IP Phone Using Raspberry Pi

Sangana Kotireddy[1], Adarsh P[2], Adarsh R Krishna[3], Lohith Narayan[4], Sunand K[5]
[1, 2, 3, 4, 5] Computer Science Department
[1, 2, 3, 4, 5] Coorg Institute of Technology, Ponnampet, Kodagu, Karnataka, India

Abstract- *VoIP stands for Voice over IP (Internet Protocol), a diversity ofmethodsthat are used for establishing two-way multi-media interactions through the Internet or any other IP-based packet switched networks. VoIP has two goals: The first is decrease in telephone charges by sending the call as packets over the data lines or internet. The next goal is to present flexible voice networks, such as allowing severalphonecalls over the same physical link. Applications of real-time VoIP communication have come into well-known use over the Internet. [2] VoIP in rooted systems is one of the blind spots, where due to its over reliance on PC environment restricts its application. In this project we have reviewed the purpose of VoIP on embedded systems and also we have proposed a system that is used to set its application using enhanced components.*

Keywords- Voice over IP, Embedded Systems, Session Initiation Protocol, Asterisk , Kernel codes, Server-Client model, Raspberry Pi.

General Terms: Session Initialization Protocol, Server-Client model, Asterisk, Raspberry Pi Board, Call Processing Language, Linux Operating system.

I. INTRODUCTION

In 1960s, when digital communication first introduced, Public Switch Transfer Network (PSTN) was widely used for communication. At the calling end office, the modern PSTN uses Pulse Code Modulation (PCM)for digital transmission.

Since PSTN has some disadvantages like expensive bandwidth and inefficient use of network channel, we go for Internet Protocol(IP). Internet Protocol (IP) telephony began with Vocaltec Inc.'s introduction of its Internet Phone software in February 1995[1].By doing multiplexing of audio, video and data, we can effectively make use of bandwidth of the communication channel. Voice over Internet Protocol (VoIP) is a type of communication that enables end-user to place phone calls over a broadband internet connection.Internet Protocol was initially designed for data networking for the purpose of its success, VoIP protocol has been personalized to voice networking [2].

II. UNDERSTANDING VOIP

Voice over IP is the technology of digitizing audio, compress it, breake it up into data packets, and sending it over an IP network where it is then reassembled, decompressed to original data, and altered back into an analog wave type [3].

Gateways are the important component for IP telephony. A gateway provides a path from an IP based phone calls to the outside PSTN network and also converts voice signals into IP packets to send over any IP network.

VoIP only sends the conversation data, not the silence period, thus we can saves the bandwidth, and we can reduce the VoIP traffic upto 50%

III. BLOCK DIAGRAM

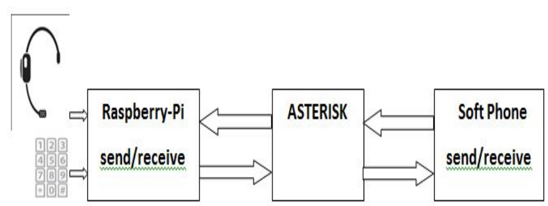

Figure 1.

The functionality of a soft phone is implemented into the raspberry pi. Asterisk is used for setting up the dial plan. The Raspberry Pi, Asterisk Server are connected to a LAN. The Raspberry Pi can initiate a call, which can be received either by another Raspberry Pi or any soft phones. The voice calls can be handled by PJSIP API.

Call is initiated using the hex keypad and a USB headset through a Raspberry device. From the Raspberry pi device the call is connected using the Asterisk software which is in an exchange (PBX). From the exchange the call is received at the soft phone which is at the receiver end. The same can be also done vice versa.

IV. SYSTEM REQUIREMENTS

The basic system requirements for VoIP transmission can be split into 3 sections.

- Hardware Requirements.

- Software Requirements.
- Protocol Requirements.

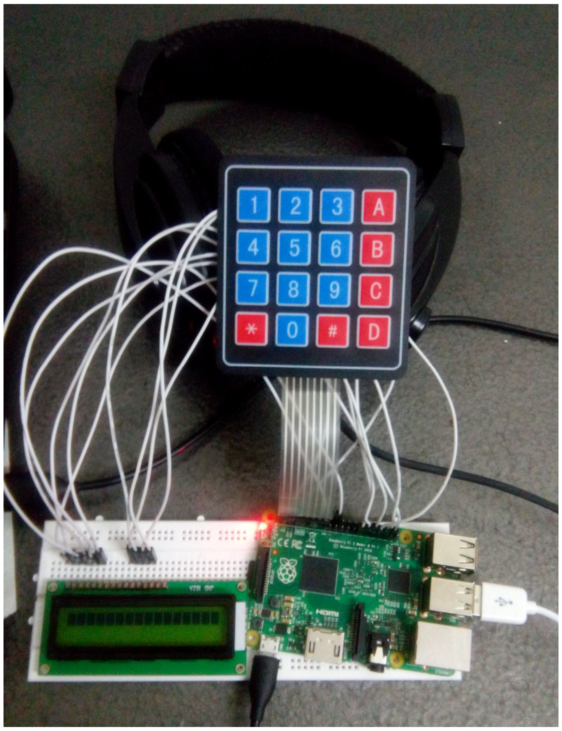

Figure 2.

A. Hardware requirements:

Raspberry pi:

Raspberry is a series of credit card sized small single board computers. Raspberry pi hardware has evolved through several versions that feature variations in memory capacity and peripheral-device support. The USB port is also connected directly to the System-on-Chip(SoC), but it uses a micro USB (OTG) port.

Figure 3.

USB Headset:

A headset is a headphone combined with a microphone. It also provides the equivalent functionality of a telephone handset with hands-free operation. Two types of headsets are available either a single-earpiece (mono) or a double ear-piece (mono to both ears or stereo).

Figure 4.

Hex Keypad:

A keypad is a set of buttons arranged in a block or “pad” which bear digits, symbols or alphabetical letters. Number keypads are found on alpha-numeric keyboards and on other devices. Keypads that contain numbers from 0 to 9, alphabets from A to F as well as a call button are called as Hex keypad

Figure 5.

LCD Display:

A liquid crystal display(LCD) is an electronically modulated optical device which uses the light-modulating properties of the liquid crystals. They do not emit light directly. LCD’s are used in a wide range of applications. Devices such as cameras, digital watches, calculators, and mobile telephones, as well as smartphones uses small LCD screens.

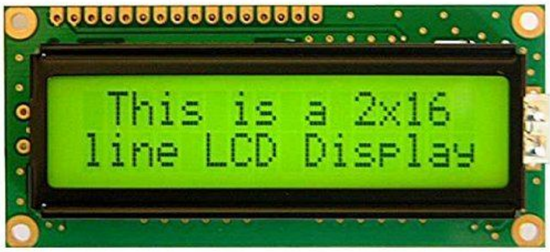

Figure 6.

B. Software Requirements :Python:

Python is a commonly used high-level programming language designed for general-purpose programming. Python has a design idea which emphasize code readability in addition to its syntax. Python features a dynamic type system and automatic memory management.

Asterisk (PBX):

Asterisk is defined as a software program that implements a traditional telephone private branch exchange (PBX). Asterisk was created in 1999 by Mark Spencer of Digium. Asterisk was designed for Linux, but now it runs a variety of operating systems. The software of Asterisk included many features available in proprietary PBX systems. Asterisk supports several standard voice over IP protocol.

C. Protocol Requirements: PJSIP:

PJSIP, a free plus an open-source small footprint multimedia communication library that is written in C language which is used to enact standard-based protocols. It combines signalling protocol (SIP) with rich multimedia framework and NAT traversal functionality into high level API. It supports audio exchange, video exchange and instant messaging, and has documentation.

The SIP protocol allows two clients to establish media sessions with each other by servers. The servers are used to analyze the request of any client and send responded information to the other clients. SIP uses e-mail-like names for addresses [4].

V. PROS AND CONS OF VoIP

A. ADVANTAGES:

The advantages of IP telephone or VoIP over a PSTN are as follows:

Low cost:

When installing VoIP in the office only a single cable is required to the desk, for both telephone and data. Eliminating separate telephone wiring.

VoIP services are cheaper than standard telephones. In many cases , they are free of cost, so can be helpful for international calls.

Portability:

VoIP services can be used anywhere as long as the internet facility is available. It is suitable for people with high mobility.

Advanced features:

While traditional PSTN companies charges for caller ID, Voice mail, call waiting, VoIP service provider gives those for free of cost.

VoIP uses “soft” switching:

By using “soft” switching we can eliminate most of the old PBX equipment.Thus the cost of infra-structure and the maintenance cost will reduce.

Bandwidth Efficiency:

VoIP can compress more voice calls into available bandwidth than traditional telephony. IP Telephony helps to bump off wasted bandwidth by not transporting the 60% of normal speech which is silence.

B. Disadvantages:

Quality of services:

Voice over IP services are vulnerable to jitters, echo, and other quality issues caused by a number of factors from hardware, internet connections, to destination of call.

Power dependency:

Unlike the traditional phone lines, VoIP depend on the electric power supply. If either the power or internet connection is lost due to outage of power supply, there will not be any phone services available.

VI. CONCLUSION

Even though, VoIP is widely used and its usage is bound to grow immensely in times to come. However, the dependence of VoIP on good internet connectivity, strong penetration, and reliance on power availability raises questions on its reliability.

In VoIP, the voices are digitized to data i.e., zeroes and ones. The problem with this is the timing. The packets arrived in the receiving end should be reassembled as the sent data, but unfortunately the packets takes different routes and thus there will be delay in delivery of data, which causes disordered voice reception

The data revolution is gaining momentum year by year. Therefore the advanced call features at negligence costs work well in the future.

VII. FUTURE SCOPE

The development of VoIP has empowered many organizations to change from existing circuit switched telephone networks, providing a new value added services. Many of them had the technological problem of quality of service. One of the major problems was packet loss, which cannot be avoided but can be minimized. VoIP provides unique and affordable services to the people.

In future an effort will be directed to the development of a good concealment algorithm with minimum overhead, a new secret sharing scheme with advanced features, optimized lifetime value of link or path with increased security, selection of optimized paths and efficient routing protocols suitable for all kinds of networks that would maintain the quality and security of voice for VoIP packets. It must meet today's cell phone technology features and functions [5].

REFERENCES

[1] Zachary D. Lund, "A VoIP Implementation on an Embedded Platform," IEEE Marquette University ,2009

[2] SamratGanguly, SudeeptBhatnagar , "VOIP: wireless, P2P and New Enterprise Voice Over IP," England; Wiley, 2008.

[3] DivyaG.S, Dr.P.C.Srikanth, "Embedded VoIP Communication System with Graphical User Interface Features" 4th ICCCNT 2013 July 4-6, 2013, Tiruchengode, India .

[4] Bill Douskalis, "IP Telephony – The Integration of Robust VoIP Services", Prentice Hall PTR 2000.

[5] Ur Goode, "Voice over Internet Protocol (VoIP)", Proceedings of the IEEE, Vol.90, No.9, September 2002, pg. 1495-1517.