Implementing the Voip Communication with Asterisk as Server using Raspberry Pi

Dr. H D Phaneendra
Professor and HOD, Department of CS&E
NIE, MYSURU, India

Mrs.Sowmya C T
PG Scholar, Department of Is &E
Nie, Mysuru,India

***Abstract* - Presently all organization having LAN they make use of PBX (private branch exchange) to process intra contact inside the organization.Seperate telephone network of your telephony service provider for making calls is required which increases the cost. IP phone makes use of existing LAN for communication within the organisation and also outside using internet.IP refers to network layer protocol which have many supporting protocols like ICMP, ARP, RARP...etc and also SIP protocol which is used in IP phone. Users of IP phone need to register to asterisk which is software implementation of PBX and connects to the desired user. This system is implemented using raspberry pi which is core of this project and also uses SIP protocol to initiate and terminate the phone calls by making the system cost effective, scalable and flexible.**

Keywords: Asterisk, Raspberry Pi, SIP protocol, Soft phones

I. INTRODUCTION

To intra communicate many organization uses PBX, which switches the calls between organization users, the main objective is to save price, PBX is owned by organization not by telephone company where all the telephone lines are terminated at PBX .PBX also consists of PC with memory that manages switching of the calls within organization user. The main aim of project is to show IP phone is more economical than PSTN and the wired PBX methodology. Whole system uses Raspberry Pi which is a card sized board to which many devices like headset, powersupply, and LAN port can be connected. Many models of raspberry pi is available, IP phone uses raspberry pi 2. Users of IP phone need to register to asterisk which is software implementation of PBX and connects to the desired user. It uses SIP protocol to initiate and terminate the phone calls; no additional network is required for communication thus making the system cost effective, scalable and flexible.

II. EXISTING SYSTEM

PBX Phone System

PBX stands for Private Branch Exchange, which is a private telephone network used within a company or organization. The users of the PBX phone system can communicate within their company or organization. The main reason of using PBX is to reduce cost for phone calls, instead of having separate phone lines to each users using telephone service provider, multiple users lines are terminated in PBX.PBX is owned by organization rather than telephone service providers.

Limitation of existing system

- Single Points of Failure: A single point of failure is a part of a system, which if it fails, will stop the entire system from working.
- Non scalable.
- Up gradation is difficult since electric wiring is involved by making the system non flexible
- Separate network is required which increases cost.

III. PROPOSED SYSTEM

By using proposed system all internal telephony is routed through the existing LAN (local computer network). This way a separate network for telephony is not required. Since IP phone is mostly using the open SIP standard, it doesn't limit the growth of a company.IP phone call uses raspberry pi and replaces PBX with asterisk which is software implementation of PBX, SIP protocol to initiate and terminate the calls. This introduces a low cost solution to connect to desired user by using LAN port. Costs include hardware requirement, training cost, which over cost for telephone services based on whether they are working on international or local level. The extended features like call forwarding, sending message to a person mail box. Block diagram is as shown in the fig.1.

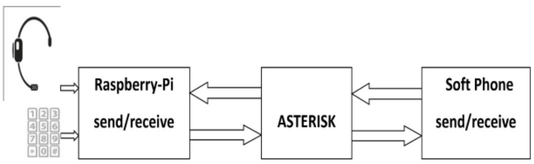

Fig1: block diagram

IV. METHODOLOGY

Raspberry pi is a small card sized board which is core of implementing IP phone. Many devices like headset, power supply LCD display are connected to GPIO of pi.All the logics are implemented using python. Raspberry pi comes with different models, IP phone uses model 2 which has 1GB RAM, Bluetooth, wifi and SD card. It does not have built in memory so it uses SD card preloaded with LINUX operating system in SD card. Many operating system are also supported in pi basically it uses raspbian which works on LINUX and is specially designed for raspberry pi.[12]

Asterisk

Asterisk is a software implementation of PBXit allows to connect call to the desired user.The user must register to asterisk to make calls, and if this server is not running then call connection is not possible. The feature of asterisk is the Ease of implementation in Unix/Linux systems.[13]

Soft phones

A soft phone is a software program for making telephone calls using a general purpose computer, rather than using dedicated hardware. The soft phone can also be installed on a piece of equipment such as a workstation, portable computer, tablet or even a cell phone and allows the user to place and receive calls without requiring an actual telephone set. Often a soft phone is designed to behave like a traditional telephone, sometimes appearing as an image of a phone, with a display panel and buttons with which the user can interact. A soft phone is usually used with a headset connected to the sound card of the PC.

SIP

The Session Initiation Protocol (SIP) is one of the supporting protocols of IP which is a network layer protocol and also communications protocol for call initiation and termination of phone calls.[9]

Advantages of proposed system

- No separate network is used, uses existing LAN.
- Asterisk is used which is PBX software and is a open source and available to anyone, which greatly reduces the cost of development of the PBX server. Hence Phone services are more economical than PSTN and the wired PBX methodology (cost effective) i.e. cheaper call rates than traditional telephony providers.
- Scalable, as they are not limited to a certain number of physical phone ports.
- Soft Phone functionality is built in Raspberry pi.
- Flexible

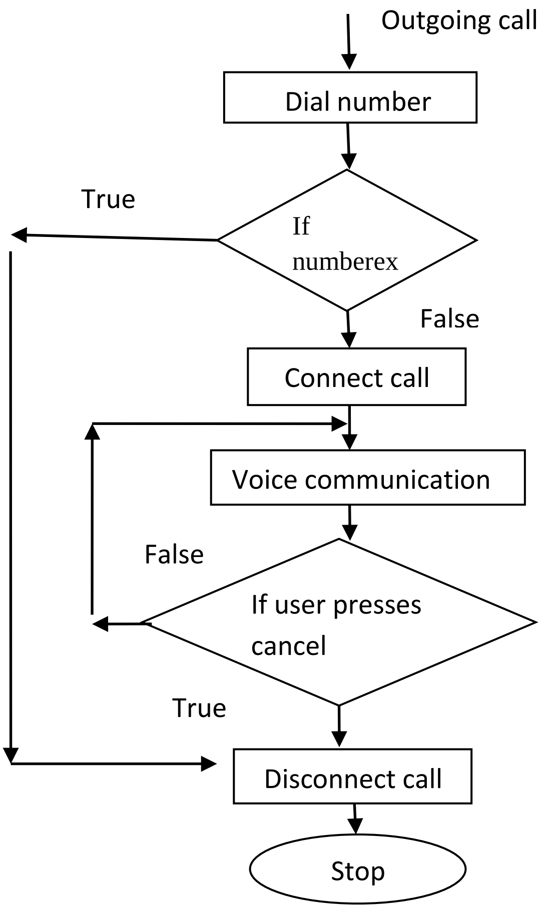

Fig.2: flow chart for outgoing call

Figure 2 and figure 3 shows flowchart of outgoing and incoming call respectively

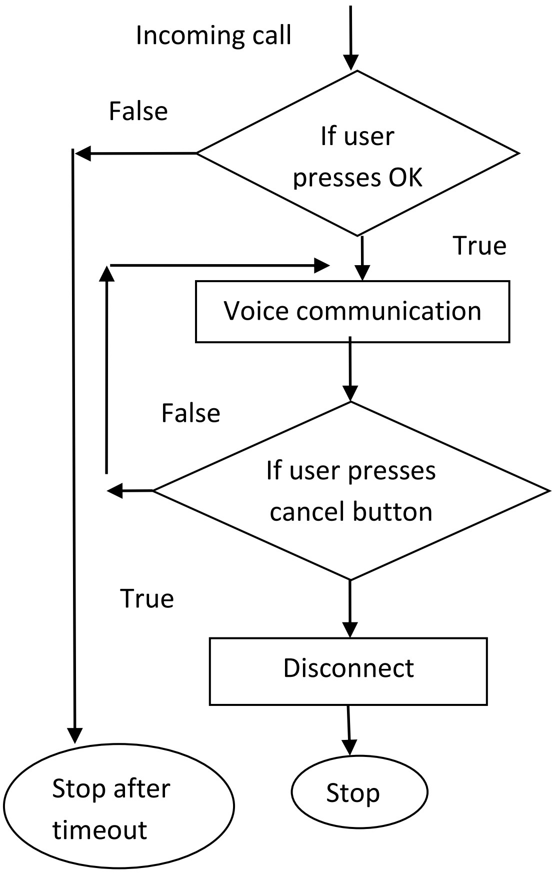

Fig.3: Flow chart for incoming call

V. RESULT

1) After the proper set of user1 and user2, they have to check for registration before calling in asterisk server and is as shown in fig.4 below.

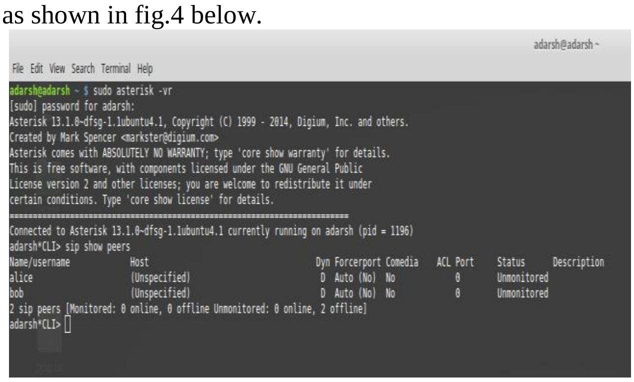

Fig.4 asterisk server screen checking for registration

2) If none of the user registers to server call set up can't be done.fig.5 Shows unregistered message on user screen.

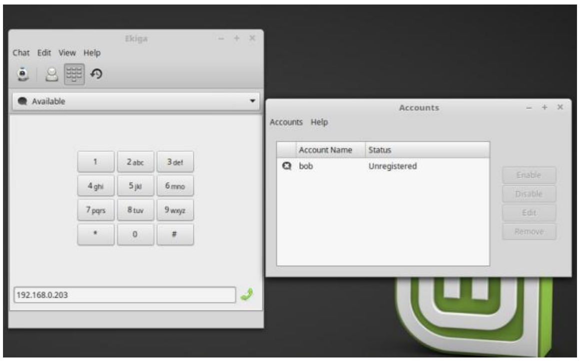

Fig.5 Unregistered message displays on user screen

3) User should register to asterisk server before calling and is as shown in fig.6.

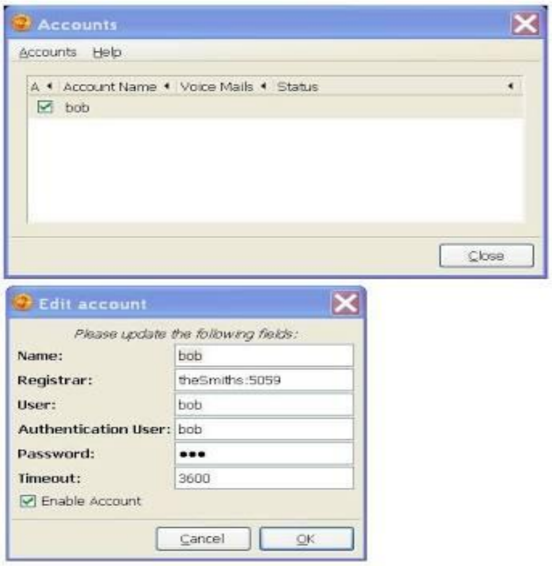

Fig.6: user registration

4) Once user register to server can call to desire user .fig.7 shows dial panel of soft phone and fig.8 shows call connection.

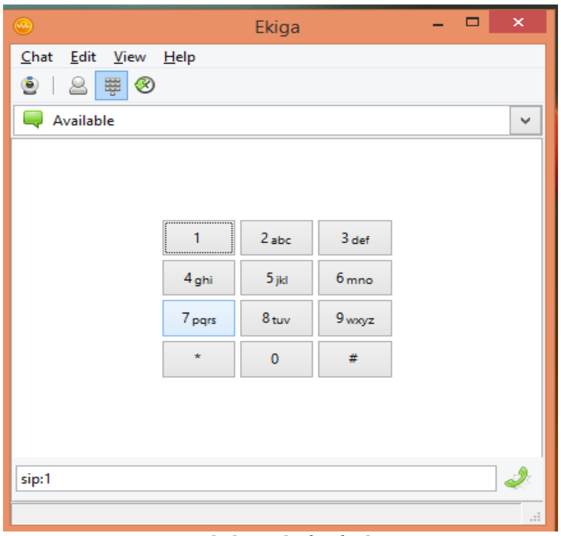

Fig.7: dial panel of soft phone

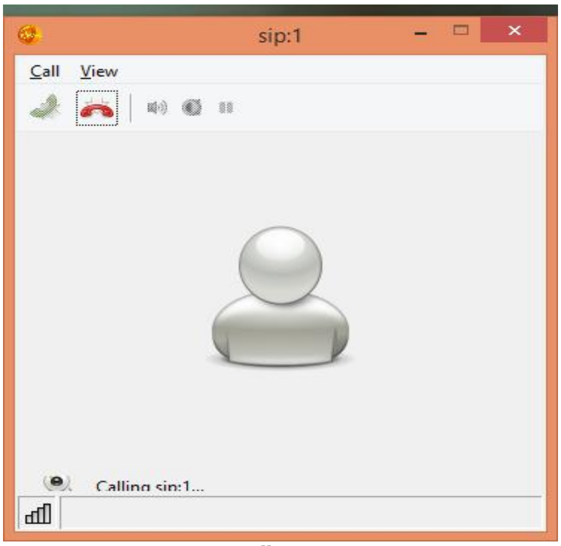

Fig.8: call connection

VI. CONCLUSION

The IP phone services more economical than PSTN and the wired PBX methodology. Instead of using traditional PSTN and PBX methodology we use IP phone call are routed through LAN port using raspberry pi and replacing PBX with asterisk. This introduces a low cost solution to connect to desired user. Costs include hardware requirement, training cost, which over cost for telephone services based on whether they are working on international or local level. The objective is to call can be made with the use of internet and intranet work may be setup in firms. The extended features like call forwarding, sending message to a person mail box. Future scope: Video calling can be executed by interfacing a camera. Call forwarding in case a person is not in a state attend the call and transferring a message to person's mail box can be put into practice.

REFERENCES

[1] Preeti V. Murkuteand V.M. Deshmukh." Implementing the VOIP Communication Principles using Raspberry Pi as Server" .International Journal of Computer Applications (0975 – 8887) Volume 124 – No.4, August 2015"
[2] Ghannam andAljabari. "Integrating VoIP Systems with The Internet of Things",2015
[3] M. UmitUyar and Samrat S. Batth."testingindustrallvoip implementations: registration for sip phones". 0-7803-9470-4/05/$20.00 ©2005 IEEE.
[4] JinheZhou,Tonghai Wu, Zemin Ma." A Research of SIP Phone Based on Embedded Platform". 978-1-4244-5265-1/10/$26.00 ©2010 IEEE
[5] VaibhavA.Vyavahare." Live Audio and Video Transmission System Using Raspberry Pi". Copyright to IJIRCCE DOI: 10.15680/IJIRCCE.2016. 0406260
[6] Henning G. Schulzrinne and Jonathan D. Rosenberg." The Session Initiation Protocol: Providing Advanced Telephony Services Across the Internet". Bell Labs Technical Journal October–December 1998
[7] StylianosKarapantazis and Fotini-NioviPavlidou. "VoIP: A comprehensive survey on a promising technology". Computer Networks, 53.12 (2009): 2050-2090.
[8] Bur Goode. "Voice over internet protocol (VoIP)". Proceedings of the IEEE, 90.9 (2002): 1495-1517.
[9] Jan Janak," SIP Introduction", Copyright © 2003 FhG FOKUS.
[10] J. Rosenberg/H. Schulzrinne/G. Camarillo/A. Johnston/J. Peterson/R. Sparks/M., "SIP: Session Initialization Protocol", Network Working Group.
[11] Mark spencer, "Asterisk", the future of IP telephony .
[12] Raspberry Pi Foundation, https://www.raspberrypi.org.
[13] Asterisk for Raspberry Pi, http://www.raspberry-asterisk.org
[14] Zachary D. Lund, "A VoIP Implementation on an Embedded Platform," IEEE Marquette University ,2009.