Project Progress Report

on

Cloud Hosted IP-PBX Phone System using Asterisk Server

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in

Computer Science

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2021-2022

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

Date: 01-04-2024

CERTIFICATE

This is to certify that Project Report entitled "**Cloud Hosted IP-PBX Phone System using Asterisk Server**" which is submitted by **Rupesh Kumar** 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: 01-04-2024

Supervisor Signature
Ms. Akanksha
(Assistant Professor

ACKNOWLEDGEMENT

It gives us a great sense of pleasure to present the project progress report of the B. Tech Major Project undertaken during B.Tech. Final 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 his/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 his/her cognizant efforts that our endeavors have seen the light of the day.

We also take the opportunity to acknowledge the contribution of Dr. Ajay Kr. 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 : 01-04-2024

ABSTRACT

This research paper examines the development of a cloud-hosted phone system using the open-source software Asterisk and the cloud service provider Oracle Cloud. The paper discusses the design, implementation, and testing of the system, as well as the benefits and challenges of using a cloud-hosted approach for phone systems. The study also evaluates the system's reliability, scalability, and cost-effectiveness, highlighting its potential as a viable alternative to traditional on-premises phone systems. The research concludes that the cloud-hosted phone system using Asterisk and Oracle Cloud can provide a flexible, efficient, and cost-effective communication solution for businesses of all sizes.Cloud-hosting changes the game. By leveraging Asterisk on a remote server, businesses can access a feature-rich IP-PBX system without the burden of managing physical infrastructure. Asterisk's open-source nature allows for customization and integration with various communication tools, creating a truly dynamic solution.

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INTRODUCTION

The evolution of telecommunication technologies has rendered traditional phone systems increasingly obsolete, paving the way for the emergence of innovative solutions such as cloud-hosted IP-PBX systems. Anchored on the robust framework of the open-source Asterisk server, these cutting-edge systems not only signify a departure from conventional methods but also herald a new era of efficiency and affordability in business communications.

At the heart of this transformation lies the IP-PBX, or Internet Protocol Private Branch Exchange, which serves as the nerve center of an organization's telephony network. Boasting an array of functionalities including voicemail management, call forwarding, and automated attendants, the IP-PBX system has long been integral to facilitating seamless internal and external communication. However, traditional setups demanded substantial investments in on-site hardware and specialized technical expertise for upkeep and maintenance.

Enter cloud-hosting – a game-changing paradigm shift that revolutionizes the landscape of business telephony. By harnessing the power of Asterisk deployed on remote servers, enterprises now have access to a feature-rich IP-PBX infrastructure without the logistical complexities associated with physical hardware management. The inherent flexibility of Asterisk's open-source architecture further empowers organizations to tailor their communication systems to specific requirements, facilitating seamless integration with a myriad of communication tools and applications.

In this exploration, we delve into the transformative potential of Asterisk-enabled cloud-based IP-PBX systems, elucidating how they empower businesses to transcend the constraints of traditional telephony and embrace a more dynamic and adaptable approach to communication infrastructure.
In this paper, the uses and performance of a Cloud Hosted IP-PBX System using Asterisk Server is used for the business communication system. And analysis the best performance of the system using the cloud.

PROJECT CATEGORY

- **Telephony & VoIP:** This is the most general category, as the project deals with creating a phone system using Voice over IP (VoIP) technology.
- **Cloud Computing & Infrastructure:** The project involves deploying the Asterisk server on a cloud platform, highlighting the cloud-hosted aspect.
- **Open-Source Software (OSS):** Asterisk is a prominent open-source framework, making this project relevant to the OSS community.
- **Business Communication Solutions:** The end goal is to provide a communication system for businesses, fitting within this category.

This project fits into the Unified Communications category by providing a cloud-based IP-PBX phone system powered by Asterisk. It leverages VoIP technology for making and receiving calls over the internet, offering features like call routing, voicemail, conferencing and more. By utilizing Asterisk, an open-source communication platform, it enables businesses to deploy cost-effective and customizable telephony solutions. As a cloud-hosted service, it provides scalability, flexibility, and accessibility.

LITERATURE REVIEW

In [1], Siddique et *al.* stated that Raspberry Pi is used to create the communication system for university communications system that is based on internet and VoIP protocols, and it is low cost as compared to existing system.

In [2], Megha S Meshram et *al.* demonstrated how Asterisk server and Raspberry Pi device is used to create the communication system at low cost for small businesses for effective communication.

In [3], Ahmed J. Aleissa et *al.* showed that how 3CX can be used to create the communication system in this paper AWS is used to create the communication system and 3CX is installed in AWS Virtual Machine so that it can be accessed anytime.

In [4], Sarwar Khan et *al.* showed the configuration of asterisk files for campus network. In this project the author used the on-premises PBX system to set up the internal communication network.

In [5], Nikitha Reddy Nalla et *al.* showed the communication system using Raspberry Pi which is like the Megha S Meshram research paper [2], both the papers are using Raspberry Pi device for their project.

In [6], Sergey Konshin et *al.* showed how the packets are transferred during communication. They used the OPNET Simulation tool to demonstrate the simulation of packets transfer and received during the middle of the communication.

In [7], Bilal Muhammad Khan et *al.* showed the analysis of using Raspberry Pi device with asterisk server and how many devices it can handle when they are connected simultaneously and how the system performance affect at that time.

In [8], Pelayo Nuno et *al.* demonstrated how to deal with the Malicious activities that can be performed by different malicious users and how to make your system more secure so that it can be protected from malicious users. They also used the Firewall at the LAN to protect and collect the logs of malicious users and activities.

Ref. No.	Year	Authors	Major Findings	Limitations
[1]	2019	Siddique Abubakr Muntaka Faiza Hussein Paul Sarfo	Created Communication System Works on Raspberry Pi device	Delay in Calls when number of clients increased
[2]	2018	Megha S Meshram, Pooja Thakare, Pranali Dandekar	VoIP calling using Raspberry Pi Communication System for Small business Organization	Required Raspberry Pi as a Hardware device.
[3]	2022	Ahmed J. Aleissa, Sabri N. Shabib, Raied K. Farrash	Cloud based phone system using AWS as a cloud provider. Using 3CX as an asterisk	3CX is not an open-source project. AWS is costly
[4]	2017	Sarwar Khan, Nouman Sadiq	Test the configuration of VoIP based PBX on campus network.	Used on premise PBX system.
[5]	2022	Nikitha Reddy Nalla, Sakthivel S, Shankar R	Low-cost voice system using Raspberry Pi	Raspberry Pi used as a server. Cannot handle large number of calls.
[6]	2020	Sergey Konshin, Nishanbayev T.N.	Asterisk IP-PBX simulation to check the packets in the simulation.	Only checking the performance in OPNET
[7]	2022	Bilal Muhammad Khan, Rabia Bilal, Muhammad Fahad	Analysis of Asterisk Server on Raspberry Pi 3	Raspberry Pi 3 is old hardware and latest Raspberry Pi is Raspberry Pi 5
[8]	2020	Pelayo Nuno, Carla Suarez	Analyzing the security of Asterisk from Malicious Activities.	Machine Learning can be used in future to predict malicious activities.
[9]	2017	Ashwani S. Gawarle	Low-Cost Communication System	Raspberry Pi cannot handle large number of extensions.

PROBLEM FORMULATION

The existing system routes all internal telephony through the existing LAN without the need for a separate network. The system uses IP phones based on the open SIP standard. The IP phone calls are handled by Raspberry Pi and Asterisk software [1], which replaces the PBX and uses SIP protocol for call initiation and termination. This connects the users through the LAN port and includes hardware and training costs. Limitations to this system are.

- **Limited processing power**: Raspberry Pi has limited processing power, which may lead to a quality issue when the number of calls is increased.
- **Limited storage space**: Raspberry Pi has limited storage space, which may limit the amount of call data and messages that can be stored.
- **Limited scalability**: While Asterisk is scalable, the Raspberry Pi hardware may not be able to support large-scale deployments.
- **Hardware compatibility**: Some hardware devices may not be compatible with the Raspberry Pi, limiting the range of phones and other devices that can be used with the system.

So, the problem in the previous methods are they are using Raspberry Pi device to setup the asterisk server which does not have the capability to handle large numbers of devices and calls simultaneously so we are using cloud to replace the Raspberry Pi device.

OBJECTIVE

To develop and deploy a cloud-hosted IP-PBX phone system utilizing Asterisk server technology, aimed at providing businesses with a scalable, cost-effective, and feature-rich telephony solution. The objective is to leverage the flexibility and customization capabilities of Asterisk to deliver advanced communication features such as call routing, voicemail, conferencing, and more, accessible from any location with internet connectivity. By offering seamless integration with existing infrastructure and ensuring reliability and security in the cloud environment, the project aims to empower businesses with enhanced communication capabilities, ultimately improving productivity, customer satisfaction, and operational efficiency.

PROPOSED SYSTEM

The proposed system uses a cloud service provider to route all the telephone calls through the cloud server. The system uses Digital Ocean as a cloud service provider. The system uses an Ubuntu server machine to install the Asterisk server. The calls are routed from this Asterisk server.

To set up the system, an instance of Ubuntu machine is created which is used to install and configure the Asterisk server. Asterisk is an open-source software that provides a complete PBX system for VoIP (Voice over Internet Protocol) applications. By using Asterisk, the system can handle the essential features for a telephony system.

After the Asterisk server is installed and set up, the system use SIP (Session Initiation Protocol) to route calls through it by configuring the different configuration files in the Asterisk server. The system can support various SIP clients, IP Phones, including softphones, hardware phones, and mobile devices. By using SIP, the system can establish and manage connections between the clients and the Asterisk server, allowing them to make and receive calls.

In the existing system Raspberry Pi device is used which is a small computing device where we need to connect peripheral to it. But we are setting up the Asterisk server on the cloud so the asterisk service can be accessed from any geographical location.

Asterisk is a LINUX based software form of PBX (Private Branch Exchange) which is used to set up the communication system inside the organization without installing any on- hardware PBX. Here we are using the Oracle Cloud platform because oracle cloud provides a lifetime free tier of up to 4 instances of ARM Ampere A1 Compute with 3,000 OCPU hours and 18,000 GB hours per month which can easily handle the calls for any small or medium sized organization.

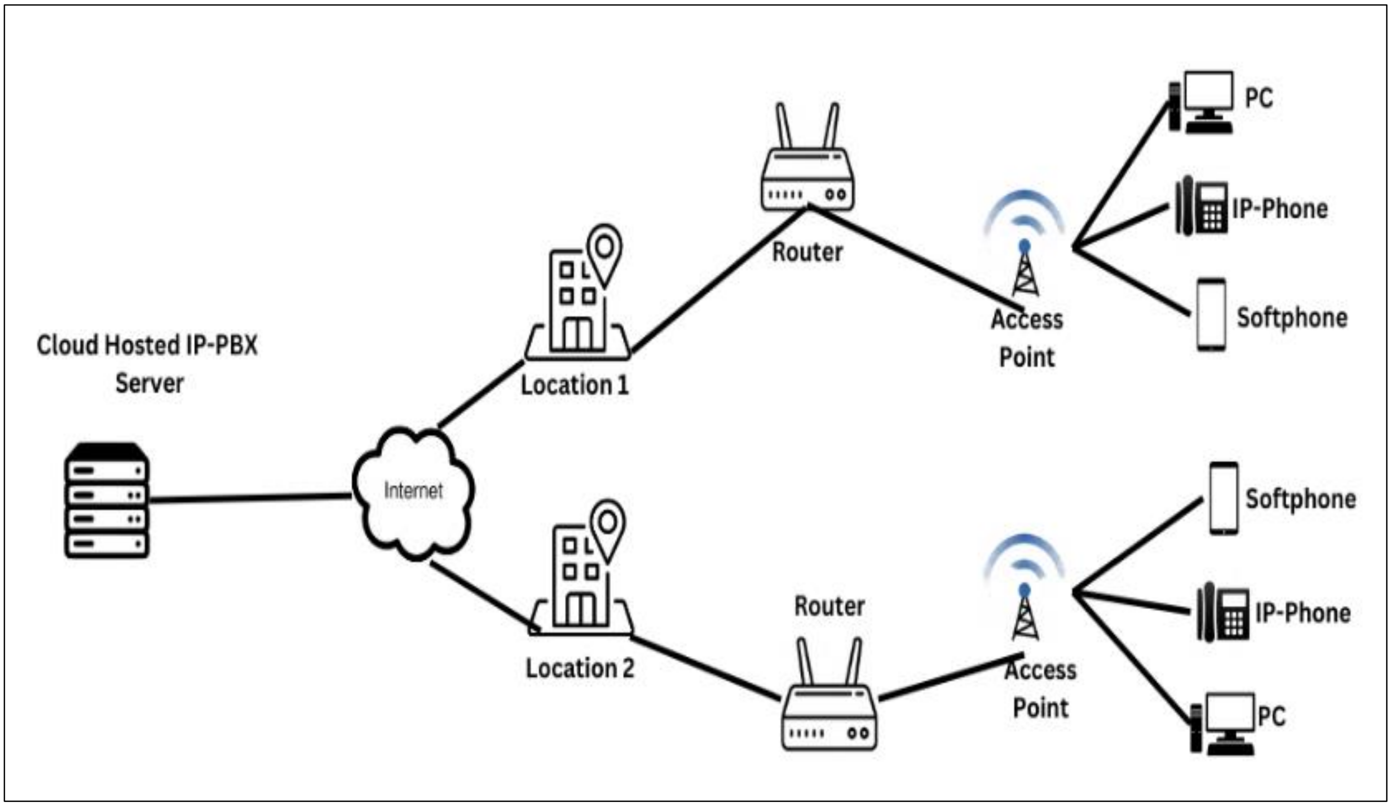

Figure 1Architecture of the Platform

FEATURES OF THE SYSTEM

Low-Resource Optimization: Tailored specifically for Raspberry Pi devices, the system is optimized to operate efficiently on low computing resources, ensuring smooth performance without compromising functionality.

Energy-Efficient Operation: Designed to minimize power consumption, the solution maximizes the battery life of Raspberry Pi devices, making it suitable for remote or off-grid deployments where energy efficiency is critical.

Intelligent Resource Management: Utilizes intelligent resource allocation algorithms to dynamically allocate computing resources based on demand, optimizing performance while minimizing resource usage on Raspberry Pi devices.

Minimal Footprint: With a compact installation footprint, the system minimizes the storage and memory requirements on Raspberry Pi devices, allowing for more efficient utilization of limited onboard resources.

Seamless Cloud Integration: Integrates seamlessly with cloud-based services to offload resource-intensive tasks, such as call processing and storage, reducing the workload on Raspberry Pi devices and enhancing overall system performance.

Remote Management and Monitoring: Provides remote management and monitoring capabilities, allowing administrators to efficiently oversee and troubleshoot the system from anywhere, minimizing downtime and maximizing uptime.

Automatic Updates and Maintenance: Implements automated update and maintenance mechanisms to ensure the system remains up-to-date and secure without placing undue strain on Raspberry Pi devices' limited computing resources.

FEASIBILITY STUDY

1. Technical Feasibility:

- **Compatibility with Raspberry Pi**: Assess whether Asterisk server can effectively run on Raspberry Pi devices given their low computing capabilities.
- **Integration with Cloud Services**: Evaluate the feasibility of integrating the system with cloud services, ensuring compatibility, reliability, and scalability.
- **Networking Requirements**: Determine the networking requirements for seamless communication between Raspberry Pi devices and cloud servers, considering factors like latency and bandwidth.
- **Security Considerations**: Address security concerns related to cloud-hosted systems, including data encryption, access control, and protection against cyber threats.

2. Economic Feasibility:

- **Cost Analysis**: Conduct a cost-benefit analysis to determine the overall investment required for implementing the system, including hardware, software, and ongoing maintenance.
- **Return on Investment (ROI)**: Estimate the potential ROI by comparing the anticipated benefits, such as improved communication efficiency and cost savings, with the initial investment.
- **Scalability**: Assess the scalability of the solution to accommodate future growth without incurring substantial additional costs, ensuring long-term economic viability.

3. Operational Feasibility:

- **User Acceptance**: Evaluate the willingness of end-users to adopt the new cloud-hosted IP-PBX phone system, considering factors such as ease of use, training requirements, and perceived benefits.

- **Operational Impact**: Assess the impact of implementing the system on existing business processes, workflows, and IT infrastructure, identifying potential challenges and mitigation strategies.
- **Support and Maintenance**: Determine the feasibility of providing ongoing technical support and maintenance for the system, including troubleshooting, updates, and upgrades.
- **Compliance and Regulations**: Ensure compliance with relevant regulatory requirements and industry standards governing telecommunication systems and data privacy, mitigating legal and operational risks.

By conducting a comprehensive feasibility study encompassing technical, economical, and operational aspects, stakeholders can make informed decisions regarding the implementation of the cloud-hosted IP-PBX phone system using Asterisk server, ensuring its viability and success in meeting business objectives.

SOFTWARE REQUIREMENT SPECIFICATIONS

Project Name: Cloud Hosted IP-PBX Phone System using Asterisk Server

1. Introduction

This document outlines the Software Requirement Specification (SRS) for a cloud-hosted IP-PBX phone system built using Asterisk server. The system will provide a central communication hub for businesses, enabling features like voice calling, voicemail, call forwarding, and more.

2. Data Requirements

- User Data: This includes user profiles (name, extension, voicemail settings), call history, and contact information.
- System Configuration: Dial plan configurations, routing rules, trunk settings, and voicemail greetings.
- Call Data: Information about each call, including caller ID, call duration, and recording (optional).

3. Functional Requirements

- **Core PBX Features:**
    - User Management: Create, edit, and delete user accounts with extensions.
    - Call Routing: Route incoming calls based on caller ID, time of day, or other criteria.
    - Call Handling: Manage call forwarding, voicemail, call waiting, and conferencing.
    - Voicemail: Allow users to record greetings, listen to messages, and manage voicemail settings.
    - Auto Attendant: Provide an automated system for directing callers to appropriate destinations.
    - Call Recording (Optional): Record calls for training or quality assurance purposes.
- **Cloud Management:**
    - User interface for managing system configuration and user accounts.
    - Secure user login and access control based on roles and permissions.

- System monitoring and reporting for call statistics and resource usage.
- Scalability to accommodate a growing number of users.

4. Performance Requirements

- **Call Latency:** Minimize call connection times and ensure smooth voice quality. (Target: < 200ms)
- **System Uptime:** Maintain high system availability with minimal downtime. (Target: 99.9% uptime)
- **Scalability:** The system should be able to handle increasing call volume and user base without performance degradation.

5. Maintainability Requirements

- Modular design for ease of maintenance and future development.
- Use of configuration files and scripting languages for flexible customization.
- Comprehensive logging and error reporting for troubleshooting.
- Well-documented code base and user manuals.

6. Security Requirements

- Secure user authentication and authorization to prevent unauthorized access.
- Encryption of sensitive data like passwords and call recordings (if enabled).
- Secure communication protocols like SIP with TLS encryption for call signalling.
- System access restricted to authorized personnel with a least privilege approach.

7. Future Considerations

- Integration with other business applications like CRM or collaboration tools.
- Support for additional communication channels like video conferencing and instant messaging.
- Mobile app functionality for making and receiving calls on smartphones and tablets.

This SRS document provides a high-level overview of the requirements for the cloud-hosted IP-PBX phone system.

SDLC MODEL USED – AGILE MODEL

For the development of the cloud-hosted IP-PBX phone system using Asterisk server, the Agile software development methodology would be well-suited. Agile methodologies emphasize iterative development, collaboration between cross-functional teams, and flexibility to adapt to changing requirements. Given the dynamic nature of telecommunications technology and the need for continuous improvement and iteration, Agile offers several advantages:

1. **Iterative Development**: Agile allows for the incremental development of features, enabling the team to deliver value to users quickly and gather feedback for continuous improvement.

2. **Flexibility**: Agile methodologies accommodate changes in requirements throughout the development process, allowing the team to respond to evolving customer needs and technological advancements.

3. **Collaboration**: Agile promotes close collaboration between developers, testers, and stakeholders, fostering a shared understanding of project goals and facilitating effective communication.

4. **Customer Focus**: By delivering working software in short iterations, Agile ensures that customer feedback is incorporated early and frequently, resulting in a product that better aligns with user expectations.

5. **Quality Assurance**: Agile methodologies emphasize continuous testing and integration, enabling the team to detect and address issues early in the development cycle, thereby improving software quality.

Overall, Agile's adaptability, focus on customer collaboration, and emphasis on iterative development make it an ideal choice for the development of the cloud-hosted IP-PBX phone system using Asterisk server.

DFD (DATA FLOW DIAGRAM)

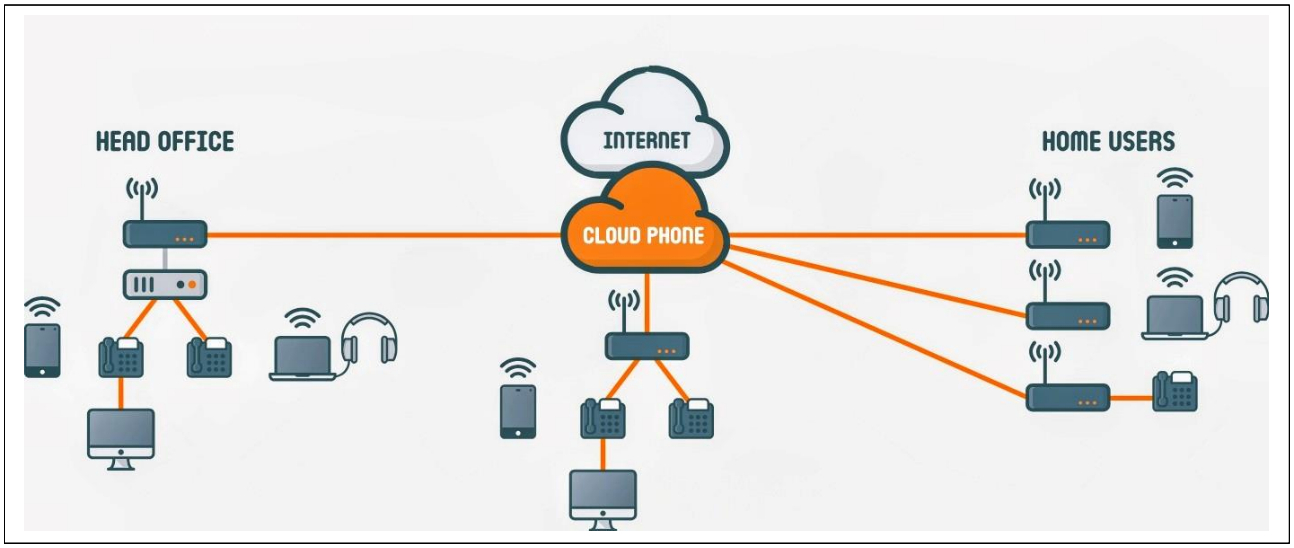

Figure 2 Data Flow Diagram

USE CASE DIAGRAM FOR VOIP SYSTEM

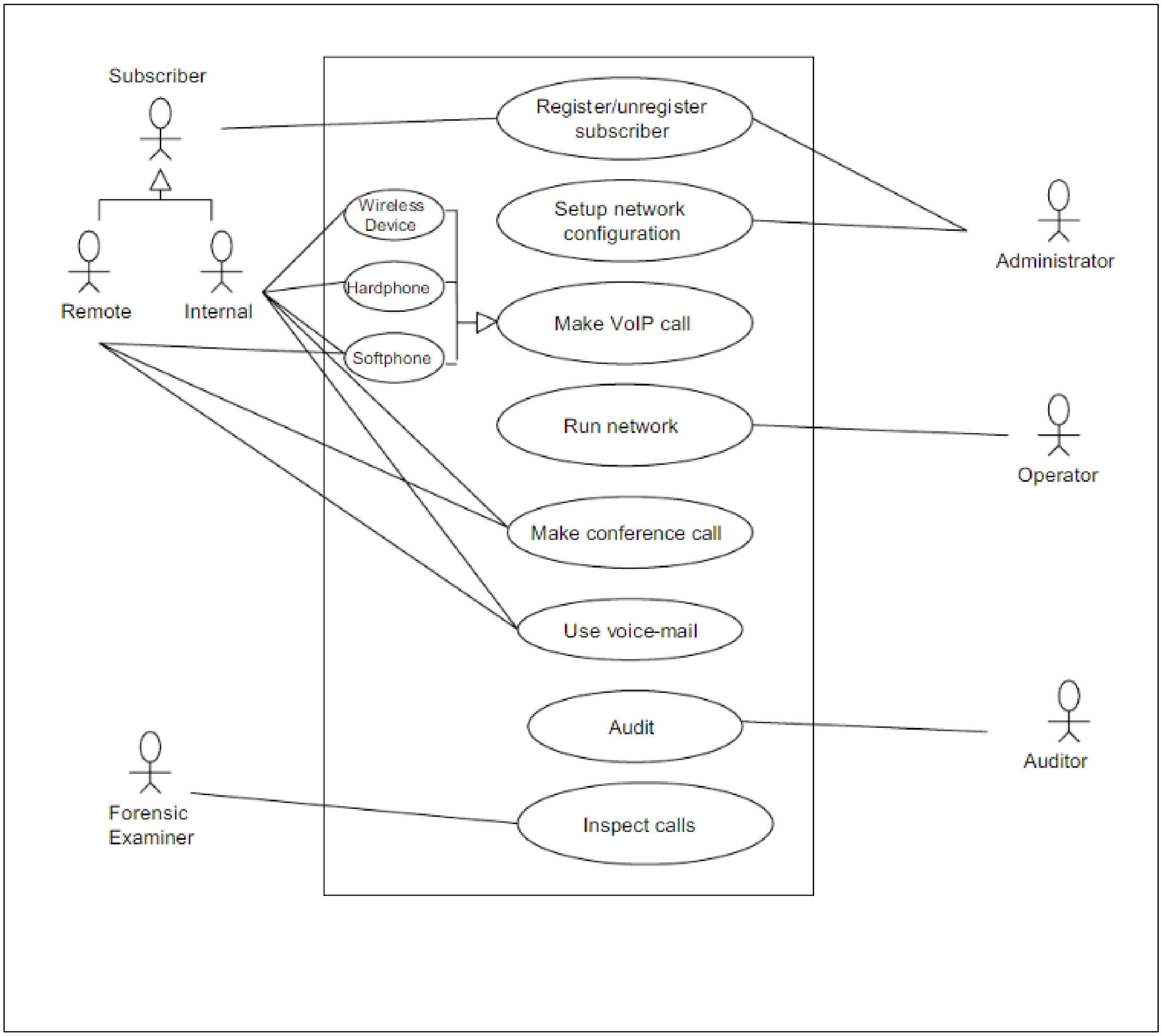

Figure 3 Use Case Diagram

IMPLEMENTATION

To set up the system, an instance of Ubuntu machine is created which is used to install and configure the Asterisk server. Asterisk is an open-source software that provides a complete PBX system for VoIP (Voice over Internet Protocol) applications. By using Asterisk, the system can handle the essential features for a telephony system.

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Asterisk

Asterisk is an open-source project which is mostly used to build the communication system or PBX (Private Branch Exchange) system. Asterisk primarily operates on the VoIP (Voice Over Internet Protocol) means it can transmit the voice on the Internet in the form of packets. It is a LINUX based project so we can use it on different Linux based distributions.

Soft Phones and IP Phones

A soft phone is a software-based phone which can be used in different devices like PC, Mobiles and Tablets etc. Soft phones can be installed normally like the other applications on your devices. The benefit of soft phones is cost effective because it does not require additional cost.

IP phones are like softphones, but they are in the hardware form. IP phones are designed to make and receive VoIP calls over the internet. IP phones provide superior voice quality as compared to softphones because they are optimized for better call clarity and have dedicated audio hardware and most IP phones are compatible with the SIP (Session Initiation Protocol), making them easy to integrate with various VoIP services. Some of the most popular Softphones are - Xphone, Zoiper, Linphone and MicroSIP.

Among those previously mentioned softphones, we have selected Zoiper.

SIP

SIP (Session Initiation Protocol) which is a Network Layer protocol used with VoIP (Voice Over Internet Protocol) to initiate, maintain, modify, and terminate the real time communication sessions. SIP clients typically connect to SIP servers and other SIP endpoints using TCP or UDP on port 5060 or 5061. Port 5060 is generally used for uninterrupted signal traffic. Port 5061 is mostly used for traffic using Transport Layer Security (TLS).

Oracle Cloud

Oracle Cloud is a very popular cloud service provider which offers IaaS (Infrastructure as a Service), PaaS (Platform as a service) and SaaS (Software as a service) and all these services can be directly accessed over the internet. Here we are using oracle free tier of Ampere A1 Compute instances which is VM.Standard.A1.Flex because the VM.Standard.A1.Flex is a flexible, we can customize the number of OCPUs and amount of memory that are allocated when we create or resize an instance. We can use all the always Free OCPUs and memory to create a single instance or create multiple smaller instances that each use a portion of the resources. We have created an instance of this specification:

- **Processor**: 4 OCPUs
- **Memory**: 24 GB
- **Image**: Ubuntu Server

Following are some key points which are discussed as benefits of the proposed research work:
Oracle reclaim its free idle instance during 7 days under these conditions:

- CPU utilization for the 95th percentile is less than 20%
- Network utilization is less than 20%

- Memory utilization is less than 20%

So, it’s important to always use your virtual machine (VM) instance to ensure its continual availability for your use.

Ubuntu Server

Ubuntu Server is a Linux based distribution which is specially designed for server environments. It is an open-source, Debian-based operating system which is used in a wide range of server applications and services. Ubuntu Server is known for its stability, security, and ease of use, making it a popular choice among businesses and individuals for hosting different services.

Design System and Configuration

This section explains about the design and configurations of our system we have created the VM instance of type ubuntu on the oracle cloud after setup the ubuntu server we have changed the security rules of the cloud CIDR so it can receive the traffic from anywhere on the internet. We allow the incoming SSH traffic from anywhere by setting up a stateful ingress rule with source CIDR 0.0.0.0/0, and destination TCP port 22 and allowed TCP and UDP port number 5060 and 5061 for SIP clients.

After that we cloned the asterisk official repository to our machine and set up the asterisk configuration files that is sip.conf file which contains the SIP endpoints, users and various parameters that are related to SIP.

And In extensions.conf file contains the dial plans for call routing and call handling the dial plans tells that asterisk about how to process the incoming and outgoing calls and what actions will be taken during the call.

Figure 4. shows the extensions.conf file in which we have created two extensions i.e., 7001 and 7002 so both the extensions can call each other.

```
[internal]
exten => 7001,1,Answer()
exten => 7001,2,Dial(SIP/7001,60)
exten => 7001,3,Playback(vm-nobodyavail)
exten => 7001,4,VoiceMail(7001@main)
exten => 7001,5,Hangup()

exten => 7002,1,Answer()
exten => 7002,2,Dial(SIP/7002,60)
exten => 7002,3,Playback(vm-nobodyavail)
exten => 7002,4,VoiceMail(7001@main)
exten => 7002,5,Hangup()
```

Figure 4 extensions.conf file

```
[general]
context=internal
allowguest=no
allowoverlap=no
bindport=5060
bindaddr=0.0.0.0
srvlookup=no
disallow=all
allow=ulaw
alwaysauthreject=yes
canreinvite=no
nat=yes
session-timers=refuse
localnet=192.168.1.0/255.255.255.0
externip=8.222.199.64

[7001]
type=friend
host=dynamic
secret=7001
context=internal

[7002]
type=friend
host=dynamic
secret=7002
context=internal
```

Figure 5 sip.conf file

- **bindaddr and bindport**: It specifies the IP address and port where asterisk listens for SIP traffic.

- **srvlookup**: Determines whether Asterisk performs DNS SRV lookups when routing SIP requests.

- **disallow and allow**: Specifies which audio codecs are disallowed or allowed for SIP communication.

- **type**: Specifies the type of SIP entity, such as "friend," "peer," or "user."

- **host and port**: The IP address or hostname of the remote SIP entity and the port to use for communication.

- **secret**: The authentication password or shared secret for the SIP entity.

- **type**: Specifies the type of SIP entity.

- **host**: The IP address or hostname for the user or peer.

- **allowguest:** Determines whether guest calls (calls from anonymous users) are allowed.

- **secret**: The authentication password or shared secret.

- **context**: The context is used for routing calls.

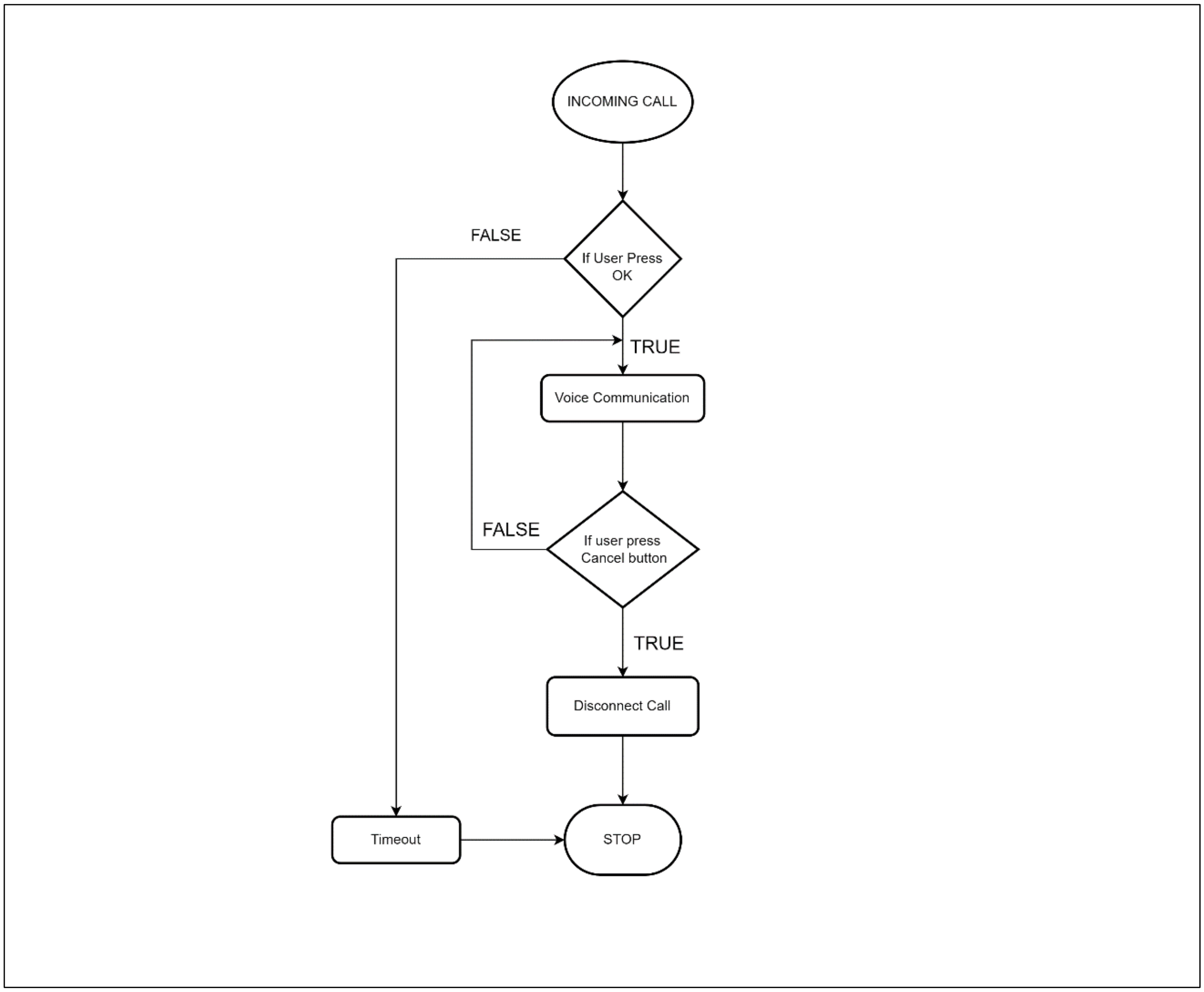

Figure 6 Incoming call flow

Figure 6. shows the flow of incoming calls to the extension.

It shows how the incoming calls will work in different scenarios like if the user does not pick the call, then the call will automatically get disconnected after timeout and if the user picks the call, then the call will establish, and the voice communication will begin. Another flow chart shows the outgoing call flow.

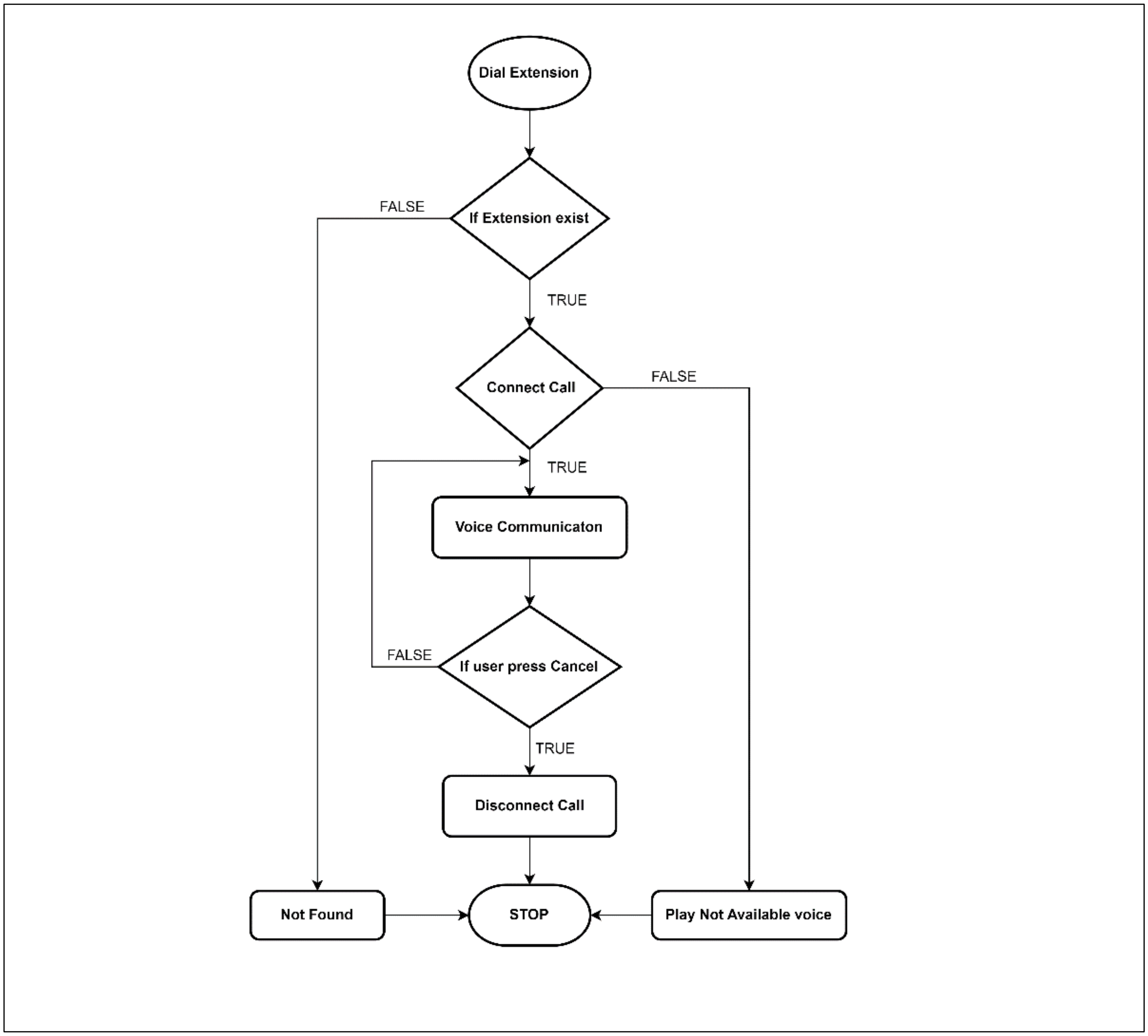

Figure 7 Outgoing call flow

Figure 7. shows the flow of outgoing calls like if a user dials an extension that doesn't exist than it shows not found and call will get disconnected and if the call will not pick by the user than after timeout it plays the not available voice, and the call will get disconnected.

TESTING

Introduction

A cloud-based phone system, or cloud phone, is a phone service that allows you to make calls over the internet rather than over a traditional analog phone line that uses copper wires or optical fibres to make a connection. Cloud phones are hosted in on or more offsite secure datacentres. Nowadays the EPBX (Electronic Phone Branch Exchange) system is used in many business organizations, Hotels, and Institutions etc. for voice communication. we are replacing the EPBX system with a Cloud based IP-PBX phone system for the use of voice communication in business enterprises, Educational Institutions etc.

Scope

In Scope

Some of the features, functional or non-functional requirements of the software that will be tested are:

- Calling Ability
- Dialling
- Registration Ability

Out of Scope

Some of the features, functional or non-functional requirements of the software that will not be tested are:

- Messaging Ability
- Call Recording

Quality Objective

Some objectives of this project are:

- To Ensure the seamless call Quality
- To Ensure the Security so that only authorised person can Register
- Fixed the calibration issue of extension with respect to user and its dial number ✓ User can easily register when provides username and password.

Test Methodology

Overview

As this project dependent on integration of different existing units so on first level unit testing was done and then integration and after this the testing of whole system was done to ensure that all major functionalities are working fine. This whole testing was done manually by the team members.

Test Levels

1. **Unit Testing**: In this Testing different units was tested.
    - Asterisk Configuration Tests: Tested individual components of the Asterisk configuration, such as dial plans and call routing rules functionality.
    - Oracle Cloud Services Testing: Tested the functionality of Oracle Cloud services like network configurations
    - User Management Tests: Verified that user account creation, modification, and deletion work correctly.
2. **Integration Testing**: Tested how different components of phone system work together. To Ensure that Asterisk properly interacts with Oracle Cloud services and that data flows smoothly between them.
3. **System Testing**: Tested the functional aspects of phone system, such as making and receiving calls, and other features to ensure they work correctly.

Test Deliverables

Here are some testcases of testing performed on the system

Test Case ID	Prerequisite	Steps	Input Data	Expected Output	Actual Output	Status
TC_01	Softphone/IP Phone	1.Enter Username 2. Enter Password 3. Click Create account	1. username@IP_address 2. Password	Account is ready	Account is ready	PASS
TC_02	Softphone/IP Phone	1.Enter Username 2. Enter Password 3. Click Create account	1. Wrong username 2. Password	Registration failed	Registration failed	PASS
TC_03	Softphone/IP Phone	1.Enter Username 2. Enter Password 3. Click Create account	1. username@IP_address 2. Wrong Password	Registration failed	Registration failed	PASS
TC_04	Softphone/IP Phone	1.Enter Username 2. Enter Password 3. Click Create account	1. Wrong username 2. Wrong Password	Registration failed	Registration failed	PASS
TC_05	Should be a Registered User	1. Dial Number 2.Make Call	1. Extension number	Call connected	Call connected	PASS
TC_06	Not a Registered User	1. Dial Number 2.Make Call	1. Wrong extension number	Extension not found	Extension not found	PASS

Decision Table for Registration Process

Input 1	Input 2	Output
Username	**Password**	**Action**
Correct	Correct	Account is ready
Correct	Incorrect	Registration failed
Incorrect	Correct	Registration failed
Incorrect	Incorrect	Registration failed

Decision Tree

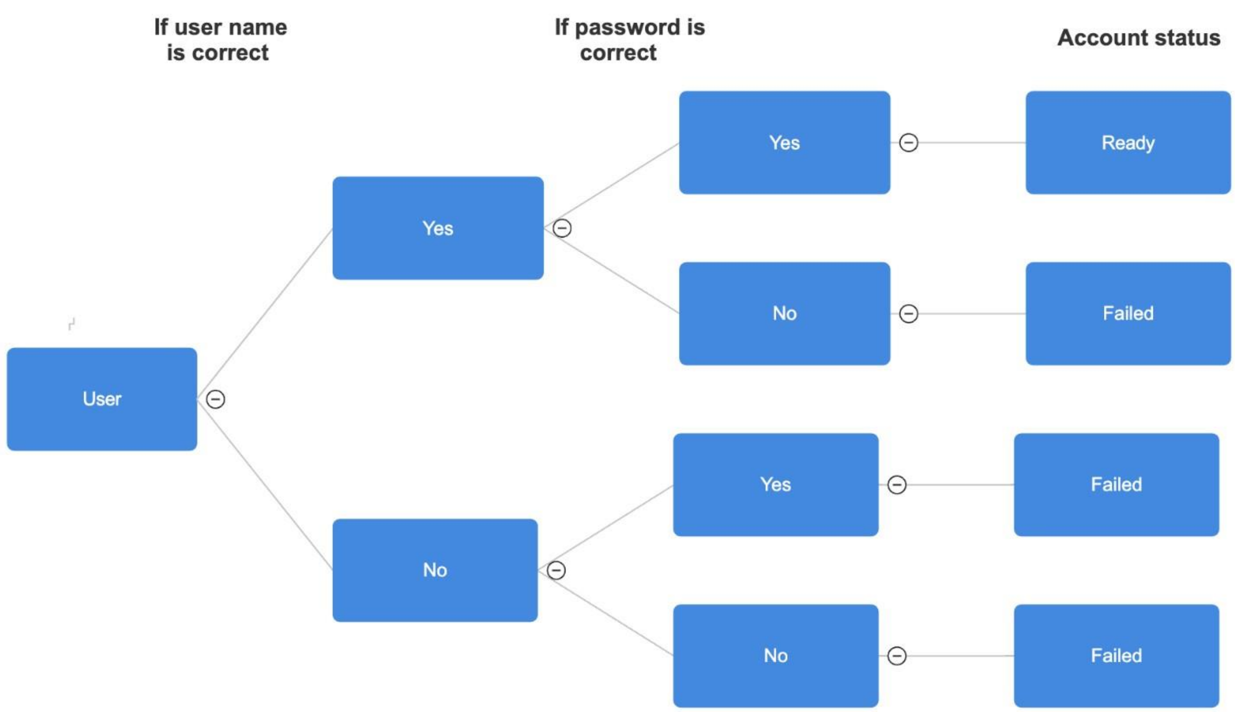

Figure 8 Decision Tree

Equivalence Partitioning		
Invalid	**Valid**	**Invalid**
<7001	7001,7002	> 7002

Resource & Environment Needs

Test Environment

The following Environment is required in addition to client-specific software.

- Windows 8 and above
- Android 5.0 and above
- Mac OS

Terms/Acronyms

Terms or acronyms used in the project

TERM/ACRONYM	DEFINITION
IP-PBX	Internet Protocol Private Branch Exchange
EPBX	Electronic Phone Branch Exchange
OS	Operating System
CIDR	Classless Inter-Domain Routing or supernetting
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
AI	Artificial Intelligence
SIP	Session Initiation Protocol
VM	virtual machine

RESULTS AND DISCUSSIONS

After the successful set up of both the user i.e., 7001 and 7002 they must check for the registration on the asterisk server before making the call.

```
root@iZt4ndmw40otwiyadrk93jZ:~# asterisk -r
Asterisk 18.10.0~dfsg+~cs6.10.40431411-2, Copyright (C) 1999 - 2021, Sangoma Technologies Corporation and others.
Created by Mark Spencer <markster@digium.com>
Asterisk comes with ABSOLUTELY NO WARRANTY; type 'core show warranty' for details.
This is free software, with components licensed under the GNU General Public
License version 2 and other licenses; you are welcome to redistribute it under
certain conditions. Type 'core show license' for details.
=========================================================================
Connected to Asterisk 18.10.0~dfsg+~cs6.10.40431411-2 currently running on iZt4ndmw40otwiyadrk93jZ (pid = 26159)
iZt4ndmw40otwiyadrk93jZ*CLI> sip show peers
Name/username             Host                                    Dyn Forcerport Comedia    ACL Port     Status      Description
7001/7001                 122.161.50.179                           D  Yes        Yes            31231    Unmonitored

7002/7002                 157.37.162.205                           D  Yes        Yes            35998    Unmonitored

2 sip peers [Monitored: 0 online, 0 offline Unmonitored: 2 online, 0 offline]
root@iZt4ndmw40otwiyadrk93jZ:~#
```

Figure 9 Showing Registered Users

Figure 9. shows the registered users and their IP addresses from where the user is using our service. If any of the users are not registered, then a call cannot be initiated between them.

Once the user is registered to our server can call to the desired extension present in our dial plans and extensions.conf file.

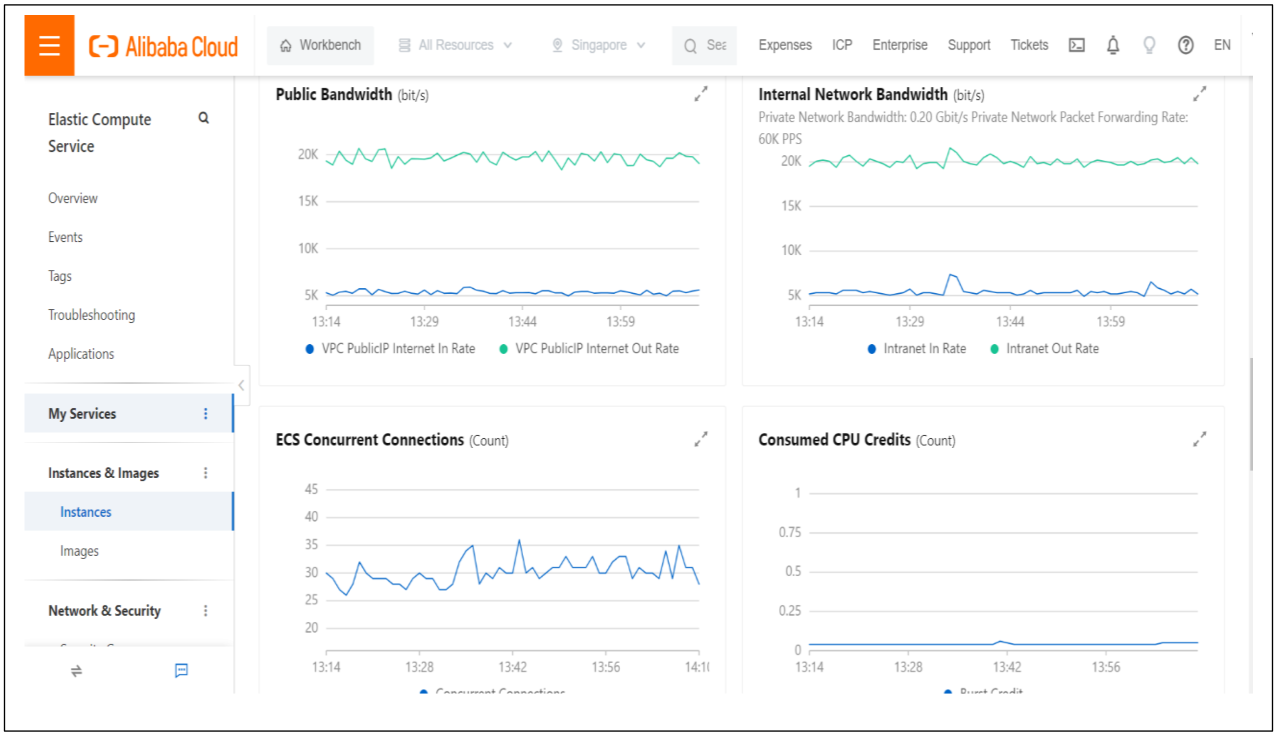

Figure 10 Realtime usage of cloud resources

Figure 10. shows the usage of the cloud where the asterisk server is running inside the ubuntu virtual machine.

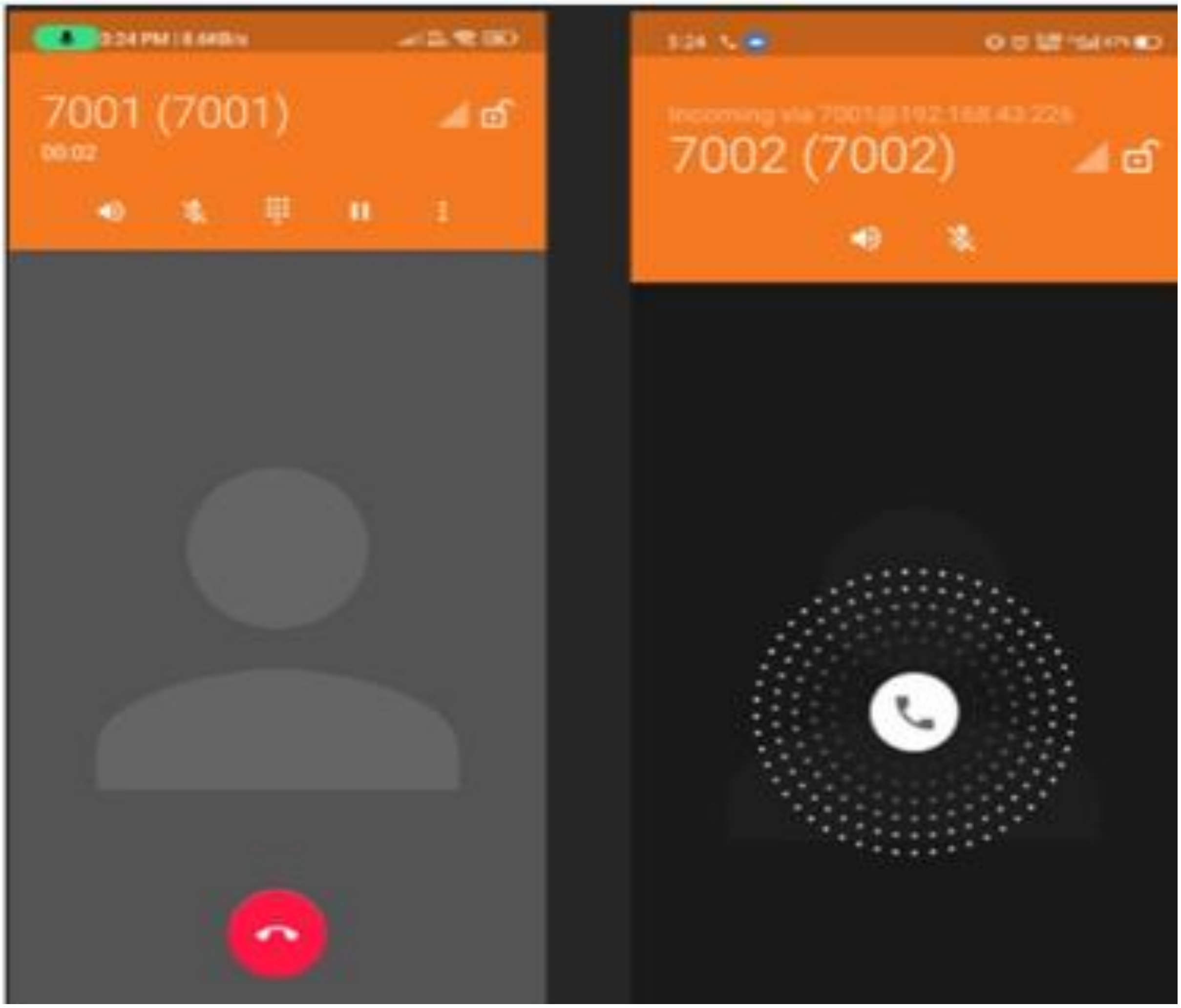

Figure 11 screen of caller and receiver device

Figure 11. shows the call connection between both the users and the sip client that is used for this call connection is Zoiper.

Zoiper is an online softphone which is available for both Android, Windows, and Mac.

CONCLUSION AND FUTURE SCOPE

In this research, the IP-PBX system setup over the oracle cloud and the result of the call connection shows that the quality of the call is good, and the voice is also clear it runs on a secure operating system like LINUX, its much less prone to viruses. The shift from traditional, on-premises PBX systems to cloud-based systems has enabled businesses to transcend geographical boundaries and streamline their operations.

The benefits are clear: reduced capital expenditures, simplified maintenance, and required less hardware and traditional wired connections. Furthermore, the scalability of cloud-hosted IP PBX systems ensures that businesses can adjust their communication infrastructure as they grow, accommodating changing demands with ease.

FUTURE SCOPE

Looking forward, the future work AI is used for automated support and adding security measures with encryption and multi-factor authentication, ensuring scalability and high availability with redundant architecture.

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