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Asterisk based VOIP in Light Fidelity

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ABSTRACT

VoIP is a term refers to Voice over Internet Protocol enables to communicate voice over the existing TCP/IP. As this overcomes the challenges and drawbacks of the PSTN traditional circuit based communication system. These reviewed papers used experimental quantitative methodology approach. By conducting simulation both on hardware and software. The software is Asterisk based IP PBX. Asterisk is a Digium sponsored open source communication platform product written in C programming language put on OS that turns ordinary PC to a telephone and it runs on Linux OS.it uses different types of communication protocols such as SIP, SCCP,H.323, MGCP. And the audio CODECS are also provides compression capability to save network bandwidth. The most popular are G.711, G.722, G.723, G729 etc. This asterisk handles call routing requested from both hard phone and softphones clients. The softphones can be installed and worked on both types of OS as well as in android Smart Phones & devices. These different solutions are proposed to overcome different types of problem. In general this open source VoIP is saves different types of costs of enterprises and an organizations such as call fee costs, installation cost, human power cost and investment cost and it adds both flexibility, scalability, reliability, &manageability. Beyond this some problems are also solved by proposing Asterisk based VoIP using Wi-Fi communication in order to have movable communication rather than communication only in an office desk, call on pocket maximizes productivity of the organizations or enterprises. The last concern issues is VoIP security problems as this protocol is a new and works over the IP it inherits all the strength and weaknesses of the IP and this leads to VoIP attack by un authorized personnel. In order to overcome this types of problems VoIP using Open VPN is proposed to interconnect different branches from different sites of the organizations. To some extent this helps to have secure VOIP Communications between the branches over the secure tunnel. But there are other still unseen gaps. These are cable, wireless costs, installation costs and maintenance costs. The second one is as the number of participants of both the LAN and WAN are increasing from time to time it results communication bottle neck. Due to this reason we will be faced to law voice communication and interruption. The last one is due to the Wi-Fi radio frequency interference with some equipment's such as airlines, hospital and nuclear and chemical plantations we will not have VoIP communications as this interferences leads to some damage and risks. Even though these gaps are observed and identified all the great works and contributions of the scholars are appreciated.

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

I. INTRODUCTION

The term VOIP stands for Voice over Internet Protocol or it can be called in an interchangeable way such as IP Telephone, IP Voice, and Internet Telephone. VOIP originated in middle 90's, as once hobbyists began to see some extent of voice information packets over a web rather than exchanging using old or standard telephone system [1, 3, 4]. Mostly standard or old or Electronic Private Branch Exchange (EBPX) is used for setting up telephone calls using a dedicated wired network and with emergence of computing technology, Internet Protocol Private Exchange (IP PBX) has been established as an optional to the Traditional EPBX System [2]. Transitioning from the circuit switching to the more efficient one packet switching model of VOIP protocols & CODECs has been enabled a great change in the transmission of voice communications.

Following the wide spreading of Internet, VoIP became very popular, especially amongst global enterprise businesses. Instead of traditional telephony services, many enterprises have adopted or plan to use VoIP. VoIP promises to provide more flexibility and cost-efficiency [5].The lower cost and high flexibility are the main advantages of the VOIP over the traditional telephone system (PSTN), even though this method converges the voice and data over the IP network, it complicates VOIP security issues and introduces new vulnerabilities [6].

II. LITERATURE REVIEW

[7] Describes the asterisk based VoIP system for an enterprise network in the intranet environment and the second concern is realizing the security issue as the enterprises network grows it rises a security concern and this problem is solved to some extent using the proposed OPEN VPN scheme.

The [7,8,9,10] asterisk IP PBX (http://www.asterisk.org) is the Open Source VoIP system written in C programming language put on an OS to turn the Pc to telephone. It runs in a Linux OS environment. It supports various VoIP protocols such as SIP, H.323, MGCP and SCCP. And it can be connected with IP network as well as traditional network such as PSTN network using adaptor.

In a similar way [8] also focuses on VoIP based on Asterisk PBX, this shown that how the voice packets are transmitted between the sender and receiver using the Asterisk PBX server and clients are both in wired and Wi-Fi based. It also conducted the SIP network analysis and network analysis are also conducted based on the aforementioned parameters to measure the capability of the existing network whether it can handle the VoIP traffics, finally based on the results observed from the graphs as the number of calls increase the performance of bandwidth also increases.

The objective of [10] is to designing LAN based server which enables two or more users to communicate within the organization by providing an extension. This study focuses on the asterisk based VoIP over a Wi-Fi by using 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." This solution enables to have wireless communication and to improve performance due to the features of ARM11.

The similarity of [7, 8, 9, and 10] is they also focused on the Open Source VoIP (Asterisk) to solve the problems. They used Experimental Quantitative methodology as this approach helps them to identify, define the characteristics of the required materials to conduct the experiment. So, this helps them to perceive the results by observing from the conducted simulations and experiments. These can be taken as their strength as all they solved different problems with having cost efficient system.

The limitations of the studies are still there is some costs needed to solve such as investment, installation & maintenance costs. Second one is communication bottle neck due to exponentially increasing the number of users from time to time. Since the bandwidth and equipment data transmission is limited, due to this reason the performance needs an improvement. The third one is the asterisk based VoIP over Wi-Fi is not applicable some areas these are in an airlines, Hospitals, chemical & nuclear plantation due to the RF interference with the equipment. As this interference leads to risk and damage both in life and business.

III. MATERIALS AND METHODOLOGY

The materials and methodology used by all researchers are almost the same. These materials are Linux platform based Asterisk IP PBX to handle the call by acting as call server installed in computer having a good performance and the clients are both in hard phone and softphone, networking switch and network cables. The Asterisk Wi-Fi portable Voice Calling System usingARM11 to improve the communication performance of the existing communication system. The softphone is that runs in all operating systems such as Zoiper Softphone for Linux OS and X-Lite for windows OS. The softphone works in a desktop computer, laptop, smartphones and smart devices [7]. This communication system works both in LAN & WAN.

Asterisk (http://www.asterisk.org) supports various VoIP protocols such as SIP, H.323, MGCP and SCCP. And it can be connected with IP network as well as traditional network such as PSTN network. This open source software have many basic capabilities such as security, scalability, powerful and rich features, integration with outside application using AGI(Asterisk gateway interface API, that compatible with programming language such as c, Java, Perl, PHP etc.) and it is cost efficient. Asterisk has channel and application modules that helps to realize the output. Channels modules are DAHDI, SIP, IAX2 and H.323. DAHDI is Digium Asterisk Hardware Device Interface card that helps for interfacing with the existent traditional telephone system .it is necessary to insert this physical telephone interface card. SIP is the Session Initiation Protocol the basic signaling protocol helps for call processing. IAX2 is Inter Asterisk Exchange version 2. It is asterisk proprietary protocol helps for connecting different asterisk severs that found in different sites over an internet. The application modules are the services or features provided by the asterisk, such as call forwarding, voice mail, voice conference and IVR (Interactive Voice Response or automatic voice responses) etc.

In addition to the above ones [7] also uses Open VPN Software for the sake of overcoming VoIP security problems between different branches of the organization. Asterisk based VoIP over a Wi-Fi by using 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[10].

The methodology used by all authors is experimental methodology approach as this comprises evaluating the existing methodology and then identification of the required materials. And enables to Design the expected communication system and conducting the experimental simulation and emulation to evaluate the expected outcome against the existing outcome.

IV. RESULTS

The result of the existing studies are designing and implementing LAN VoIP communication system using Asterisk IP PBX in order to save the costs incurred within the LAN and to get rich communication system [1][7][8][9]. And the other one is designing and implementing LAN to WAN VoIP using Open VPN Software in order to establish a secure VoIP communication between enterprises or different asterisk server found in different sites or branches of the organization [6]. And the last one is setting up Asterisk Wi-Fi portable Voice Calling System usingARM11within the organization LAN in order to provide communication movability or portability with a high performance due to ARM11 features[2][10].All these studies saves call fee cost, installation and wiring costs, and investment cost and human power costs such as maintenance cost.

V. CONCLUSION

As the EPBX is needs high investment, complex to management and not scalable. To overcome these challenges researchers developed VoIP. Hence through time VoIP is improved and used for both LAN and WAN implementation. As the VoIP is simple to manage, flexible and saves cost. Even though many scholars conducted studies and brought many result, Sill It have some gaps I found. Based on my critics the strength of the papers are aforementioned as they follows scientific research method and experimental quantitative research methodology even though some papers methodology approach and gaps are not clearly stated. On the other hand the weakness of these papers are mentioned as a research gap in the following statements. The first research gaps is in the near future the number of participants of the internets are growing from time to time so this growth leads to decrease the VoIP communication performance as the given bandwidth and data transmission is limited and also both the data and voice are also trunked together. The second problem is huge amount of cables and installation, maintenance cost. The third one is the technology barrier due to radio frequency interference with the organization equipment's as it leads to health endanger and risk. Due to this reason some areas are unable to use VoIP using Wi-Fi such as Hospitals, Airlines, and chemical & nuclear plantation.

On the other hand, in order to overcome these challenges and problems there is ongoing research proposed as a solution to fill the existing gaps mentioned in the above. The research is designing and Implementing Asterisk Based VoIP Using LiFi in case of Adama Referral Hospital. From this solution any hospitals, airlines, chemical plantations can benefit as the proposed solution is a most fast, reliable, flexible, safe, and cheap. Due to this reason, anyone can also do further solution in different areas using the LiFi and Asterisk Technology such as designing smart transportation, Smart home automation and IoT.

VI. REFERENCES

[1]. Sonaligolhar, Prof. "Asterisk VoIP Private Branch Exchange" V.S Dhamdhere Computer Department G.H. Raisoni COEM, Pune, Maharashtra, International Journal of Advanced Research in Computer Science and Software Engineering , Vol 5, Issue 6, ISSN: 2277 128X, June 2011 , ISSN: 2277 128X ,India

[2]. Harshada Jagtap, Prof. D.G.Gahane, Student, & Professor "Asterisks Internet Protocol Private Branch Exchange with Smartphone" Dept. of Electronics and Telecommunication, Priyadarshini College of Engineering, Nagpur, International Journal of Innovative Research in Computer &Communi,cation Engineering, Vol. 3, Issue 5, ISSN(Online): 2320-9801, ISSN (Print): 2320-9798. 2015, India

[3]. Alireza Heravi, Sameera Mubarak "EVALUATION OF USERS' PERSPECTIVE ON VOIP'S SECURITY VULNERABILITIES" School of Computer and Information Science

University of South Australia, Australian Information Security Management Conferences, Symposia and Campus Events Conference, Edith Cowan University, 2011

[4]. White Paper Voice over Internet Protocol (VoIP). Optimizing business processes using modern communication, T-Systems Enterprise Services GmbH, Corporate Marketing & Communications, Uwe Werner, Mainzer Landstrasse 50, and 60325 Frankfurt am Main, Germany

[5]. Lian Wu Anders Hagelskjær Aasgaard "Migration of VOIP/SIP Enterprise Solutions towards IMS" Information and Communication Technology, Agder University College Faculty of Engineering and Science, Grimstad Norway, 2006

[6]. Piero Fontanini "VoIP Security" Department of Computer Science and Media Technology, Gjovik University College, 2008, Norway

[7]. Fumikazu Iseki, Yuki Sato, Moo Wan Kim, "VoIP System based on Asterisk for Enterprise Network" Department of Information Sciences, Tokyo University of Information Science, Japan, Feb. 13~16, 2011 ICACT2011, ISBN 978-89-5519-154-7

[8]. Trupti Datarkar, Assistant Professor N.P.Bobade, & Professor M.A. Gaikwad "Voice over Internet Protocol (VOIP) Based On Asterisk" Electronics Engineering, BDCOE, Sewagram, International Journal of Applied Research 2015; 1(8): 499-502, ISSN Print: 2394-7500, ISSN Online: 2394-5869, www.allresearchjournal.com

[9]. Dr.Ghossoon M.Waleed Al-Saadoon "Asterisk Open Source to Implement Voice over Internet Protocol" School Of Computer and Communication Engineering, University Malaysia Perlis, IJCSNS International Journal of Computer Science and Network Security, VOL.9 No.6, June 2009

[10]. Sayyad Nikhat Parveen, Prof. Tirupati M. Goskula "Asterisk WiFi portable Voice Calling System using ARM11" Electronics Engineering(Communication), Anjuman College of Engg & Technology, Nagpur, International Journal of Applied Research 2016; 2(4): 567-570 ISSN Print: 2394-7500 ISSN Online:2394-5869 IJAR 2016; 2(4): 567-570.