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Voice IP and SIP based Astrerisk PBX

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Abstract - The ability to communicate voice over an existing TCP/IP network is referred to as Voice over Internet Protocol (VoIP). As a result, the traditional PSTN's limits and downsides are no longer relevant.The experimental quantitative technique approach was utilised in these peer-reviewed works. By using both hardware and software simulations. The IP PBX software is based on Asterisk.

Asterisk is a Digium-sponsored open source communication platform product written in the C programming language and installed on a Linux-based operating system that converts a regular PC into a phone. It employs a variety of communication protocols, including SIP, SCCP, H.323, and MGCP. The audio CODECS also include compression capabilities, which helps to reduce network bandwidth. G.711, G.722, G.723, G729, and others are the most popular. This asterisk handles both hard phone and softphone customers' call routing requests.

Softphones can be installed and used on both sorts of operating systems, as well as Android Smart Phones and devices. These various remedies are presented to address various types of issues. In general, open source VoIP reduces a variety of costs for businesses and organisations, including call fees, installation costs, human resource costs, and investment costs, while also providing flexibility, scalability, reliability, and manageability. In addition, by combining Asterisk-based VoIP with Wi-Fi connection in order to have moveable communication rather than communication solely at an office desk, call on pocket maximises the productivity of businesses or enterprises. The final point of concern is VoIP security issues. Because this protocol is new and runs over IP, it inherits all of the IP's strengths and weaknesses, making it vulnerable to VoIP attacks by unauthorised individuals. VoIP utilising Open VPN is proposed to interconnect different branches from different sites of the corporation to overcome these types of challenges.

This aids secure VOIP communications between the branches over the secure tunnel to some extent. However, there are certain gaps that have yet to be discovered. These include the price of cable, wireless, installation, and maintenance. The second reason is that when the number of participants in both

the LAN and the WAN grows, a communication bottleneck develops. As a result, we shall be subjected to laws governing voice communication and interruption. The last reason is that owing to Wi-Fi radio frequency interference with various equipment, such as aircraft, hospitals, and nuclear and chemical plants, we will not be able to use VoIP conversations because of the harm and risks that this interference causes. Despite the fact that these gaps have been discovered and reported, all of the researchers' tremendous work and contributions are valued.

***Key Words*:** VoIP, IP voice, IP Telephone, Internet Telephone, IP PBX, EPBX, CODEC, SIP, MGCP, H.323, PSTN, OPEN VPN, SCCP.
Asterisk Keywords: VoIP, IP voice, IP Telephone, Internet Telephone, IP PBX, EPBX, CODEC, SIP, MGCP, H.323, PSTN, OPEN VPN, SCCP, Asterisk.

1. INTRODUCTION

The name VOIP refers for Voice over Internet Protocol, but it can also be referred to as IP Telephone, IP Voice, or Internet Telephone. VOIP developed in the mid-1990s, when enthusiasts noticed that voice information packets could be sent over the internet rather than utilising an old or standard telephone system. The traditional or old Electronic Private Branch Exchange (EBPX) system is mostly used for setting up telephone calls over a dedicated wired network, but with the advancement of computing technology, Internet Protocol Private Exchange (IP PBX) has emerged as an alternative to the Traditional EPBX System. The transition from circuit switching to the more efficient one packet switching model of VOIP protocols and CODECs has enabled a significant transformation in voice communication transmission.

VoIP became extremely popular after the widespread adoption of the Internet, particularly among multinational corporate firms. Many businesses have adopted or plan to employ VoIP in place of traditional telephony services. VoIP claims to offer greater flexibility and cost savings. The main advantages of VOIP over the traditional telephone system (PSTN) are lower costs and greater flexibility. However, because this method converges voice and data over the IP network, it complicates VOIP security issues and introduces new vulnerabilities.

2. LITERATURE REVIEW

The Body of the Paper describes an asterisk-based VoIP system for a business network in an intranet environment, and the second worry is addressing the security issue, which arises when the enterprise network grows, and which is partially addressed by the suggested OPEN VPN approach.

The asterisk IP PBX (http://www.asterisk.org) is an Open Source VoIP system developed in C and installed on an operating system to convert a computer into a telephone. It runs on the Linux operating system. It supports SIP, H.323, MGCP, and SCCP, among other VoIP protocols. It can also be connected to an IP network as well as a traditional network, such as a PSTN network, with the help of an adaptor.

In a similar vein, concentrates on VoIP based on Asterisk PBX, demonstrating how voice packets are exchanged between the sender and recipient utilising the Asterisk PBX server and clients, which can be cable or wireless. It also performed SIP network analysis and network analysis based on the aforementioned parameters to determine whether the existing network is capable of handling VoIP traffics, and finally, based on the results observed in the graphs, as the number of calls increases, so does the performance of bandwidth.

The goal is to provide a LAN-based server that allows two or more users to interact within an organisations by allowing them to use an extension. The "Raspberry Pi contains a Broadcom BCM2835 system on a Chip (SoC), which incorporates an ARM1176JZF-S 700 MHz Processor, with 512 MB of RAM," according to the report. Due to the capabilities of ARM11, this method enables wireless connection while also improving performance.

The similarities are that they all used Open Source VoIP (Asterisk) to tackle their issues. They utilised Experimental Quantitative methodology because it helped them identify and specify the qualities of the items they needed to conduct the experiment. As a result, they are able to perceive the results by seeing the simulations and tests that have been completed. These can be considered a strength because they all handled diverse difficulties while maintaining a cost-effective system.

The research' limitations remain that some expenditures, such as investment, installation, and maintenance, must be resolved. The second is a communication bottleneck caused by the constantly increasing number of users. Because the bandwidth and equipment for data transmission are limited, the performance needs to be improved. Due to RF interference with the equipment, asterisk-based VoIP over Wi-Fi is not usable in particular situations, such as aircraft, hospitals, chemical and nuclear power plants. As a result of this interference, there is a danger of harm and loss in both life and business.

2. METHODOLOGY

All researchers utilises nearly identical materials and technique. These materials include a Linux platform based Asterisk IP PBX for handling calls by functioning as a call server installed in a computer with good performance, clients in both hard and soft phones, networking switch, and network cables. The Asterisk Wi-Fi portable Voice Calling System uses ARM11 to increase the existing communication system's performance. The softphone is compatible with all operating systems, including Linux's Zoiper Softphone and Windows' X-Lite.

The softphone can be used on a desktop, laptop, smartphone, or other smart device. This communication technology can be used in both a LAN and a WAN environment.

SIP, H.323, MGCP, and SCCP are among the VoIP protocols supported by Asterisk (http://www.asterisk.org). It can also be connected to an IP network as well as a traditional network like the PSTN. This open source software has a number of basic features, including security, scalability, powerful and rich features, integration with external applications via AGI (Asterisk gateway interface API, which is compatible with programming languages such as C, Java, Perl, PHP, and others), and cost effectiveness.

Asterisk includes channel and application modules that assist in achieving the desired result. DAHDI, SIP, IAX2, and H.323 are the channel modules. Digium Asterisk Hardware Device Interface (DAHDI) is a card that connects to an existing traditional telephone system. This physical telephone interface card must be inserted. The Session Initiation Protocol (SIP) is a basic signalling protocol that assists with call processing. Inter Asterisk Exchange version 2 (IAX2) is the second version of the Inter Asterisk Exchange. It is an asterisk proprietary protocol that aids in the connection of several asterisk servers throughout the internet. The application modules are the asterisk's services and features, such as call forwarding, voice mail, phone conference, and IVR (Interactive Voice Response or automated voice answers), among others.

In addition to the aforementioned, employs Open VPN Software to address VoIP security issues across the company's many branches. The "Raspberry Pi has a Broadcom BCM2835 system on a Chip (SoC), which includes an ARM1176JZF-S 700 MHz Processor, with 512 megabytes of RAM" is also used and proposed to improve performance. Asterisk-based VoIP over Wi-Fi using the "Raspberry Pi has a Broadcom BCM2835 system on a Chip (SoC), which includes an ARM1176JZF-S

All of the writers employed an experimental methodology approach, which entails reviewing existing methodology and then identifying the requisite materials. And it allows you to design the intended communication system and run experimental simulations and emulation to compare the predicted and actual results.

2.1 SYSTEM ARCHITECTURE

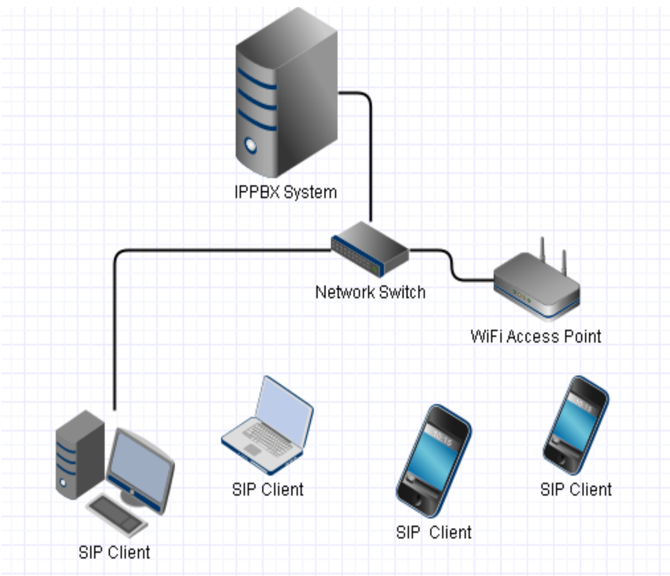

Fig -1: System Architecture

3. WORKING

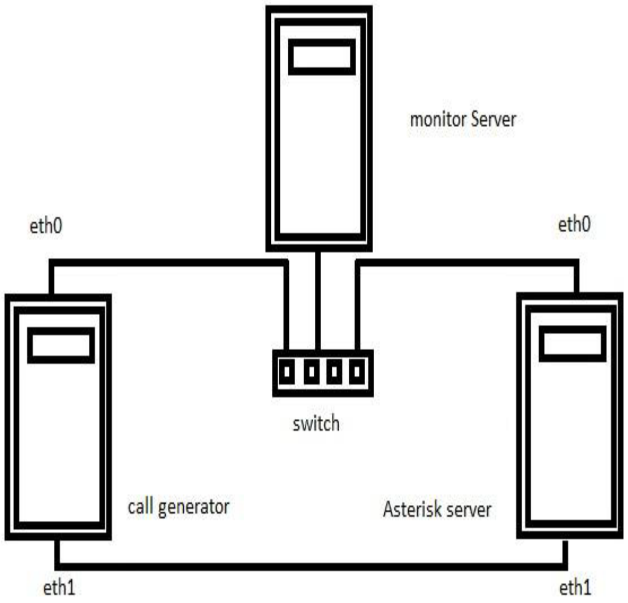

Fig -2: Test Procedure

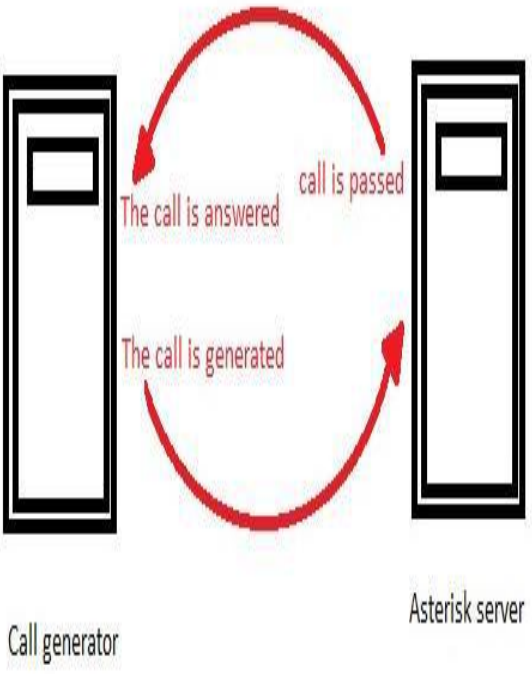

Fig -3: Call generator

4. FLOWCHART

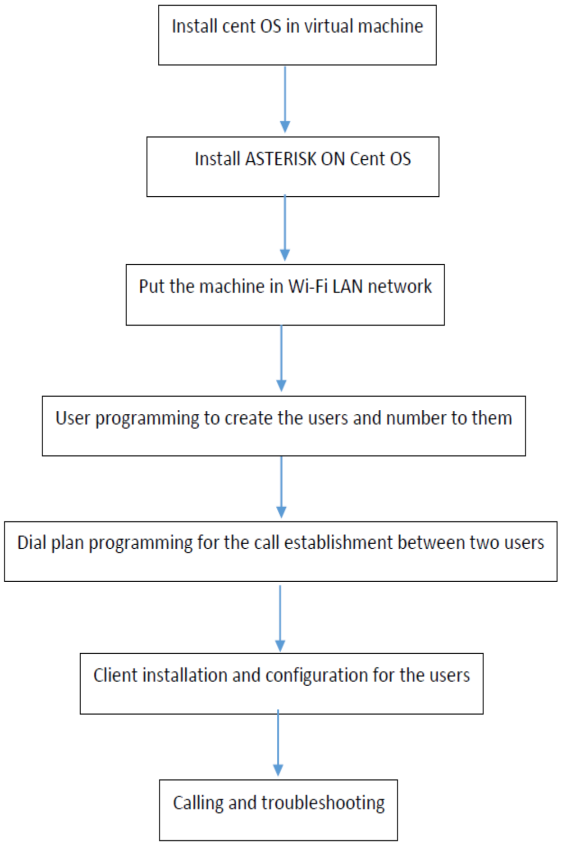

Fig -4: Process Flowchart

5. OUTCOMES

The previous investigations have resulted in the design and implementation of a LAN VoIP communication system utilising Asterisk IP PBX in order to reduce LAN costs and provide a rich communication system. The other is to develop and deploy LAN to WAN VoIP utilising Open VPN Software in order to create secure VoIP connection across enterprises or between different asterisk servers located in different sites or branches of the organisation.

Finally, within the organization's LAN, set up an Asterisk Wi-Fi portable Voice Calling System using ARM11 in order to give communication mobility or portability with excellent performance due to ARM11 features. All of these studies save money on call fees, installation and wiring costs, as well as investment and human resource expenditures like maintenance.

6. CONCLUSIONS

The EPBX requires a significant investment, is difficult to administer, and is not scalable. To address these issues, researchers devised VoIP. As a result, VoIP has developed through time and is now used for both LAN and WAN implementation. Because VoIP is simple to use, adaptable, and cost-effective. Despite the fact that many researchers conducted studies and produced numerous results, there are some gaps that I discovered. The papers' strengths, according to my critics, are that they follow scientific research methods and experimental quantitative research methodologies, despite the fact that some publications' technique approaches and gaps are not explicitly disclosed.

The following remarks, on the other hand, refer to the flaws of these articles as a research gap. The first research gap is that the number of internet users is expected to increase in the near future, resulting in a deterioration in VoIP communication performance due to restricted bandwidth and data transmission, as well as the fact that both data and voice are trunked together. The second issue is the large number of wires and the associated installation and maintenance costs. The third barrier is the technological one, which is caused by radio frequency interference with the organization's equipment, which puts people's health at risk.

As a result, some places, such as hospitals, airlines, and chemical and nuclear power plants, are unable to use VoIP over Wi-Fi.

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