Project Synopsis

on

**Cloud Based Phone System : VoIP**

Submitted as a part of course curriculum for

**Bachelor of Technology**

in

**Computer Science**



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**2022-2023**

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**DECLARATION**

We hereby declare that this submission is our work and that, to the best of our knowledge and belief, it contains no material previously published or written by another person nor material which to a substantial extent has been accepted for the award of any other degree or diploma of the university or other institute of higher learning, except where due acknowledgement has been made in the text.

Signature of Students:

Name:

Roll No.:

Date:

**CERTIFICATE**

This is to certify that Project Report entitled “**Cloud Based Phone System**” which is submitted by **Rupesh Kumar, Vinay Kumar and Yash Surya** in partial fulfilment of the requirement for the award of degree B. Tech. in Department of Computer Science of Dr A.P.J. Abdul Kalam Technical University, Lucknow is a record of the candidates own work carried out by them under my supervision. The matter embodied in this report is original and has not been submitted for the award of any other degree.

**Date: Supervisor**

**Signature:**

MS. AKANKSHA

(Assistant Professor)

**ACKNOWLEDGEMENT**

It gives us an enthusiastic sense of pleasure to present the synopsis of the B. Tech Mini Project undertaken during B.Tech. Third Year. We owe a special debt of gratitude to **MS. AKANKSHA Assistant Professor**, Department of Computer Science, KIET Group of Institutions, Delhi- NCR, Ghaziabad, for her constant support and guidance throughout the course of our work. Her sincerity, thoroughness and perseverance have been a constant source of inspiration for us. It is only her cognizant efforts that our endeavours have understood something clearly at last of the day.

We also take the opportunity to acknowledge the contribution of Dr. Ajay Kumar Shrivastava, Head of the Department of Computer Science, KIET Group of Institutions, Delhi- NCR, Ghaziabad, for his full support and assistance during the development of the project. We also do not like to miss the opportunity to acknowledge the contribution of all the faculty members of the department for their kind assistance and cooperation during the development of our project.

Last but not the least, we acknowledge our friends for their contribution to the completion of the project.

Signature:

Date:

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Roll No:

**ABSTRACT**

Currently almost all organizations having LAN(Local Area Network) they make use of PBX (private branch exchange) to make within the organization communication system

with the help of Asterisk Server. Separate telephone network of your voice communication service provider for making calls is required which increases the cost of the communication system. IP telephony uses your existing LAN for communication within your organization and the Internet outside your organization. IP refers to the network layer protocol with many supporting protocols such as ICMP, HTTP, FTP, ARP, RARP, and the SIP protocol used by IP telephony. Users of IP phone need to register to asterisk which is software implementation of PBX and connects to the desired user. This system is implemented using a cloud-based Asterisk server which is core of this project and also uses SIP protocol to initiate and terminate the phone calls by making the system cost effective, scalable, and flexible is higher.

**LIST OF FIGURES**

1. **Flow chart for outgoing call**
2. **Flowchart for incoming calls**
3. **shows the Dial plans**
4. **shows the extensions.conf file**
5. **shows the number of Online and Offline devices connected with our server.**
6. **shows the dialer and call receiver's screen**

**LIST OF ABBREVIATIONS**

1. VoIP : Voice over Internet Protocol
2. SIP : Session Initiation Protocol
3. LAN : Local Area Network
4. ICMP : Internet Control Message Protocol
5. HTTP : Hypertext Transfer Protocol
6. API : Application Programming Interface
7. PSTN : Public Switched Telephone Network
8. IP : Internet Protocol
9. PBX : Private Branch Exchange
10. EPBX : Electronic Private Automatic Branch Exchange
11. ARP : Address Resolution Protocol
12. RARP : Reverse Address Resolution Protocol
13. **INTRODUCTION**
    1. **Introduction**

Voice over Internet Protocol (or "VoIP") technology converts voice calls from analog to digital and transmits them over digital data networks. Voice over IP (VoIP) defines how voice calls are routed over IP networks, including digitizing, and packetizing the voice stream. IP Telephony leverages the VoIP standard to create a telephony system that allows for advanced call routing, voicemail, contact centers, and other high-level features.

Voice over IP is revolutionizing the world of communications. This is used for making and receiving calls over the Internet or IP networks much cheaper than traditional fixed telephone networks. Also, many improved features and expanded possibilities make your communication experience richer and more beautiful.

The equipment used in PBX depends on the complexity of the system. For example, is it a traditional PBX connected to a copper landline, does the PBX support a mix of analog and digital lines, does it support voice over, uses IP (VoIP) whether or not Whether it is company-hosted or cloud-based PBX system.

VoIP calls do not use the phone network. They route calls via the internet. To send voice across the internet, the voice information is coded into a digital format and transmitted in packets of information in the form of data. The data packets are then sent across the internet and reassembled into sound at the other end for the receiver to hear.

* 1. **Problem Statement**

Nowadays the EPBX system is used in many business organizations, Hotels, and Institutions etc. for voice communication.

The EPBX (Electronic Private Automatic Branch Exchange) is used for communication.

Now we are replacing the EPBX system with Cloud based IP-PBX phone system for the use of voice communication in business enterprises, Educational Institutions etc.

The benefit of using cloud-based phone system is –

* It does not require a lot of hardware.
* Work with different devices like Android, Windows, Linux and VoIP or IP based phones.
* It easily supports remote work.
* It keeps logs into the cloud server.
* New services and features are added easily.

**Higher Cot**

* 1. **Objective**
* **Lower Costs of Service :** It lower the cost of the hardware because we are using cloud server to communicate. The server is replacing the phones that are connected with the wires in the institution.
* **Cloud Implementation:** By adding our phone system to cloud server it will not only lower the hardware cost also it will increase area of communication because it uses the Ubuntu Server so we can access Currently almost all organizations having LAN (Local Area Network) they make use of PBX (private branch exchange) to make within the organization communication system with the help of Asterisk Server. Separate telephone network of your

voice communication service provider for making calls is required which

increases the cost of the communication system.

* Allowing Users to be able to communicate effectively, speedily, and most importantly, securely there by enhancing the privacy and confidentiality of mobile communication.
* **Monitor Calls:** We can also monitor the calls and analyze them to find strong and weak communication points.
  1. **Scope**

According to a recent survey, nearly 90% of large enterprises have switched to VoIP technology. The VoIP revenues worldwide have increased significantly from $13 billion (about $40 per person in the US) in 2002 to $197 billion (about $610 per person in the US) (about $610 per person in the US) in 2007. In terms of usage, we have experienced 8.3 billion minutes (about 15780 and a half years) in 2002 and 823 billion minutes (about 1,600,000 years) in 2007. VoIP offers free calling over the Internet at as low as $0.09 per minute over the network, making it an excellent choice for businesses and business organizations including government agencies, non-governmental organizations, and small growth companies. have provided. With a lot of research and practice, VoIP has consistently improved voice quality. Increased speed and bandwidth and the proliferation of high-speed networks such as LTE and upcoming 5G have improved Voice over Internet protocols. Ensuring the best voice quality can truly eliminate communication barriers in your business. Cost and quality have drawn the capital of business organizations.

1. **LITERATURE REVIEW**

In [1] , Dr. H D Phaneendra etal. Mrs. Sowmya C T states that IP telephony services are more economical than PSTN and wired PBX methods. Instead of using the traditional PSTN and PBX method, replace the PBX with an asterisk and have IP phone calls routed to the Raspberry Pi via LAN port . This introduces an inexpensive solution for connecting the to its intended users. Costs include hardware requirements, training costs in excess of phone service costs. The goal is to make calls over the Internet, and an intranet can be set up at your company. Advanced features such as call forwarding, sending messages to your personal mailbox. Future Prospects: Video calls can be made by connecting a camera. You can call forward when the person is unavailable and forward the message to the person's mailbox.

In [2], Author H. Nadella etal. C. Nagamalla, N. B. Sai Shibu, D. Arjun and S. N. Rao states that A VoIP server was configured on a Raspberry Pi and a enabled a cost-effective maritime communication system for fishermen. The VoIP service uses the commercial Ocean Net network to provide communication services. The system has been validated in a laboratory setup and proven to be efficient and reliable. In the future, he plans to test the service to sea fishermen in identified villages and develop his prototype of a VoIP client that will enable a standalone system for communications.

In [3],Author Meshram etal. Megha S., Pooja Thakare, and Pranali Dandekar States that e IPPBX system utilizes organizational features such as call routing, music on hold, voicemail, and audio conferencing. Reduce cabling costs by using your existing network infrastructure. Setting up a softphone is easy. It is also flexible not only for users but also for administrators. Creating and removing extensions is easy. The nature of communication in a global world is an area of growing interest as people increasingly become relays on the Internet. Voice over Internet Protocol (VOIP) is this case and has become the most useful technology for long distance communication. VoIP is a rapidly growing technology on IP networks and is a time sensitive application, requiring real-time support. His VoIP over IP network is designed for data communication, but achieving reliable, high-quality voice over IP networks is a technical challenge. Designing a high-quality his VoIP implementation on the Asterisk PBX system requires choosing the best codecs and applying perfect techniques.

In [4], M. A. Qadeer etal. A. Imranauthor states that The success of any business is directly related to customer satisfaction, but the most important factor in customer satisfaction these days is excellent customer service. Unfortunately, in our country this service is largely neglected, especially by state-owned enterprises. This white paper has shown how to provide responsible customer service using an open source IP PBX solution. It also shows how such systems can facilitate intelligent business decisions, thereby improving service quality and customer satisfaction. Therefore, it should prove sufficient to justify and persuade our companies to adopt and deploy Asterisk-based IP PBX systems in order to provide responsible customer service.

In [5], author Mohammad Azam Khan Khaled Mahmud Shahriar etal. states

The success of any business is directly related to customer satisfaction, but the most important factor in customer satisfaction these days is excellent customer service. Unfortunately, in our country this service is largely neglected, especially by state-owned enterprises. This white paper has shown how to provide responsible customer service using an open source IP PBX solution. It also shows how such systems can facilitate intelligent business decisions, thereby improving service quality and customer satisfaction. Therefore, it is expected to prove sufficient to justify and persuade our nation's businesses to adopt and deploy Asterisk-based IP PBX systems in order to provide responsible customer service.

In [6], Kurniawati, Nazmia, et al. states that In this study, we deploy a VoIP system using a Raspberry Pi and Kamailio SIP server operating in an ad-hoc network using the OLSR routing protocol. Experiments show that the network performance is satisfactory when compared with the Indonesian Ministry of Communication and Information standards. Considering the experimental results, it is possible to use the Raspberry Pi as an indoor node in his VoIP system in rural areas to provide optimal coverage in residential areas.

In [7], author Muntaka, Siddique Abubakr, Faiza Hussein, and Paul Sarfo. et al.T states that he research project aimed at creating a communication system to enhance campus communication. Testing the system reveals that communication over the wireless network was successful with relatively some appreciable level of delay as the number of client connected to the Asterisk PBX Server increases. The delay was understandable since the prototype used Raspberry pi which comes with a limited computing resources. The call quality was however very good. The researchers further enhanced the setup by integrating the system with public switch telephone networks in Ghana like Vodafone, MTN, Tigo, Airtel, etc. This was achieved with Huawei USB Modern. This research project was very difficult, especially since the system is Linux-based. The main objective was achieved and the results were satisfactory.

1. **PROPOSED METHODOLOGY**

We are making the IP-PBX cloud-based phone system using Asterisk Server which is an open-source tool used for building real-time communications applications and can make real time call routing decisions. By using asterisk server, we can also build the simple office network with few phones.

The asterisk server can be installed in any Linux Distributions like Ubuntu OS, Kali Linux, and Parrot OS. But we are using the Ubuntu 20.02 Server to install the asterisk server and we are using the cloud server provider that provide the Linux based cloud servers**.**

1. **Flowcharts**

**Diagram

Description automatically generated**

Figure 1 flow chart for outgoing call

Diagram

Description automatically generated

Figure 2 Flowchart for incoming calls

1. **TECHNOLOGY USED**

**VoIP**

A VoIP phone is a hardware - or software-based telephone designed to use voice over Internet Protocol (VoIP) technology to send and receive phone calls over an IP network.

**Asterisk Server**

Asterisk is an open-source framework for building communications applications. Asterisk turns an ordinary computer into a communications server. Asterisk powers IP PBX systems, VoIP gateways, conference servers and other custom solutions.

**Ubuntu Server**

Ubuntu Server is a server operating system, developed by Canonical and open-source programmers around the world, that works with nearly any hardware or virtualization platform.

**SIP**

SIP stands for Session Initiation Protocol. The Session Initiation Protocol (SIP) is a signaling protocol used for initiating, maintaining, and terminating communication sessions that include voice, video, and messaging applications.

1. **DIAGRAMS**

Text

Description automatically generated

Figure 3 shows the Dial plans

A screenshot of a computer

Description automatically generated with medium confidence

Figure 4 shows the extensions.conf file

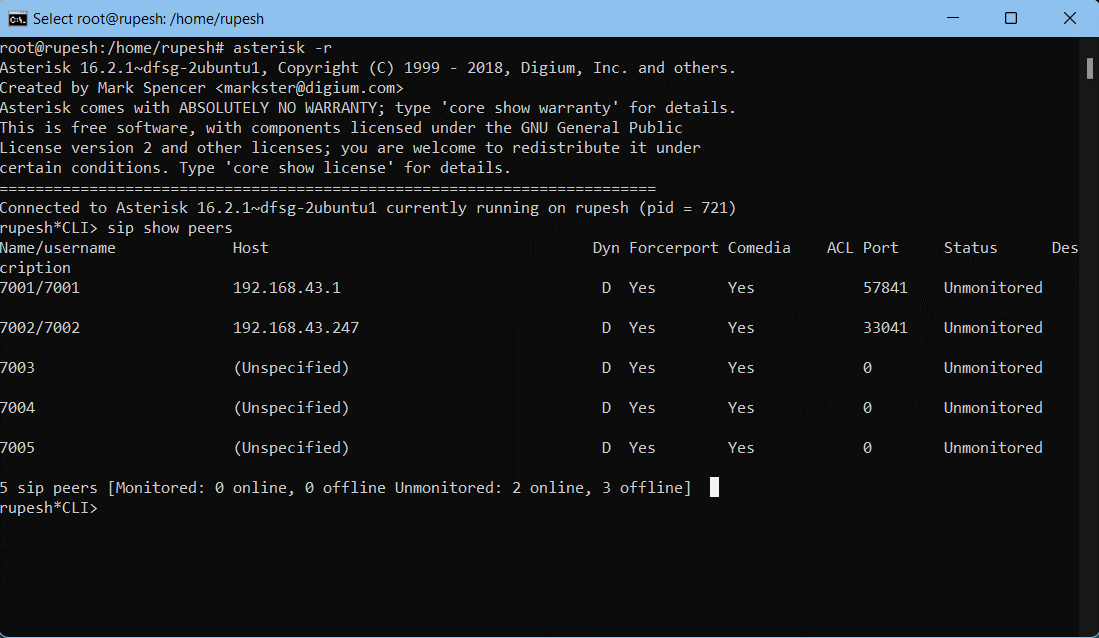


Figure 5 shows the number of Online and Offline devices connected with our server.

Graphical user interface, application

Description automatically generated

Figure 5 shows the dialer and call receiver's screen

1. **CONCLUSION**

This research project aimed to create communication System, to enhance the voice communication inside any organization. The previous system uses Raspberry Pi device in different research papers to implement the Asterisk PBX system. That system causes a sort of delay Because the system Used the Raspberry pi which comes with a limited computing Resources. But now we are replacing the Raspberry Pi device with the Cloud Server because in cloud we have a suitable number of resources available.

The research project was particularly challenging, especially because the system is Linux based. The result was satisfying since the main objectives was realized.

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