DEPLOYMENT AND TESTING OF OPENSOURCE IP-PBX SOLUTION FOR REALTIME COMMUNICATION

INT 400 – INTERNSHIP 3

PROJECT REPORT

Submitted by

KEERTHANA S – E0322005

In partial fulfilment for the award of the degree of

BACHELOR OF TECHNOLOGY

in

COMPUTER SCIENCE AND ENGINEERING

(Artificial Intelligence and Data Analytics)

Sri Ramachandra Faculty of Engineering and Technology

Sri Ramachandra Institute of Higher Education and Research, Porur, Chennai - 600116

OCTOBER 2024

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(Category - I Deemed to be University) Porur, Chennai

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BONAFIDE CERTIFICATE

Certified that this project report **“DEPLOYMENT AND TESTING OF OPENSOURCE IP- PBX SOLUTION FOR REALTIME COMMUNICATION”** is the bonafide record of work done by **“KEERTHANA S – E0322005”** who carried out the internship work under my supervision.

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ABSTRACT

This project explores the deployment and testing of an open-source IP PBX solution designed for real-time communication, offering a cost-effective alternative to traditional telephony systems. It involves the installation, configuration, and performance evaluation of the IP PBX system in a simulated business environment. Key features such as VoIP, call routing, voicemail, and conferencing are implemented to assess the system's capability. The deployment process details the integration of IP phones and gateways, while performance testing examines metrics like call quality, latency, and system reliability under various traffic loads.

The testing phase validates the system's ability to handle multiple concurrent users, maintaining high call quality and low latency. The results show that open-source IP PBX platforms are viable for small to medium-sized businesses, offering flexibility, scalability, and significant cost savings compared to proprietary alternatives. Additionally, security measures such as encryption and firewall configurations are evaluated to safeguard communication channels. This project focuses on the deployment and testing of an open-source IP-PBX (Internet Protocol Private Branch Exchange) solution aimed at enhancing real-time communication systems. IP-PBX integrates VoIP (Voice over IP) technology, enabling seamless voice, video, and messaging communication over the internet. The deployment process involves setting up the PBX system, configuring extensions, trunks, and dial plans to support various communication protocols. Testing is conducted to ensure system reliability, call quality, and scalability under different network conditions. This solution provides a cost-effective, scalable alternative to traditional PBX systems, suitable for both small businesses and large enterprises.

Keywords: IP-PBX Deployment, Real-time Communication, Open-source VoIP, Network ScalabilityTesting

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CHAPTER 1

INTRODUCTION

1.1 INTRODUCTION TO IPX-SYSTEM

In the modern digital era, the demand for effective, scalable, and reliable communication systems is growing rapidly, particularly in the field of computer networks. As organizations expand and adapt, they require advanced solutions capable of handling complex real-time interactions while maintaining high performance, security, and reliability. This shift has driven the adoption of IP-based communication systems, which provide greater flexibility and integration compared to traditional telecommunication methods. The project, titled **"Deployment and Testing of Open-source IP-PBX Solution for Real-time Communication,"** aims to implement and thoroughly assess an IP-PBX (Private Branch Exchange) system that operates over Internet Protocol (IP) networks.

IP-PBX systems are transforming how businesses manage their communication infrastructure. Unlike conventional phone systems that rely on dedicated hardware and circuit-switched technology, IP-PBXutilizes an organization's IP network to manage voice, video, and messaging services. By leveraging the same network infrastructure used for data transmission, IP-PBX systems allow for the convergence of voice and data traffic, reducing operational costs, improving scalability, and simplifying management. However, the successful deployment of an IP-PBX system requires meticulous planning and extensive testing to ensure that it can handle real-time communication demands, while maintaining minimal delays and ensuring optimal performance.

This project focuses on deploying an open-source IP-PBX system, which offers a flexible andbudget- friendly alternative to proprietary systems. Open-source platforms provide the advantage ofcustomization, allowing organizations to modify and adapt the system according to their specific needs, without the burden of licensing fees or vendor constraints. The primary objective of this project is to showcase the successful deployment of an open-source IP-PBX system that facilitates internal and external communications in a real- world environment. Additionally, the project will explore how well the solution integrates with existing network infrastructure, including legacytelephony systems, VoIP (Voice over IP) devices.

The testing phase will be an essential part of this project, involving a thorough evaluation of the IP- PBX solution in various operational conditions. The project will conduct performance tests to assess voice and video quality in real-time communications, security tests to ensure the system is resistant to potential cyber threats (such as DoS attacks or call interception), and scalability tests to examine how the system performs as the number of users and call volumes increase. Furthermore, the project will evaluate the system's compatibility with various VoIP protocols, such as SIP (Session Initiation Protocol), to ensure it works effectively with a broad range of communication devices and software.

Key challenges to address in real-time communications, essential for minimizing issues like latency, packet loss that can deteriorate call quality. Additionally, the project will focus on ensuring the system's high availability, incorporating failover and redundancy features that maintain service continuity during network failures or hardware malfunctions.

The outcome of this project will offer important insights into the practical implementation of open- source IP- PBX systems and their efficiency in supporting real-time communication across computer networks. The findings will be especially useful for small and medium-sized businesses looking for cost-effective communication systems that deliver robust performance and flexibility. The project will conclude by offering best practices for optimizing the deployment of IP-PBX solutions and recommendations on ensuring their long-term reliability, security, and scalability.

In essence, this project not only explores the technical aspects of deploying an open-source IP-PBX system but also delves into its broader impact on communication infrastructures, making it a significant study in real- time communication over IP networks.

Keywords: IP-PBX Deployment, Real-time Communication, Open-source VoIP, NetworkScalabilityTesting

1.2 TECHNOLOGY INVOLVED

VMware

VMware is a powerful virtualization platform that allows the creation of virtual machines (VMs) on a single physical server, reducing the need for multiple hardware units. It provides an environment for testing and deploying various operating systems and applications without hardware limitations. In this project, VMware is used to host the CentOS server running FreePBX.

VMware is a global leader in cloud computing and virtualization technology, offering software solutions that enable organizations to efficiently manage their IT infrastructure. At the core of VMware's technology is virtualization, which allows a single physical machine to run multiple virtual machines (VMs) simultaneously, optimizing hardware utilization. VMware's are designed to manage these virtual environments by allocating resources like CPU, memory, and storage across the VMs.

FreePBX

FreePBX is an open-source, web-based GUI that controls and manages Asterisk, the most widely used IP telephony engine. It provides features like call routing, voicemail, conferencing, and IVR, making it a comprehensive PBX solution. FreePBX is highly customizable and integrates seamlessly with VoIP services, allowing users to configure telephony features according to their business needs.

Being open-source, FreePBX is highly cost-effective, eliminating licensing fees and allowing businesses to customize the system according to their specific needs. It supports additional modules, such as call recording and CRM integration, which enhance its functionality. FreePBX can be deployed both on- premises and in the cloud, making it suitable for businesses of all sizes.

CentOS

CentOS is a free, enterprise-class Linux distribution derived from Red Hat Enterprise Linux (RHEL). It is widely used for server environments due to its stability, security, and long-term support. In this deployment, CentOS serves as the operating system for hosting FreePBX, offering a robust and secure foundation for running the IP PBX system. CentOS allows users to freely access, modify, and distribute the software, embodying the principles of open-source development.

The vibrant and active CentOS community contributes valuable documentation and support, which, although not commercial, is often sufficient for many users. Its versatility makes it suitable for a wide array of applications, ranging from web servers to cloud computing platforms, allowing it to fit various operational environments.

WinSCP

WinSCP is a free, open-source file transfer tool that supports FTP, SFTP, and SCP protocols. It enables secure file transfers between a local machine and remote servers, making it useful for managing files on the CentOS server where FreePBX is installed. It features a user-friendly graphical interface that facilitates secure file transfers between a local computer and remote servers, making it a favored choice among web developers and system administrators.

The interface is designed to resemble Windows Explorer, which makes it easy to navigate both local and remote file systems. Users can efficiently transfer files using drag-and-drop functionality. Additionally, WinSCP includes a built-in text editor, allowing quick modifications to remote files without the need to download them first.

Wireshark

Wireshark is a network protocol analyzer that captures and analyzes network traffic in real time. It helps in diagnosing network issues, monitoring VoIP communications, and ensuring data security by inspecting packet-level details. Wireshark is essential for troubleshooting communicationproblems and optimizing network performance. With its ability to support thousands of protocols, Wireshark can decode and analyze a wide array of data formats, making it essential for understanding the interactions within a network.

The intuitive graphical user interface (GUI) displays captured packets in a structured manner, allowing users to view detailed information about each packet, including headers and payloads. Wireshark offers robust filtering capabilities, enabling users to isolate specific packets based on various criteria, such as IP addresses and protocols. It also provides protocol analysis features that allow users to drill down into packet details, helping to identify anomalies and troubleshoot network issues effectively.

Soft Phones

Zoiper is a multi-platform softphone that supports VoIP, enabling users to make and receive calls via their IP PBX system. It is used in this project to test and validate VoIP communication, 3CX is utilized alongside Zoiper to provide additional testing options and enhance the user experience. Zoiper is a versatile VoIP softphone that supports SIP and IAX protocols, making it easy to connect with various VoIP providers and PBX systems. It offers features like call hold, transfer, voicemail, and conferencing.

3CX is a software-based PBX providing unified communications solutions, integrating voice, video, and messaging. It can be hosted on-premises or in the cloud, with advanced featureslike call queuing, IVR, and CRM integration.

PuTTY:

PuTTY is a terminal emulator that allows secure remote access to servers via SSH, Telnet, and other protocols. It is used to manage and configure the CentOS server hosting FreePBX from a local machine. PuTTY is critical for performing command-line operations, system updates, and troubleshooting in real- time during deployment. PuTTY facilitates secure file transfers through its companion application, PSCP, and supports public key authentication for enhanced security. Additionally, it can log session activities for auditing and troubleshooting.

It is a popular open-source terminal emulator that allows users to connect securely to remote servers and network devices using protocols like SSH, Telnet, and SCP. Primarily designed for Windows, it also supports other operating systems, making it versatile for network administration.

CHAPTER 2

LITERATURE REVIEW

The literature review chapter provides a comprehensive analysis of existing research on the deployment of IP-PBX systems in business communication. This chapter explores key studies that discuss the technological advancements, benefits, and challenges associated with IP-PBX, highlighting its role in replacing traditional telephony systems. The review covers critical areas such as cost- effectiveness, scalability, enhanced collaboration features, and the growing adoption of cloud-based solutions.

Mr.Mohammad Azam Khan concludes that [1]"Success of any enterprise is directly connected to customer satisfaction and the factor that greatly influences customer satisfaction now-a-days is good customer support service. Unfortunately such service is largely neglected in our country, particularly by government enterprises. In this paper we have shown how accountable customer service can be provided using open source IP-PBX solution. We also demonstrate how such a system can facilitate intelligent business decision to improve service quality and customer satisfaction thereby. Thus we expect that this should prove sufficient to justify and convince enterprises of our country to adopt and deploy Asterisk based IP-PBX system to provide accountable customer support service"

Bilal Muhammad Khan et al [2] examines the evolution and rising popularity of IP PBX systems due to their cost- effectiveness, scalability, and hardware efficiency. It details the implementation of an IP PBX on a Raspberry Pi 3 Model B, capable of handling a limited number of calls. Load testing using the SIP tool assessed performance metrics, revealing that CPU load stayed under 15% for most codecs at 100 simultaneous calls, while iLBC significantly increased after 40 calls. Memory utilization approached 95% under high load, causing instability and increased response times. Round-trip time (RTT) was highest for iLBC and G722, with H264 showing the lowest RTT. Disk utilization remained stable throughout the tests. The study concludes that an IP PBX on a Raspberry Pi 3 is suitable for small-scale applications, such as home use or small offices, particularly with the right codec for low traffic handling.

Mr .F.Abid et al concludes that [3] this paper deals with an embedded IP-PBX (IP-Private Branch exchange) /VoIP (Voice over IP) gateway, around the OpenRISC processor and an open source Asterisk PBX running on an embedded Linux kernel. In this work we leverage the Opencores reuse strategy, which provides a lower cost solution to realize a portable SoC (System on Chip) for VoIP application. In this approach both hardware and software components are Opensources. The achieved part of the system has been implemented on a Virtex5 FPGA circuit. The hardware part uses 50% of BRAM memories, 27% of slice registers and 48% of slice LUTs, the power consumption is 4.233 W. Regarding the software part, a Linux kernel has been ported to the Virtex5 FPGA, and the network functionality of the system is successfully tested.

Pradeep Mullangi states that [4] Static code analysis proved to be a potent and efficient method of checking code for vulnerabilities. Many Security flaws were found in the Asterisk 1.8 source code, and all five open- source static analysis tools and the commercial programme LDRA found them. In contrast to open- source alternatives, the commercial programme LDRA found more Security vulnerabilities and provided a more thorough analysis report. In addition, LDRA encourages a wide variety of programming languages and security standards, while most open-source tools only support a single language and a single security standard (such as C, C++, PHP, JAVA, ADA, or NETRINO). As a consequence of this effort, we can now use Static code analysis to examine a particular C/C++ source file.

Nguyen Tai Hung et al states that [5] This article outlines an experimental approach to implementing PBX-like telephony supplementary services using IMS (IP Multimedia Subsystem) technology, focusing on the example of call forwarding when busy. This approach offers several advantages, including a seamless transition from traditional PBX services in enterprises to an all-IP operator environment and demonstrating that IMS operators can provide legacy telephony supplementary services to corporate clients. Currently, the implementation only covers call forwarding upon busy as a proof of concept. Plans include expanding to a full suite of PBX-like supplementary services on the Telephony Application Server (TAS).

CHAPTER 3
PROPOSED METHODOLOGY

The proposed methodology chapter outlines the approach for deploying and evaluating an IP-PBX system within a business environment. This section details the research design, including the selection of appropriate tools and technologies, and the steps involved in the implementation process.It covers both hardware and software requirements, such as server configuration, network infrastructure, and IP phones. A key focus is on the testing phase, where performance metrics like callquality, system reliability, and security are measured.

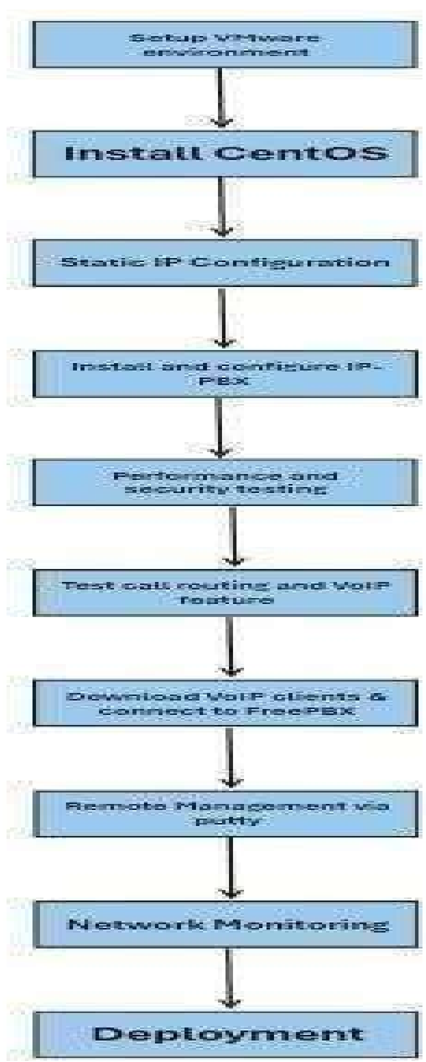

Fig.3.1.Methodology
[The set up of the whole project]

Environment Setup

Install VMware: Set up a virtual environment to host the CentOS VM.
Install CentOS: Deploy the CentOS operating system on the VMware virtual machine.
The environment setup for deploying the open-source IP-PBX solution involves configuring the required hardware and software components. This includes installing acompatible Linux-based operating system, setting up a virtual machine or dedicated server, and ensuring network connectivity with adequate bandwidth for VoIP traffic. Necessary software dependencies, such as a database server and web interface, are installed alongside the IP-PBX software. Security configurations, including firewall settings and encryption protocols, are implemented to protect the system during testing and deployment.

Static IP Configuration

Configure Static IP: Assign a static IP address to the CentOS VM to ensure consistent network connectivity.
Configuring a static IP address is crucial for the stability and accessibility of the IP-PBX server within the network. This process involves assigning a fixed IP address to the server to ensure consistent communication with other network devices, such as phones, routers, and gateways. The static IP is configured directly in the server's network settings or through the router's DHCP reservationfeature. This ensures that the IP-PBX server remains reachable for VoIP services,preventing connection drops and ensuring reliable communication flow.

Installation of IP-PBX Solution

Download FreePBX: Obtain the FreePBX installation package.
Install FreePBX: Follow the installation instructions to set up FreePBX on CentOSVM. The installation of the IP-PBX solution begins with selecting a suitable open-source platform, such as Asterisk, FreePBX, or similar, based on feature requirements and community support. The chosen software is downloaded and installed on the configured server, ensuring compatibility with the operating system.

This process includes setting up necessary modules for call handling, voicemail, SIP (Session Initiation Protocol) trunking,and user management. Configuration files are edited to customize settings like dialing plans, extension numbers, and network parameters.

Tool Installation

Install WinSCP: Use WinSCP for file transfers between your local machine and the CentOS VM.

Install Wireshark: Set up Wireshark on your local machine for network traffic analysis. Tool installation involves setting up various software utilities required for the management, monitoring, and testing of the IP-PBX system. These tools include network analyzers like Wireshark for monitoring SIP traffic, bandwidth management tools to ensure optimal VoIP performance, and softphone clients for testing call functionalities. Additionally, firewall management tools are installed to configure and monitor security settings, while logging tools are set up for tracking system performance and diagnosing potential issues.

Download VoIP Clients

Download 3CX: Obtain the 3CX softphone client for managing calls.

Download Zoiper: Install the Zoiper softphone for additional VoIP functionalities.

VoIP clients are essential for testing and utilizing the IP-PBX system, as they serve as endpoints for making and receiving calls. Various softphone applications, such as Zoiper, MicroSIP, or Linphone, are downloaded and installed on desktop and mobile devices to simulate different user environments.

Integration and Configuration

Configure FreePBX: Set up extensions, SIP trunks, and user accounts within FreePBX. Connect Softphones: Configure 3CX and Zoiper to connect to the FreePBX system.

Integration and configuration involve connecting the open-source IP-PBX system with various network components and setting up the necessary communication protocols. This includes registering SIP trunks with service providers to enable external calling, configuring internal extensions for users, and setting up call routing rules. The process also involves integrating the IP-PBX with existing IT infrastructure.

Network Monitoring

Use Wireshark: Monitor network traffic to troubleshoot any connectivity issues and ensure call quality.

Network monitoring is essential for ensuring the stability and performance of the IP-PBX system within the network. This process involves using tools like Wireshark,

Nagios, or Zabbix to monitor VoIP traffic, bandwidth usage, and latency, whichcan directly affect call quality. Monitoring helps in identifying packet loss, jitter, or any network congestion that may impact real-time communication. Alerts and logs are configured to detect any abnormal activities or potential security threats, such as unauthorized access attempts.

Testing and Validation

Perform Functional Testing: Test basic functionalities such as making and receiving callsusing 3CX and Zoiper.

Assess Performance: Evaluate system performance and call quality under load.

Testing and validation are crucial steps to ensure the IP-PBX system functions correctly and meets the intended communication requirements. This phase involves conducting various tests, including call quality assessment using metrics like MOS (Mean Opinion Score), stress testing to evaluate the system's handling of multiple concurrent calls, and verifying the functionality of features like voicemail, IVR, and call forwarding. Compatibility testing is also performed with different VoIP clients and endpoints to ensure seamless integration. Security testing, such as checking for vulnerabilities in SIP authentication, is conducted to safeguard against potential threats.

Deployment

Deployment involves transitioning the configured and tested IP-PBX solution from the testing environment to a live production setting. This process includes migrating the server configuration, user extensions, and SIP trunk settings to the organization's network infrastructure. Detailed planning ensures minimal disruption to existing communication services during the switch-over. Deployment also involves setting up backups and disaster recovery plans to ensure data integrity and system availability. User training is provided to familiarize staff with the new system, covering essential features like voicemail access and call transfers.

After testing, employees receive training on how to use the new system, ensuring a smoothtransition. Post-deployment, continuous monitoring is done to maintain system performance and security, with regular updates applied as needed. This process ensures the IP-PBX system operates efficiently and provides enhanced communication capabilities forthe business

CHAPTER 4

PROJECT IMPLEMENTATION

SYSTEM REQUIREMENTS

The system requires both hardware and software components. On the hardware side, a dedicated server or virtual machine will host the IP-PBX solution, alongside VoIP-enabled end- user devices like SIP phones, smartphones, or laptops. Network infrastructure components such as routers and switches will also be crucial. For software, an open-source IP-PBX solution like Asterisk or FreePBX will be installed on a Linux- based operating system, preferably Centos. VoIP clients like Zoiper will be used for communication testing. Adequate bandwidth and a stable internet connection will also be required to ensure uninterrupted VoIP traffic.

IP-PBX SETUP

The setup of the IP-PBX solution begins with preparing the environment by installing a Linux-based operating system on the server, ensuring all updates are applied and network settings are configured correctly. Once the environment is ready, the chosen IP-PBX software (Asterisk, FreePBX, or similar) will be installed along with required dependencies such as Apache, MySQL, and PHP. After installation, SIP extensions will be configured for each user/device, and unique usernames, passwords, and extension numbers will be assigned. The next step involves configuring trunks to establish communication with external VoIP service providers, followed by setting up dial plans to manage how calls are routed withinthe system, including internal, outbound, and emergency call routing.

SECURITY CONFIGURATIONS

To secure the IP-PBX system, several layers of security will be implemented. First, the firewall will be configured to restrict unnecessary access to the server, allowing only essential traffic. SIP encryption via TLS will be enabled to secure signaling, while SRTP (Secure Real-Time Transport Protocol) will be used to protect voice data during transmission. Additionally, strong password policies will be enforced for all users, and access to the system will be restricted to trusted IP addresses to minimize the risk of unauthorized access.

TESTING THE IP-PBX SYSTEM

The testing phase will focus on verifying both the functionality and performance of the IP-PBX solution. Functionality testing will involve making internal calls between extensions, checking inbound and outbound calls, and ensuring features like voicemail, call forwarding, and conferencing are working as expected. Performance testing will simulate various network conditions to test the system's scalability and call quality, especially under heavy load. Security testing will include penetration tests to identify potential vulnerabilities such as unauthorized access, call hijacking, or eavesdropping attempts, ensuring the system is resilient against common VoIP threats.

MONITORING AND LOGGING

To maintain optimal system performance, continuous monitoring will be implemented using tools to track server health, CPU usage, memory consumption, and call metrics. Logging will also be enabled to record call details, system events, and any error logs that occur during operation. These logs will provide valuable insights into system behavior and help in troubleshooting issues that arise. Real-time monitoring will ensure that the system is performing well under varying loads, and any abnormal activity can be detected and mitigated immediately.

ADVANTAGES

Implementing an open-source IP-PBX solution offers numerous benefits. Primarily, it reduces costs compared to traditional PBX systems since the software is free, and there are no recurring licensing expenses. This makes it particularly appealing to small and mid-sized businesses that want to upgrade their communication infrastructure without heavy financial investments. The system's flexibility and scalability make it easy to modify as needed, allowing businesses to adjust to growing communication demands or incorporate new features. Moreover, IP-PBX supports various forms of communication such as voice, video, and messaging over the internet, serving as a comprehensive, unified communications platform.

The open-source nature of the software also allows for greater customization, letting companies adapt the system according to their unique needs. Another significant advantage is improved mobility, as employees can remain connected through softphones or mobile devices from anywhere with internet access.

Lastly, the solution is future-ready, with continuous updates and improvements available through the open- source community, ensuring the system evolves with new technological advancements.

DISADVANTAGES

However, there are also some drawbacks to using an open-source IP-PBX system. One major concern is the complexity of installation and configuration. Unlike commercial alternatives that often include built-in support, open-source solutions may require a deep understanding of networking, VoIP protocols, and Linux operating systems, which can complicate both the deployment and the ongoing management of the system.

Additionally, although the software itself is free, businesses might incur other costs related tohardware, security enhancements, and the time required for proper maintenance and updates.

Security risks also pose a challenge, as VoIP systems are vulnerable to threats like call interception, unauthorized access, and denial-of-service attacks. Open-source systems may not come with robust security features pre-configured, which requires additional effort to secure the system properly.

Lastly, relying on community support can sometimes be slower in resolving technical issues compared to the dedicated customer service offered by commercial providers, potentially causing delays in fixing problems.

CHAPTER 5

RESULTS AND DISCUSSION

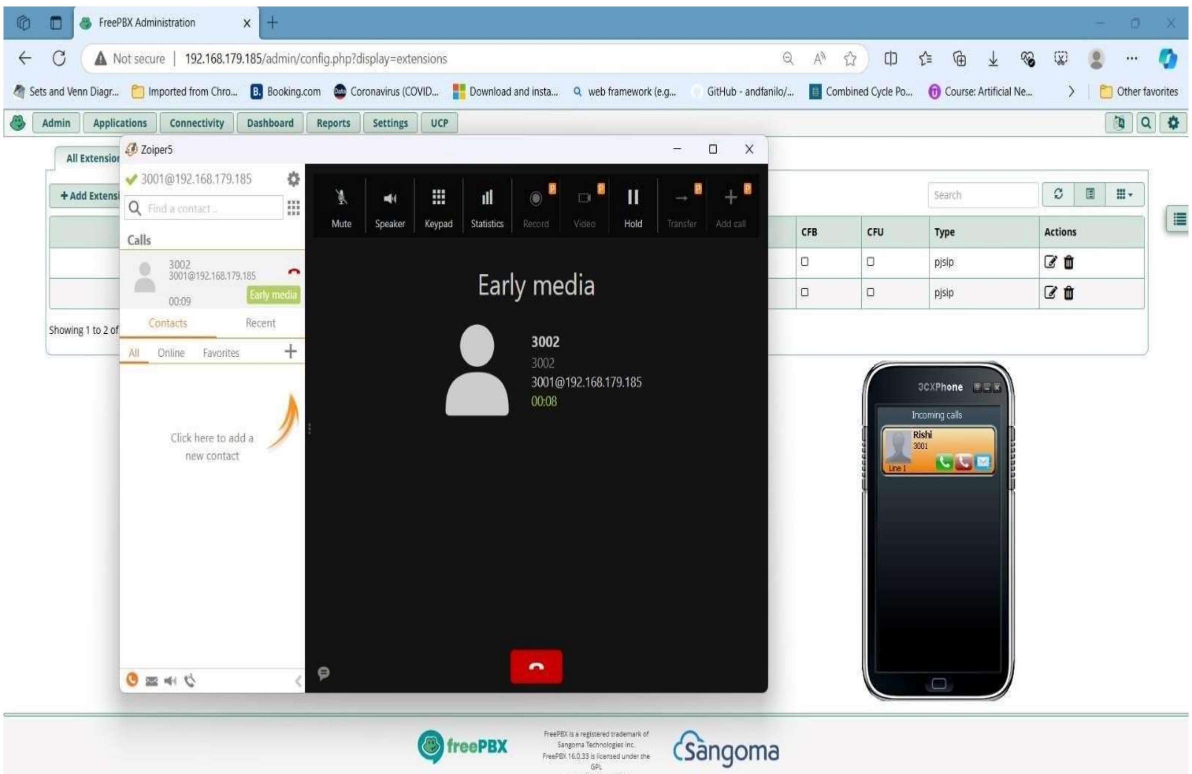

Fig.5.1.Final Result

[FreePBX Administration interface with a softphone call in progress displaying "Early media" between two extensions.]

The deployment and testing of the open-source IP-PBX solution delivered positive outcomes, efficiently supporting real-time communication for voice, video, and messaging. Internal calls and external connections through configured trunks operated seamlessly, and the system successfully handled multiple simultaneous calls with minimal performance impact. Security features such as SIP TLS encryption and SRTP ensured that communication remained secure, with penetration testing confirming the system’s ability to withstand common VoIP threats. The project highlighted the affordability of open-source IP-PBX systems, providing advanced functionalities similar to commercial options without the burden of licensing costs. However, the setup process was complex, requiring specialized technical knowledge, especially in securing the system. Ongoing monitoring and regular updates are essential to maintain security, given the susceptibility of VoIP systems to cyber threats.

The solution demonstrated scalability and adaptability, making it a suitable option for businesses of different sizes, but it demands continuous oversight to prevent potential risks. In conclusion, the project confirmed that open-source IP-PBX is a reliable and cost-effective choice for modern communication needs. Furthermore, the flexibility of open-source systems allows for customizations, making them easily adaptable to specific organizational requirements. As the system evolves, businesses can integrate new features as needed, ensuring the communication infrastructure remains future-proof.

In terms of security, the system demonstrated strong resilience against potential threats, thanks to the implementation of firewall rules, encryption, and continuous monitoring. No majorvulnerabilities were identified during testing, and the security measures helped protect against unauthorized access and data breaches.

Overall, the deployment of the IP-PBX system proved to be a significant upgrade over the legacy telephony infrastructure. It delivered on promises of scalability, flexibility, and enhanced communication features. Continuous monitoring and routine updates will ensure the system remains efficient, secure, and adaptable to future needs.

Additionally, the broad support of the open-source community provides ongoing improvements and patches. With proper management, the system offers long-term value for enterprises seeking modern, scalable solution.

CHAPTER - 6
CONCLUSION

In conclusion, the deployment of the IP-PBX system has successfully transformed the business's communication infrastructure, offering a more modern, cost-effective, and scalable solution compared to traditional telephony systems. The project delivered noticeable improvements in call quality, system reliability, and enhanced features such as voicemail, call forwarding, and video conferencing, significantly improving internal collaboration and external communication.

The flexibility of the system, particularly through cloud integration and remote access for mobile and off-site employees, has made the organization better equipped for the demands of a dynamic work environment, including remote work and global communication needs.

The project also addressed key security concerns, with robust encryption, firewalls, and monitoring systems in place, ensuring data integrity and protection against cyber threats. While there were minor challenges during deployment, such as configuring network settings and fine- tuning certain features, these were quickly resolved, demonstrating the adaptability of the system. The positive reception from users, who found the system intuitive and efficient, further underscores its success. As the business continues to grow, the IP-PBX system's scalability will allow for easy expansion and integration of future technologies, ensuring long-term sustainability. Continuous monitoring, updates, and support will be essential to maintaining system performance and security, making this deployment a critical and forward-thinking investment for the company's communication needs.

APPENDICIES

APPENDIX-1: CODE COMPILER

```
Ifconfig

eth0: flags=4163<UP,BROADCAST,RUNNING,MULTICAST> mtu 1500
        inet 192.168.1.100 netmask 255.255.255.0 broadcast 192.168.1.255
        inet6 fe80::f816:3eff:fe41:2eec prefixlen 64 scopeid 0x20<link>
        ether fa:16:3e:41:2e:ec txqueuelen 1000 (Ethernet)
        RX packets 24510960  bytes 15329895613 (15.3 GB)
        TX packets 15319128 bytes 2435072278 (2.4 GB)

sudo nano /etc/sysconfig/network-scripts/ifcfg-eth0
TYPE=Ethernet
BOOTPROTO=none
NAME=eth0
DEVICE=eth0
ONBOOT=yes
IPADDR=192.168.1.100
PREFIX=24
GATEWAY=192.168.1.1
DNS1=8.8.8.8
DNS2=8.8.4.4
sudo systemctl restart network
```

APPENDIX – 2 : SCREENSHOTS

Fig 6.1. Free PBX Command
[VMware Workstation Player window running Rocky Linux 64-bit, indicating the system is awaiting a command.]

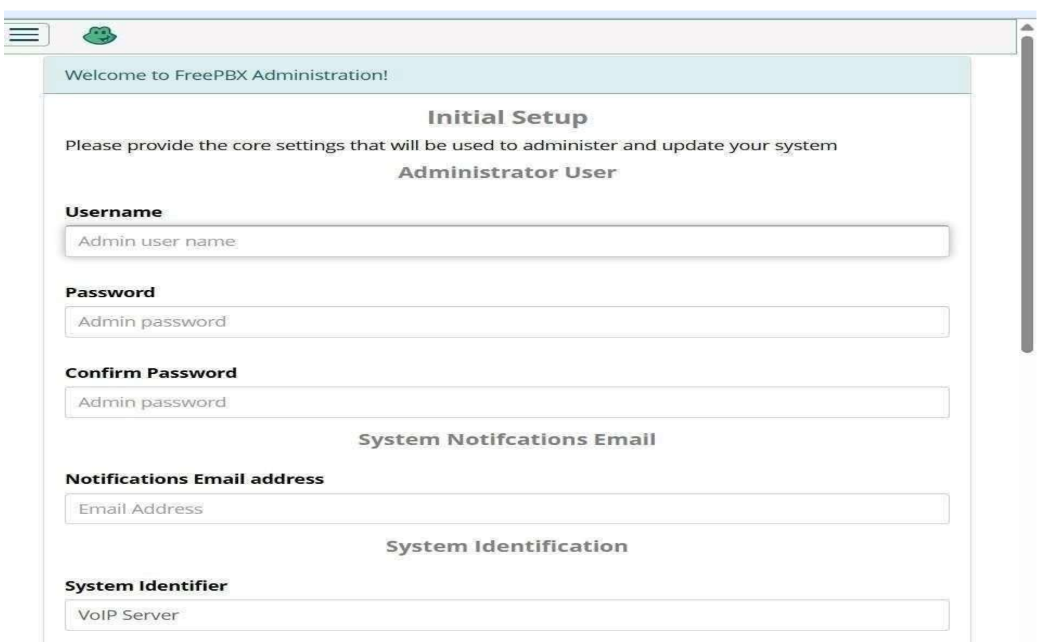

Fig 6.2. Initial Setup
[Initial setup screen for FreePBX Administration, prompting the user to create an administrator account]

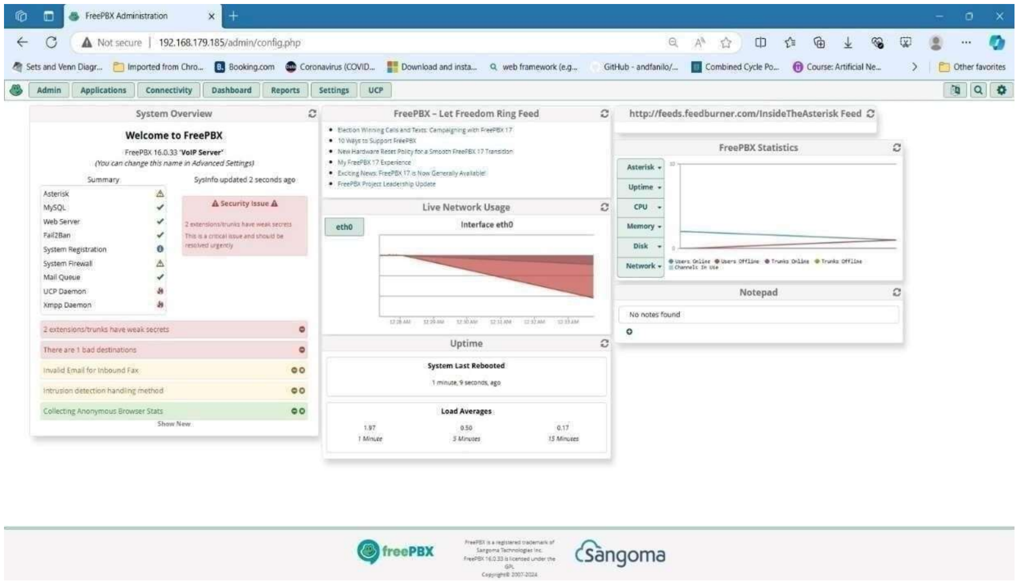

Fig 6.3. Home Page

[the FreePBX Administration dashboard, providing system status, network usage, and various statistics, including memory, CPU, and uptime.]

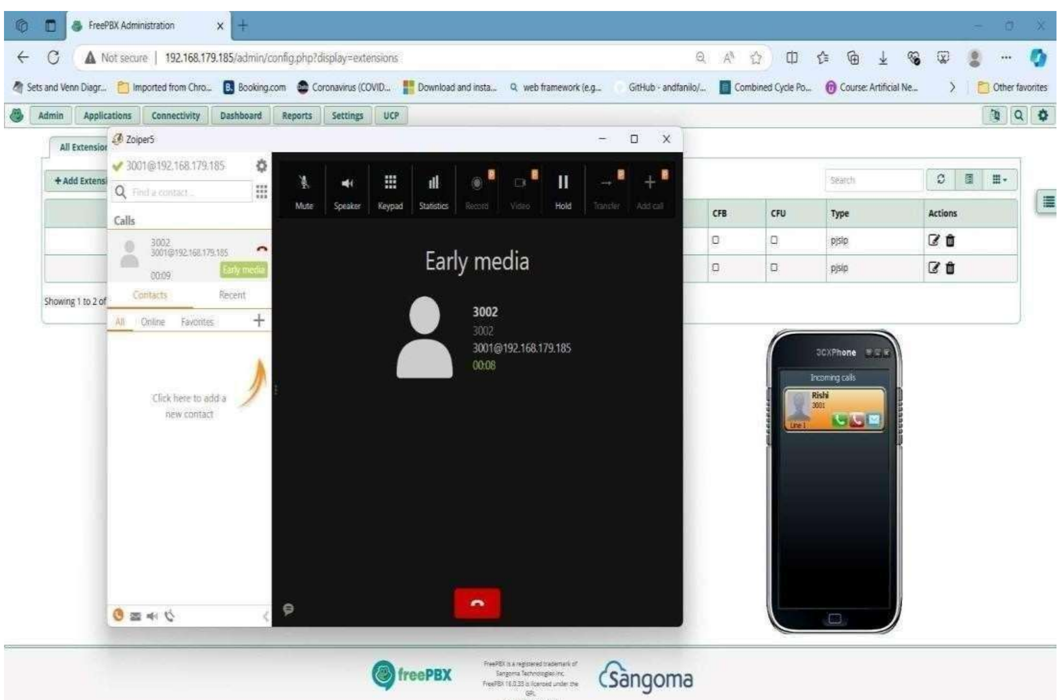

Fig 6.4. Working of Soft Phones

[Interface of a softphone call in progress within the FreePBX system, demonstrating how calls are handled]

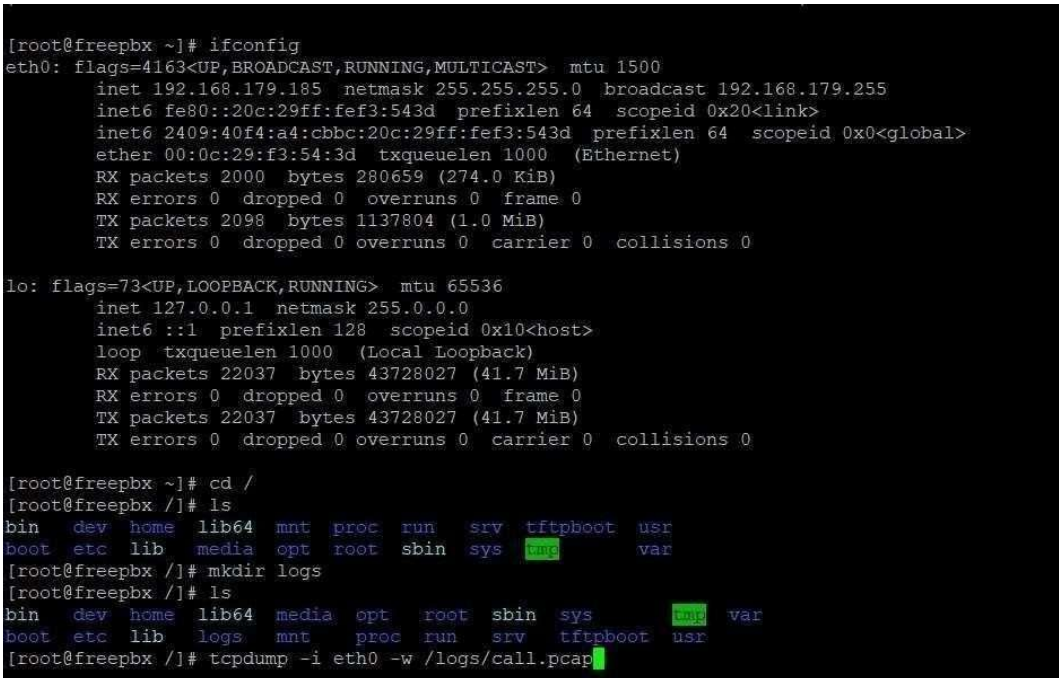

Fig 6.5. Execution of Linux Commands

[Execution of Linux networking and file management commands.]

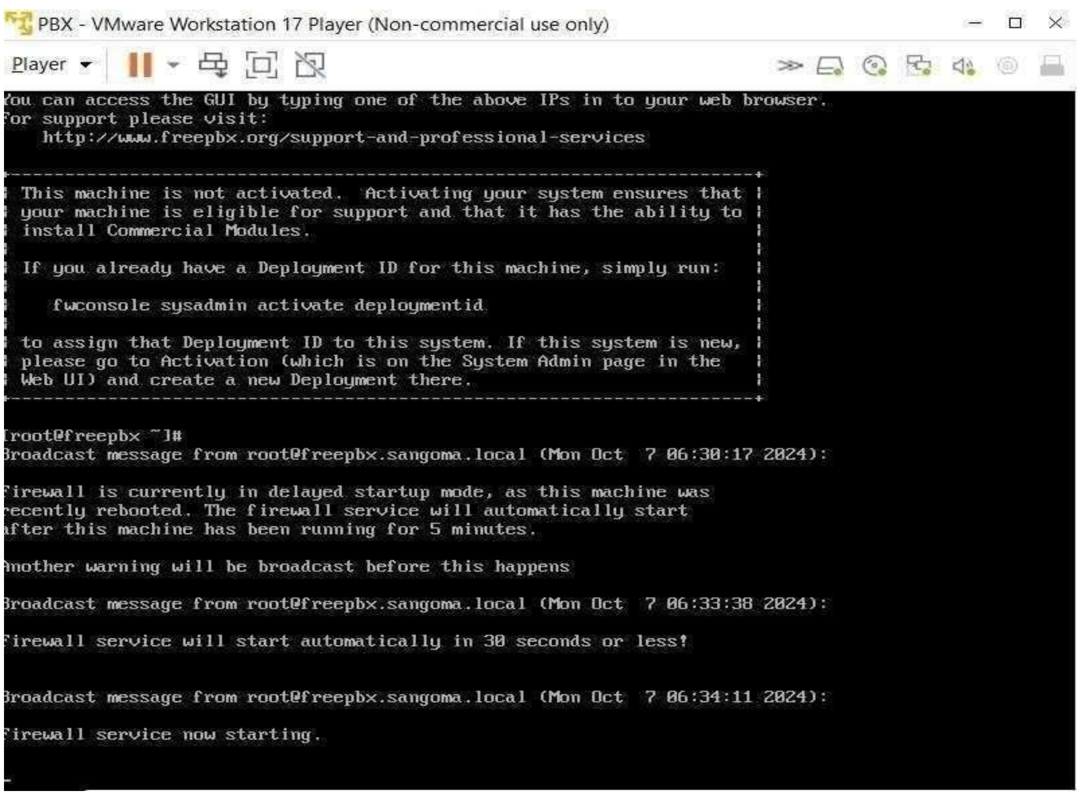

Fig 6.6. Free PBX Server

[Configuration of FreePBX server with firewall activation messages]

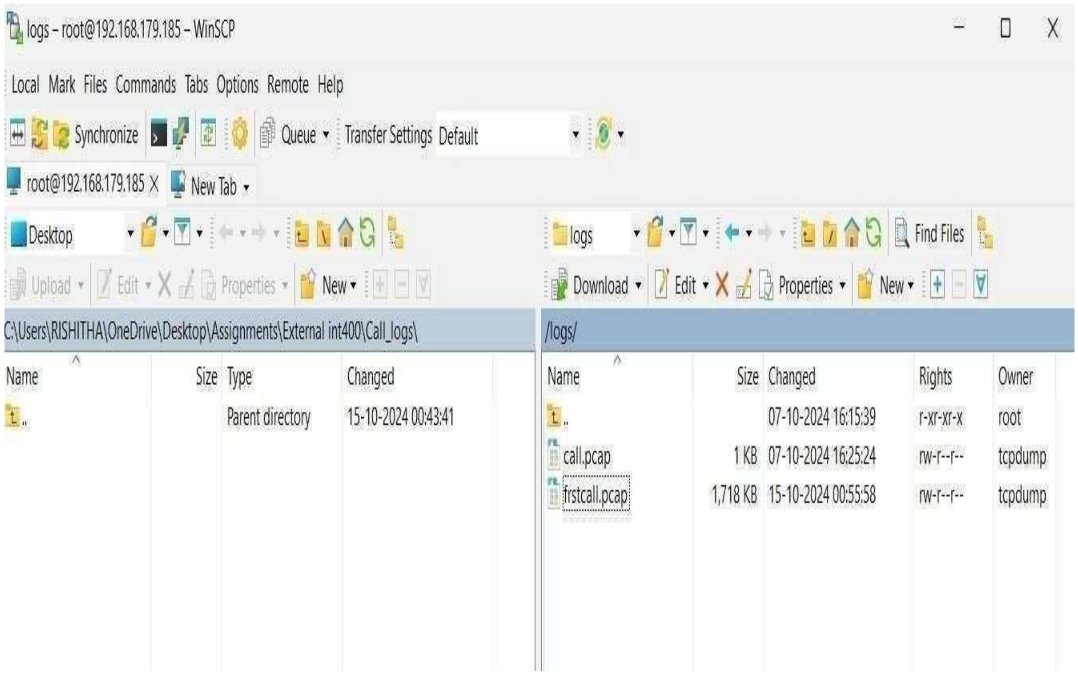

Fig 6.7. File Transfer

[File transfer session using WinSCP, transferring logs between a local machine and a remote server.]

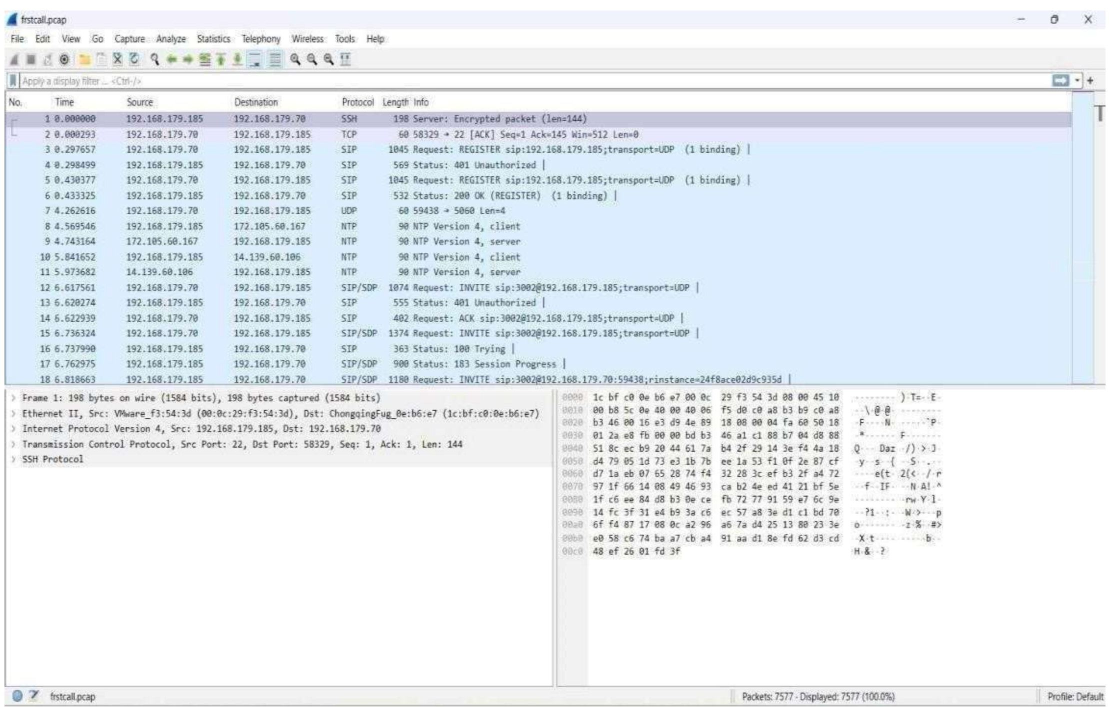

Fig 6.8. WireShark

[Wireshark interface capturing and analyzing network traffic, displaying packet details for protocols like SIP, NTP, and TCP]

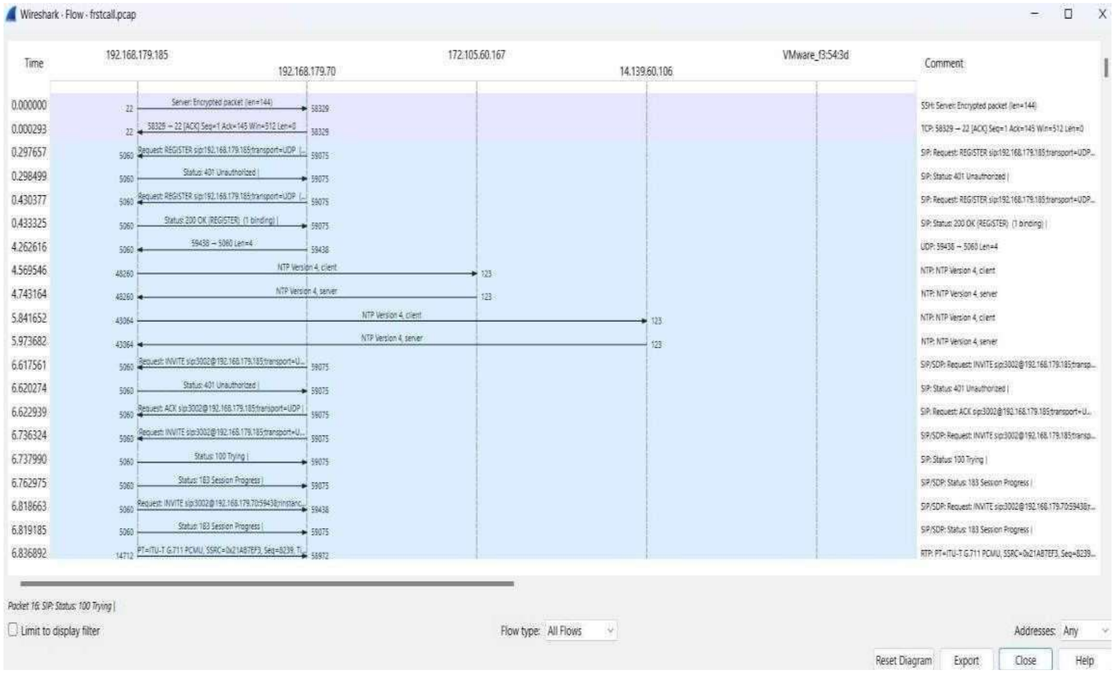

Fig 6.9. SIP Flow 1

[SIP flow diagram in Wireshark, visualizing the interaction between different IP addresses over time for SIP]

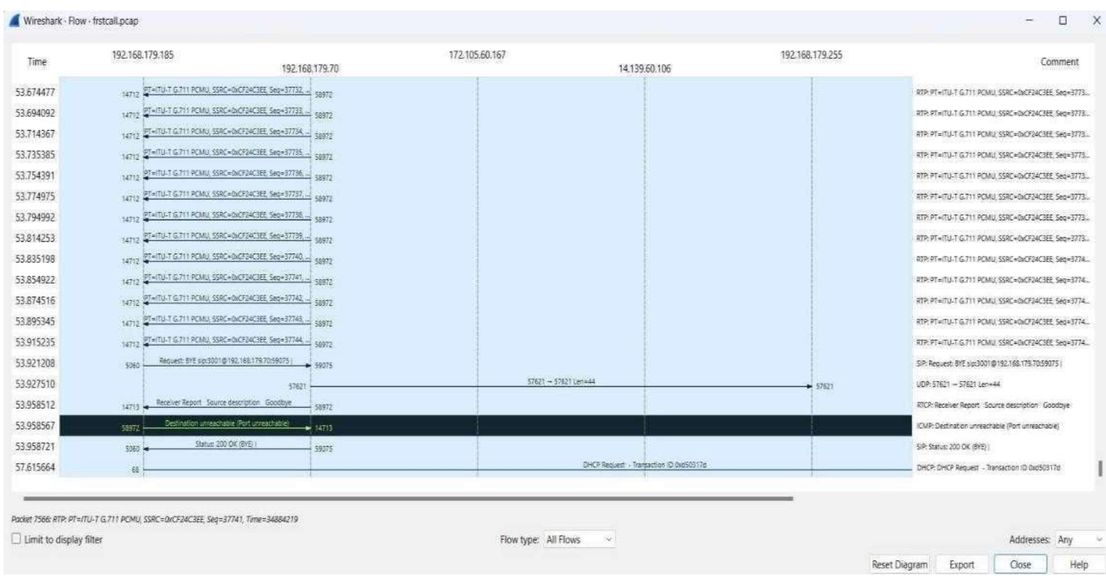

Fig 6.10. SIP Flow 2

[Wireshark capture displaying SIP message flow between multiple IP addresses, illustrating call signaling in a VoIP network.]

REFERENCES

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WORKLOG

DATE	TASK / DESCRIPTION
21-08-2024	Introduction about the domain & the company
22-08-2024	Software Testing (Testing models, Testing lifecycle, Test cases)
23-08-2024	Task 1 presentation – Positive & Negative Scenarios
26-08-2024	CentOS and Linux Commands
27-08-2024	Installation Process
28-08-2024	Testing Linux Commands
29-08-2024	Task 2- Test cases Presentation
30-08-2024	Learnt about Communications (Telecom)
02-09-2024	Telecommunication Basics
03-09-2024	Basics Of networking
04-09-2024	OSI Layers & TCP/IP
05-09-2024	Routing & Doubt Clearing session
06-09-2024	Ip address & Mac Address
09-09-2024	Subnetting, CIDR
10-09-2024	Network Assignments
11-09-2024	First Review
12-09-2024	Static IP & Dynamic IP
13-09-2024	Setting up static IP
16-09-2024	Static IP Configuration
17-09-2024	Session initiation protocol
18-09-2024	Sip Servers(proxy,b2b,register,location,redirect)

DATE	TASK / DESCRIPTION
19-09-2024	Sip stack & message headers
20-09-2024	Sip session, Transactions & Dialogs
23-09-2024	Sip messages & responses
24-09-2024	Session description Protocol
25-09-2024	SDP headers & Codecs
26-09-2024	Basic Call flows & call hold
27-09-2024	Forking, Call forwarding ,Call transferring
30-09-2024	Doubt Clearing session
1-10-2024	FreePBX installation
3-10-2024	Setting up FreePBX
DATE	TASK / DESCRIPTION
4-10-2024	Installation of softphones(Zoiper & 3cx)
7-10-2024	Setting up the extensions & trunks
8-10-2024	Installation of WinSCP & Wireshark
14-10-2024	WinSCP & Wireshark usage demo
15-10-2024	Doubt Clearing session
16-10-2024	Second Review

OFFER LETTER

CERTIFICATE OF COMPLETION