DESIGN, PLANNING AND IMPLEMENTATION OF VOICE DATA INTEGRATED SERVICE FOR A UNIVERSITY

By

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This Report Presented in Partial Fulfillment of the Requirements for the Degree of Master of Science in Computer Science and Engineering.

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APPROVAL

This Thesis titled "**Design, Planning and Implementation of Voice Data Integrated Service for A University**", submitted by "**Md. Shamim Badsha**" ID NO: "**213-25-047**" to the Department of Computer Science and Engineering, Daffodil International University, has been accepted as satisfactory for the partial fulfillment of the requirements for the degree of M.Sc. in Computer Science and Engineering and approved as to its style and contents. The presentation has been held on 21-09-2022.

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DELARATION

I hereby declare that, this thesis report paper has been done by me udder the supervision of **Professor Dr. Md. Ismail Jabiullah, Professor, Department of CSE,** Daffodil International University. I also declare that neither this thesis nor any part of this thesis has been submitted elsewhere for award of any degree or diploma.

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ABSTRACT

Voice Data Integrated IP Network is a set of standards for concurrent digital voice, data, and other network services transfer over the IP data network. Over the network, every conversation is transmitted as a data packet. The technology offers robustness and scalability with a healthy dose of worry-free functionality, in addition to enhanced communication features. The IP PBX exchanges local calls in this manner through the company's internal data network, enabling all users to utilize the same external phone lines. In the last ten years, phone and data services through the Internet have replaced the outdated wired media as the primary means of communication. Currently, Voice over Internet Protocol (VoIP) technology is used to provide voice communications over the internet or intranet using packet switching technology, which allows voice conversations to leverage the current Internet protocol. Unlike IP phones, which use a voice frequency of 7 kHz rather than the standard 4 kHz of a regular phone. So, it is clearer than an analog phone. The ability to communicate voice data between individuals via the IP framework is provided by IP telephony, which expands this idea. Enabling voice service via the internet will significantly reduce expenses associated with placing calls using regular telephone lines because it is a cheap medium. This would allow people to communicate with one another at a fraction of what it would cost them to use regular telephone lines. This can be due to the globally accessible, non-uniform Internet services. H.323 and SIP are two signaling protocols that are utilized. Although they are relatively rare, VoIP services can significantly reduce our monthly phone expenditures.

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CHAPTER 1

INTRODUCTION

1.1 Introduction

In today's enterprise network the demand is high in communication and the ability to communicate in all aspects, data, voice and video is an essential part of the communication. Until recently enterprise networks were divided by two main infrastructures one of which is data and the other the voice network [17]. Maintaining data and a voice network is costly and resource wasting. Internet Telephony (IPT), sometimes referred to as VoIP (Voice over Internet Protocol), uses internet protocol to enable voice calls to be made using a new or existing communications network rather than, in most cases, a voice network separated from data. Voice communication can be made possible by encapsulating the voice into the internet Protocol (IP). In other words, voice calls can be made over the same network that carries Data, mail or video whether this network is local-area (LAN) or wide-area (WAN). IPT can be a replacement to the legacy-used telephony system. It can replace a private automated branch exchange (PABX) system

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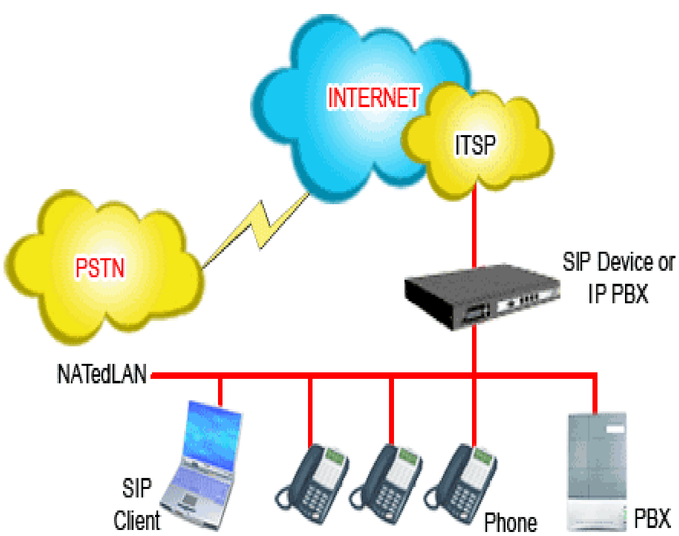

Figure 1.1: Basic idea of IP Telephony

Internet Protocol is the name of the internet protocol that controls how data is transferred between computers on the network (IP). On the Internet and many other networks, TCP/IP—the combination of IP and the Transport Control Protocol (TCP)—is employed. Using IP, computers on a network can have unique addresses. The majority of networks adhere to the IPv4 standard, which has four-byte IP addresses (32 bits). The 16-byte-long upgrade to the Internet Protocol (IPv6) standard (128 bits). IP is specifically the addressing system and the packet format, often known as datagram. A higher-level protocol called the Transmission Control Protocol (TCP) establishes a virtual connection between a source and a destination., is typically used with IP in most networks.

1.2 Motivation

Nowadays, technology has become an indispensable part of human life and among all its inventions; the internet has turned out to be the most significant creation. While I believe that the communication over internet has numerous benefits to the individuals and the society as a whole. As a part of technology, employees use a PBX system to call co-workers at one or more branches for free and make external calls. Unlike standard phone services, which supply one phone line per device, a PBX assigns user extensions so multiple phones can share lines.

PBX business phone systems are a core technology for communications, providing scalability and enterprise-grade features, such as an auto-attendant and business-hours routing. However, costs vary by deployment method, number of lines, and capabilities.

1.3 Rationale of the study

PBX and VoIP have been used for many years. It can be designed to fit into numerous positions inside an enterprise and be many things to many different people. Wherever it is required, Asterisk can be set to work. In the field of telecommunications, Asterisk also enables the integration of current applications. Instead of using their current interfaces, such a Web page or a data terminal, users can communicate with existing apps using telephones. This provides benefits in terms of flexibility and usability. Asterisk is used by both big and small businesses in the current industry [6]. Small businesses can use it to discover a PBX that won't lock them into a vendor or require a substantial upfront investment, while large businesses can use it to take advantage of the existing infrastructure and save money by not relying on the telephone company.

1.4 Research Questions

The goal of this study is to identify the factors that are delaying the widespread adoption of this technology and to inform the reader about the factors that should be taken into account and how to address any issues that may be preventing widespread VoIP adoption.[18] The chance to improve the adoption of this technology is given to the technology's service provider by identifying the main reasons for its delayed uptake as well as solutions for addressing those reasons or fixing those issues. We have the following research question, which is based on the aforementioned goal.

- What is Asterisk?
- Asterisk is developed by which company?
- Asterisk is written in which programming language?
- Asterisk runs on which type of operating system?
- Enlist the features of Asterisk?
- Asterisk supports which type of IP protocols?
- How many is the maximum number of phones supported by Asterisk?
- Does Asterisk support Caller ID from a user database?
- What is SIP in asterisk?
- Explain what is a context?
- Where does Asterisk store its logs?
- How is voicemail set in Asterisk?
- How do you ODBC with Asterisk & MYSQL?
- In Asterisk can reinvite is used for?
- Explain what is a hang-up priority?

By analyzing the above research question, we can support and clarify the research problem.

1.5 Expected Output

- Uninterruptible voice
- Unlimited User Extensions
- Operator Panel for Monitor call activities
- Call Detail Report (CDR) Analysis
- Cost Affordable
- Low cost of hardware maintenance
- Stable, Scalability and Disaster recovery
- CISCO IP Phone Voice and Video Calling
- Call Center solution
- T&T or Traditional Telephone integration

1.6 Report Layout

Chapter 1, Here I written reasons for select the title, how the project will be fulfilled, the motivation of the project, the expected results, etc. are briefly discussed, Chapter no. 1 details the role of the project**.**

Chapter 2, The works shows that have been studied in this area and also findings, Limitations are summarized and so are the research opportunities and challenges.

Chapter 3, The research methodology discusses about research topics and materials, data collection methods, statistical analysis and implementation requirements.

Chapter 4, Results of experiment and results of discussion experiment and descriptive analysis are described**.**

Chapter 5, Conclusion and References.

CHAPTER 2

BACKGROUND

2.1 Preliminaries

The technology that enables telephone conversations to be held via the Internet or a dedicated Internet Protocol (IP) network as opposed to dedicated voice transmission lines is known as IP telephony, also referred to as Internet telephony. Circuit switching and the resulting bandwidth waste are therefore made unnecessary. Only when data needs to be delivered, rather than when packet switching is employed, are IP packets including voice data sent over the network.

Due to its status as the universal transport, IP is now used for practically all international data and video communications. It is progressively taking on the role of voice traffic's infrastructure. Because of built-in IP, today every communications provider is required to have an IP backbone for at least some of its voice services. Furthermore, big businesses are either planning to deploy IP for some internal voice traffic, are already using it, or are setting up test environments.

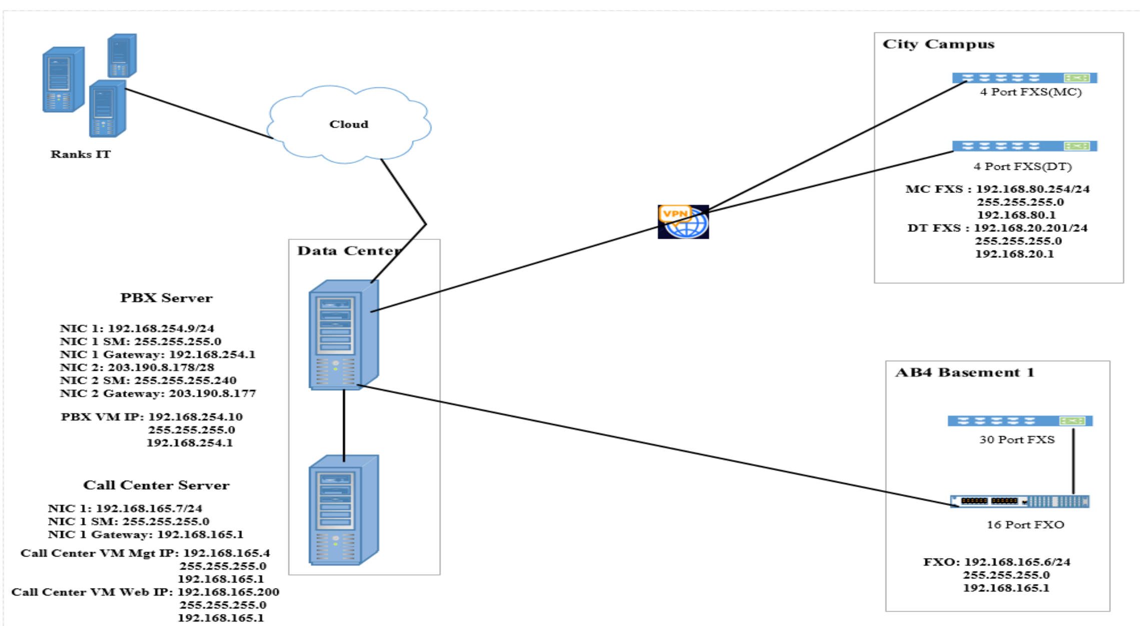

Figure 1.1: DIU IPBX Network Diagram

The two-way voice transmission over a packet-switched IP network is performed using the TCP/IP protocol suite. VoIP (voice over IP) and IP telephony are interchangeable terms. While IP telephony frequently refers to the technology, VoIP is a word that is frequently used to describe the actual services.

IP Telephony Server(s) – This is the fundamental component of IP telephony systems, which provide complete Call Control, Dial Plan Control, and all the essential voice applications (In case of smaller systems, all the functionalities of the below mentioned application servers can also be bundled with this)

Application Servers – IVR (Interactive Voice Response - Auto Attendant), call recording, voice mail, and data base integration are some applications that may need to be housed on separate servers, especially for larger VOIP systems.

IP Phones – These IP Phones include all the voice functionalities previously offered by analog phones, such as caller ID displays, speaker phones, speed dial keys, memory, etc. They link directly to the IP Network (RJ-45 based UTP Cables).

Soft Phones – These are essentially telephony-capable software utilities that make use of a computer, a headset, and a microphone to place and receive calls.

Wi-Fi Phones/ Dual Mode Cell Phones – Wi-Fi phones connect to the wireless network and function as mobile extensions because they are based on IP Technology. Some cell phones can be used as Wi-Fi phones since they come with Wi-Fi adapters (if the manufacturer supports the same) [17]. In order to place a VOIP call, cell phones can also establish a connection with the IP Telephony server using 3G or CDMA networks.

Analog Telephony Adapters (ATA) – These are specialist devices with LAN connections on one end and FXO (Analog Trunks) or FXS (Analog Extensions) connections on the other.

PRI Cards – These are used to link IP Telephony Servers to PRI/E1/T1 Trunk Lines; typically, they connect directly with the PCI/PCI Express Slot in the server.

Computer IP Network – The voice signals are transmitted across the business and, on occasion, even to remote areas, via an IP-based computer network. Comparing the prices of IP phones to analog phones reveals a significant price difference.

The quality of voice calls over IP networks is influenced by a number of factors, including network latency, jitter, bandwidth availability, and the proper setting of QoS parameters.

For IP Telephony services to be available continuously, IP Networks must be established with enough redundancy and security [15]. If the network is subjected to a denial of service attack, for instance, both the computers and the telephones go offline.

It is important to carefully prepare for scaling IP telephony systems; otherwise, the server may not be able to manage a high volume of concurrent calls. Hardware and license restrictions provide a cap on the number of calls that a single server can handle simultaneously and the number of endpoints that can connect to a single server.

2.2 Related Works

The problem of VoIP network security is a broad one that involves many different technologies, organizations, and attack vectors [9]. For instance, [10] looks at an analysis of the vulnerabilities to deal with in an H.323-based IP telephony scenario made up of various PBXs and gateways. The two primary assaults on the VoIP infrastructure that target the PBX, according to the authors, are gatekeeper registration and gatekeeper DoS.

SIP is currently the dominant signaling protocol in VoIP networks, hence there are several research studies in this area. A thorough classification of SIP attacks according to several factors, such as origin and repercussions, may be found in [11], while countermeasures are discussed in [12, 13]. Asterisk's improved SIP security is the subject of some research projects. [14] reports the discovery of a call hijacking attack that jeopardizes the SIP implementation of Asterisk. Threatened are both legitimacy and the inability to retract calls. In [15], a SIP-based protected architecture employing a cryptographic token for authentication is created, according to the idea of Single Sign-On. This proposal calls for the SIP standard to be given a new security layer. To make an Asterisk PBX compatible with the suggested SIP implementation, the authors expand its functionalities.

A defense mechanism against distributed DoS (DDoS) assaults based on captchas for programmable PBXs is proposed in [16], one of numerous research works recently released with the goal of enhancing the security of an Asterisk PBX. Only human users can successfully complete these captchas, therefore calls from zombie VoIP phones are blocked and only authorized users are permitted to access the PBX. [17] develops a strategy for preventing SPIT (Spam over IP Telephony) in Asterisk systems. They implement a real-time caller blacklist that is updated based on information obtained from call detail records (CDRs). Following replays, the duration of the ban for spammers is gradually increased. To further prohibit the IP address linked to spammers, however, using Asterisk ACLs or Linux iptables is not an option.

An intrusion prevention middleware is presented in [18]. Data marshalling and remarshaling are employed to stop data theft while caller and call information are separated from Asterisk and maintained in a separate relational database. [19] presents a secure Asterisk-based mobile VoIP service platform. The proposed secure mobile VoIP platform uses real-time migration and secure voice coding to provide defense against DoS attacks and eavesdropping. Additionally, [20] describes an architecture based on a honeypot to recognize assaults and alert Asterisk. This platform can identify VoIP MAC spoofing, SIP port scanning, and VoIP service abuse attacks. Asterisk PBX is not, however, dynamically reconfigured.

Asterisk has been upgraded to provide an online platform for corporate VoIP risk management [8]. To reduce security assaults, this platform dynamically modifies the PBX's exposure through the use of immediate countermeasures. In order to analyze the network's vulnerability, a risk model is established, and the network is regularly evaluated against it.

When detecting unauthorized login attempts, SPIT calls, and eavesdropping, the platform can take action. The platform dynamically initiates security countermeasures including call redirection, call hanging up, or call holding whenever a possible danger is detected. It is composed of four components: a call monitor for recording VoIP calls, a threat detector for spotting potential attacks, a risk manager for evaluating potential risks to the model, and finally a configuration manager for disseminating security measures to VoIP devices throughout the network.

According to the authors of [8], the performance of risk management and the assessment of the VoIP network exposure may be enhanced by the automatic identification of configuration vulnerabilities. As a result, this article suggests a platform with a proactive approach to finding vulnerabilities. The suggested technology not only has real-time attack detection and response capabilities, but it also proactively assesses the configuration of the Asterisk PBX by parsing its configuration files. This enables the Asterisk PBX to identify dangers originating from dangerous coding practices and defects in the dial plan as well as the configuration files. Furthermore, these risks are reduced by automatically resolving such problems, hence preventing vulnerabilities that would allow possible VoIP network attacks.

2.2.1 Comparative Analysis between Cellular Operator and IP Phone Operator:

Below are the basic differences between cellular operators and IP phone operators

Area	Cellular Operator	IP Phone Operator
Last Mile Connectivity	GSM, CDMA, LTE, etc.	Over IP (SIP, H.323)
User Device	Mobile Phone, etc.	IPPhone, VOIP Supported Device
Core Architecture	MSC, IMS etc.	Class 5 Softswitch
User Authentication	SIM, RUIM	Password or IP
Failover	No	High Availability
Voice Interruption	Frequently happen due to weak signal	As it's running over IP, so it depends on the net bandwidth and redundant connection
Cost	High	Cost effective compared to mobile operator

Table 2.1: Cellular Operator and IP Phone Operator

2.2.2 Compare between IP Telephony VS Analog Telephony

Single Network: One of IP Telephony systems' greatest benefits is this. Traditional PBX require their own network, and building a separate phone network that covers the entire building is rather expensive. With IP telephony, audio calls may be transmitted across computer networks alongside data, and IP phones can be connected directly to network switch ports utilizing Cat 5/6 cables. Most IP phones have a built-in two-port switch, which eliminates the need for additional switch ports and cables for connectivity. Redundancy is a problem, and if the computer network goes down, IP Phones become inoperable.

However, redundancy may be added to IP networks using technologies like Link Aggregation and RSTP.

Inter-branch Calls: If IP Telephony has been implemented in numerous branches (in various locations) of the same company, it is possible to use MPLS Networks/Internet Leased Lines (With Unlimited Usage plans) to transmit voice calls across the WAN IP Network. In this way, the inter-branch calls wouldn't incur any additional fees.

Long Distance Calls: International and long-distance calls can be made via the Internet for less money since SIP Trunks from ITSPs (Internet Telephony Service Providers) can be terminated directly to the IP PBX (Direct Inward Station Access).

Easier Management: Managing the analog/digital PBX is challenging. Some of them are only able to be managed by utilizing intricate CLI commands that are unique to each vendor. However, an IP PBX typically features a GUI (Graphical User Interface) panel that is web-based and can be used to manage, set, and alter many of its functionalities. Users may also be provided with a bespoke web page that they may use to login and set their own preferences, which makes it simpler for administrators to administer an IP PBX.

Soft-Switch: Some IP PBX models come with software that may be downloaded and installed on common computer servers. They are referred to as Soft-Switches. IP PBXs that are based on soft switches offer many benefits. Some open-source-based Soft-switches are available for free download (Like Asterisk, Trixbox, FreePBX, etc).

Cell Phone/ Land Line Integration: You can use a Wi-Fi network to accept landline calls directly on your cell phone by downloading a SIP client that supports this functionality. Therefore, your cell phone can act as an extension of your mobile landline!

Fixed Mobile Convergence: To further the argument made above, a technology known as Fixed Mobile Convergence allows users to seamlessly switch between Wi-Fi and cellular networks. As a result, if you are utilizing Wi-Fi to attend a landline call on your mobile device and then leave the workplace, the call will still be able to go through on a cellular network.

Wi-Fi Phones: Nearly every business has wireless networks (Wi-Fi), which are fairly common. Within the corporate grounds, you can utilize specialized Wi-Fi-based phones to handle landline calls from wherever you are (Wi-Fi Zone). Similar functionality is also provided by the DECT standard used by Digital PBX, although doing so requires the use of a separate digital wireless network. When compared to DECT, VoWLAN (Voice over Wireless LAN) technology has many benefits (Digital Enhanced Cordless Technology).[19]

IP Phones/ Soft Phones: Soft-Phones, which are computer software applications and have a number of advantages over IP Hard-Phones, are supported by IP PBX. Despite being more expensive, IP phones have many benefits over analog phones, including ease of transfer from one location to another (while still retaining the extension number),connecting to the internet using inbuilt browsers, downloading ring-tones, etc. These soft phones allow you to take all of your landline calls from your desktop PC while using a headset and microphone.

Encryption: By utilizing methods like sRTP, IP PBX and IP phones can encrypt communications (Secure Real Time Protocol). The IP Network can be used to prevent hackers from listening to voice calls, despite the fact that this technology is not frequently employed.

IP Faxing: With the help of IP PBX, it is possible to send and receive faxes from a computer directly, among other things.

Analog Trunks/ Analog Phones: Analog Fax Machines and Analog Phones can both be connected to an IP PBX using equipment known as Analog Telephony Adapters, as well as Analog/Digital Trunks from the Telephony Service Provider such as ISDN / PRI Lines / FXO Lines, etc (ATA). The primary benefit of ATA is that it can exist anywhere on the network. For instance, the ATA can be installed in a department where an optical fiber cable links to directly connect the FXO/FXS trunk and subscriber terminals while talking with the IP PBX over the IP Network (Using the Optical Fiber Cable).

Video Calls: Making video calls via the IP network is possible with an IP PBX in addition to making voice calls. This functionality is supported by some IP PBX providers, and video phones can be utilized for this.

Unified Communications: Unified Communications (or UC) is a new area of IP telephony that combines various forms of communication, including voice calls, video calls, voice mail, email, fax messaging, instant messaging, cell phones, and more. It enables users to seamlessly switch between one form of communication and another.

Call Recording: Frequently, you might want to record specific voice calls for your records. Each phone must have its own line connected (in parallel) to the pricey Voice Logger device in an analog/digital PBX system. However, with some IP PBX versions, call recording is an integrated feature that can be turned on by the user at any time through their IP phone or PC interface.

Speech Recognition: Another cutting-edge technology, speech recognition, has been successfully integrated with IP PBX. It identifies voice input (speech) immediately from a caller and takes the appropriate action. For instance, a caller may dial a company's board number and identify the individual they need to speak with, and the IP PBX would immediately redirect their call to that individual's extension after retrieving the extension number from a database.

Meet Me Conference: With Analog/Digital EPABX, there are only basic built-in conference facilities available, and even those are limited to 3–8 party conference calls at a given time. However, IP PBX can provide a Meet-Me Conference (sometimes included as a default option) that enables many users to dial into a conference room to enable multi-conference calls. Such conference rooms may be numerous, and each conference room may accommodate numerous callers. Even a security passcode that allows people to join the conference call could be provided.

QoS & Voice Compression: Since IP telephony uses the same network as computers and other IT equipment, it is frequently necessary to divide the available bandwidth among various kinds of devices. But thankfully, today's IP Network backbone has up to 10 Gigabit

Ethernet bandwidth, which is more than enough for the majority of enterprise applications. Network switches also enable QoS policies to be applied to IP phones and voice-related apps, allowing for the prioritization of voice traffic above data traffic for real-time latency-sensitive applications. For the purpose of compressing voice signals over an IP network, various voice compression CODECS are available.

Remote Maintenance: To make configuration changes and for monitoring reasons from any location on the Internet, IP PBX and IP Phones can both be accessed remotely (with necessary authorization, perhaps using VPN Networks). Analog/digital phones cannot be accessed from a remote location, while the analog PBX might provide some remote access support.

Hosted IP PBX: With an IP PBX, you can register your IP Phones to the hosted IP PBX in their facilities through the Internet and host the IP PBX Software (with a service provider). While charging a monthly charge for the service, the service providers handle the hosted IP PBX's hardware, software, and maintenance.

Voice Mail/ IVR: Although an analog or digital PBX can also provide voice mail capability, there is a time limit on how long voicemail can be stored. IP PBX have larger voice mail storage capacity since they employ servers or computer-based drives to store voice mail. When a new voice mail is recorded, an IP PBX can even email the user a notification. Advanced Interactive Voice Response (IVR) construction / modification can be done relatively easier and the tree structure modified frequently using IP PBX.

Help-desk/Call Center functionalities: IP PBX can provide basic call center/help desk features like Call Queuing, Group Ringing, Automatic Call Distribution, etc.

Database Integration: When users request certain database entries, an IP PBX can be configured to retrieve them when they press the right key combinations in response to an Interactive Voice Response (IVR). Integrating databases (such MySQL, etc.) and IP PBX can result in a variety of useful applications. An application that is made possible by IP PBX through database integration is mobile banking.

Application Programming Interface (API): External software and applications can interface with IP PBX by using the Application Programming Interface, which is provided by many IP PBX suppliers. For instance, the IP PBX can extract recent orders and their status and display them on the employee's screen when a customer calls a company's help desk. This is done because the IP PBX can tell from the caller's phone number that they are a regular customer. By integrating the Customer Relationship Management Application with an IP PBX, this is made possible. API before the employee even picks up the phone.

2.3 Scope of the problem

At beginning of shifting of Daffodil International University campus from Dhanmondi to Ashulia was facing interruption while communication through mobile phones. It was very much difficult to communicate with admin, faculties, staff and students using mobile communication due to very weak signal of cellular mobile network. Moreover, the students couldn’t reach accounts department, admission department and student hall using mobile communication and also higher authority couldn’t get the department wise call logs and reports. Hence, the campus needed a Unified Communication System to solve this issue. I have analyzed the number of users in admin, faculties, staff and student and build a PBX server through which admin, faculty, staff and students can easily communicate with university

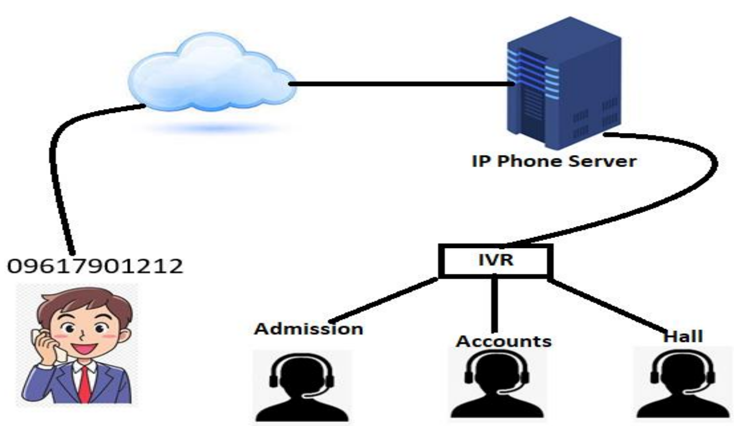

Figure 2.2: DIU Call Center

2.4 Challenges

There are, however, challenges with using this program. For one, configuring the Asterisk engine directly takes a great deal of skill – sometimes taking years to master. To quote Asterisk's official website: "Asterisk is built by and for communication systems developers." That should give you an idea of the level of expertise required to manage this software.

An additional challenge is the lack of support. Most PBX solutions purchased from a vendor will not only have some kind of warranty, but also a team of support professionals that can be engaged should you run afoul of their software [3]. Asterisk comes with no implicit warranty, and unless purchased separately from Digium or a third-party, no support. You're on your own!

Despite these challenges, there are many successful implementations of Asterisk in the wild today, fueling collaboration in industries like software development and telecommunications, and providing an excellent sandbox for aspiring developers to learn the ins and outs of protocols such as SIP and H.323. For those of us looking for something a little more 'pre-packed' though, there are still several options to choose from. Major challenges businesses face and IP PBX helps in resolving

Poor customer service: The companies usually use traditional telephony or a mobile phone. In both cases, a single line can assist one customer at a time. Therefore, businesses either need to distribute multiple numbers, or customers need to wait for their turn if an executive is already on another call. Also, this increases some challenges related to connecting with the right executive and getting round the clock assistance [14].

Privacy concerns: If a business chooses to let its customer care, sales, and other team members distribute their numbers to customers, there is no way to make sure the employees will not leak the information of customers knowingly or unknowingly. This is the biggest concern in a business. The same applies when a business uses a traditional phone line.

Assuring fair usage of telecommunication resources is challenging: With the traditional communication and collaboration solutions, it is not possible for a company to assure that all calls made by executives are related to work. Even all internal communications are work oriented or cannot be tracked.

Work life balance of employees cannot be managed: This happens when a company distributes individual numbers of the team to let customers reach out to them. This can disturb the work life balance of the team.

CHAPTER 3
RESEARCH METHODOLOGY

3.1 Research Subject and Instrumentation

Users of technology, especially those who utilize Asterisk PBX, must consider the substantial implications of being able to transmit and process voice via Internet protocol (VoIP) networks. Many companies are coming to market right now with a variety of VoIP devices that perform a wide range of tasks This essay's primary subjects will be the introduction of VoIP and its implementation using an Asterisk PBX, after outlining the project's objectives and some fundamental VOIP theory.

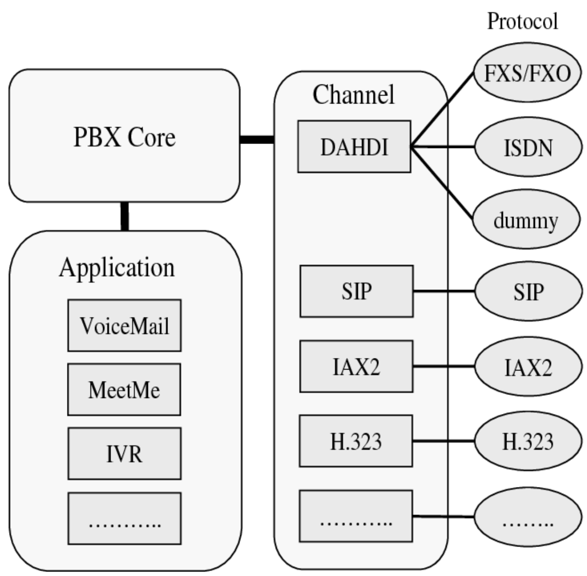

Figure 3.1: Asterisk Architecture

A report on the viability of using the Asterisk PBX as a foundation for conducting VoIP research performance tests is the project's second element. Finally, the project is demonstrated utilizing actual experimental examinations of SIP voice traffic[7,4]. The article's goal is to pinpoint the VoIP technological characteristics that best meet consumer expectations. The study empirically assessed the performance of voice calls made using a SIP simulator to test the SIP protocol's performance and found that Asterisk PBX provided significantly greater stability and accuracy. To show how swiftly this technology is evolving and becoming ready for mass VoIP adoption, the open-source PBX of choice, Asterisk, is used. Asterisk PBX VoIP implementation is possible.

3.1.1 Hardware Requirements

From the calculation of users and concurrent calls I have specified the server computer.

1. CPU: Intel Xeon

   Core: 4

   RAM: 16GB

   HDD: 1TB

   Network Connectivity: Dual LAN connectivity
2. IPPhone Set
3. FXS Gateway (Optional)
4. Analog Phone Set along with ATA device (Optional)

3.1.2 Software Requirements

1. Linux Based Operating System (CentOS7)
2. Asterisk (Voice Engine)
3. Soft Phone (3cx, Zoiper etc.)
4. Web Framework (OmbuTel)
5. IPTSP Number: Buy from IPTSP provider

3.2 Statistical Analysis

In this time almost every industry or organization using IPBX server. Its high price and paid version so need to be purchased. I have created an IPBX server that is free to use. In addition, IPBX and call center servers will run together in one server, which free of cost than others. I have installed Cisco IP phone in this server, also I have done analog telephone integration with this server.

3.3 Proposed Methodology

I have created a new IP PBX server for easy communication on campus. To implement this IP telephony server, I used Asterisk 4.0, an open-source Unified Communications IP Telephony software. I have installed VMware Virtualized Linux Machine inside a physical server, then I have installed Asterisk PBX Software inside that VMware. Asterisk PBX was needed for the Graphical User Interface (GUI) and network address was integrated as part of PBX, PHP, and MySQL Server. I've done everything using command line interface (CLI) and the server's web interface. Below I have given more details about IP Telephony Server Web Interface

The proposed system routes all internal calls across the current LAN (local computer network). This eliminates the need for a separate network just for phone calls. Since IP phones mostly adhere to the open SIP standard, business expansion is not constrained by them. Asterisk, a software implementation of a PBX, is used for IP phone calls instead of a PBX and the SIP protocol is used to place and receive calls. In this, a low-cost method of connecting to desired users via LAN ports is introduced[17]. Hardware requirements, training expenses, and the cost of t elephone services vary depending on whether they are

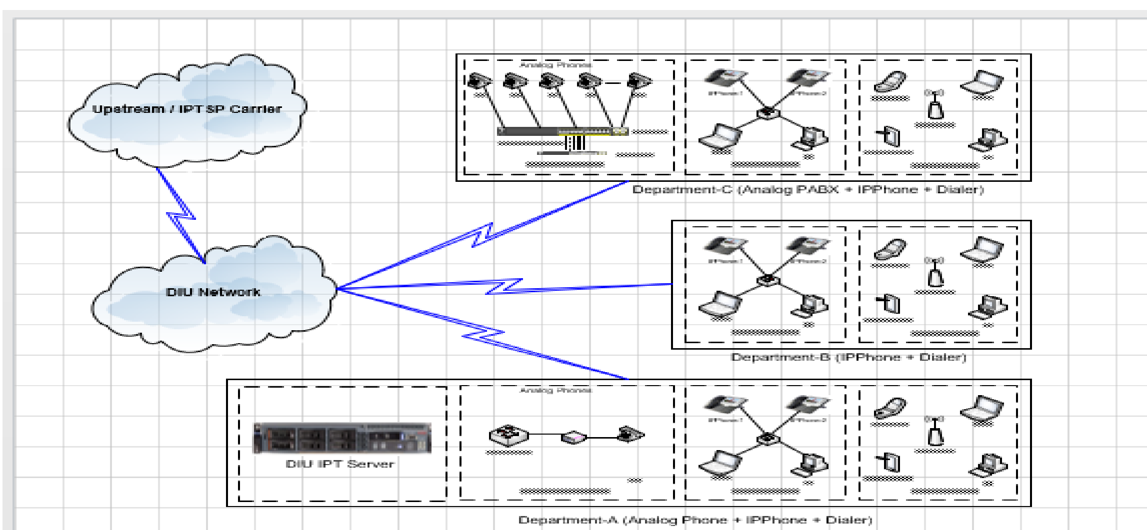

Figure 3.2: Centrex IPPBX Network Diagram

being used locally or internationally. The added capabilities, such as call forwarding and message delivery to a person's mailbox. Block diagram is as shown in the fig.5.

3.4 Implementation Requirements

Asterisk PBX is open-source unified communication system. Its implementation needs some hardware and Software.

3.4.1 Hardware Requirements

Hardware is utilized to support the system and enable it to function. In a virtual machine, the system is put into operation. The virtual machine used has the specifications from the calculation of users and concurrent calls I have specified the server computer.

- CPU: Intel Xeon
    - Core: 4
    - RAM: 16GB
    - HDD: 1TB
    - Network Connectivity: Dual LAN connectivity
- IP Phone Set
- FXS Gateway (Optional)
- Analog Phone Set along with ATA device (Optional)

3.4.2 Software Requirements

Asterisk PBX system requires software to build servers and other supporting components. Asterisk PBX use Issabel 4.0, Call center and Soft Phone Micro SIP.

- Linux Based Operating System (CentOS7)
- Asterisk (Voice Engine)
- Soft Phone (3cx, Zoiper etc.)
- Web Framework (OmbuTel)
- IPTSP Number: Buy from IPTSP provider
- Soft Phone (3cx, Zoiper etc.)
- Web Framework (OmbuTel)

CHAPTER 4

EXPERIMENTAL RESULTS AND DISCUSSION

4.1 Experimental Setup

Open-source Unified Communications PBX software called Asterisk offers web-based configuration, management, and reporting for telephone systems that are user-friendly. Predictive dialing modules can be included. The method to install asterisk PBX is covered in this tutorial.

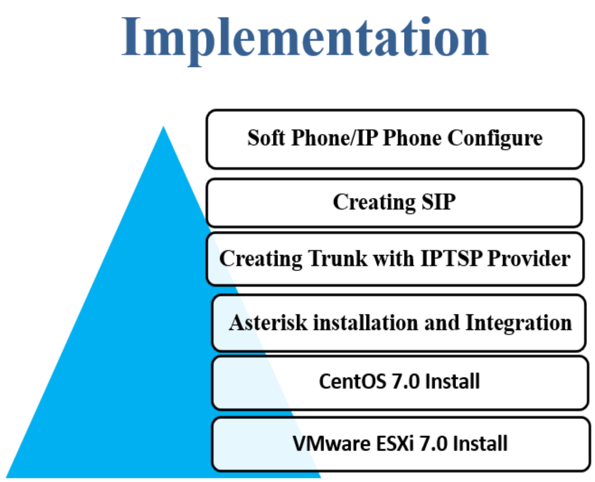

Figure 4.1: Implementation

4.1.1 IPBX WEB Interface

DIU Asterisk Server WEB interface is shown

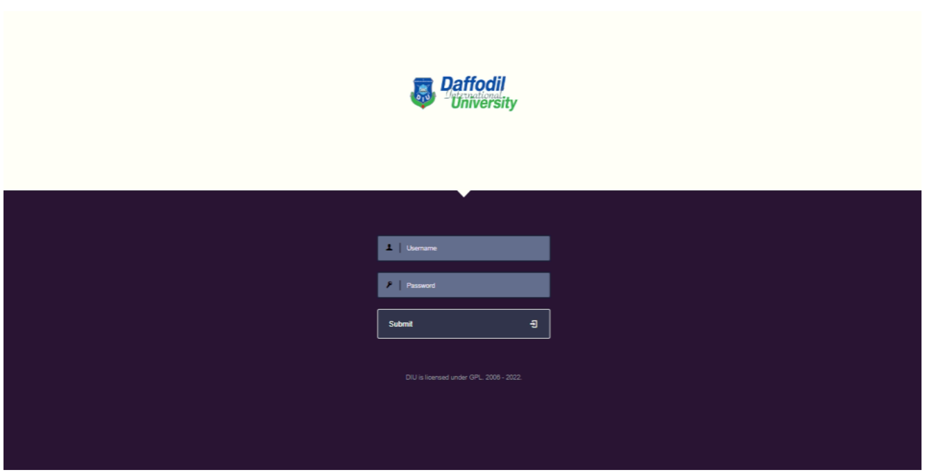

Figure 4.2: Web Interface Page

4.1.2 Create Extensions

For my case I have create some users for test PABX system. Such as 201, 202, 1001, 1002, 1003 etc.

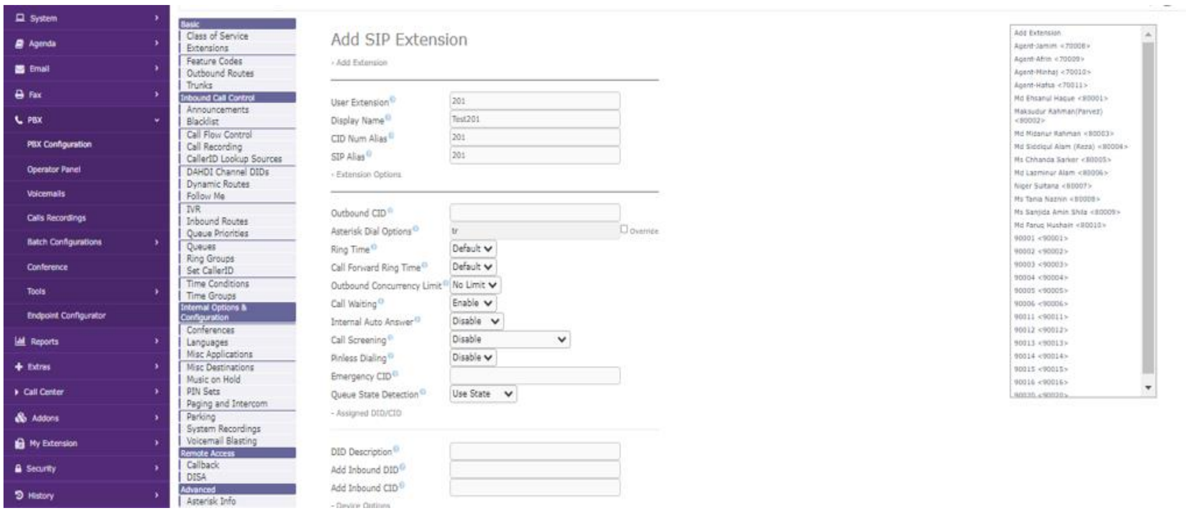

Figure 4.3: Extensions Create Page

4.1.3 Create Conference Room

For my case I have created a conference room. Such as 1212 is our test conference room.

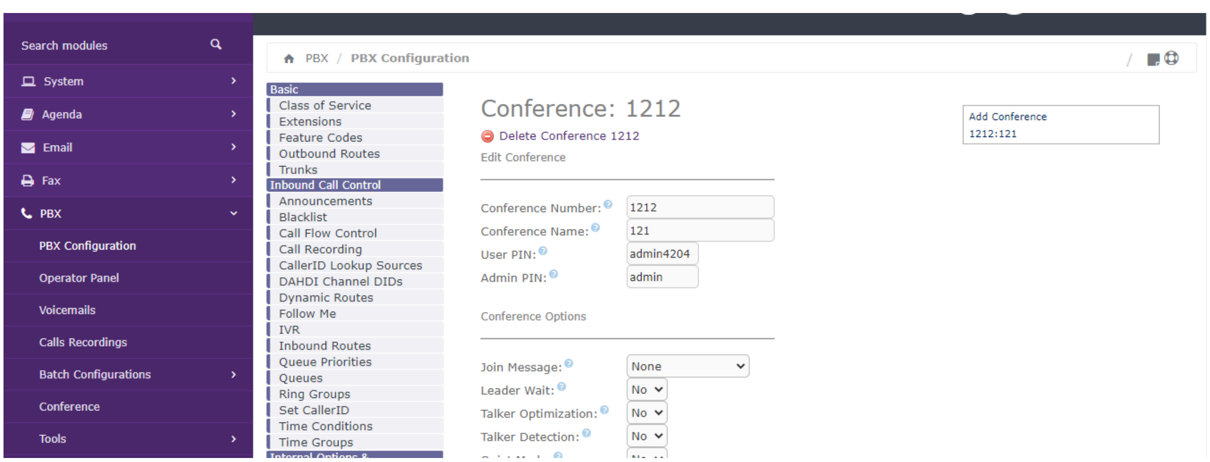

Figure 4.4: Conference Room Create Page

4.1.4 Add a SIP Trunk

For make outbound call to other ANS a have add SIP trunk with an IPTSP provider. For my use RanksITT IPTSP SIP Trunk.

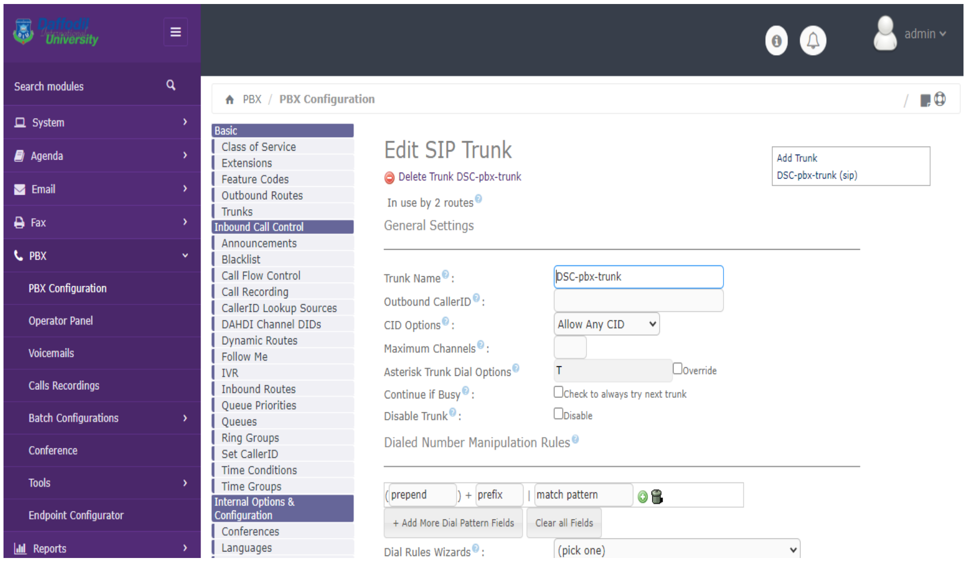

Figure 4.5: Add SIP Trunk Page

4.1.5 PABX Dashboard Panel

The Asterisk PBX module (System Dashboard) offers a system information dashboard that displays data on calls, CPU, memory, disks, the network, and processes.

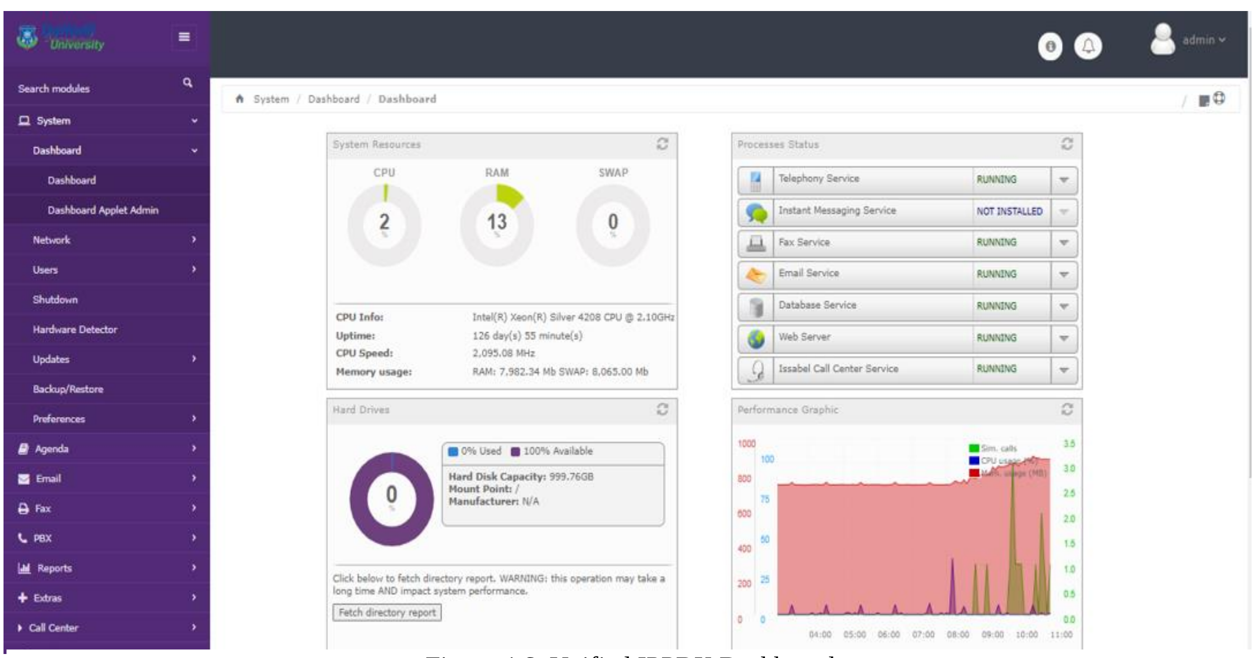

Figure 4.6: Unified IPPBX Dashboard

4.1.6 Asterisk CLI

Using CLI can run asterisk core command from GUI.

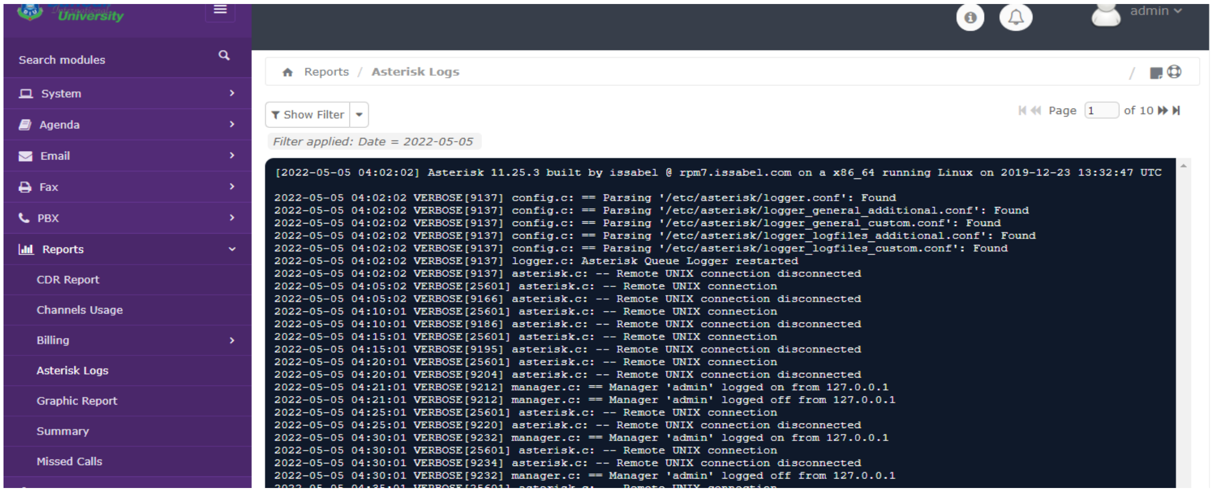

Figure 4.7: Asterisk CLI

4.1.7 Operator Panel

It can show any number of lines per phone and held call status, so you can see exactly what is going on.

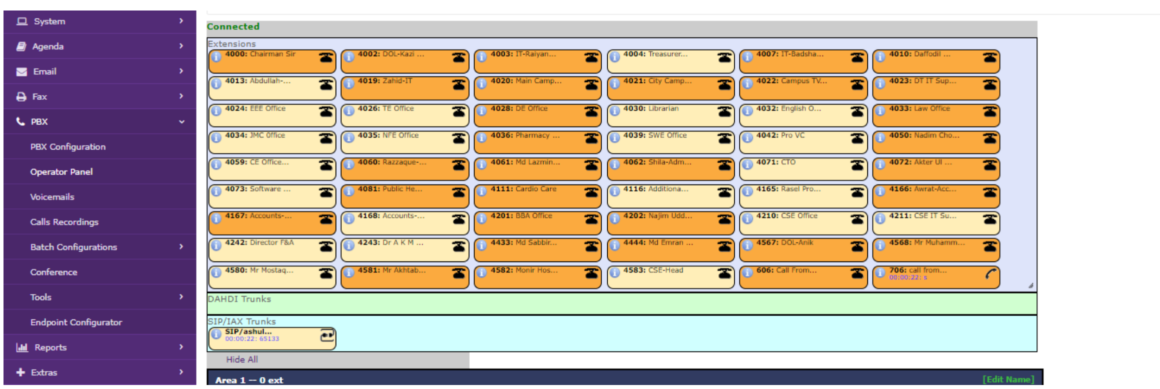

Figure 4.8: Operator Panel

4.1.8 Soft-Phone Zoiper Setup

Here for the 1st user: I configured Zoiper softphone with 121 extensions. Configuration details below. From the above figure, the softphone shows Ready means 121 extension phone has been registered with our server. Similarly, I setup another softphone Micro SIP and the configuration details shown below:

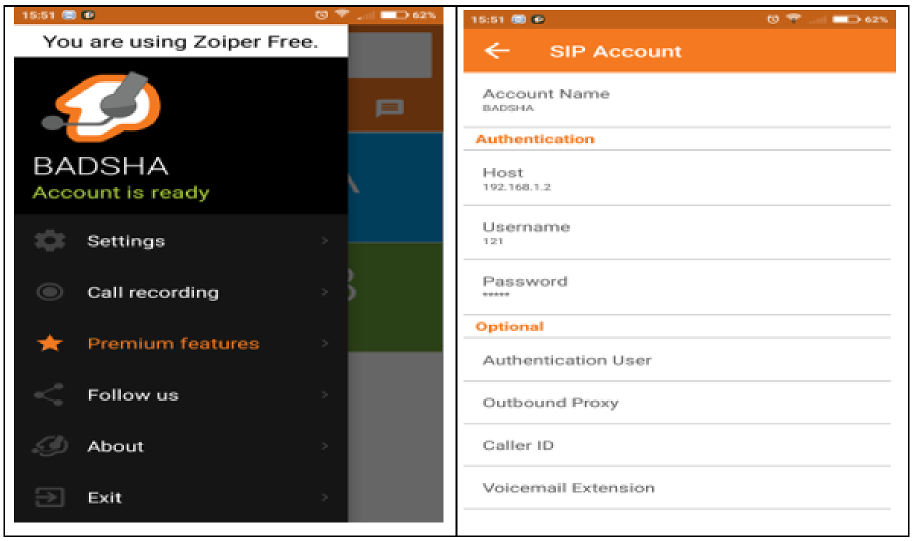

Figure 4.9: Zoiper Setup

4.1.9 MicroSIP setup

From the above soft-phone, I saw that the soft-phone "On hook" means it has been registered on our server. Now I can call between202 to 90020, without any cost

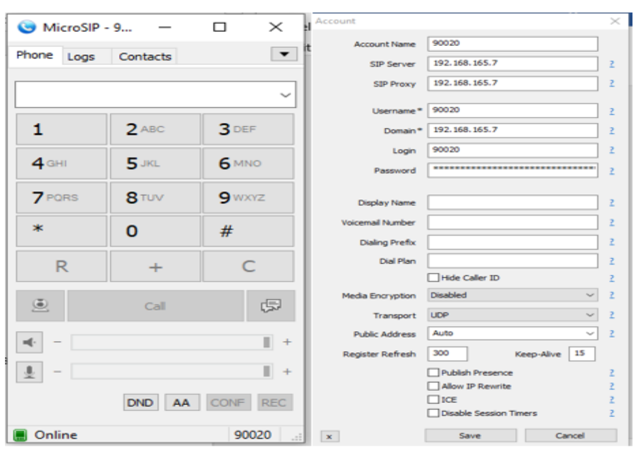

Figure 4.10: MicroSOP

4.2 Experimental Results & Analysis

To evaluate the performance of a PBX system, it is necessary to describe and quantify its workload, average call arrival rate, and average call duration during peak usage. These measurements are employed to gauge how well the system performs during high usage. The Erlang is a unit of measurement for the volume of phone calls made during an hour. This component is typically used to determine the size of a telephone PBX. An hour of continuous voice channel usage is represented by an Erlang according to Equation.

4.2.1 CPU Load test performance

The project is analyzing the server stress load test result.

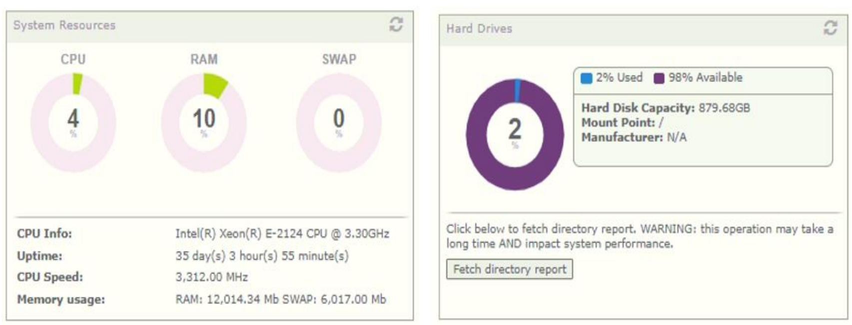

Figure 4.11: CPU Load Test

4.2.2 Network stress load test result

The GSM codec performance cannot be evaluated just using the server stress load because Asterisk refused to provide GSM codec voice to the SIP simulator. Analyzing the results of the Network Stress Load Test is done in the project below.

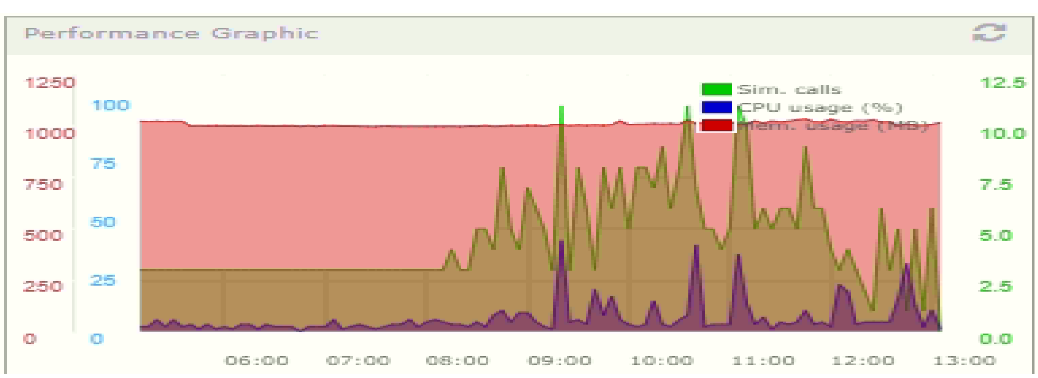

Figure 4.12: Network Load Test

4.2.3 Graphical view Call Per Hour

Incoming and Outgoing Call from Mobile Phone to IPBX server

Figure 4.13: Call Per Hour

4.2.4 IPTSP Incoming and Outgoing Call

IPTSP operator incoming and outgoing Call graph is given below.

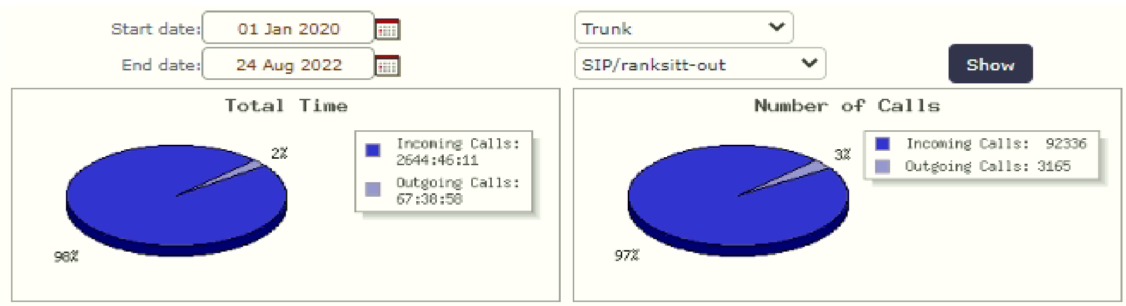

Figure 4.14: IPTSP Incoming and Outgoing Call

4.2.5 Call Center Incoming and Outgoing Call

Call Center Incoming & Outgoing Call graph is given below.

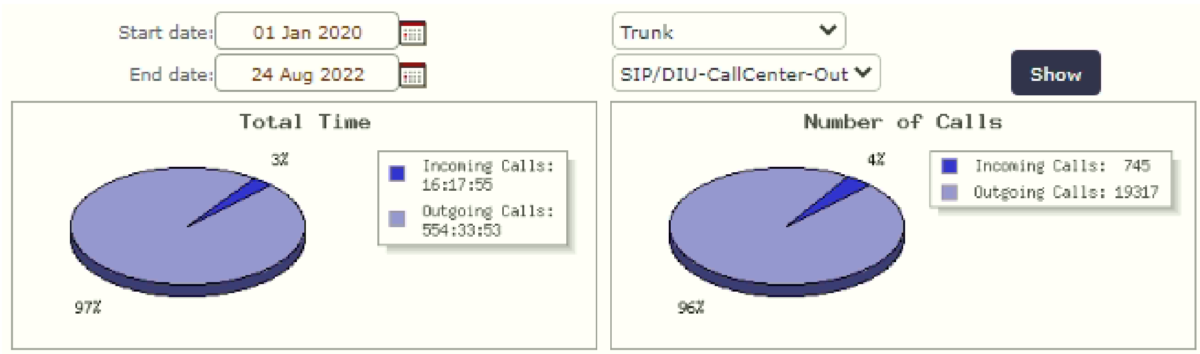

Figure 4.15: Call Center Incoming & Outgoing Call

4.2.6 T&T Operator Call

T&T Operator Incoming and Outgoing Call graph is given below.

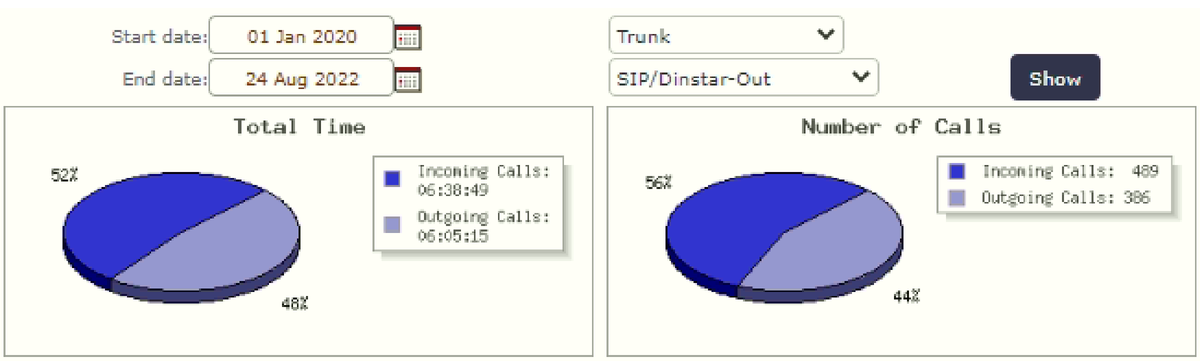

Figure 4.16: T&T Call

4.2.7 Internal Extension Call

The internal extension local call graph is given below

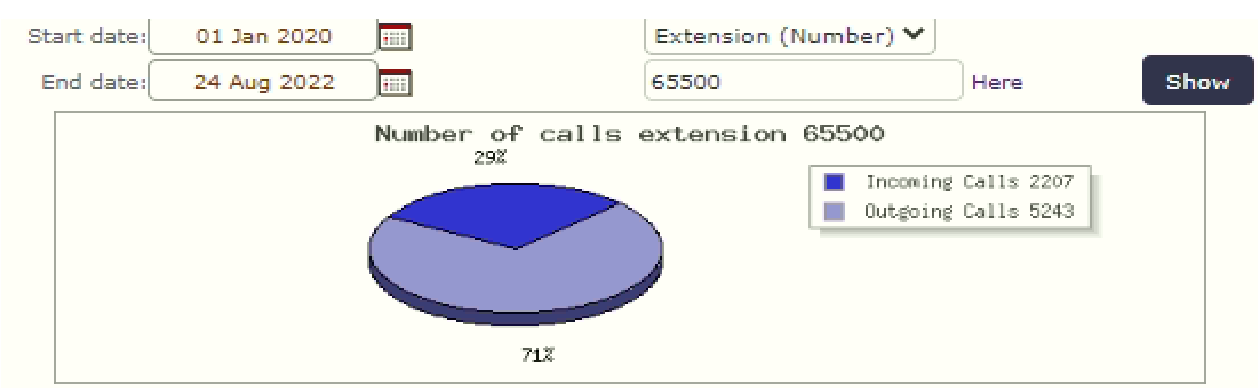

Figure 4.17: Internal Extension

4.3 Discussion

Nowadays, getting PBX and VoIP numbers is not easy. People are seeking an uninterrupted, affordable and easy way of getting PBX extensions and VoIP numbers for their University, Office or home. Our main objective is to develop a web-based portal by which the users will be able to get PBX extensions and a wide range of PBX services without any hassle at an affordable cost and smooth communication.

CHAPTER 5

SUMMARY AND IMPLICATION FOR FUTURE RESEARCH

5.1 Summary of the Study

The suitability of the Asterisk PBX server to offer VoIP communication services with a respectable MOS quality to a sizable user base is examined in this article. In order to achieve this, both analytical and empirical findings are used to evaluate the server's capability. The blocking probability metric is used to gauge the VoIP server's capacity, and the Mean Opinion Score is used to gauge the effectiveness of voice calls. On the basis of the experimental setting, this article demonstrates that the analytical model based on the Erlang-B can efficiently characterize the capacity of the Asterisk PBX, not only in terms of the number of calls being handled by the server but also in terms of the quality of the voice calls [9]. The Asterisk PBX system workload could be accurately described by the model we employed in this study, which was validated. Additionally, the testing outcomes demonstrate that the Asterisk PBX employing SIP successfully handled more than 160 concurrent voice calls in the situation under consideration with a blockage probability of less than 5%.

5.2 Conclusions

In this paper, we have presented a new IP Telephony server-based on-campus of university environment. The suggested remedy offers a new communication tool in which data packet streams are delivered over IP in order to carry all telephony signals. in order for the Daffodil International University to have its own unique VOIP server, therefore can make communications over campus easier and more affordable. This based solution takes into consideration the specific context and the local architecture of the computer network over the campus. It gets along very well with the different devices from traditional phones to softphones, even IP phones, and IP Wi-Fi phones while providing a conference meeting function too. Some tests are accomplished using simulation ways and during the actual phone calls. With the user-friendly management tool, the System administrators can monitor and control servers along with call flows in a first and simple way. For future investigations, securing IP Telephony calls and assuring the quality of service will be among our future works to make this solution more practical to use on a larger scale over campus.

5.3 Implication for Further Study

Asterisk is obviously here to stay and is not a passing fad. The main developer of Asterisk, Digium Inc. in Huntsville, Alabama, estimates that 1 million downloads of the software have already taken place. The corporation claims that 130 business partners are developing Asterisk-based systems globally. Additionally, Digium recently completed its first round of venture capital funding, receiving $13.8 million from Matrix Partners, a venture capital firm that oversees over $2.5 billion in assets and has previously made investments in companies like Sycamore Networks Inc., Apple Computer Inc., and JBoss Inc [13].

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APPENDICES

AAL-ATM Adaptation Layer

ACD (Automatic Call Distribution)

ACM-Association for Computing Machinery

ADC- Analog to Digital Converter

ADPCM-Adaptive Differential Pulse Code Modulation

ADSL- Asymmetric Digital Subscribers Line

AIN- Advanced Intelligent Networks

ARPANET- Advanced Research Projects Agency Network

ASP-Active Server Pages

ATA- Analog Telephone Adaptor

ATM-Asynchronous Transmission Mode

Bps- Bits per second

CEO-Chief Executive Officer

CTI -Computer Telephony Integration

DSL-Digital Subscriber line

DTel-Deepija Tele networks

H.323- Application Protocol by International Telecommunication Union (ITU)

IEEE- Institute of Electrical and Electronics Engineers

IETF- Internet Engineering Task Force

IP- Internet Protocol

IP/PBX-Internet Protocol Public Branch Exchange

IPv4 & IPv6-Internet Protocol version 4 and 6

ISDN-Integrated Service Digital Network

ISP- Internet Service Provider

ITU- International Telecommunication Union

IVR- Interactive Voice Response

MS- Milli seconds

MIT- Massachusetts Institute of Technology

NAT-Network Address Translation

OSI (Open Systems Interconnection)

PBX- Public Branch exchanges

PC-Personnel Computer

PCM- Pulse Code Modulation

PPP-Point to Point Protocol

PSDN- Packet Switched Data Network

PSTN- Public Switched Telephone Network

QoS- Quality of Service

RFC- Request for Comment

RSVP- Resource Reservation Protocol

RTCP-Real Time Control Protocol

RTP- Real Time Protocol

RTSP-Real Time Streaming Protocol

SIP-Session Initiation Protocol

TCP-Transmission Control Protocol

TTS- Text to Speech

UDP-User Datagram Protocol

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