

Efficient and Economic IP Private Branch Exchange for Organization

Sayyad Nikhat Parveen ¹, Prof. Tirupati M. Guskula²

¹M.Tech Scholar, Electronics Engineering (Communication), Anjuman College of Engg & Technology, Nagpur, India

²Assistant Professor, Electronics Engineering (Communication), Anjuman College of Engg & Technology, Nagpur, India

Abstract - A private branch exchange PBX is an old telephone exchange which requires huge manpower, extra wiring for new connection and extension is difficult to handle. This old PBX system was replaced by IPPBX system, which is the internet protocol based private branch exchange system based on voice over internet protocol, which carries voice as a data over internet. The advantage of IPPBX system is cost efficient which uses only one computer system as a server with Linux based operating system called "centos". The system uses the USB handset as an interfacing device like telephone. The clients are connected to server using hard phone and soft phones. Here the communication is possible using Wi-Fi also.

Key Words: Asterisk, SIP, VoIP, Raspberry, USB handset.

1.INTRODUCTION

A private branch exchange PBX, is telephone exchange that serve a particular business or office, as opposed to one that a common carrier or telephone company operates for many business or for the general public. PBX make connection among the internal telephones of a private organization usually a business and also connect them to the public switched telephone network i.e. PSTN via trunk lines. Because they incorporate telephone, fax machines, modems and more. It is a mini telephone exchange that connects you to the extension. The extension is connected with copper cables to the central electronics system [3]. PBX require lot of maintenance and manpower. It is less secure and less flexible. It require extra wiring for new extension which is very expensive and it does not support advance features like voicemail, call waiting, caller ID etc. The main disadvantage of PBX system is that the change of extension is very difficult. To overcome this problem the PBX is replaced by internet protocol private branch exchange i.e. IPPBX.

As the modern telephone networks begun to take shape, private companies saw a greater reliance on telephone

communication. Many decide to implement their own service so that they could handle calls internal to the organization. The drawback of PBX system is that it require huge manpower, extra wiring for new connection and the extension are very difficult to manage. so instead of using PBX system the use of internet protocol which carries voice as a data called voice over internet protocol. Voice over internet protocol deals with the conversion of analog audio signal into the digital data and using internet protocol the digital data can be transmitted over the internet. VoIP turns the standard internet connection into the phone calls. The IPPBX run on VoIP telephony which provides the organization the sophisticated installation and configuration of user extension [2]. IPPBX system has many features like call forwarding, alarms, reminders etc. These are very useful for reminding any meetings. Voicemail, IVR systems, least cost routing (LCR) etc. which not only increases efficiency and output in office, but also reduces expenses.

1.1 IPPBX System

Internet protocol based private branch exchange is based on voice over IP telephony. Communication can be done by computers, IP phone and smart phones having Wi-Fi facility. VoIP-PBX system uses local area network on which the extensions were configured. The advantage of this system is that it uses the existing installed network connection for the organization and the management of this extension is through web browser. Enterprises don't need to disrupt their current external communication infrastructure and operations. With IP PBX deployed, an enterprise can even keep its regular telephone numbers. This way, the IP PBX switches local calls over the data network inside the enterprise and allows all users to share the same external phone lines

1.2 Working Of IPPBX

An Internet Protocol Based Private Branch Exchange System consists of one or more SIP (Session Initiation Protocol) phones. The IP PBX server functions in a similar manner to a proxy server: SIP clients, being soft phone or hardware-

based phones, register with the IP PBX server, and when user wish to make a call user ask the IP PBX to establish the connection. The IP PBX has a directory of all phones/users and their corresponding SIP address and thus is able to connect an internal call. An IP PBX is a complete telephony system that provides telephone calls over IP data networks. All conversations are sent as data packets over the network. IP PBX can enable each and every employee/person in an organization to be provided with a voice extension, thereby multiplying his/ her productivity. Easier to manage the routine operations because of web/Graphical User Interface based configuration interface. It eliminates phone wiring as it uses existing network. There is no need to disrupt current external communication infrastructure and operations.

2. SYSTEM REQUIREMENT

IPPBX system uses one computer as a server with Linux based operating system called "Centos". These operating system consist of software called Asterisk which consist of telephone package .The package consist of several feathers such as caller ID, call-waiting, call hold, call transfer etc. Asterisk support session initiation protocol i.e. SIP which is the audio protocol used for audio communication. IPPBX support analog phones, soft phones and IP features phones. Soft phones are Specific application installed on user computers so that user can use the computer itself as IP Telephone to make voice calls. The VoIP PBX system for the organization use the backbone of Local Area Network on which the extensions were configured using computer system. The "CentOs" server is the Linux based and the clients were the windows based or Linux based using the "soft phone" for the communication. The VoIP telephone devices can be used instead of soft phone such as USB handset and Hard phone. The USB handset is the plug & play telephone which doesn't requires any specific driver to install. It uses the soft phone to run. The IP-PBX may use for a LAN where the outgoing VOIP calls will be send and the incoming calls will be come through the PBX system. We used only PC to PC communication for simulating the whole task. VOIP can be achieved on any data network that uses IP, like the Internet, Intranets and Local Area Networks (LAN). USB phone is a telephone handset just like ordinary landline phone set but connected to the client PC via USB. It is plug & play device i.e. device driver is automatically installed when plug-in. It only required soft phone to run.

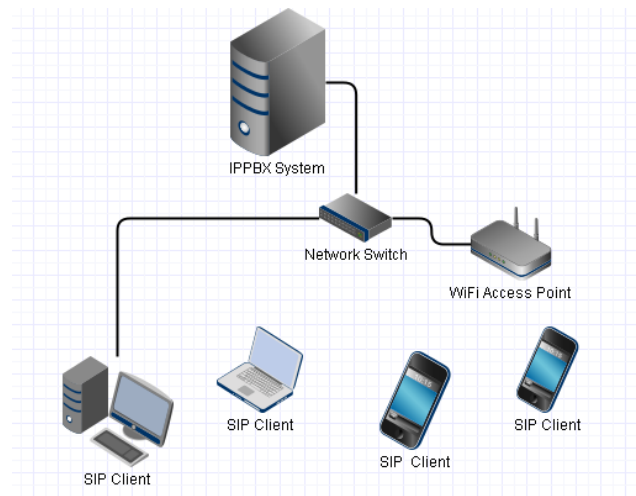


Figure 1: System Architecture

3. METHODOLOGY

3.1 Asterisk

Asterisk is basically a telephony toolkit .It is the fully open source, hybrid TDM, Packet Voice PBX and IVR Platform. It was developed by' Mark Spencer' in 1999who later founded a company called 'Digium' which is now a major provider of open source solution in telecommunication field. It enables developers to create numerous types of applications that interface with telephone networks. The most obvious application is that of a PBX. Asterisk can also be used as an IVR (Interactive Voice Response) system, for teleconferences and as a voicemail system. Asterisk is, however, most commonly used to build hybrid PBX systems that utilize modern PCI cards instead of banks of switches and relays, and software instead of custom hardware. By using relatively simple PCI cards in a standard x86 computer system running on Linux, the cost to build a working system is greatly reduced as compared to the often expensive and inflexible traditional PBX.

Asterisk PBX is just software. While different hardware connectivity components are available, all of the features and routing is done through software. This is an amazing technical breakthrough considering that even the most modern PBX systems still rely completely on proprietary hardware and electronic switches and relays, and require specialized technicians to install and maintain. The costs for a telephone engineer to work on these systems can be extremely expensive.

3.2 Voice over Internet Protocol

Voice over Internet Protocol (VoIP) is one of a family of internet technologies, communication protocols, and transmission technologies for delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet. Other terms frequently encountered and often used synonymously with VoIP are IP telephony, Internet telephony, voice over broadband (VoBB), broadband telephony, and broadband phone.

Internet telephony refers to communications services—Voice, fax, SMS, and/or voice-messaging applications—that are transported via the Internet, rather than the public switched telephone network (PSTN). The steps involved in originating a VoIP telephone call are signaling and media channel setup, digitization of the analog voice signal, encoding, packetization, and transmission as Internet Protocol (IP) packets over a packet-switched network. On the receiving side, similar steps (usually in the reverse order) such as reception of the IP packets, decoding of the packets and digital-to-analog conversion reproduce the original voice stream. Even though IP Telephony and VoIP are terms that are used interchangeably, they are actually different; IP telephony has to do with digital telephony systems that use IP protocols for voice communication while VoIP is actually a subset of IP Telephony. Voice over Internet Protocol (VoIP) is a technology for voice communication that uses the ubiquity of IP-based networks to rely VoIP client devices such as desktop IP phones called Session Initiation Protocol (SIP) phone or soft phone, in an increasing number of businesses and homes around the world because IP is the protocol connecting almost all devices. Voice over IP (VoIP) uses the Internet Protocol (IP) to transmit voice as packets over an IP based network.

VoIP that is Voice over Internet Protocol deals with the conversion of analog audio signals into the digital data that can be transmitted over the internet by using internet protocol. VoIP turns the standard network connection into the phone calls. Many organizations uses Electronic Private Branch Exchange System for the communication using extension numbers assigned to the users. It utilizes the man power and extra wiring for the installation as well as it doesn't support the advance facilities like call waiting, voicemail, caller ID etc.

3.3 Use of Raspberry Pi

The Raspberry Pi hardware has evolved through several

versions that feature variations in memory capacity, and peripheral device support. The SoC used in the first generation Raspberry Pi is somewhat equivalent to the chip used in older smart phones (such as iPhone / 3G / 3GS). The Raspberry Pi is based on the Broadcom BCM2835 system on a chip (SoC), which includes an 700 MHz ARM1176JZF-S processor, Video Core IV GPU, and RAM. It has a Level 1 cache of 16 KB and a Level 2 cache of 128 KB. The Level 2 cache is used primarily by the GPU. The SoC is stacked underneath the RAM chip, so only its edge is visible.

Raspberry Pi 2 is based on Broadcom BCM2836 SoC, which includes a quad-core Cortex-A7 CPU running at 900 MHz and 1 GB RAM. It is described as 4–6 times more powerful than its predecessor. The GPU is identical to the original.

4. ANALYSIS

Figure 2 shows the Voice data packets analysis of SIP which is 5.4% of the total data speed of the IPPBX. This is for four user analysis in call.

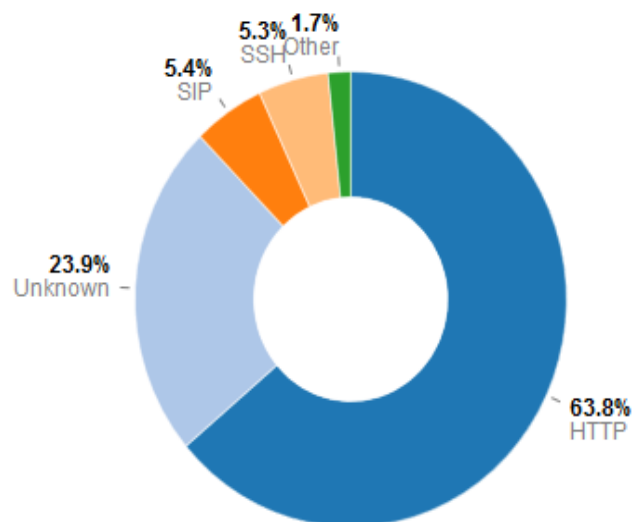


Figure 2 SIP data analysis

192.168.1.2	UDP	Source port: 19000 Destination port: sip
Broadcast	ARP	Who has 192.168.1.1? Tell 192.168.1.4
192.168.1.5	UDP	Source port: 19088 Destination port: sip
192.168.1.5	UDP	Source port: 60227 Destination port: sip
Broadcast	ARP	Who has 192.168.1.2? Tell 192.168.1.1
192.168.1.5	SIP	Request: REGISTER sip:192.168.1.5
192.168.1.2	SIP	Status: 401 Unauthorized (0 bindings)
192.168.1.5	SIP	Request: REGISTER sip:192.168.1.5
192.168.1.2	SIP	Request: OPTIONS sip:6003@192.168.1.2:60227;ob

Figure 3: Client IP for SIP

Data packets send and received by the client 192.168.1.5 which is UDP and using protocol SIP from the Asterisk server

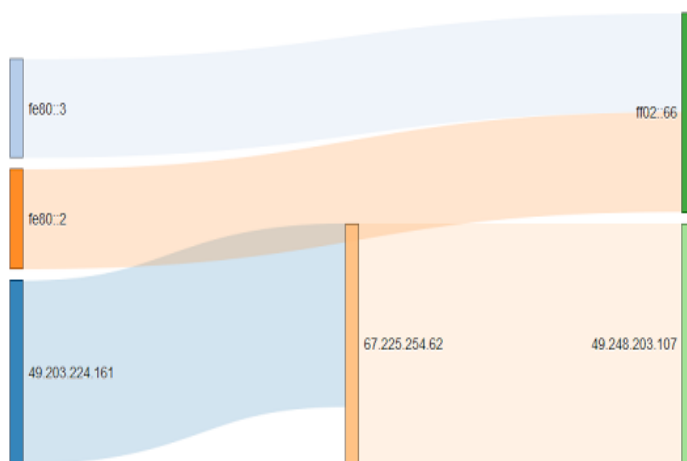


Figure 4: Voice data packet flow

In the above figure 4 the voice data packet transmission from Source to destination via Asterisk SIP server is shown. Here the Source IP address or two user calling each other IP address are 49.203.224.161 and 49.248.203.107 the Asterisk server IP address is 67.225.254.62.

4. CONCLUSIONS

The IPPBX run on VoIP telephony which provides many facilities to the organization. The sophisticated installation and configuration of user extension [2].IPPBX system has many features like call forwarding, alarms, reminders etc.

These are very useful as a reminder for any meetings, Voicemail, IVR systems, least cost routing (LCR) etc. which not only increases efficiency and output in an organization but also reduces operating expenses along with facility to modify and change the existing setup.

This work describes one solution for a local PBX based on existing LAN hardware infrastructure. It is a low-cost and modular solution that completely meets its basic function, transporting of voice packets over IP networks. Since most of the processing is performed on a PC it provides a number of additional features.

The main advantage of the system, that it reduces the wiring cost as EAPBX system. The extensions can be easily created, deleted or shifted without disturbing the other communication. We can configure the system from any computer in the network using internet browser.

REFERENCES

[1] Md. Zaidul Alam, Saugata Bose, Md. Mhafuzur Rahman, Mohammad Abdullah Al-Mumin "Small Office PBX Using Voice Over Internet Protocol" in ICACT 2007. [2] BUR GOODE, SENIOR MEMBER, IEEE "Voice Over Internet Protocol (VoIP)" in PROCEEDINGS OF THE IEEE, VOL. 90, NO. 9, SEPTEMBER 2002 [3] J. Rosenberg, H. Schulzrinne, Camarillo, Johnston, Peterson Sparks, Handley, and Schooler, "SIP: Session initiation protocol v.2.0," IETF RFC 3261, 2002 [4] "A CRM model based on Voice over IP". Y.S. Moon, C.C. Leung, K.N. Yuen', H.C. Ho, X. Yu Department of Computer Science and Engineering, Dept. of System Engin. and Engin. Management, The Chinese University of Hong Kong Shatin, N.T., Hong Kong. IEEE 2000 [5] "SMALL OFFICE PBX USING VOICE OVER INTERNET PROTOCOL (VOIP)" Md. Zaidul Alam, Saugata Bose, Md. Mhafuzur Rahman, Mohammad Abdullah Al-Mumin ICACT 2007 [6] "Conferencing, Paging, Voice Mailing via Asterisk EPBX" Ale Imran1, Mohammed A Qadeer2, 2009 International Conference on Computer Engineering and Technology

[7] "trixbox made easy" Barrie Dempster and Kerry Garrison , Packet Publication 2006

[8] Mr.Sonaskar,S.D.Giripunje,"Low Cost IP Private Branch Exchange(PBX)",International Journal Of Computer Application ,Volume23-no.3,June 2011.

[9] Ms.Harshada Jagtap,Prof.D.G.Gahane,"Asterisk Based IP-PBX Cost Efficient Server for Small Organization",International Journal On Recent and Innovation Trends in Computing and Communication,vol.3issue.2,February 2015

[10]NdaohialyMandaVyRavonimanantsoa,AndryAuguste and MalaladianaHajaso , "The impact of calls to the Processor on a server PABX",International Journal Of Future Computer And Communication,Vol.1,No.1,June 2012

[11]Mr.Mukunda.Ghogale,Dr.Prashand V.Ingole,"IP-PBX:Architecture and Protocols" IORD Journal Of Science And Technology E-ISSN:2348-0831 Vol.1,Issue 4,Sept-Oct 2014

[12]G.Kavitha,K.T.KanyaKumari,D.R.SanthoshKumar, "Internet Protocol Private Branch Exchange",International Journal Of Recent Trends in Engineering,Vol1,no.2,May 2009