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Asterisks Internet Protocol Private Branch Exchange with Smartphone

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ABSTRACT: A telephone conversation can be done over a data network byVoice over Internet Protocol(VoIP). VoIP products are cost efficient, more versatile and also provide good voice quality as compared to proprietary system and promise converged telecommunication and data services. This paper describe Internet Protocol Private Branch Exchange (IP-PBX)calling on smartphones. IP-PBX is a complete telephony system that provides both wireless and wired calling without using SIM card. IP-PBX server is implemented by using CentOS Linux based operating system where Asterisk open source package is used. By implementing IP-PBX server wireless communication can be done in WIFI range using laptops and smartphones.

KEYWORDS: VOIP, IP-PBX, CentOS, Asterisk.

I. INTRODUCTION

Mostly Electronic Private Branch Exchange (EPBX) is used to conduct telephone calls over a wired network. With the development of computing technology, Internet Protocol Private Branch Exchange (IP-PBX) has been established as an alternative to traditional EPBX system. Evolving from circuit switching to a more efficient packet switching model the VoIP protocols and codecs has enabled a remarkable change in the transmission of vocal communications.[5][6].

While implementing Internet Protocol Private Branch Exchange Linux based operating system is required. In this paper, CentOS Linux based operating system is used as it is user friendly. After the installation of CentOS operating system open source package Asterisk is installed. With Asterisk the IP-PBX can be designed according to the requirement of the organization, the required features can be added according to the need. For communication hardphones or softphones installed in computers or laptops can be used. For communication using smartphones an application called CSIPSimple is used.

Wireless IP-PBX utilizes WIFI technology for communication, the same wireless infrastructure used for your corporate network. Just as we use mobiles and laptops within this wireless infrastructure to gain access to information, now we can use wireless IP phones system as this system uses the telephony function directly into an already existing data network. This provides an advantage that voice and data network can be used together using single system.
One of the major advantages of the IP-PBX wireless phone is thatyou can carry your extension with you inside a wireless networked environment. The wireless IP phone carries your personal extension and is part of your corporate phone system.

In large or small organisations, IP telephony offers many benefits. The major benefit will be productivity. By extending mobile communication in the organisation, the productivity of the users will be increased when they are not at their desk using wireless IP telephony. This improves the business response results by enabling users to answer critical business calls anywhere anytime within the wireless campus.

Another advantage of this system is that it is cost saving. Calling can be done free of cost within the campus. Expanding the communication system is easier and cheap. As it uses wireless infrastructure there is no need of additional hardware or wiring for new user.

IP-PBX system is easy to install as compared to proprietary phones. Proprietary phones may be cumbersome and somewhat difficult to install and configure. A computer savvy person can easily install Internet Protocol Private Branch Exchange.

II. RELATED WORK

In [1] authors presented some theoretical and practical results while setting voice over internet protocol server using open source VoIP server Asterisk. Authors have set up server based on Asterisk where the clients where connected to the server with the help of softphone or hardphone and then the configuration of server with the softphone, where the have used SIP protocols. In [2] authors have implemented features like conferencing, voice mailing and paging. Along with this the have configured hardphones on the client side with the server. All the applications are compatible with Linux operating system and are developed in C language. In [3] authors have described features of VoIP in internet environment implemented by using Asterisk and also high security Open VPN is described for enterprise network. In [4] authors have described the Internet Protocol Private Branch Exchange over Electronics Private Branch Exchange (EPBX). Authors have used Linux based operating system 'trixbox' for implementation of the server. Accessing of the server, extension configuration and configuration of SIP phones is also described by the author. In [5] authors have implemented Voice over Internet Protocol in university campus and have studied the results. Authors have also described performance parameters such as Latency, Jitter (delay variability), packet loss, Bandwidth requirements. In [6] authors have described main characteristics of Voice over Internet Protocol. Also comments previous work. Then the implementation of test system has been described with the analysis of the result such as low processor utilization. Results show the existence of the relationship between the protocols. Memory utilization is in the range of 33MB to 42MB, ie there is low requirement of memory.

III. PROPOSED SYSTEM

A. *Implementation*

Initially we need to implement IP-PBX sever. Firstly we have to install the Linux based operating system. In this paper we are using CentOS Linux based operating system which is user friendly and easily to install. Second step is to install Asterisk- the open source package. With Asterisk the IP-PBX can be designed according to the requirement of the organization, the required features can be added according to the need.

For installation of Asterisk following package sources are required.

- Asterisk main program
- Zapata Telephony Driver (zaptel)
- PRI libraries (libpri)
- Asterisk sound package

X-lite software is used for communication through laptops and personal computers. X-lite has a dialing pad and all other calling options which are normally found in a phone. After the registration of X-lite peer with the server calling can be done.

B. *SIP framework*

Session Initiation Protocol (SIP) is used in IP-PBX for setting up calls between the peers. Its purpose is to allow two end points talk to each other but does not deal with media of the call. Figure1 shows sequence of steps followed while making a call between two users. First the users are registered with Asterisk server; once the users are registered call can be placed between the users. While calling an invite request is first sent to asterisks server then via asterisk server it is send to the called party. SIP only takes and makes call while the media session is carried by another protocol RTP. RTP protocol is used to deliver voice.

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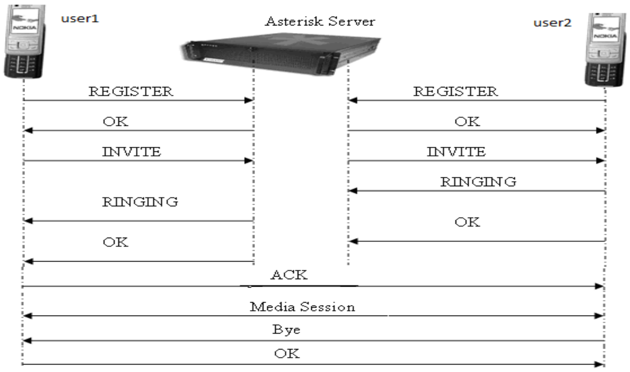

Figure1. IP-PBX SIP calling sequence.

IV. CONFIGURATION ON SMARTPHONE

In IP-PBX for communication using smartphone CSipSimple application is used.CSipSimple is android application which uses Session Initiation Protocol (SIP). Generally normal calling using smartphone requires SIM card whereas for IP-PBX calling there is no need of SIM card. The only things required are CSipSimple application and Wi-Fi facility in the smartphone. Within the range of Wi-Fi call can be done from anywhere. Figure2 shows demonstration of call using smartphone.

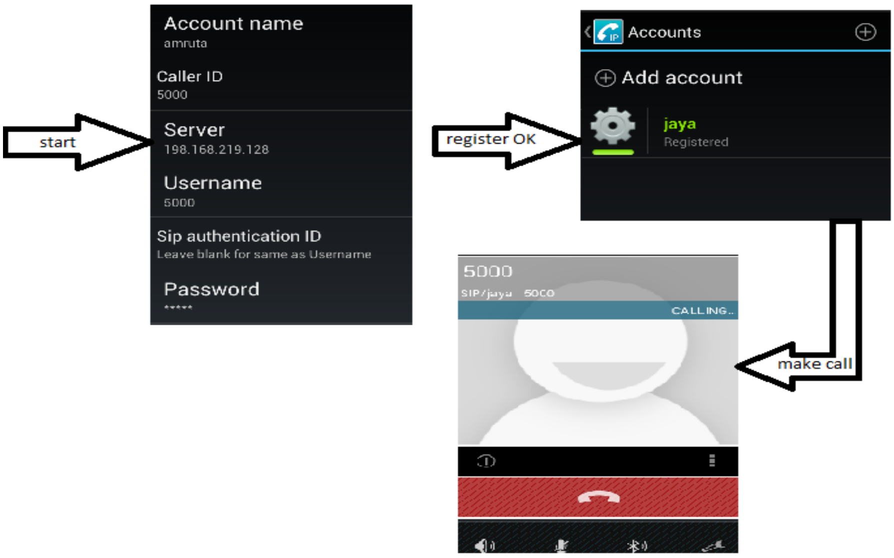

Figure2. Demonstration of call

V. RESULTS

System shows that Asterisks Internet Protocol Private Branch Exchange with Smartphone is feasible. This system is easy to install as compared to proprietary phones. As compared to proprietary system this system is cost efficient and maintenance work is reduced. With smartphones user can receive the call from anyplace with the Wi-Fi range.

Figure3. Shows Asterisk console in which successful registration of user 5006- smartphone user is shown. In the next line it shows information of the peer 5006, Csipsimple_GT-I9100 i.e. the name of the application and the smartphone number. Next it shows that peer 5006 is reachable which means call can be placed. Asterisk console gives every single detail of the user.

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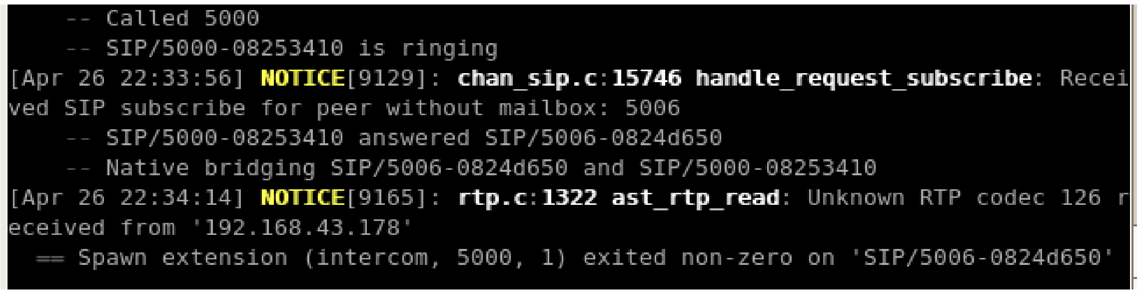
Figur3. Asterisk console showing registration of user 5006

Figure4 shows Asterisk console showing calling activity of the user. User 5006 is calling user 5000. Once the request is send then 5000 starts ringing. Then 5000 answers the call and media session starts. All these activities are shown in Asterisk console.



Figur4. Asterisk console showing calling activity of the user

VI. CONCLUSION AND FUTURE WORK

This paper describes Internet Protocol Private Branch Exchange with smartphones. This system is cost efficient and reliable as calling is done within the range of Wi-Fi and there is no need of SIM card for calling. Internet Protocol Private Branch Exchange is easy to configure and install only few steps are required for configuration and installation. As far as future work is concerned code for addition features can be designed like call center functions, ring group etc.

REFERENCES

1. Mohammed A Qadeer, M J R Khan, Ale Imran,"Asterisk VoIP Private Branch Exchange" IMPACT-2009, in IEEE,2009.
2. Ale Imran, M. A. Qadeer, "Conferencing, Paging, Voice Mailing via Asterisk EPBX", International conference on Computer Engineering and Technology, 2009.
3. Fumikazu Iseki, Yuki Sato, Moo Wan Kim, "VoIP System Based on Asterisk for Enterprise Network", ICACT 2011.
4. Mr. Sonaskar, S.D.Giripunje,"Low cost IP Private Branch Exchange (PBX)", International Journal of Computer Applications, volume 23-no.3, June 2011.
5. Shirish V. Narnaware, "Implementation of VoIP communication model with open source technologies", International Journal For Engineering Application And Technology ISSN:2321-8134.
6. Pablo Monotoro, Eduardo Casilari, "A Comparative Study Of Standards with Asterisk" 2009 Fourth International Conference on Digital Telecommunication,2009 IEEE, DOI 10.1109/ICDT.2009.8
7. BUR GOODE, SENIOR M EM BER, IEEE "Voice Over Internet Protocol (VoIP)" in PROCEEDINGS OF THE IEEE, VOL. 90, NO. 9, SEPTEM BER 2002