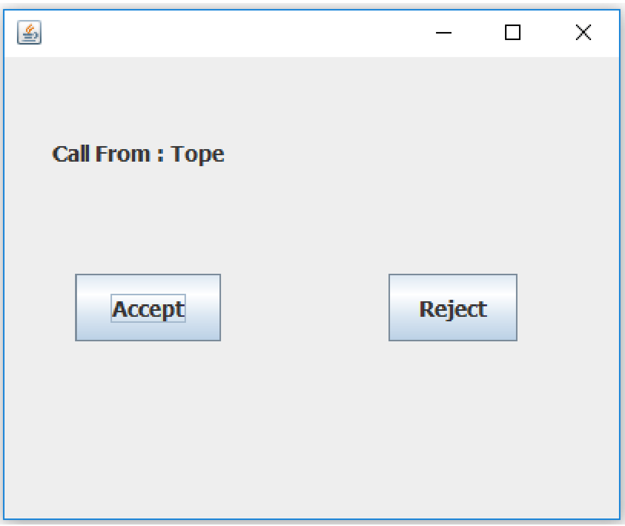

Figure 14 Accept and reject interface.

This interface signifies where the call is coming from. The name of the person calling is displayed. It is now left to the recipient to accept or reject the call.

The message in Figure 15 indicates that the recipient has accepted the call and that there is connection from both caller and recipient.

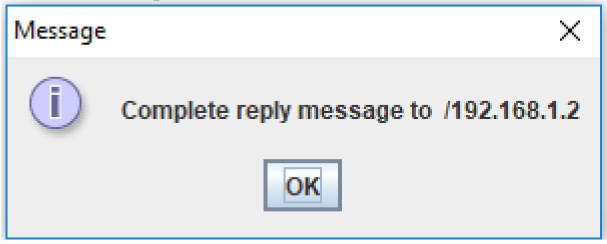

Figure 15 Complete reply Interface

Other OTTs include data, television, video, radio and audio services which make it difficult for voice calls to have clear quality due to competition of packets along the line, trying to meet each of these services. Voice calls, which require real time, suffer from packet loss which cannot be re-sent in real time unlike data, television, radio and video, where packets can be retransmitted and still have a good quality output. The VPBX model is developed for only voice streaming. This gives it the advantage of not competing with services on the network, hence having a clear voice quality. To the best of our knowledge, there is no other OTT service dealing with only voice. The VPBX model adapts to instant updates on the network as-well-as adapt to IP changes as the clients moves from one location to another during voice calls. No running of cable from one building to another is required and calls are made with clarity in voice call, giving it the needed quality.

V Conclusion and Recommendation

The developed VPBX app model is designed to mimic the traditional analogue Private Branch Exchange (PBX) and other OTT services. The study is a management scheme for the improvement of call quality proposed and method of saving cost of calls made by workers within and outside the organization or institution. The VPBX software call app saves cost of equipment’s and materials by transmitting voice over the existing internet network. This research work is a pilot study conducted at Babcock University where from where it can be extended to other institutions and industries. The proposed model is recommended for voice quality calls and will serve as an avenue to improve the institutions intranet and internet communication system.

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