The International Journal of Multi-Disciplinary Research
ISSN: 3471-7102

Implementation of Open-Source and Secure VoIP systems with integrated Softphone Software on Personal Computers and Mobile Devices

(Conference ID: CFP/458/2017)

Chisanga Kapinda,
Student-Information and Communications University.
SMART Zambia Institute,
P.O. Box 31998,
chisangakapinda@gmail.com
Lusaka, Zambia.

Abstract

As Voice over IP communication is taking over from traditional communication systems worldwide, one of the main related challenges is the dependency on expensive commercial VoIP systems that are restrictive and fixed to the manufacturers design. This calls for more research in cheaper and more flexible alternatives such as open-source VoIP systems. This paper seeks to report on the project whose aim was to prove that implementation of open-source VoIP systems with integrated Softphone Software on Personal Computers and Mobile devices would help reduce on cost of hardware VoIP systems and at the same time increase on features and services through virtualized capabilities. The project's main objective was to prove that reliability, quality and security would be maintained by open-source and virtual VoIP systems, similar to commercial physical VoIP systems. The project also aimed at providing a solution for an integration problem for already existing physical IP phones. To address these problems, a 3CX server was installed on a Linux-Debian Virtual Machine, hosted on a Physical Server at the Main Data Center at the Ministry of Finance, Zambia. The Softphone Software were also installed on employees' Personal Computers and Mobile Devices. These employees under Smart Zambia Institute (SZI) were given time to test the quality, reliability and security of the system and were provided with questionnaires in order for them to give feedback on the comparison of open-source VoIP against commercial VoIP systems. The installed VoIP provided more features and flexibility and it was concluded that there was need for more research and tests of open-source VoIP systems to address the problems of over dependency on expensive, prohibitive VoIP systems.

Keywords: *Secure, Open-source, Voice over IP, PBX, Call Manager, IP Phone, Softphone*.

1.0 Introduction

The birth of e-governance and ICT in Zambia has come with many challenges. These challenges have delayed the realization of successful e-governance systems. One of the main challenges is absence of implementation of available and cheaper information and communication technologies (ICTs). These technologies include use of open-source solutions in government institutions. However, despite the many challenges, there has been very little research on implementation of such solutions in order to address the problem. This paper seeks to report on the project whose objective was to provide a cheaper alternative of commercial, expensive and inflexible VoIP systems in government institutions, through implementation of open-source Private Branch Exchange (PBX) with open-source Softphone Clients on existing Servers, Personal Computers and Mobile Devices. The project was conducted at the Smart Zambia Institute formerly known as the Centre of Excellency for e-Government and ICT (CEEGICT). The project aimed to prove that open-source solutions were equally as reliable and secure as commercial products with the benefit of reducing on cost hardware systems and duplication of similar VoIP technologies.

In order to archive the objective of this project, software PBX and software phones were installed on one (1) shared Virtual Machine hosted by a physical server. On the user end, softphones were installed on Personal Computers and Mobile Devices of employees. The employees were required to use the VoIP system similar to other systems in order to make comparisons. The main issues of comparison were quality of services, reliability, cost, features, integration and security.

Firstly, a brief background review of similar studies and other related literature was collected through internet sources, to collect more facts about open-source software with regards to PBXs and softphones. In the review, both negatives and positive were cited in order to give the paper a less biased approach. The problem statement to justify the importance of this project was also given followed by the objectives. The resulting data was analyzed, presented and discussed in the results and discussion sections. Lastly, a conclusion was made on the upsides and downsides of open-source PBXs and softphones against commercial PBXs and physical IP Phones.

Open-source software (OSS) is computer software with its source code made available with a license in which the copyright holder provides the rights to study, change, and distribute the software to anyone and for any purpose. Open-source software may be developed in a collaborative public manner. According to scientists who studied it, open-source software is a prominent example of open collaboration (OpenSource.Com, 2017).

In the recent years, almost all information systems are shifting to virtualized systems and therefore reducing the demand for expensive hardware. This however, doesn’t solve the cost on licenses and software upgrades hence the solution provided by open-source software (OSS) technologies. Further, with the cost of hardware VoIP systems, commercial licenses and costly upgrades, open-source and virtual systems have proven as the best way forward to address this issue.

1.1 Background

Government through the Ministry of Transport and Communications (MTC) had in the past been implementing Cisco IP phone systems (Ministry of Transport and Communication, Zambia, 2017). In 2015, MTC through the Smart Zambia Institute, engaged Microsoft to implement Microsoft Skype for Business Services that came with HP IP phones and similar services on the older Cisco IP phones (SZI, 2016). Recently, in 2016 MTC and SZI had engaged Huawei Technologies for similar VoIP systems in various government Ministries (SZI, 2016). These three (3) systems had their own call managers that did not support communication with each other to enable communication of their different IP phone brands. This led to duplication of similar VoIP systems as a result of this failed integration.

Despite there not being any recorded research from the local perspective on the similar subject, many researchers globally, have argued on whether open-source PBXs would replace commercial PBXs. However, there are different factors that provide both successes and failures to this shift. According to (Daniel, 2016), during a presentation on his migration project at the VON show in Boston in 2016 "Some organizations consider taking the plunge off of big iron PBX platforms into IP telephony as being pretty daring, but that was nothing compared to what Sam Houston State University had done. He added that "the south Texas school was boldly moving thousands of users off a Cisco VoIP platform to an open-source VoIP network based on Asterisk because it was more cost effective in the long run to go with an open source solution due to the massive amounts of licensing fees required to keep the Cisco Call Manager network up and running". He also added that "in the Cisco model, each phone attached to the Call Manager required a separate licensing fee to operate. His presentation justified that a Cisco model, where existing Cisco phones were kept, were instead attached to Asterisk servers on the back end and the phone licensing costs were eliminated".

Daniel further argued that more security and control over the IP PBX software and servers was another reason for the shift. This was because Commercial PBXs were more susceptible to hacks, since only approved server updates and patches could be installed on them. Meanwhile, on open-source PBXs, any identified exploit could be fixed in the source code.

(Gilbert, 2010) argued that open Source is a great concept for those businesses that had the expertise and resources required for a successful implementation. However, businesses that were looking to get into a VoIP solution and see open-source as a way to save money were going to be very disappointed. He explained that the notion that anyone could, with minimal effort, deploy a secure, reliable Open Source PBX, was vain. This was because dealing with a commercial and hosted VoIP service provider implied spending less, in terms of both time and money, by focusing on core business and letting a reputable provider take care of delivering VoIP service.

(Avad Technologies, 2010), also states that open source deployment also led to a hosted VoIP solution. As this was noted in the "Open source PBX does not mean it's free" blog on the cited Avad Technologies website, which said “though open source software tools were free, the complexity of coding and scripting Asterisk and other open source communication platforms into a viable, customized communication platform can be costly”.

As an addition to the solution of open-source software (OSS) against expensive licensed software, virtual systems such as software PBXs and software phones, whether free or licensed are still far cheaper than physical PBXs and IP phones without compromising on quality, security and reliability. The main benefit that comes with open-source is that it gives an organization full control and flexibility of their VoIP system at a lesser cost compared to commercial products (Smith, 2017).

By design, open source software licenses promote collaboration and sharing because they permit other people to make modifications to source code and incorporate those changes into their own projects. They encourage computer programmers to access, view, and modify open source software whenever they like, as long as they let others do the same when they share their work (Gentek, 2016). Gentek adds that part of the allure of open-source software is its affordability. The software itself costs nothing to run, besides the associated hardware costs. Without licensing fees, resellers can create their own pricing structures for their clients.

On the other hand, Softphones simulate physical phones and have proven to maintain the same quality, reliability and security similar to hardware IP phones with the benefit of providing more features through free upgrades and flexibility compared to hardware IP phones.

Free and open source software (FOSS) communication engines, such as Digium's Asterisk may sound like the perfect fit for budget-strained organizations looking to transition from traditional TDM-based telephony to IP telephony. Promoted as the "telephony glue" that ties VoIP to TDM, Asterisk can be especially beguiling to companies looking to gradually migrate to VoIP by combining legacy PBXs with IP PBX servers (Digium, 2017).

1.2 Problem Statement

VoIP is one of the most expensive technologies ever since there has been an increased demand for IP phones to replace ordinary telephone systems both in the private and public sectors worldwide. Similarly, in government institutions, the migration to Desk IP phones has increased in the past decade (Werbach, 2005).

Most commercial VoIP PBXs are prohibitive towards integration of their systems with other vendors. These include, Cisco, HP and Huawei as the top three (3) products in the target government institution. Due to the cost of these systems, a larger number of employees have no access to IP phones and have continued to rely on none or ordinary telephone systems.

In addition to the cost of their hardware, licenses and upgrades, these commercial products also attract an over dependency of government institutions on outsourced expertise and consultation and maintenance. This challenge has led to the duplication of similar technologies such as commercial VoIP call managers that cannot communicate with each other. Another issue is security concerns in call managers that come with inflexibilities and do not allow for modifications to suit our specific organizational needs. This therefore, calls for Zambia and other developing countries to invest more in ICT research and initiatives to address these problems.

1.3 Objectives

This project aimed to prove that implementation of simple and free software such as open-source PBXs on available hardware systems in government institutions would reduce hardware duplication costs of both physical IP phones and physical PBXs.

In addition, the project would prove that open source VoIP systems would address security and reliability concerns using security protocols and encryption. This would be provided by flexible open source and license free upgrades.

The project also aimed at reducing on the demand for physical IP phones through integration of softphones that provide more services such as Voice, Video, Instant Messaging and Conferencing that may not be available on prohibitive and physical IP hard phones.

This project's overall objective was to justify the replacing of Desk IP phones with Softphones on PCs and Smart phones resulting in increased access to majority government employees with access to PCs, Tablets and Smart Phones.

The project also aimed to provide a foundation for further research into open-source alternatives during the implementation of e-governance systems and service in the SZI.

2.0 Methodology and Design

The research was mainly an experimental research. In the project, the main population of focus was the Smart Zambia Institute (SZI) formerly known as the Centre of Excellence for E-government and ICT, E-government Division. From this population, a sample of fifty (50) employees was targeted as participants from a total of eighty (80) available employees. As an employee of the SZI, the Author selected the participants from various departments on basis of availability. These participants were approached and agreed to participate in the project. Participants were required after using the Softphones, to give independent feedback on the system through interviews and simple scale questionnaires based on the factors of comparison. Main factors were the reliability, quality and security of the systems compared with Desk IP Phones. Based on the feedback from the participants, a conclusion was made on whether if softphones would replace desk IP phones.

ISSN: 3471-7102

The participants were given a scale of 1-10 on factors of comparison of the VoIP system with other existing VoIP systems. The factors of comparison included: - Quality of Voice and Video services, Reliability of communication, cost of implementation, Features of the VoIP system and Integration capabilities with other end devices. These results were collected from the 50 users and the averages were recorded in the results section of this paper.

With authority from the National Coordinator of the E-Government Division, a 3CX/Asterisk server was installed on a Linux-Debian Virtual Machine, in the Government Wide Area Network. This was done on a Physical Servers at the Main Data Center at the Ministry of Finance.

The SIP client Software (Virtual phones) were installed on Personal Computers that were connected to GWAN via Wired and Wireless Local Area Network at E-government Division. Mobile SIP client software were also installed and configured on the Participants' mobile devices. In addition, a few models of desk IP phones were linked and provisioned to the PBX server. The three (3) category of end devices were linked and tested for successful integration with softphones.

Tools required included: Linux based Server Operating system, Virtual Machine on a Physical Server, 3CX/Asterisk PBX server, X-lite/Zoiper SIP client, three (3) Cisco/Huawei/HP IP phones, and Connection link from the Server to the clients through the GWAN.

2.1 Installing 3CX on Debian Server

In this chapter, steps of installing Debian Server on a virtual machine are presented. The Debian Server was installed on a physical server machine in the Data Centre. Other tools such as Secure Shell and http server were installed to enable the Administrator to access the server from a remote location. A static IP address was also allocated to the Server. Debian was chosen as the host Operating systems because Debian is an operating system based on the GNU/Linux build. Also like most other Linux distributions, its free and open source. It is a popular operating system for both desktop and server use, spawning several notable offshoots such as the Ubuntu operating system.

Below is a screen shot of the initial installation of debian with 3CX. **Figure 1:0** below shows the initial screen during the installation of Debian-Linux host server for the 3CX PBX server.

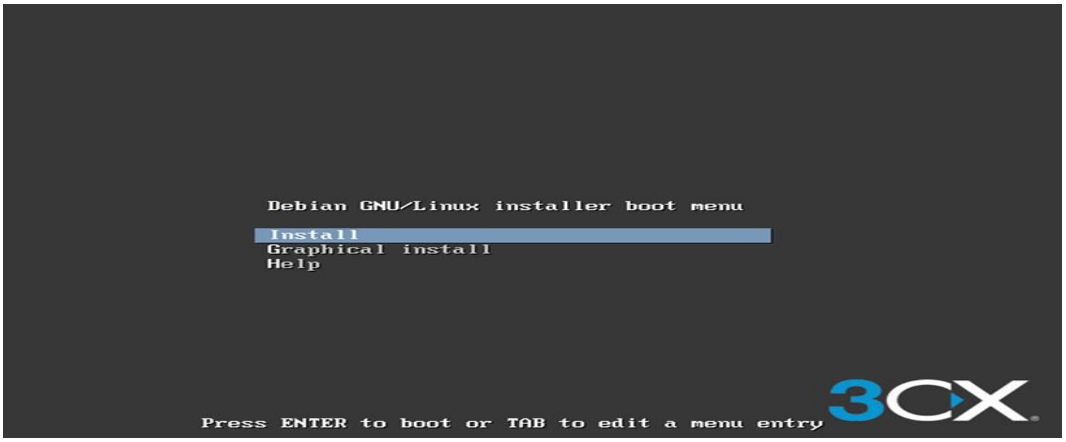


Figure 1:0 Initial Installation of Debian Server

During the installation of Debian host server, a hostname of the server was preconfigured to enable identification of the host server in the network. **Figure 1:1** below shows the hostname set to default 'debian'. However, it can be set to any name.

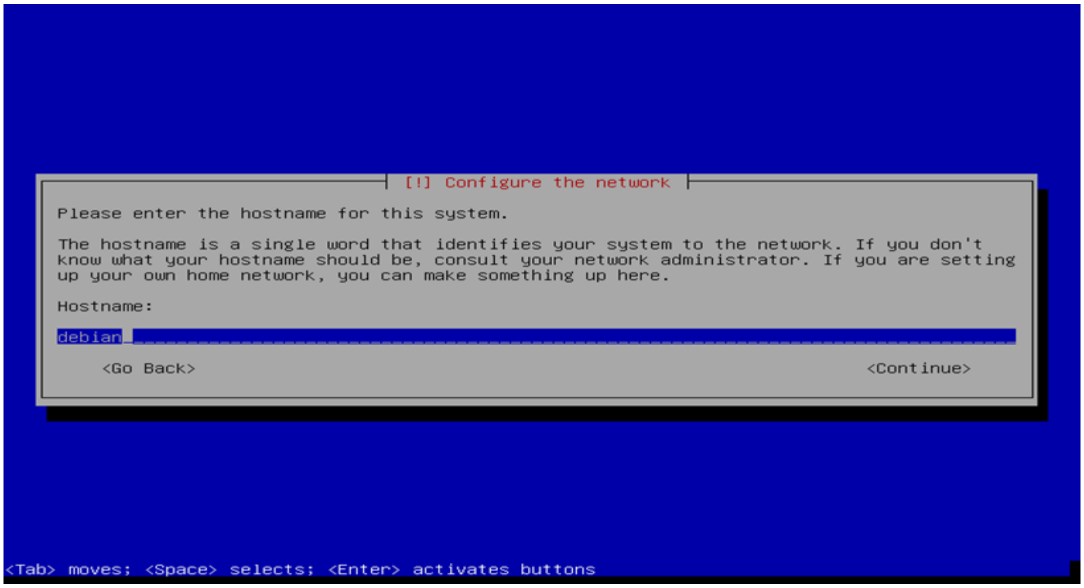


Figure 1:1 Network configuration of the host Debian

The network of the host server was set to static configuration. This is because all Session Initiation (SIP) Clients require to be configured with the server IP address in order for them to communicate with the server. This however, could be left or configured later through the network files of the server. From the screen shot in **Figure 1:2** below, the network was configured manually with the host server IP address.

ISSN: 3471-7102

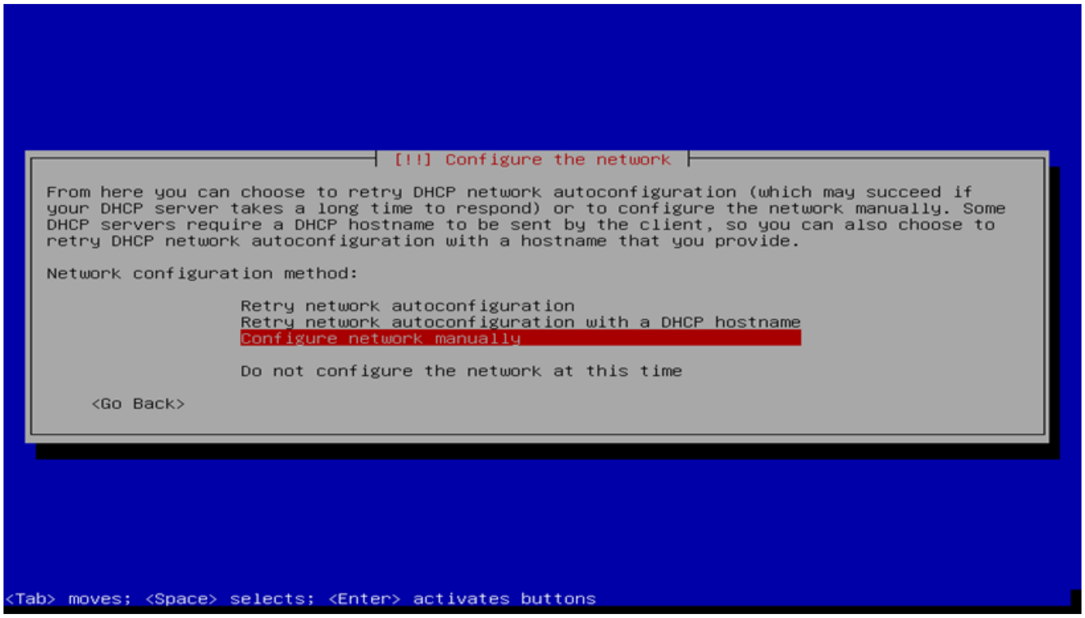


Figure 1:2 Set network configuration to manual

The Figure 1:3 below shows the step of partitioning the host server storage. Since the partition was previously done when allocating the virtual machine storage, the installation was done on the entire allocated space as shown below. This was justified with the fact that the virtual machine could only use its allocated storage without affecting other VMs that were hosted on the main server.

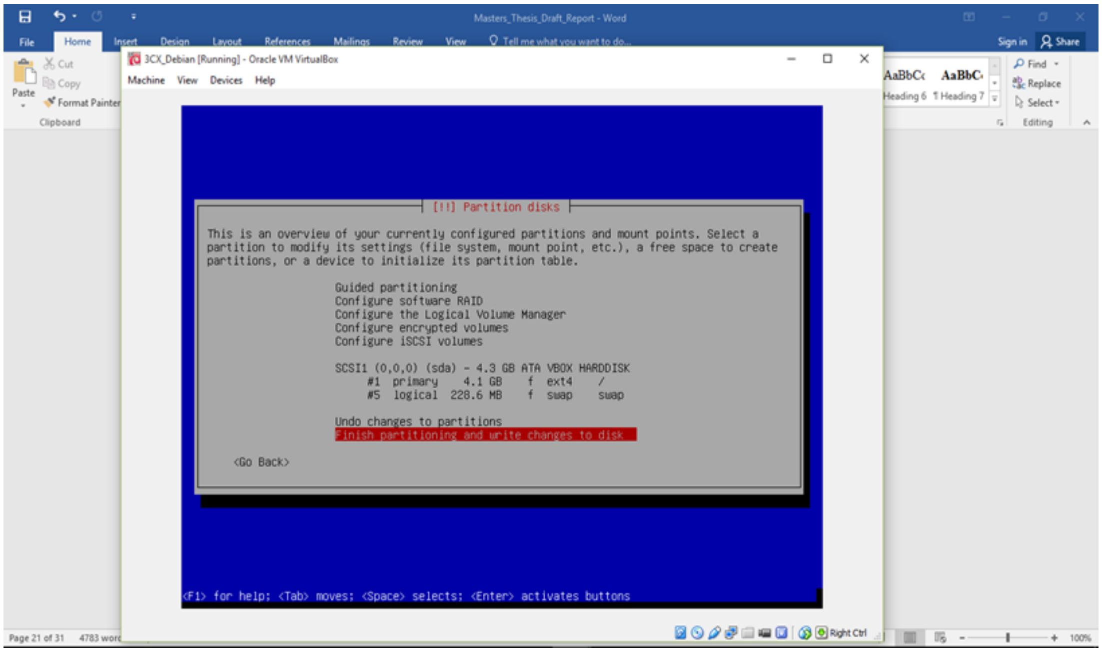


Figure 1:3 Setup disk partitions for the virtual machine

Figure 1:4. In the Debian Host server, the next step was to download the required version of 3cxPbx with the highlighted command in the screen shot below. This allows the host operating system to download the needed files for the PBX.

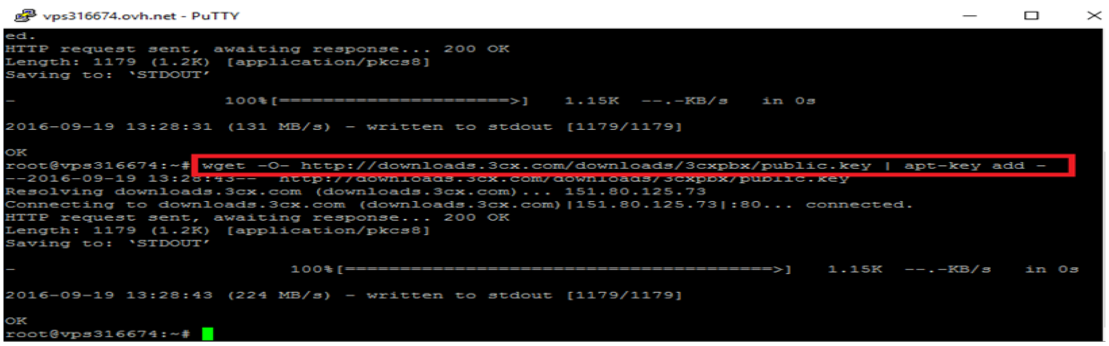


Figure 1:4 Download 3CX files

Figure 1:5. Once all the necessary 3cx files were downloaded and installed. 3CX required initial configuration. The two available options are 1) Using a Web browser (GUI) and 2) Command Line. This initial setup involves the setting up of Administrator login credentials, security settings and extension number parameters for the PBX server.

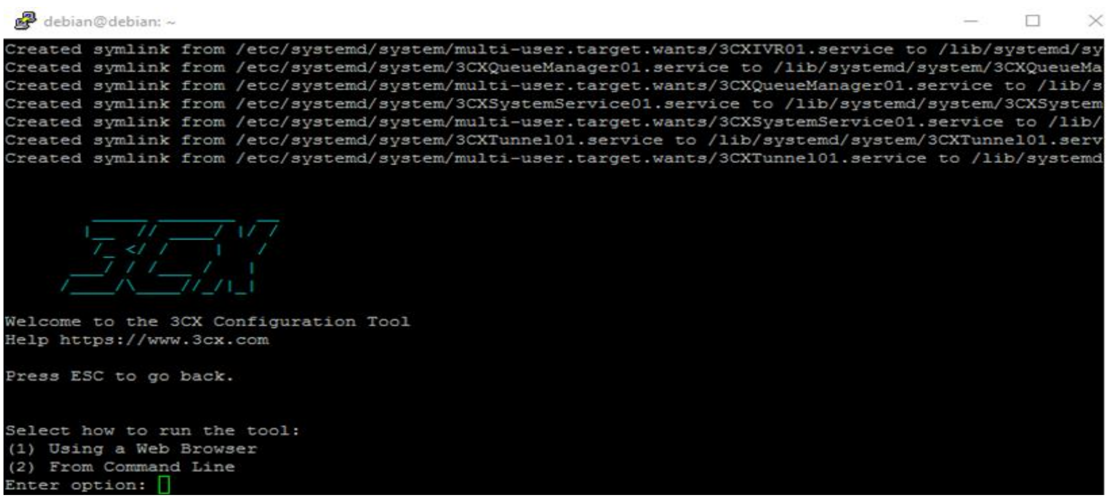
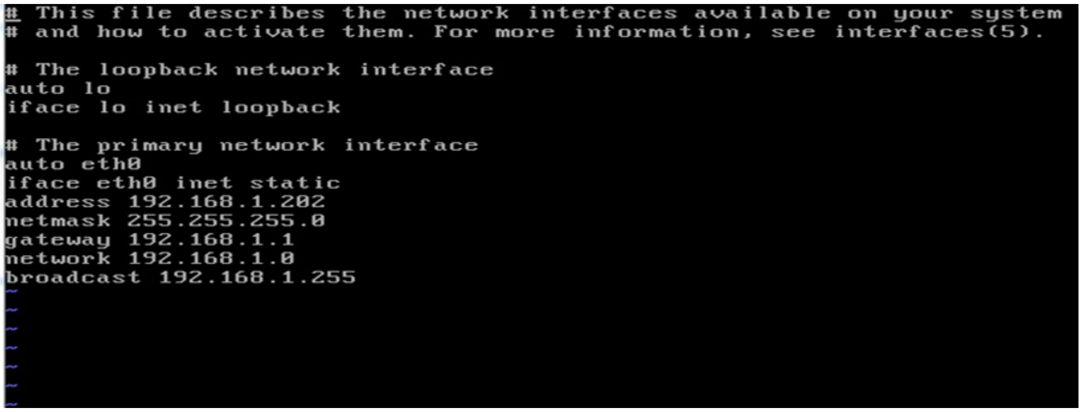


Figure 1:5 Initial Configuration Options

The remote server, using the 'vi /etc/network/interfaces command'. The network configuration file was edited to give the remote server a static IP address as shown in **Figure 1:6** below.

ISSN: 3471-7102



```
# This file describes the network interfaces available on your system
# and how to activate them. For more information, see interfaces(5).

# The loopback network interface
auto lo
iface lo inet loopback

# The primary network interface
auto eth0
iface eth0 inet static
address 192.168.1.202
netmask 255.255.255.0
gateway 192.168.1.1
network 192.168.1.0
broadcast 192.168.1.255
~
~
~
~
~
~
~
```

Figure 1:6 Network configuration file

In the remote host Server via a terminal, an update was required to get the latest updates and patches for both the host server and the 3CX server. This was done with the apt-get update and upgrade commands as shown in **Figure 1:7**. below.

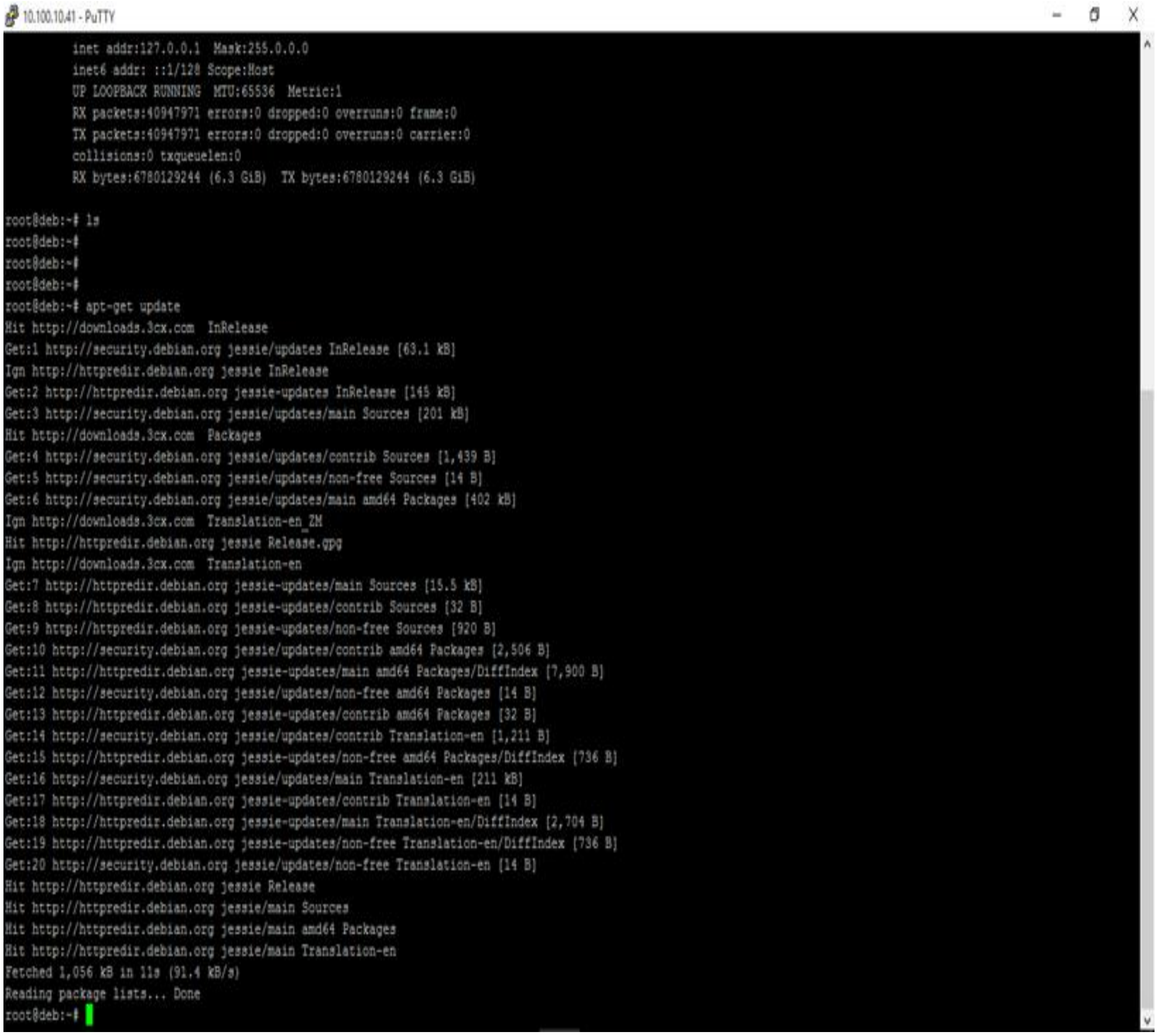


Figure 1:7 In the remote host Server via a terminal

The update and upgrade concludes the successful installation of the servers. The username is set by default as '**root**' and the hostname in the project was set to '**deb**' as shown above. Remote access to the remote server was administratively protected by username and password login via **ssh** or **https**.

3.0 Results

This section provides a report on the results of the VoIP system as well as the Feedback from the users of the system. Following the successful installation of the VoIP backend system and in order to access the web server via **http** for internal or **https** for external administrators. The administrator, with previously configured credentials was able to browse to the 3CX server from a remote machine web browser as shown in Figure 2:1 below.

The International Journal of Multi-Disciplinary Research
ISSN: 3471-7102

3.1 3CX Graphical User Interface

Figure 2:0. Once the host server and 3cx server have been installed and configured successfully. Through a remote machine's web browser, the GUI of the 3cx can be accessed through the credentials that were set in the initial setup. The figure shows the dashboard view of the 3cx server.

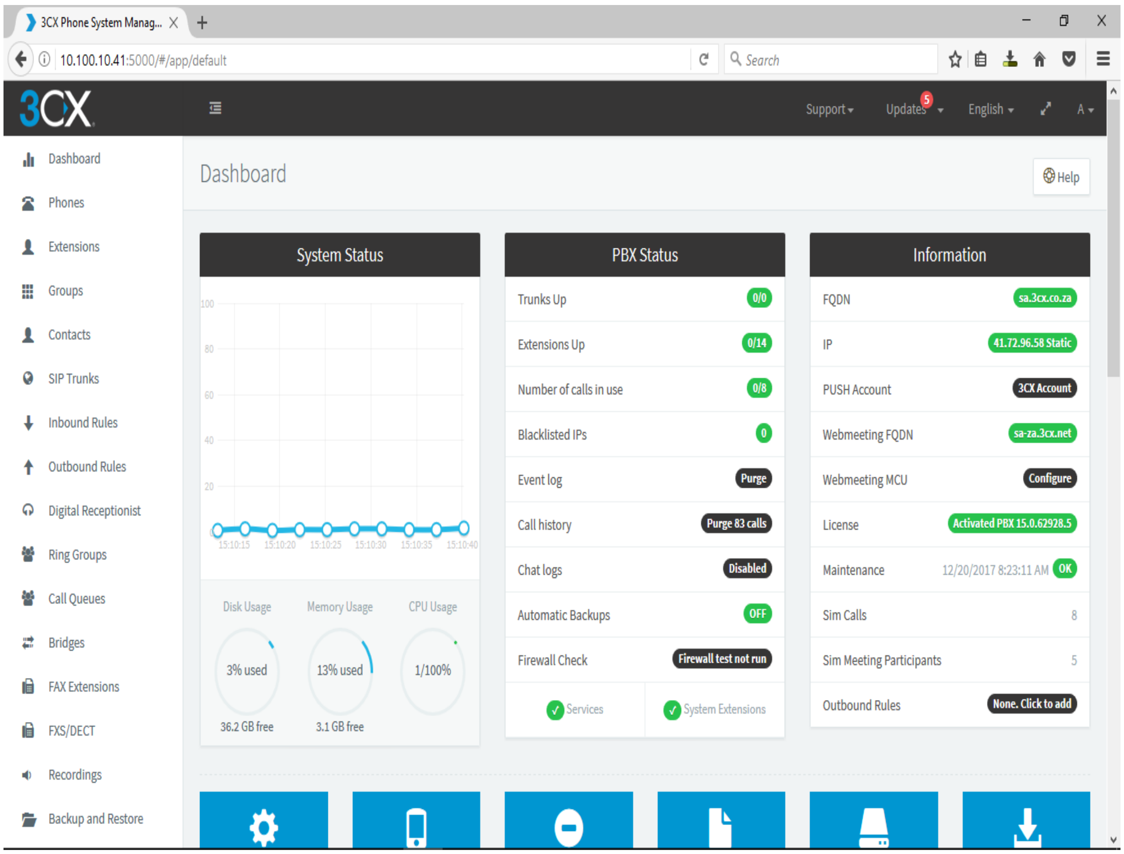


Figure 2:0 Web GUI

From the web-based GUI, the server provides a variety of options that can be accessed in the settings page as shown in **Figure 2:1** below.

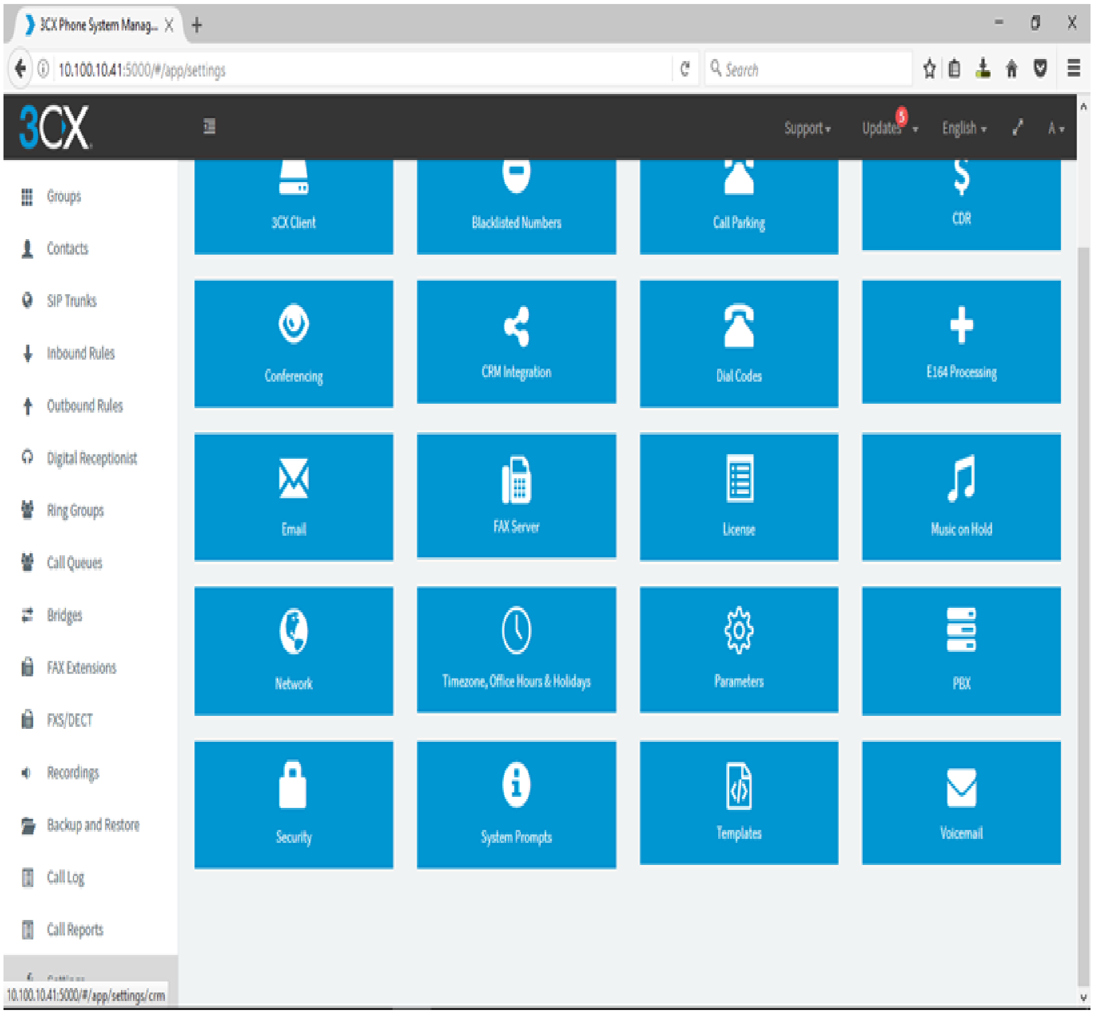


Figure 2:1 Settings page

3.2 3CX Command Line Interface

The command Line Interface provides the administrator with more access to server files than the GUI preconfigured options. This is where the 3cx server files can also be accessed and edited instead of using the web-based GUI as shown in **Figure 3:0** below

ISSN: 3471-7102

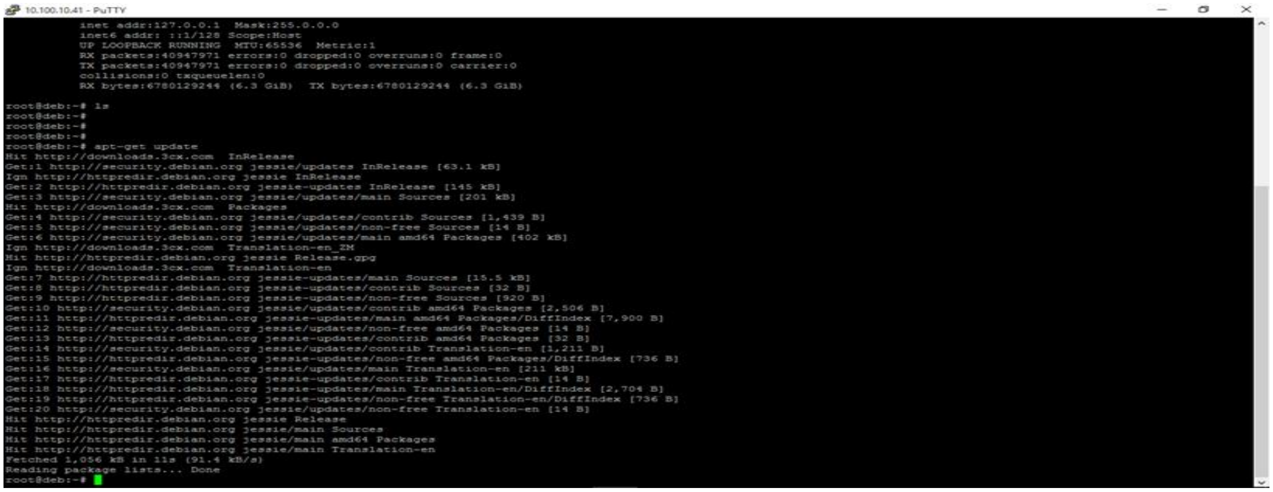


Figure 3:0 The command Line Interface

Figure 3:1 Shows the Debian host server being accessed through a remote windows machine to verify the network configurations. The remote server can be accessed through a LAN, WAN or internet connection depending on the location of the administrator.

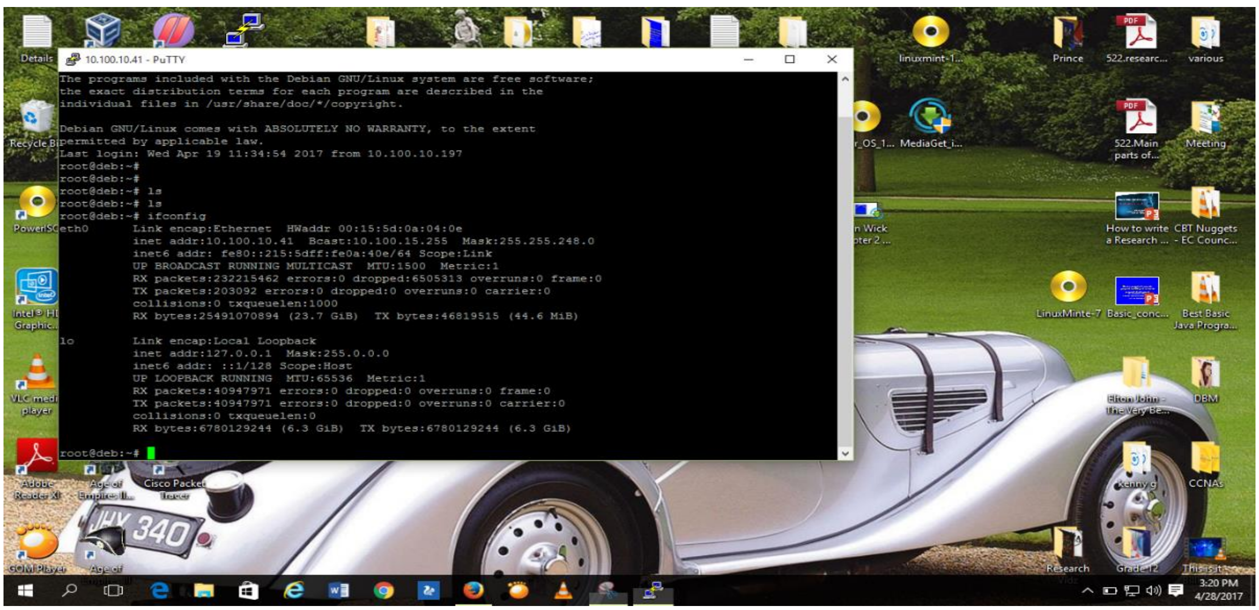


Figure 3:1 Debian remote host server

3.3 Softphone Extensions

ISSN: 3471-7102

One of the main objectives of this project was to ensure that the PBX server was accessible by softphone clients as well as physical IP phone clients. This chapter shows how softphone clients are added in the server through editing extension files or via the web-based GUI. The PBX can be accessed both through the local area networks or via internet depending on the private and public IP addresses that are mapped to the host VM server. This was because the server is allocated a private and public IP address.

Figure 4:0 shows a list of extension numbers for users that had been added to the PBX server. The extension numbers operate as caller IDs for specific softphone clients. The same details were required to be entered into the clients Personal Computers and Mobile devices in order for them to access the server. Four (4) main categories of information include, unique ID (extension number), username (login id) and user password and server IP address.

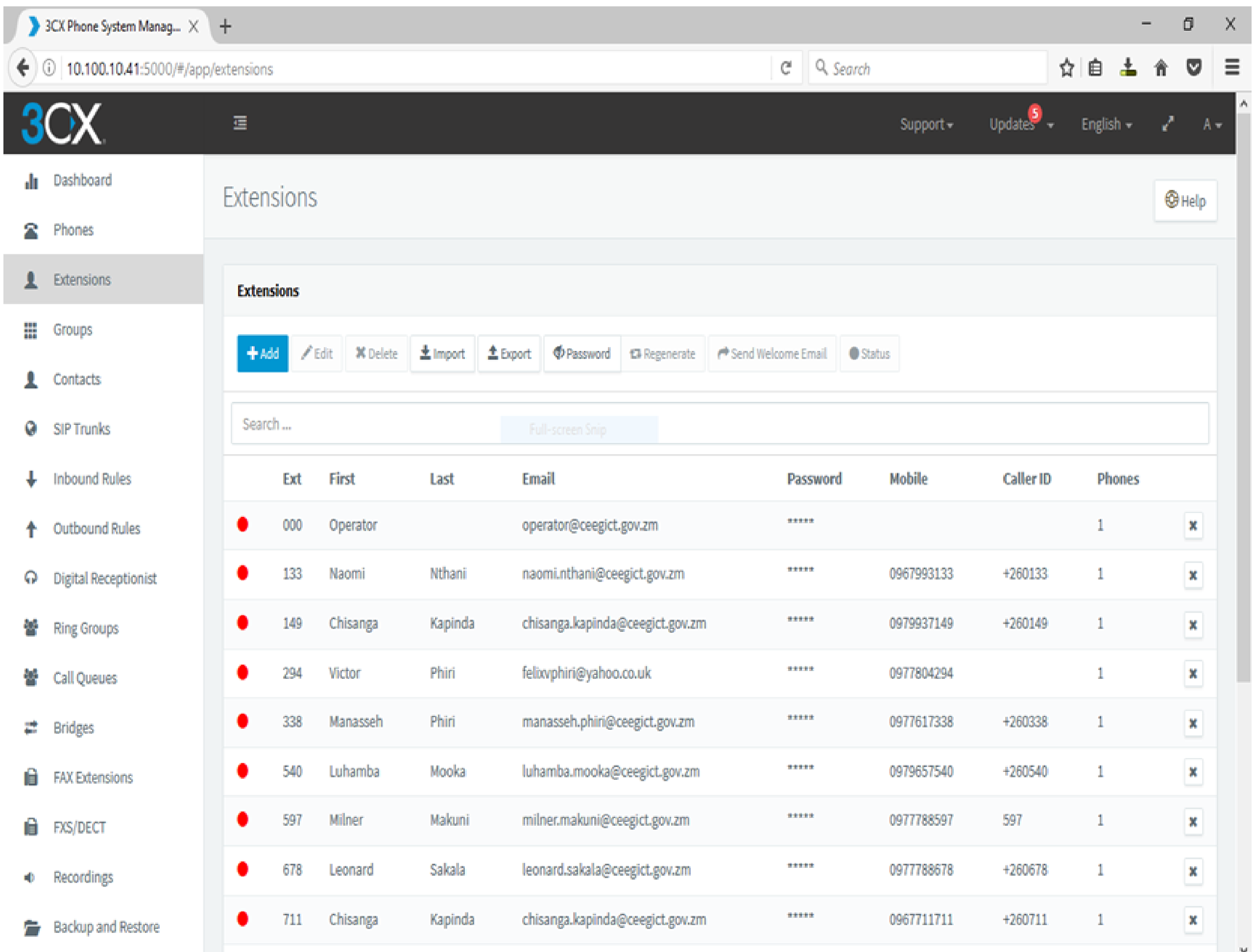


Figure 4:0 A list of extension numbers that are registered in the server.

ISSN: 3471-7102

3.4 Softphone Configuration

On the user end, softphone clients that were installed on Personal Computers and Mobile Devices required to be configured with matching information as on the server in order for them to be registered and provisioned. The Figures below show configuration steps for a Zoiper softphone on an Android mobile phone.

Figure 4:1 Step one, configuration of a SIP account.

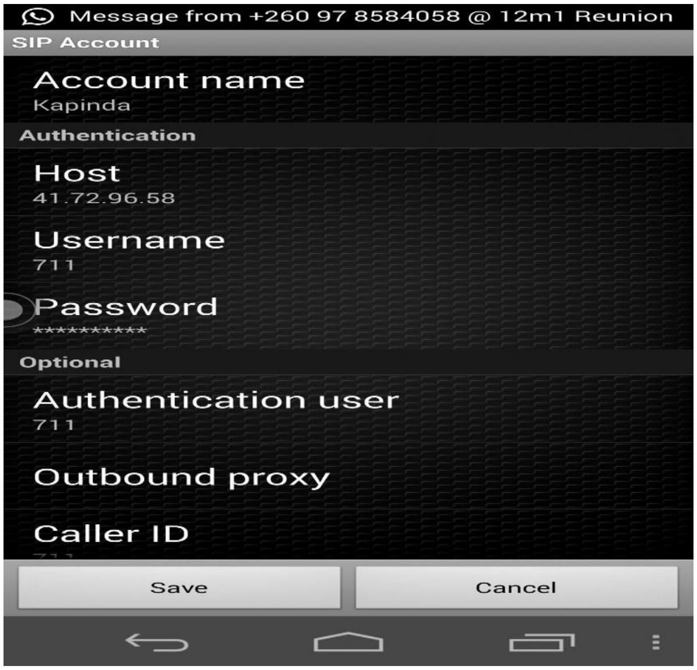


Figure 4:2 Step two, adding server details

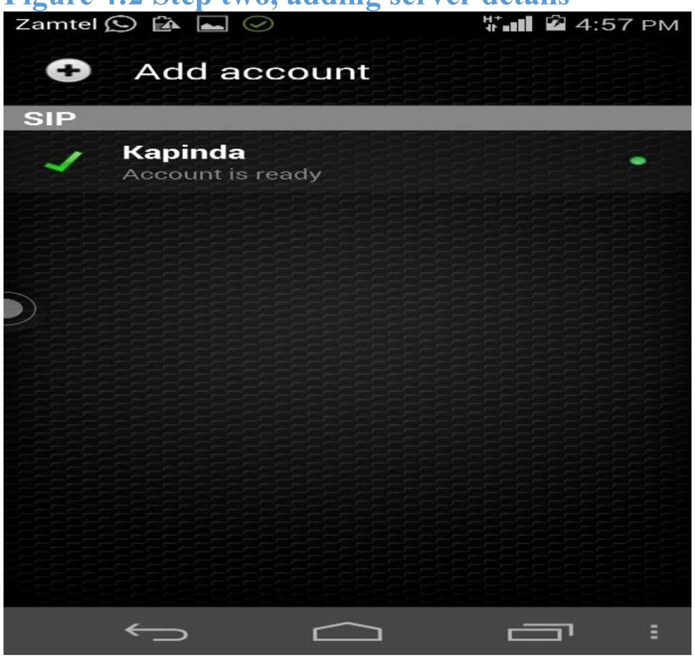


Figure 4:3 Step three, SIP account registration with remote server

ISSN: 3471-7102

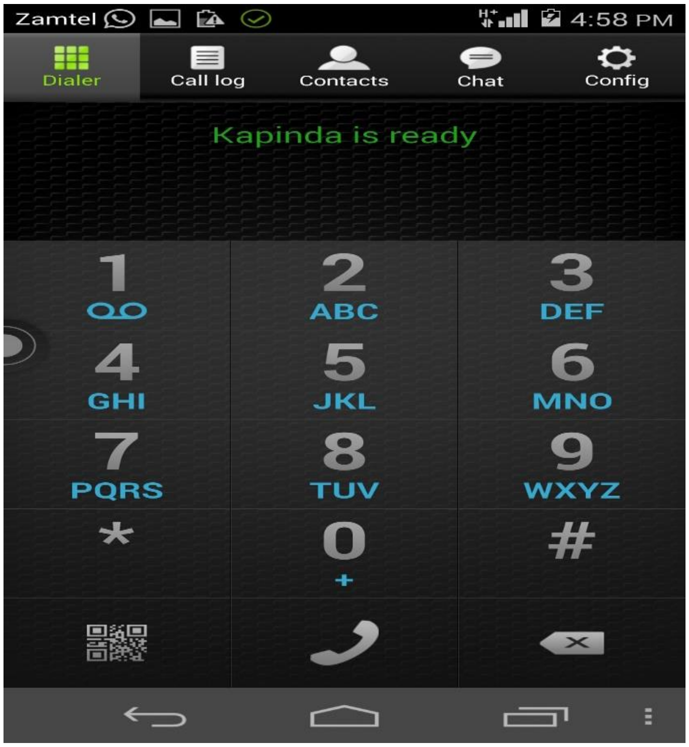


Figure 4:4 Step four, Softphone ready to communicate

3.5 PC Softphone

Both PC and Mobile softphone require the same procedures of configuration in order to register with the 3CX server. **Figure 5:0** shows an X-lite softphone on a windows desktop PC

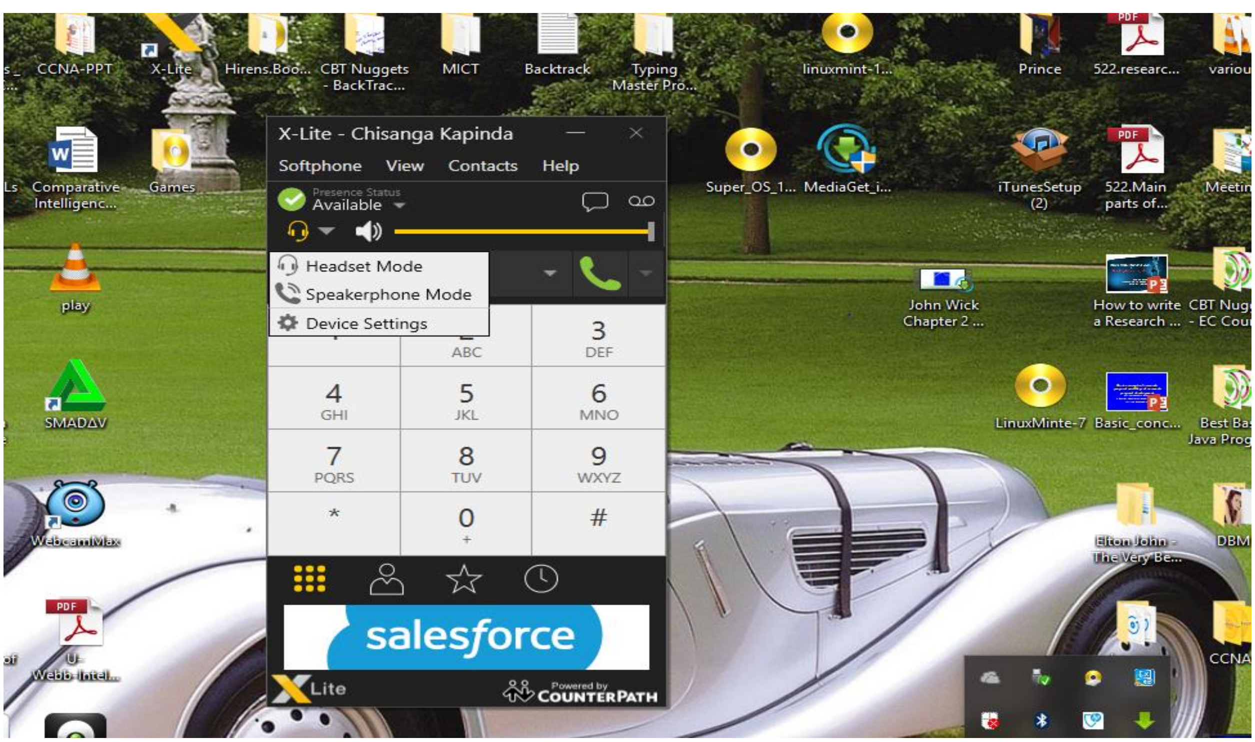


Figure 5:0 PC softphone

3.6 IP Phone Provision and Integration

Another objective of the project was to ensure that desk IP phones such as Cisco were successfully provisioned and integrated with softphones. This was to allow already exiting IP phones to be able to communicate with other models and softphone clients on Personal Computers and Mobile devices. **Figure 6:0** below shows a page where physical IP phone models could be added into the 3cx server. The 3CX server has a provision for addition of over 100 models of IP phones that can be integrated.

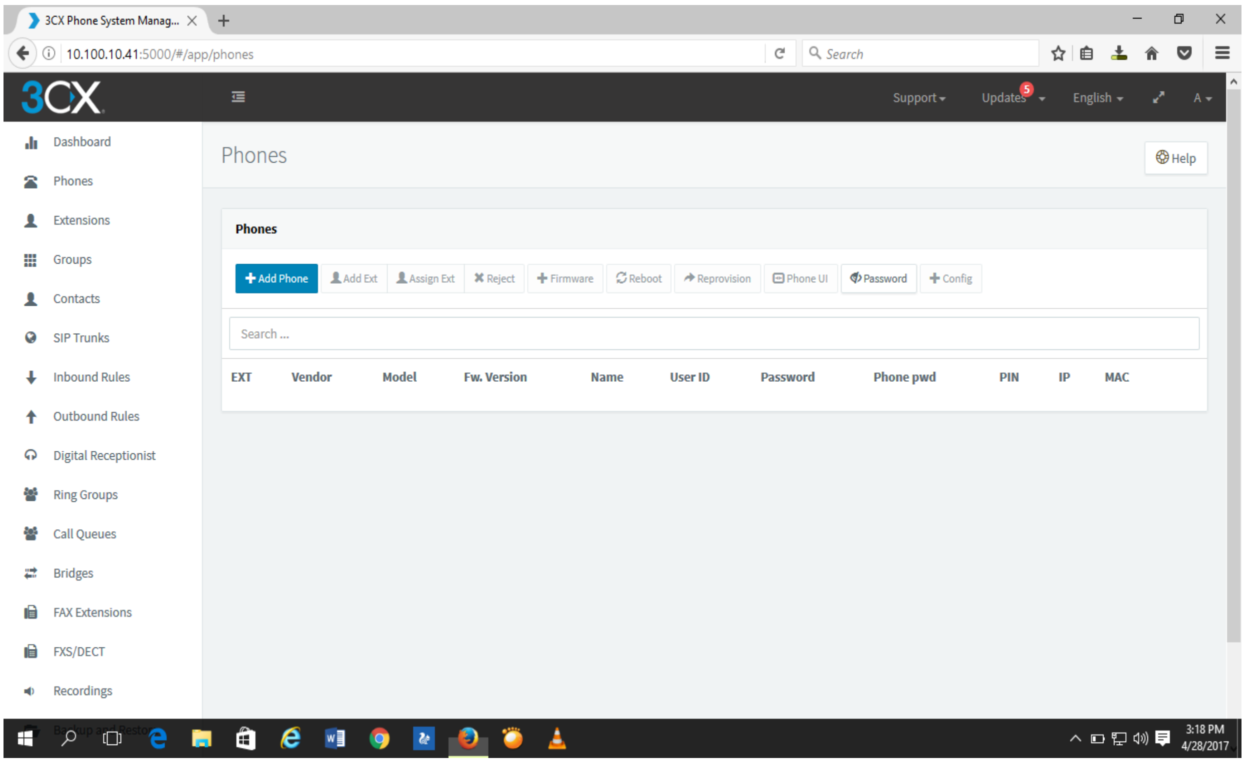


Figure 6:0 shows a page where physical IP phone models could be added into the 3cx server.

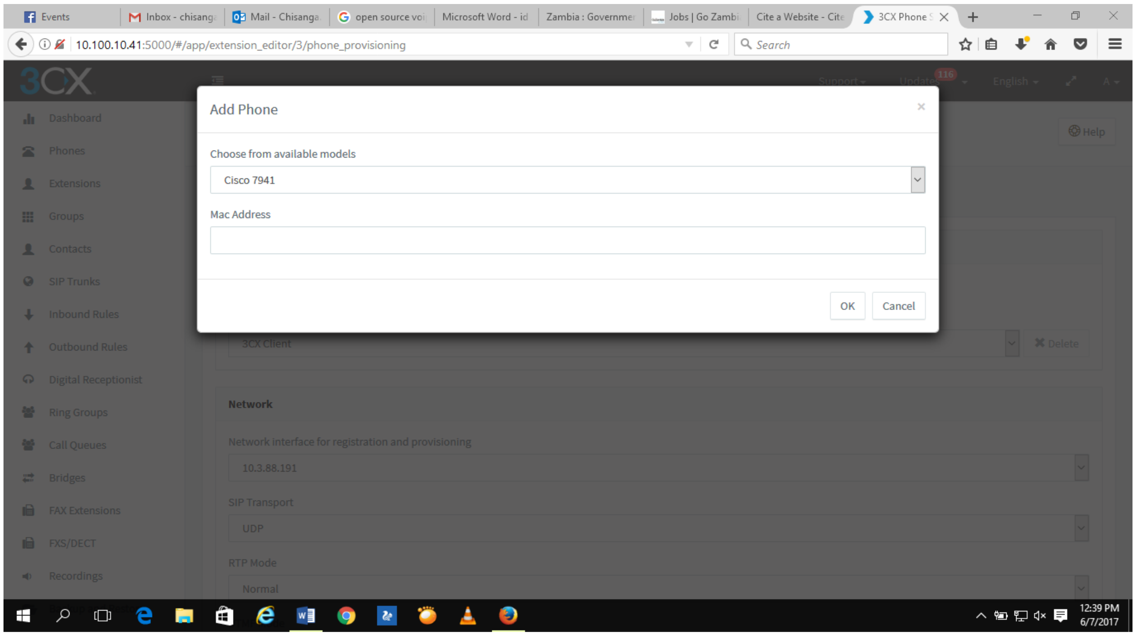


Figure 6:0 shows a model of Cisco IP phone added into the 3cx server.

3.7 Successful Call between PC and Mobile Softphone

Figure 7:0 Below shows a successful call between the X-lite softphone on the PC and the Zoiper Softphone on the Android Mobile Phone.

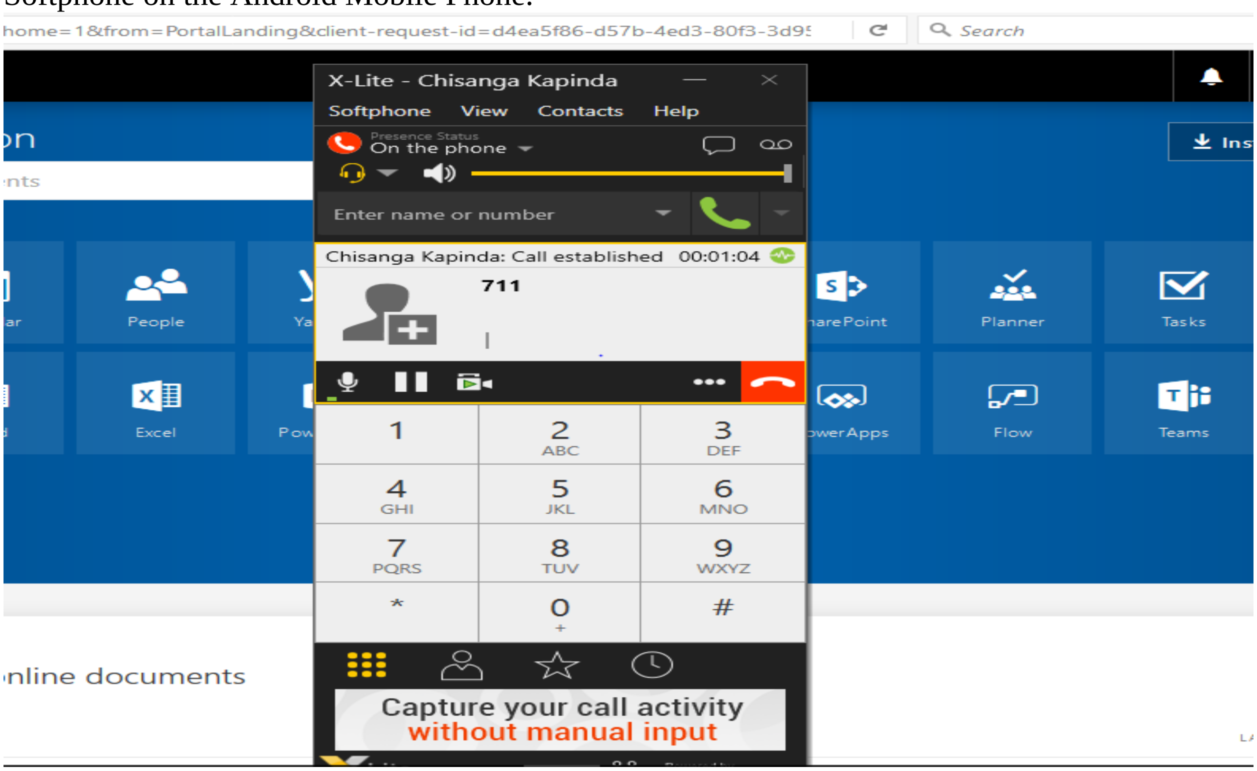


Figure 7:0 X-lite softphone with extension 149

Figure 2.11. Zoiper softphone with extension 711

3.8 Feedback from Users

Following the successful installation and testing of the systems. It was imperative that the participants independently gave objective feedback of the VoIP systems. The participants who had previously been using physical IP phones were required to test the open-source PBX and softphone clients on their PCs and Mobile Devices. After testing the systems, the participants through a structured questionnaire and interviews were asked to give a comparison of the system with the Cisco, Huawei and HP VoIP system that was being used in the target institution. Main factors of comparison included, quality, reliability, security, cost, features and integration.

1. **Quality**

Overall from a scale of **1-10** on call and other service quality from the 50 users, an average of **5** was recorded. This however, proved that softphones would replace IP phones with above average quality. This is to put in consideration that softphones were running on host Personal Computers and Mobile Devices with other shared services. The quality was mainly dependent on available processing power on such devices.

2. **Reliability**

Compared to physical IP phones, softphones and software PBX's uptime entirely depends on their host devices. In the case of the project, the systems were installed on a host server that had high availability due to other services being hosted by it. On a scale of 1-10, from 50 users an

ISSN: 3471-7102

average of **4** was recorded. The users generally felt the system was less reliable that physical systems that hard dedicated hardware.

3. Security

Provided that 3CX had an option to either use, Transport Control Protocol (TCP), User Datagram Protocol (UDP) and Secure Socket Layer (SSL) for transportation and initiation of VoIP traffic, which was similar to physical systems. In addition, due to the open-source nature of the project system, more features could be modified in order to add more customized security features. A feature that would not be available on prohibitive call managers. From a scale of 1-10, an average of **8** was recorded from users.

4. Cost

The system was installed on already existing and shared hardware. There, having implemented the systems at completely no cost on software, and hardware a **10** on a scale of 1-10 was recorded with regards to cost.

5. Features

3CX/Asterisk is open, freely available, and developed by a community of developers committed to constant improvement of the product. This ensures that the latest enhancements and fixes are available without the purchase of new software, licensing, or hardware. In many cases a Commercial Call Manager upgrade will require to purchase new software, pay for the same licenses again, and often times buying new rebranded HP servers that have been marked up by Cisco three times their original retail price. Similarly, users had a variety of softphone clients to install on their end devices, which provided different features compared to physical IP phones that only supported limited features. These included, video recording and instant messaging among others. An average of **8** was recorded from users.

6. Integration

Cisco and other call mangers makes great phones, however, 3CX/Asterisk allowed us to choose any phones or softphones wanted to use with our system. It is practically very prohibitive to manage anything on systems like Cisco's Call Manager that does not have a Cisco logo emblazoned on it. Following successful integration of old Cisco IP phone with softphone clients, users recorded an average of **8** on a scale of 1-10 with regards to integration.

4.0 Discussion

The questions this project addressed include: -

- would softphones and software PBXs replace desk IP phones and physical PBXs with equal or more reliability and security?

- would softphones and open source call managers lessen cost on government?
- would open-source Call Managers provide the integration of different IP phones and softphones?
- would softphones have more flexibility and mobility than IP phones?
- would government employees adopt to softphones in place of IP phones?

In order to address the discussed questions, there was need to compare 3CX/Asterisk with Commercial PBXs such as Cisco Call manager. With an analysis of feedbacks from users combined with facts about 3CX/Asterisk, the following were highlighted: -

Future Proof Technology

3CX/Asterisk is open, freely available, and developed by a community of developers committed to constant improvement of the product. This ensures that the latest enhancements and fixes are available without the purchase of new software, licensing, or hardware. In many cases a Commercial Call Manager upgrade will require to purchase new software, pay for the same licenses again, and often times buying new rebranded HP servers that have been marked up by Cisco three times their original retail price.

It is also predicted that more services are likely to be integrated on more flexible virtual phones. These services include, Voice, Video, Instant Messaging and Conferencing that may not be available on prohibitive and fixed IP hard phones.

Cost

Even if commercial call managers undercuts their cost upfront, they will make it up on the backend. If you buy a Cisco system you will pay a license for every extension on the system, the phones are more expensive, and we have to buy Microsoft exchange licenses for each voicemail box on the Unity voicemail system. On top of this, we need to pay for annual support from a Cisco partner who will charge 20% - 30% of the total cost of the system yearly.

Features

In some areas the Cisco phone system really excels. The distributive architecture of the system is quite nice. All in all, it is a great phone system. However, out of the box if we compare features, 3CX/Asterisk can do much more due to its flexibilities. In addition, due to the openness of its architecture we can make Asterisk do pretty much anything that you want.

Voicemail Systems

The voicemail system that comes free with 3CX/Asterisk is 100% better than the Unity voicemail system that Cisco uses. Unity relies on a Microsoft Exchange mail system to manage voicemails. This is a needlessly complex design that does not provide any enhancements to the overall features of the voicemail system. In addition, on the Cisco system voicemail administration is separate from user/extension administration. Therefore, in addition to logging into the Cisco Call Manager to manage the user and extension, the administrator has to log on to a completely separate system to administer voicemail. With 3CX/Asterisk, combined with our device management software, the User, Extension, Voicemail and device configuration are all managed from one screen as shown in **Figure 1.0** in the results sections.

Integration of Phones

Cisco makes a great phone, however, 3CX/Asterisk allows us to choose the phones we want to use with our system. It is practically very prohibitive to manage anything on Cisco's Call Manager that does not have a Cisco logo emblazoned on it.

In summary of the discussion sections 3CX/Asterisk can replace commercial call managers because of: -

- very rich functionalities due to its virtualization capabilities;
- ease of integration with other call managers and IP phones with software phones;
- no license or maintenance costs required;
- very IT-oriented solution which can be deployed in a virtualized infrastructure;
- ease of migration as it can be phased and allows Cisco and other IP telephones to be retained.

The drawback however, that made this project fail in the quality and reliability tests, comes in when hardware and dedicated PBXs are compared with software PBXs that share hardware with other server services. This therefore compromises on memory, shared network interface and processing power of their host servers. Despite this, the benefits outweigh the losses. Some other challenges of the project included the following: -

- Installation of the 3CX server in a Virtual Machine in a Government Physical server required authorization and support from Management.

- Installation of the softphone clients on individual PCs and Mobile Devices required Administrative privileges and cooperation from participants.

- The network to be used for the VoIP traffic was a live network currently used by other intranet and internet services hence the expected challenge on bandwidth.

- There were always security concerns within government to rely on open-source technologies.

5.0 Conclusion

The dependency of government institutions on expensive systems such as commercial call managers, calls for more research into other open-source alternatives. These alternatives would help lessen the cost on licenses without compromising on quality, reliability and security.

The main difference between an IP Phone and Softphone is that the Softphone is a virtual or software IP Phone that uses the same concept as a physical IP Phone. As technologies are shifting to virtualized systems, Softphones are preferred due to their flexibility to support more services through cost free upgrades. On the other hand, Open-source call managers provide more efficiency in hardware usage as they don't need dedicated hardware unlike commercial call managers.

The significance of this project was to provide a cost free alternative that ensures that all government employees that have access to the Government Wide Area Network through their mobile devices such as smart phones and personal computers, were connected via Voice over IP.

Secondly, this project would help on reducing the demand of commercial IP phone services and eventually cut down on the cost for Government through the Wide Area Network.

Thirdly, this project would see the replacing of Desk IP phones with Softphones on PCs and Smart phones resulting in increased access to majority government employees with access to PCs, Tablets and Smart Phones.

The project also aimed at ensuring that the VoIP systems would be linked with already existing IP phone systems through integration of Desk IP phones with Softphones.

From the tests and feedbacks from participants in the project. The main factors of comparison between open-source and other call mangers such as Cisco, which are dominant in our target institutions were cost, reliability, security and quality.

The motivation and significance of this project was that it was conducted in one of the key institutions mandated to spearhead the implementation of ICTs in government.

This paper ends with a recommendation that government institutions in Zambia should look more towards cheaper alternatives in the enhancement of ICTs in their governance systems. Such alternatives include open-source VoIP systems as one of them.

Acknowledgement

First, I would like to thank the Creator for giving me an opportunity to be one of the first students of the Information and Communication University (ICU) in Zambia. My special gratitude also goes to the entire Zambia Research and Development Centre (ZRDC) team for their partnership with ICU.

Further, I would like to thank Dr. Richard Silumbe, my lecturer and supervisor for his guidance and support in previous studies as well as this one.

I would like to extend my gratitude to all my lecturers and reviewers in this paper for their guidance and contribution. I also wish to thank all my students and fellow students for their motivation.

I wish to also say a big thank you to the Smart Zambia Institute (SZI) Management and all participants in the project.

Bibliography

[1] Avad Technologies, 2010. *Open Source PBX Does Not Mean Its Free.* [Online] Available at: http://blog.avadtechnologies.com/2010/04/open-source-pbx-does-not-mean-its-free.html [Accessed 6 June 2017].

[2] Daniel, A., 2016. *Sam Houston State University dumps Cisco VoIP for open-source Asterisk.* [Online] Available at: http://www.networkworld.com/article/2307107/lan-wan/university-dumps-cisco-voip-for-open-source-asterisk.html [Accessed 5 June 2017].

[3] Digium, 2017. *Asterisk Open Source Communications Solutions.* [Online] Available at: https://www.digium.com/products/telephony-solutions [Accessed 4 June 2017].

[4] Gentek, 2016. *Open-source versus proprietary IP PBX - Gentek VoIP, Broadband and Network Products.* [Online] Available at: https://www.gentek.com/blogs/open-source-vs-proprietary-pbx/ [Accessed 06 June 2017].

[5] Gilbert, N., 2010. *What are the advantages and disadvantages of open source.* [Online] Available at: http://www.ictinnovations.com/waht-are-the-advantages-and-disadvantages-of-open-source [Accessed 6 June 2017].

[6] Ministry of Transport and Communication, Zambia, 2017. *Ministry of Transport and Communication.* [Online] Available at: http://www.mtc.gov.zm/index.php/component/content/article/58-psc-latest-news/132-communications-sub-sector.html [Accessed 4 June 2017].

[7] OpenSource.Com, 2017. *What is open source.* [Online] Available at: http://opensource.com/resources/what-open-source [Accessed 5 June 2017].

[8] Smith, G., 2017. *The upsides and Downsides of Open Source VoIP Systems - VoIP Insider.* [Online] Available at: http://www.voipsupply.com/blog/voip-insider/open-source-voip-systems-pros-cons/ [Accessed 5 June 2017].

[9] SZI, 2016. *ICT STAFF Meeting SMART Z Presentation.* Lusaka, SZI.

[10] SZI, 2016. *SMART Zambia Phase I Project Overview- 13_07_2016 ver2-2,* Lusaka: SZI Microsoft.

[11] Werbach, K., 2005. *Using VoIP to Compete.* [Online] Available at: https://hbr.org/2005/09/using-voip-to-compete [Accessed 6 June 2017].