Low-Cost VoIP Solution for University PBX by Using SPA3102

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Abstract— *The benefits of implementing an open source VoIP system at General Sir John Kotelawala Defence University (KDU) where an analog system is still being used, and VoIP system provides not only voice communication but also voicemail, call conference, call waiting and Instant Messaging (IM). As KDU is one of the largest defence universities in the South Asian region, the security of the system should be very high. The proposed system offers the overall security as it required an authorized person or group of people to monitor all the conference and the data exchange through the system. The implemented system is a server-based software-design, "AsteriskNow" software PBX Version 6.12.6522. If KDU ever decided to go with commercial VoIP based servers such as CISCO, 3CX; then KDU also has to use proprietary equipment such as CISCO servers, CISCO switches, and IP phones. For that, KDU will have to spend extremely higher capital expenditure. The system provides a softphone to every user. When a person is not available on the analog line the call automatically transfers to his/her mobile softphone. A database is maintained to store all the contacts and those contacts can be checked through the softphone. The most important feature in the system design is that the system uses a public IP (Internet Protocol) address, which connects the system with the outside world, and also any registered person who is outside of the KDU can connect with the analog extensions of the KDU. VoIP is considered as real-time communication and time sensitive service. After implementing the system network performance and voice quality have been tested. Compared to the existing system, clarity of the voice is greatly increased and the number of calls that can be handled simultaneously by the exchange is increased. Post research questionnaires have proven that the system is very useful, reliable and cost effective method for KDU.*

Keywords— ***VoIP, PBX, AsteriskNow***

I. INTRODUCTION

In this universe of fast growing technological developments and the telecommunications, the need for fast-paced, integrated services are needed. Whereby, the Voice over Internet Protocol (VoIP) provides a sound environment for integrated services combining voice and data.VoIP is a well-suited replacement for current networks such as circuit switched based PSTN analog networks.

This research deals with implementing a VoIP network at General Sir John Kotelawala Defence University. The thought of utilizing an IP-based PBX is quite useful. Asterisk software is selected due to three reasons. First, it is Open Source (Hence Free). Then, it is easy to implement. And finally, it has many internet-support-forums. Proposed system integrates the corporate telephony system with the shared computer network, removing the need for two separate networks. PBX itself becomes just another server or group of servers in the corporate LAN and/or WAN, which helps to facilitate voice/data integration. The proposed scheme has been built up by implementing a soft PBX which acts as the server that performs call-routing functions, replacing the traditional legacy PBX.

This digital Asterisk PBX would allow a number of attached softphones to initiate calls to one another. Moreover, it connects to the existing analog telephone system and directly connects IP phones to this digital PBX system. This Open Source basic software would contain various features available in proprietary PBX systems: voice mail, conference calling, call waiting, and automatic call diverting, call forwarding, etc. The proposed system also facilitates the user in building the dial plan for the network.

The scope of this research task is to enable novel configurations of telecommunication and engaged in the lecture room, allowing extensions and variations of the informal interactions for collaboration across geographic boundaries.

II. BACKGROUND

In the world communication is more important to share information with others and for solving problems. Ancient people use different methods to communicate and after the invention of the telephone in the world, communication becomes more complex. All the people in the world using the telephone and world become more connected. While this communication technology is growing, the mobile phone was invented in the world. This all communications are using analog signals, but people try to change it into digital signals. Then billions of people use mobile phones to communicate with each other and later they move to data related communications such as data calls and messages (Krings and others, 2012). In these times, VoIP becomes more popular in the world. VoIP works with a variety of protocols, including Real-time Transport Protocol (RTP) for transfer of the multimedia data and Session Initiation Protocol (SIP) for establishing signals, and controlling sessions (Arora and Jain, 1999).

A. PBX

Private Branch Exchange (PBX) telephone systems have been in use in business since the early 1960's and have developed over time to provide for the needs of productions as they adjust and grow with new technologies and shifting methods of business communication (Channabasavaiah, n.d.). An earlier PBX system referred as Private Automatic Branch Exchanges (PABX). PBX system was adopted because they streamlined business operations and reduced costs by allowing a telephone network to be dedicated to an organization or entity rather than being managed through telephone providers such as SLT, Dialog, Mobitel. Internal calls within a PBX are free. KDU has an "Iwatsu ADIX-E" PBX system and it costs about 4 million Rupees in Sri-Lanka. This system provided by the Lanka Telephone Company (Pvt) Ltd in October 2010.It has 256 ports and can use only 220 ports. Because some ports are busy with the networks provider.

B. SIP

The Session Initiation Protocol (SIP) was originally designed by (Thompson et al., 2013) Henning Schulzrinne and Mark Handley in 1996 as a signaling-telecommunication-protocol, commonly used for controlling telecommunication sessions. SIP is used to transfer video and voice calls over Internet Protocol (IP) networks. SIP uses fewer resources and designed for real-time transmission, so it is less complex than other protocols. The main purpose is to establish, coordinate and disassemble a communication session between end devices. Also, SIP has the capability to forward calls over User Datagram Protocol (UDP) and Transmission Control Protocol (TCP). But VoIP only works with UDP because to achieve fast and great efficiency. SIP uses some identification tool named as Universal Resource Locator (URI). User Agents (UA) and SIP servers are the two parts of the SIP. Endpoints such as mobile phones, laptops, IP phones are going under User Agent. This can be either an agent client or an agent server. A UA client initiates a session by sending a SIP request. A UA server can accept, terminate or redirect the request as responses to this SIP request. SIP registrar servers, SIP proxy servers, and SIP redirect servers are the SIP server types. SIP has the capability to connect IP networks to the PSTN or other IP networks. Carriers are donated SIP Trunking packages to permit an enterprise to connect to the PSTN or to another IP network by means of a session border controller (SBC) device. SIP has several advantages such as working independently of the type of session, giving flexibility, easy troubleshooting, accommodating several users with differing capabilities, clear text messages and simplicity. They have several disadvantages such as processing text messages to initiate a session, hence putting a higher workload on router gateways.

III. PROBLEM DEFINITION

There are two main problems in the present analog PBX system. First, number of extensions cannot be increased due to few reasons. All the line expansion slots are occupied and KDU has to spend more than US$50,000 to expand the network. With the rapid expansion of KDU, it's hard to rewire new tall buildings and besides it will cost lots of manpower and money. Secondly, when choosing a network with internet facility to set up the system, the team had to join two networks together, since KDU has two Wi-Fi networks. They are own Wi-Fi network of KDU provided by Lanka Education and Research Network (LERN) and Wi-Fi network which is owned and maintained by Dialog, one of leading mobile operator in Sri Lanka. But these networks are not connected each other hence creating seamless connectivity was very difficult. These difficulties were faced when implementing this IP based PBX system.

IV. LITERATURE REVIEW

Internet telephony with VoIP (Voice over Internet Protocol) reaches critical mass, but there's already tremendous movement in this direction (Bryant, 2013). Many organizations are not only attracted to VoIP's assurance of cost savings, but its capability to move data, images, and voice traffic over the same connection (Rahman and Islam, 2014). With the fast pace of development and growing demands of the users for a higher quality of voice, data transfer at lower bandwidths and better performance over faster growing networks (Al-Saadoon, 2009), the efficiency that network users can reach with VoIP is almost mind boggling and cost effective. These can be achieved by a system which is built with open source software like Asterisk. There are profitable VoIP options out there, but many of them are expensive, running old, complicated code on obsolete hardware. Asterisk runs on Linux and can interoperate with almost all the standards-based telephony equipments, providing the opportunity to program it to one's liking (Bryant, 2013).

V. SYSTEM OVERVIEW

This research deals with implementing a VoIP network KDU. The thought of utilizing an IP-based PBX is quite useful and timely. The proposed system which is an Asterisk server (Wickramanayake and Brayant, 2011), is implemented as a soft PBX that acts as the server guarantee for performance routing functionalities replacing the traditional legacy analog PBX or key system (Jayasundara et al., 2015). The proposed system has been developed in the fast growing, cutting edge technology in the world.

Figure1. System Architecture

As in figure1 shows that the Asterisk PBX Server and SPA3102 device have plugged with the Switch to get the internet connection. The entire analog system has connected to the Asterisk server through the SPA3102 device. The server is connected to the internet using a public IP, so the users can access into the server from anywhere in the world.

A. Asterisk

Figure2. Installing Asterisk

The asterisk one of the world's leading open source telephony engine and tool, and a free software application of a telephone Private Branch Exchange (PBX). Like any PBX, it permits a number of attached telephones to make calls to one another and to connect to other telephone services together with Public Switched Telephone Network (PSTN). The Asterisk software consist of many features available in proprietary PBX systems: voice mail, interactive voice response (phone menus), conference calling and automatic call distribution. Asterisk supports a wide range of Voice over IP protocols. By supporting a mix of conventional and VoIP telephony services, Asterisk allows developers to build new telephone systems, or gradually migrating existing systems to new technologies.

B. Softphone

Initially 'Zoiper' softphone is used and then developed a unique softphone to the KDU by using Ozeki VoIP SIP SDK ("Ozeki C# SIP Stack - Ozeki VoIP SIP SDK Home Page," n.d.). The developed softphone can easily install to windows platform and its interface is more affordance to the user. It is a light weighted softphone and also maintains a small database to store other user's phone numbers. The Zoiper ("Zoiper - Free VoIP SIP softphone dialer with voice, video and instant messaging :: Zoiper," n.d.) VoIP softphone will look everywhere for your contacts and will display them in a combined list for easy access. It offers free encryption for all your text, voice and video communications and runs on a multitude of different platforms. It needs low memory and CPU usage and maintains audio quality even on older hardware. Zoiper is compatible with most VoIP service providers and PBX's. It offers free calls between Zoiper users. It offers free encryption for all your text, voice and video communications and runs on a multitude of different platforms. It needs low memory and CPU usage and maintains audio quality even on older hardware. Zoiper is compatible with most VoIP service providers and PBX's.

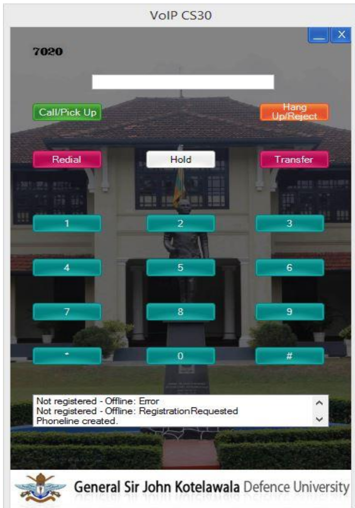

Figure3. Developed Softphone

C. Analog Card

'TDM400P 4 port FXS, FXO asterisk ("Digium TDM410P Configurator - VoIP Supply," n.d.) quad span analog Voice Board tdm410p' card is the analog card that used in our server for connecting analog phones. This card is with 2 FXO + 2 FXS modules installed and it supports any PBX system. The two FXO (Foreign exchange Office) ports to connect to two analog lines and two FXS (Foreign exchange Subscriber) ports connect to two analog phones.

Figure4. TDM410p Analog Card

A. Linksys SPA3102

Figure5. The Linksys SPA3102

The Linksys SPA3102 ("Linksys SPA3102 Configuration and Review," n.d.) SIP VOIP Phone Adapter enables feature-rich telephone service over the cable or DSL Internet connection. The SPA3102 also supports one PSTN FXO port to connect to a Telco or PBX circuit. The SPA3102 includes 2 100BaseT RJ-45 Ethernet interfaces to connect to a home or office LAN, as well as an Ethernet connection to a broadband modem or router. The SPA3102 FXS and FXO lines can be independently configured via software controlled by the service provider or the end user. It is compatible with all common telephone features: Caller ID, Call Waiting, Voicemail, etc. The Linksys SPA3102 Internet Phone Adapter enables high-quality feature-rich VoIP (voice over IP) service through the broadband Internet connection. By simply plugging it into the Router or Gateway, it allows the two standard telephone ports to connect analog phones or use one of the ports for a fax machine.

E. Specifications of Linksys SPA3102

- Call Waiting, Cancel Call Waiting, Call Waiting Caller ID
- Caller ID with Name/Number (Multinational Variants)
- Caller ID Blocking
- Call Forwarding: No answer, Busy, All
- Do Not Disturb
- Call Transfer
- Call Back on Busy
- Music on Hold

F. Features

1) Quality of Service: The expectations of Quality of Service (QoS) differ from person to person. For some people, the quality is the standard of the voice that gets through the telephone. On the other hand, others less background commotion and resounding they hear via telephone is the quality that they anticipate. Additional to these elements simple access to the service can likewise be characterized as a quality that a few people anticipate. Conversely, quality can be referred to the security of a conversation and reliability. A study illustrates that the majority of people preferred lower cost of service over responsiveness, value added service, reliability and voice quality. Furthermore, security and privacy of phone calls are another issues on QoS. The said issues, develop into extremely vital matters for law enforcement officials. There are many differences between security measures for public switched telephone networks (PSTN) and VoIP.

2) Cost: Cost is a matter that has more involvement than the end of the month phone bills. As costs include hardware requirements and training, switch over costs, and costs involved in potential or loss of business in transition. Companies have different cost strategies for their telephone service. This will be based on whether they are dealing intentionally, nationally or locally. The company CISCO provides a lot of hardware for VoIP phone service and claims on their website that companies have saved millions of dollars by using their technology.

VI. CONCLUSION

As mentioned above KDU has two Wi-Fi networks which are not integrated and hence making seamless connectivity between clients attached to the both networks, was the biggest issue was faced by the team. The LERN network has high bandwidth and it is divided into segments according to IP addresses. When installing Asterisk to the PC (the server) the team faced some difficulties because of this segment of the IP-based network. The second network, Dialog Wi-Fi has very low bandwidth, hence cannot get clear calls through this network. We use a specific IP address (10.10.30.1) for Asterisk server which is on the LERN network. In this stage server can communicate only with the devices which are on LERN network. Hence, the communication was very limited. Therefore, the team gets a public IP address (192.248.104.22) and combined these two IP addresses. Then end devices can access the public IP address and communicate with the server by using any network. Asterisk cannot run on Windows platform so that the team needs to buy a separate PC for installing Asterisk. Asterisk itself is an OS. Also, when using Asterisk two OSs cannot run on the same PC. When installing the TDM410p card, had to buy a new machine because the analog card has a PCI-Express slot and the previous machine had only mini-PCI slots. The team was unable to install the card because the TDM410p card manufacturer doesn't provide technical support in Sri Lanka. In order to avoid this complication, the team had to use a SPA3102 device to connect the analog telephones to the digital Asterisk server and also to connect the existing Analog network to the new digital Asterisk server. The team was unable to install the Short Message Service (SMS) facility to the server though the team spent reasonable time to researching on the forums. Some forums say that there are some modules to be purchased for installing features like SMS. However, the team was able to install the most important features to have a working digital Asterisk server at KDU.

VII. SUMMARY

Implementing a VoIP system for KDU was very timely since KDU was using a very outdated, un-expandable ana-log telephone system. As KDU, which is the defence Univer-sity of the Country and also is expanding very rapidly, it needs a communication system which is fast, reliable and more secure. This is a qualitative research design. After the implementation, the system has been tested on the univer-sity premises and it fulfills the needs of the current user base such as the lecturers, staff, and students. Users' feed-back was positive and they agreed that the system is us-er-friendly and efficient.

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