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Design a Real-time Communication System using 3CX Software-based Private Branch Exchange Phone System on Raspberry Pi Device

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Authors' contributions

This work was carried out in collaboration among all authors. All authors read and approved the final manuscript.

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ABSTRACT

With the advancement of technology, the telephone network has become the primary mode of communication worldwide, and private businesses have increased their reliance on telephone communication. Many organizations choose to establish their own service in order to manage internal calls. Voice over Internet Protocol (VoIP) is one of the emerging technologies that may provide low-cost service with high-quality and availability. VOIP technology enables the transfer of multimedia data such as audio and video. While some VoIP services require a computer or a dedicated VoIP phone, others allow you to make VoIP calls using your landline phone via a special adaptor. Rather than using a traditional private branch exchange (PBX), we used a Raspberry pi, which is a set of credit card-sized single-board computers, as a server for handling voice and video call communications over a wired or wireless LAN network while monitoring the entire system. Wireshark is a software application that is used to capture packets in a network and present information about certain packets in as much detail as possible.

Keywords: VOIP; PBX; wireshark; Internet of Things (IOT); SIP; mean opinion score.

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1. INTRODUCTION

The introduction of Morse code, single-line transmission, and later wired telephones brought in the modern era of communication. A network was built by multiplexing several such telephones on a single line. The age of wireless communication via cellular phones began after that. Now, the Internet of Things (IoT) will change the face of telecommunication [1].

Due to the increasing development of Internet availability (and subsequent price drops), organizations and end users alike are increasingly relying on IP-based technologies to meet their communication needs. One of these requirements is voice communication via telephony, which was one of the first and most widely deployed services over information networks [2]. The majority of private businesses rely on these telephone networks to operate, as they would cease to exist without them. As a result, a significant number of them have chosen to operate their own telephone service. So, they could manage their own internal calls. In the meantime, voice telephony is available via a mobile service provider such as GSM or via an IP service provider at a lower cost. [3]. Many businesses use a PBX (Private Branch Exchange) to communicate internally, which routes calls between organization users. The primary goal is to save money; the PBX is controlled by the organization, not the telephone company, and all telephone lines terminate at the PBX. In addition, a PBX has a computer with memory that manages call switching for an organization's users [4].

IP-based communications network technology, such as Voice over Internet Protocol (VoIP), has been developing as one of the most effective telecommunication technologies (VoIP). VoIP is an IP-based communication system that allows users to make voice calls over the internet. The analogue speech data is transformed to digital data before being packetized and sent via a packet-switched network [5]. Using VoIP has several advantages. VoIP is less expensive and more flexible than traditional telephones since it may be placed at any Ethernet or IP address [6].

3CX is indeed a software version of a telephone private branch exchange (PBX) that allows telephones to connect to telephony services such as the public switched telephone network (PSTN) and voice over Internet Protocol (VoIP) via a range of hardware technologies. An inexpensive single-board computer (SBC) like the Raspberry Pi can be used to run a system like this. It can then be used to run the telecommunications service [7].

2. VOIP

In 1995, VoIP innovation began. Around that time, specialists began to recognize the potential of transferring voice information bundles over the Web rather than over traditional phone frameworks [5]. VoIP is an acronym for Voice over IP (Internet Protocol), and it refers to a range of techniques for creating two-way multi-media communications over the Internet or other IP-based packet switched networks. VoIP serves two purposes: The first benefit is a savings in telephone charges due to the call being transmitted as packets across data lines or the internet. The second objective is to provide flexible voice networks, such as those that support many calls over a single physical link. Over the Internet, applications for real-time VoIP communication have gained significant use [8]. Audio and video data transmission. VOIP systems, in general, send voice information in digital form in discrete packets rather than the traditional circuit committed protocols of the public switched telephone network (PSTN) [7].

To facilitate the transmission of audio data over the Internet, the continuous audio stream is segmented. These packets must navigate the Internet independently, seeking bandwidth with packets from all other Internet applications. These packets must be efficiently transported in order to achieve high data throughput with minimal latency. Typically, huge buffers are employed to produce a high data flow, resulting in a simulated delay of the packets. Thus, by applying a data compression algorithm, the number of bytes can be minimized. However, most audio compression methods have a significant inherent algorithmic delay, resulting in excessive audio latency. Even though excessive latency isn't a big deal for normal VoIP, it's very important for some applications [9].

A significant advantage of VoIP and Internet telephony is that they eliminate the fees associated with traditional telephone service. When you start an Internet protocol telephone connection, the analogue voice signal is converted to digital format and the signal is compressed for transmission over a network or the Internet. At the other end, the process is reversed [7].

3. PBX

An organization's internal phone system is known as a PBX, or private branch exchange. The PBX phone system enables internal communication among its users. Multiple customers' lines are terminated in a PBX to reduce phone call costs rather than having individual phone lines with their telephone service provider. An organization retains ownership of the PBX rather than the phone company [4].

4. SIP

The Session Initiation Protocol (SIP) is one of IP's supporting protocols, which is a network layer protocol and communications protocol used for call initiation and termination [4]. SIP is the most widely used protocol for voice communication and is located in the OSI model's applications layer. SIP has the same features as the HTTP protocol and can be used in a variety of web services or e-mail. TCP and UDP are two IP protocols that are used to transport SIP messages over the transport layer [10].

5. INTERNET OF THINGS

The Internet of Things (IoT) is regarded as a critical enabler technology for the future. The Internet of Things refers to the billions of devices that are linked to the Internet. In the Internet of Things, a thing is a smart object that can be given an IP address and the ability to share data over a network. Smart objects typically have limited computation and memory capabilities and use low-power to operate and communicate [11]. Information can be exchanged and exchanged through the interconnection of various goods and networks, such as sensors, brilliant meters, advanced cells, and intelligent cars. The interconnection of multiple devices enables advanced IoT applications such as item tracking and weather forecasting. This technology could also be used for patient surveillance and energy management for CEOs. Ready Home is an Internet of Things application that allows customers to open their garage door, relax, order a coffee, and automatically control lights, TV, and other devices. IoT is used in smart cities, light networks, healthcare, transportation, mechanical robotization, and disaster response [12].

6. RASPBERRY PI

The technology can use a collection of minicomputers as a server and an access point that serves as a network provider, making it flexible and portable, similar to the Raspberry Pi, which is a minicomputer product for a variety of purposes. Despite its small size, this device performs well. The device is compatible with a variety of operating systems, including Linux, Windows, and even Android. When compared to other types of computers, the Raspberry Pi is superior in terms of size, making it portable and requiring lower wattage. As a result, the Raspberry Pi can be used as a VoIP server while being powered by a battery [13].

A small credit card sized single board called the Raspberry Pi, which is a standalone system and does not require any single PC to take or receive the data from one user to another user. It supports live audio streaming as it has an onboard audio codec. It is possible to connect to another user by just accessing the IP address of another user [14]. As shown in (Fig. 1, the Raspberry Pi is a microcomputer framework created by the Raspberry Pi establishment in the United States that is used for training and research in basic software engineering. The Raspberry Pi 4 Model B is powered by a 1.5 GHz Broadcom BCM2711, quad core Cortex-A72 (ARM v8) 64-bit SoC, and has on-board 802.11ac Wi-Fi, Bluetooth, and USB ports, as well as a gigabit Ethernet port and 4GB of LPDDR4-3200 SDRAM. The operating system is installed on a micro-SD card, which serves as the computer's hard disk. A minimum of 16 GB of memory is required to install the operating system as well as the 3CX Server packages. The framework has an Ethernet port as well as Wi-Fi to connect to the association's system. The system has an HDMI port for connecting the monitor and a USB port for connecting the keyboard and mouse. The system is powered by a mobile charger and runs on a 5V/2A 2amp supply. A device with such low power consumption can also run-on batteries to avoid power outages. [15]. The Raspberry Pi is used because it is less expensive, has more functions, is portable, and is compatible with a wide range of open-source software when compared to other embedded devices on the market [16] [17].

7. WIRELESS ROUTER

Even though many wireless LAN technologies and standards were developed in the 1990s, one class of standards has clearly emerged as the winner: the IEEE 802.11 wireless LAN, also known as Wi-Fi [18]. A wireless router is an electronic device that performs routing functions

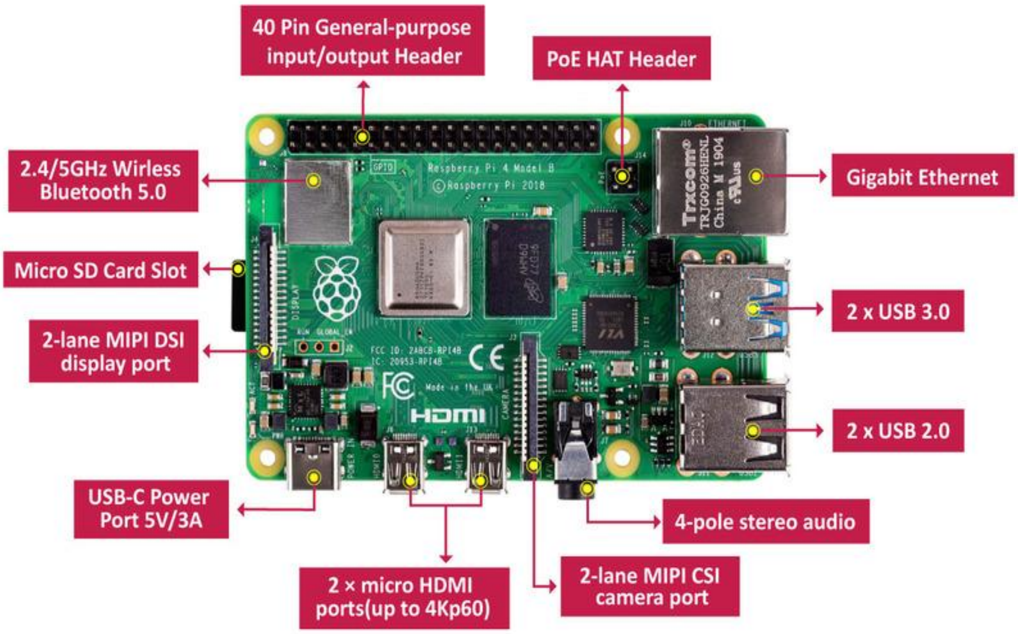

Fig. 1. Raspberry Pi 4 Model B Architecture

and may also perform the functions of a wireless access point. It's used to connect to the Internet or a private computer network. Depending on the manufacturer and model, it can operate on a wired local area network with a limited number of LAN communication ports. It supports wireless-only LAN as well as mixed, wired, and wireless networks. The primary goal is to establish a network [7]. A data packet is typically routed from one router to another across the networks that comprise the internetwork until it reaches its destination node. A router is a device that connects two or more data lines from different networks. When a data packet arrives on one of the lines, the router reads the address information to determine the ultimate destination. The packet is then routed to the next network on its journey using information from its routing table or routing policy. This results in the formation of an overlay network [19].

8. WIRESHARK

There has been a rise in attacks over the internet that try to hack or illegally change data that is available over networks [20]. Wireshark is a piece of software that can be used to record the packets in a network and show as much information about them as possible. Wireshark shows the time, source and destination IPs, communication protocol, packet length, and other information. The tool Wireshark also shows information about delays, packet drops, and jitter, among other things. Based on these statistics, the network's performance can be looked at and judged. The information can also be used to troubleshoot the network, investigate security issues, test network applications, and investigate protocol implementation. The information can be taken from different types of network adapters, such as Ethernet, Wireless Local Area Network, Bluetooth, and USB to get the information [21].

9. NETWORK PERFORMANCE PARA-METERS

Before the data packet reaches its destination, it may encounter some disruptions. Thus, network performance metrics such as throughput, latency, jitter, and packet loss should be monitored to determine the network's quality [13] [9] [22]. The International Telecommunication Union-Telecommunication (ITU-T) G.114 recommended standard for network performance is shown in Table 1.

Table 1. ITU-G.114 Recommendation Standard

Call Quality	MOS
Best	Over 4.34
High	4.03 – 4.34
Medium	3.6 – 4.03
Low	3.1 – 3.6
Poor	Below 3.1

This way of testing a VoIP system's call quality is called the MOS test, which stands for the mean opinion score, or MOS. Over time, the number of extensions/users and simultaneous calls will decide if a call is declining. The MOS score is based on how users rate a stimulus in a subjective quality evaluation test.

$$MOS = \frac{\sum_{n-1}^{N} R_n}{N}$$

Where R are the ratings for a single stimulus made by N people. In the next Wireshark capture, you can see how the signaling protocol SIP and the voice packaging protocol RTP work together. The RTP packets are decoded and played back to get a good MOS score for the captured packets.

Any network's dependable functioning ensures that services are available to clients. As a result, several factors must be measured in order to determine the system's network performance. Jitter, packet loss, and latency are the parameters.

- **Delay** is an accumulation of delay times from sender to receiver.
- **Jitter** (delay variation) is the variation of arrival time intervals between packages in the receiver device. The difference in the arrival time can be caused by some factors, such as the capacity of the network and congestion.
- **Packet Loss** is the overall loss of packets when sending packet data between source and destination. The packet loss can be caused by collisions, the full capacity of the network, or packet drops caused by the endless TTL packets. The standards of (Delay, Jitter, Packet Loss) by ETSI: TIPHON can be seen in Table 2.

The European Telecommunication Standards Institute (ETSI) has developed QoS standards for telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) (ETSI). The table depicts the five levels of communication transmission quality. The highest network quality is perfect quality, which occurs when clients communicate normally over a network. The modest latency (251), jitter (76 ms), and packet loss (4%), ensure that the clients communicate effectively. As the values of the parameters increase, the network's service quality degrades from perfect to poor [6].

10. RELATED WORK

Ashwini S.Gawarle suggests installing a telephony program that uses WIFI in p2p (Peer-to-Peer) or WLAN (Wireless Local Area Network) mode as a free means of communication between mobile phones. To convert phone numbers into IP addresses, the asterisk software will use a correlation between current address books available in mobile phones. The system will allow the user to converse via voice. The current system supports only one call per connection and does not support call waiting or conference calls. Various VoIP-related security services are presented [19].

Table 2. ETSI: TIPHON Standard for QoS

Quality	Delay (ms)	Jitter (ms)	Packett Loss (%)
Perfect	0-150	0	0
Good	151-250	1-75	1-3
Fair	251-350	76-125	4-15
Poor	351-450	126-225	16-25
Bad	>450	>225	>25

DS Pandithage discusses, in a simplified form, a system developed with the use of VoIP, a prototype that could be used as a communication method between a couple of IP phones or PCs with VoIP enabled software using the raspberry pi. It also examines the benefits and drawbacks of VoIP, as well as some aspects of VoIP applications [3].

A. Muntaka propose the use and power vested in the open-source asterisk PBX to deploy a prototype of a communication system for Garden city university college, using open source liblinphone library and Software development kit to create a softphone for both inbound and outbound communication among staff and the university administration. The prototype of the system is based on a Raspberry Pi. The same project can be recreated on a powerful computer architecture for considerably better capacity of communication. Evaluation of the system is done to ascertain information regarding the project. The evaluation of the prototype assists in constructing similar system for industrial use [7].

P. V. and V. M. Deshmukh The raspberry pi board and VoIP protocols are used in the proposed hardware. There is a protocol called SIP, which is used to start a new session. It also has a protocol called IAX/IAX2 called SIP2 (Inter Asterisk Protocol: Open-source trucking protocol). For free, the system creates a small intranet that can be used by any company to communicate with each other. Programming for the raspberry Pi is done at the kernel level of Linux, so it can be used to run things like web servers. The system makes use of free telephony software like Asterisk and Free-PBX to make codes, which makes the system more cost effective.

Dr. H D Phaneendra and Mrs. Sowmya offer a system that is based on the raspberry pi and utilizes the SIP protocol to originate and terminate phone conversations, hence making the system cost effective, scalable, and customizable [4].

The system proposed by G. Aljabari is composed of hardware and software components. The hardware is based on the Raspberry Pi [12], which enables the implementation of a low-cost smart VoIP system. The Raspberry Pi is a credit card-sized computer with physical world interaction capabilities that has been utilized in a wide variety of automation applications [11].

G. P. and J. Soundararajan suggest the design and implementation of a high-compression rate Voice over Internet Protocol (VoIP) system. The device is powered by a Raspberry Pi B+ ARM 11 microcontroller, which is capable of processing audio signals more quickly with the use of CODEC2 software. The Raspberry Pi is interfaced with an Audio Adapter (CM-108 or 109) that functions as a bridge, supplying the processed voice signal to the Raspberry Pi in the appropriate PCM format. The configuration is implemented in a Local Area Network (LAN), which is used for data transfer. Throughout the data transmission, the wireshark program was utilized to analyze the network traffic [9].

W. Wardi, A. Ahmad, and D. Aprianto propose a research project aimed at developing portable wireless mesh networks using IP PBX technology. The system is powered by a combination of solar cells and batteries. The system makes use of the Asterisk FreePBX software as the configuration medium for the server, while the clients use laptops and smartphones [13].
According to M. S. Meshram, P. Thakare, and P. Dandek's article, building a completely working voice exchange entails setting up an Asterisk server, connecting clients to the server via softphones, and then configuring the softphones via a server [23].

11. DEVELOPMENT TOOLS AND PLATFORM

The performance and output that a system is expected to deliver have a significant impact on its design. As a result, the efficiency of a system is determined by factors such as hardware reliability and software processing. This lays the groundwork for the development of software and systems. The design of the system has a significant impact on efficiency, performance, and maintainability. The tools and softwires utilized in the project were as follows:

- 3CX PBX Account
- Raspbian image OS (for raspberry pi)
- Micro SD card as storage Disk for Raspberry Pi
- Enther Tool for extracting Raspbian OS to Micro SD card
- Raspberry pi 4 model B
- USB cable for power
- Wireless router
- Network cables

- Wireshark For Monitoring and Analyzing System Package

12. DESIGN AND IMPLEMENTATION

The goal of this study is to determine how well the Raspberry Pi and 3cx IPBX system perform as a communication exchange in terms of mean opinion score (MOS). This research will also be optimized, and it is expected to be used as a reference and alternative communications device.

In order to use the system, A diagram showing the process of work as shown in (Fig. 1) is followed by making an account on the 3CX website, which gives us a year's worth of free VOIP services. Next, we must download and install Raspbian Image OS on a Raspberry Pi 4 Model B using Ether Tool. Finally, we must assign the Raspberry Pi a static IP address.

Then we need to setup the IP phones through the 3CX PBX terminal with the IP address given to the pi as shown in (Fig. 3), giving the phones separate names so that it is easier to identify each and giving the IP phones static IP addresses so they won't change when the network is disconnected and connected again (for assurance). We can also use this PBX system with PC's by using a software that gives the VoIP service over calls (e.g. ExpressTalk), install it on your PC and set it up with the IP address of the Raspberry pi, so that you can call any phone or PC that is connected to the pi and the same network. One thing that we should also do, is give a common reference number to the connected devices so that any device can call another using this reference number. This number should also be given during the setup process.

By using the proposed system shown in Fig. 4, all internal telephony is routed through the existing LAN (local computer network). This way, a separate network for telephony is not required. Since IP phones mostly use the open SIP standard, it doesn't limit the growth of a company. The IP phone call uses the Raspberry Pi and replaces PBX with 3CX, which is a software implementation of PBX and uses the SIP protocol to initiate and terminate the calls. This introduces a low-cost solution to connect to a desired user by using a LAN port. Costs include hardware requirements, training costs, and the cost of telephone services based on whether they are working at an international or local level. The extended features like call forwarding, sending messages to a person's mail box.

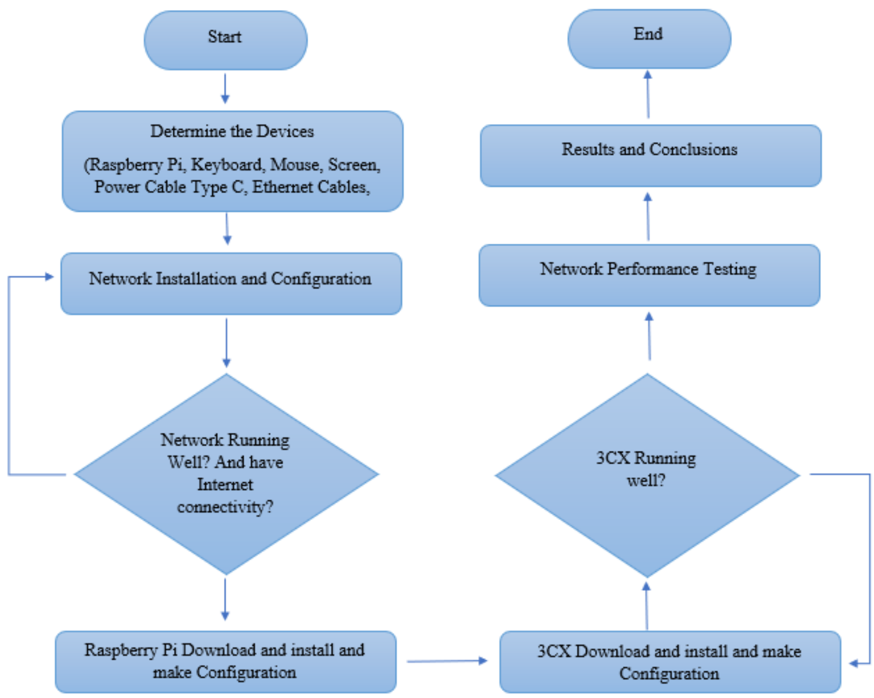

Fig. 2. Diagram of system design

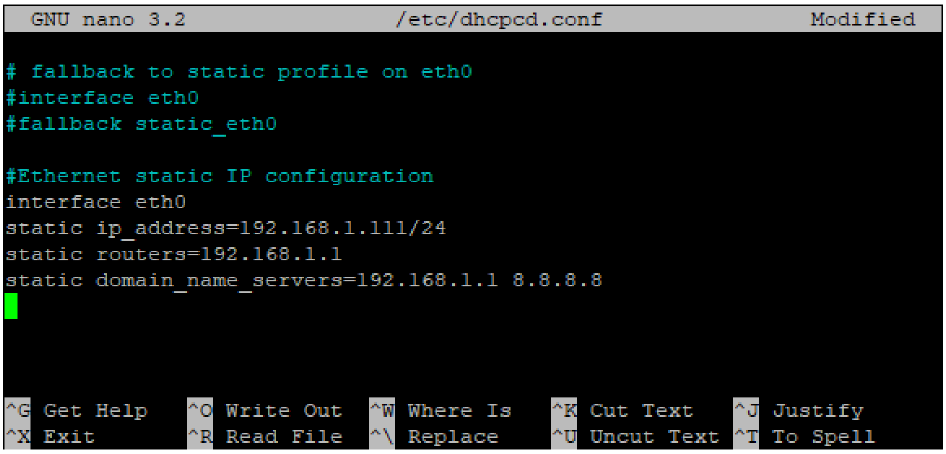

Fig. 3. Static Network Configuration on Raspberry Pi using GNU

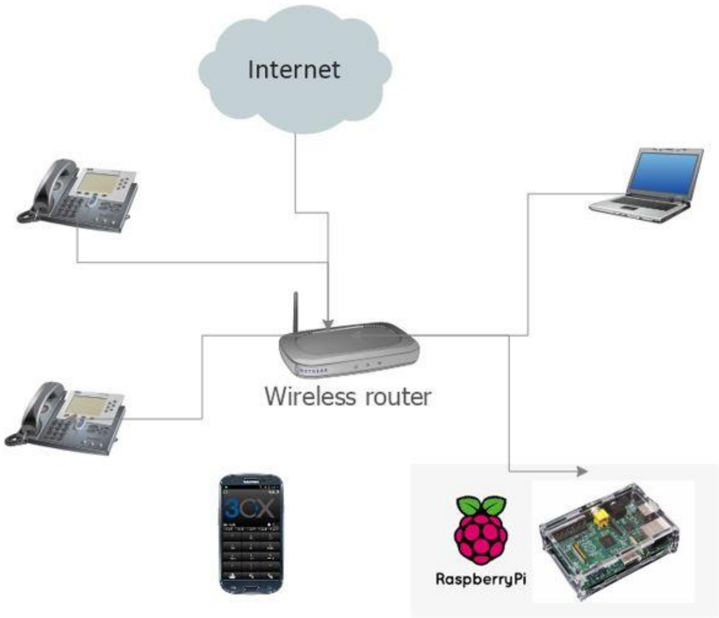

Fig. 4. Overview of the system

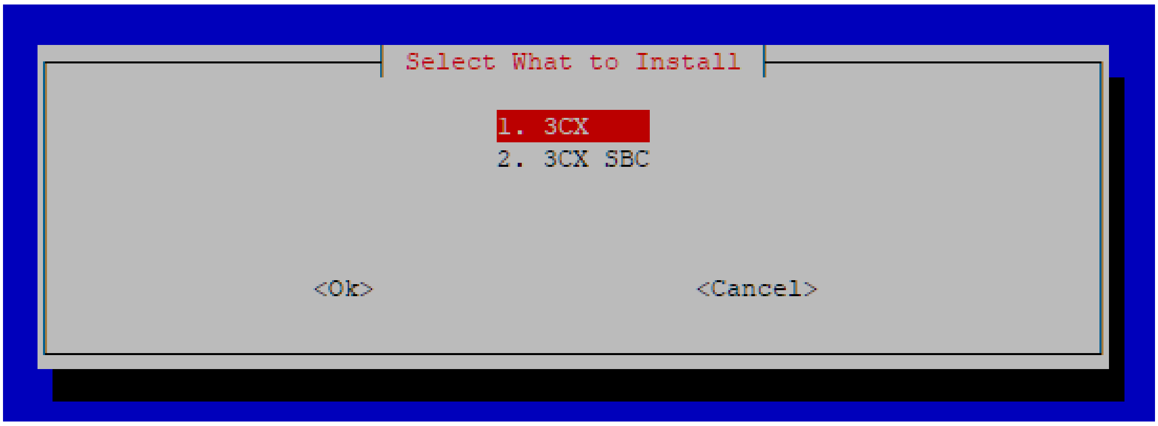

Fig. 5. Installing 3CX PBX System on Raspberry Pi

By using the free OS available online, you can download and install the 3CX PBX system shown in Fig. 5 with a one-year free trailer from the online store by using the below command: wget https://downloads-global.3cx.com/downloads/misc/d10pi.zip; sudo bash d10pi.zip

Once 3CX is installed, the 3CX PBX Web Configuration Tool guides you through important network and security settings. To run it, open the IP of the machine 3CX is installed on, followed by port 5015 in a browser, e.g. http://192.168.1.1:5015.

Then a series of wizard pages will appear as you go through the configuration of adding IP phones and mobile devices to the 3CX system and give each device a number and voice mail box to be used in the VOIP system. Once the 3CX PBX System on Raspberry Pi is configured, an IP and a Mac address are assigned to the SIP phones from the router, as seen in the above figure. The SIP phones used for this project are the Yealink T21P E2. The Yealink SIP phone supports up to two accounts and provides various features such as open VPN and IPV6 compatibility. The Yealink SIP phone can also be accessed and configured by entering the IP address of the phone into a browser.

Once an account is created on the SIP phone, it is linked to the PBX server using the IP address 192.168.1.1 on UDP port 5001 that was created during the installation of the 3CX. Once the phone configuration is complete, as shown in Fig. 6.

13. ANALYZING CALL QUALITY USING WIRESHARK (MOS TEST)

The Wireshark Network Tool shown in Fig. 7 is used to record five calls to produce a computed mean opinion based on the calculations given above.

The SIP connection between the two IP phones was established with the status ACK and 200 OK, indicating the successful acknowledgment of a received invitation and response, as shown in the preceding packet sample. When the call is picked up, RTP begins transporting voice packets over the network. Each of the five tests took place for a total of 16 seconds. The findings are shown in the following table. The overall quality of the call is determined by calculating the average MOS score.

$$MOS = \frac{3.9 + 4.3 + 4.5 + 3.8 + 4.4}{5} = 4.18$$

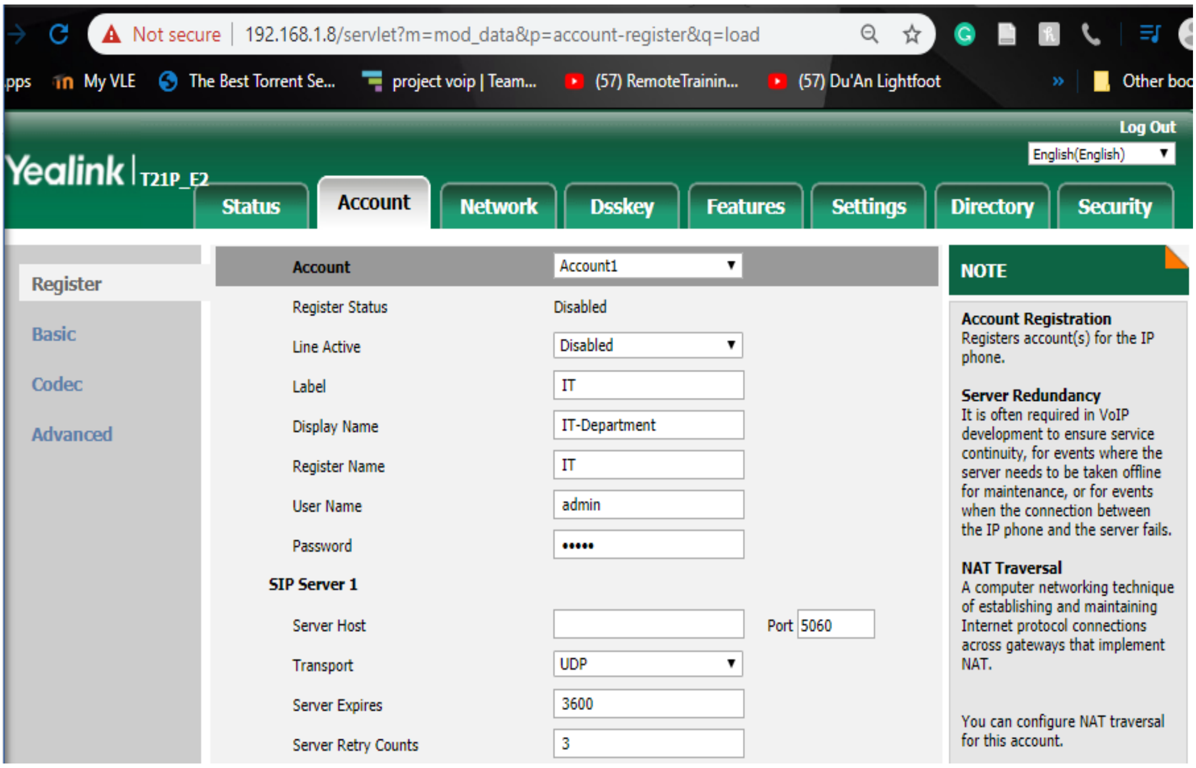

Fig. 6. Phone registration

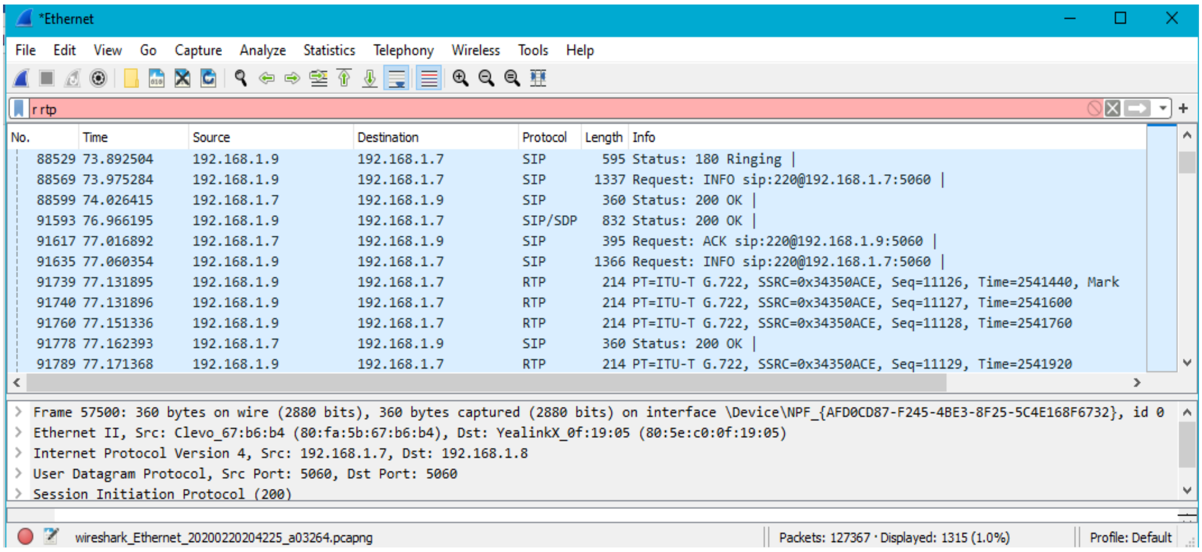

Fig. 7. SIP & RTP capture

Table 3. MOS Test Result and Comparisons

Number of calls	Duration of call	MOS SCORE	MOS SCORE by B. Mekonnen [10]
1	16s	3.9	3.9
2	16s	4.3	4.2
3	16s	4.5	4.0
4	16s	3.8	3.7
5	16s	4.4	3.8

14. CONCLUSION

Voice over IP, also known as VoIP, allows users to make phone calls over LANs (local area networks) or the internet. This cutting-edge technology can convert analogue voice signals into digital data packets. Our goal of building a system that is both cost-effective and makes use of VoIP communication in an embedded system was achieved after analyzing and researching the basics of VoIP. The session initialization protocol (SIP) serves as the system's conduit for data transmissions. One must consider the practicality of SIP implementation considering the numerous possibilities currently accessible for its use in projects and products. There hasn't been much of an open-source revolution in the telecommunications business, so we decided to make an effort to incorporate it into embedded systems. It's also possible to speed up implementation by leveraging tools like 3CX, which can reduce processing overheads while doing so. Open-source software is used to create a cost-effective system with minimal VoIP communication capability that operates on embedded hardware like Raspberry PI, which is a low-cost server. Intranets can be set up in companies so that calls can be made. Call forwarding and message delivery to an individual's personal mail box are examples of optional services. The breadth of the future: A camera can be used to make a video call. It is possible to use call forwarding and message transfer to a person's mail box while that person is not in a state to answer the call. As a result, the system we developed can be deduced as a very basic yet comprehensive call system using 3CX with Voice over IP Installed on Raspberry Pi, where only a microscopic amount of network is used during calls, and it can be concluded as a very useful system with many advantages and benefits.

DISCLAIMER

The products used for this research are commonly and predominantly use products in our area of research and country. There is absolutely no conflict of interest between the authors and producers of the products because we do not intend to use these products as an avenue for any litigation but for the advancement of

knowledge. Also, the research was not funded by the producing company rather it was funded by personal efforts of the authors.

COMPETING INTERESTS

Authors have declared that no competing interests exist.

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