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Application of Asterisk based Automatic Dialer And IVR system employing Raspberry Pi

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***Abstract---* Voice Over Internet Protocol is the developing innovation according to the present business necessity. Such a large number of organizations utilizing VoIP answers for their systematic call communities, communication promoting called auto dialers frameworks and voice message calling frameworks. VoIP utilizes the Internet as the primary correspondence media to transfer the voice as data packets on the system. These dat packs are move to the normal PSTN network utilizing VoIP Gateways called FXS or GSM Gateway. Electronic Private Branch Exchange system (EPBX) is communication framework with wireline correspondence. This customary PBX framework for example EPABX was supplanted by IP-PBX framework, the web convention is found PBX framework dependent on (VOIP), which passes on voice as an information over the web. An IP-PBX framework is a finished communication framework that gives liberated from cost, without SIM card wireless calling. There is one significant specialized gadget called VoIP Gateway and for utilizing SIM card called GSM Gateway which is the go between the IPPBX dialer and the general public switched telephone network.**

***Keywords*--- PSTN, IPPBX, VoIP, asterisk, GSM gateway**

I. INTRODUCTION

A private branch exchange build relationship between the inside telephones of a independent association by and large a trade and moreover interface them to people in general exchanged phone organize for instance PBX requires part of upkeep and labor. It is substantially less impervious and considerably less adaptable .It needs further wiring for new augmentation that is indulgent and it is not reinforce propelled alternatives like phone message, communication ,guest ID and afterward forward. The Electronic Private Automatic Branch Exchange (EPABX) used by an enormous bit of the relationship for correspondence with inward agents and with the outside world. It is a phone line partition device and associate with the expansion. It is an augmentation littler than typical telephone exchange that interfaces you to the expansion. The augmentation telephone is associated with wires to the PBX framework. . Also this PBX is not suitable for telemarketing as well as general voice based calling as it does not consist of any call details record or voice recording. So the IP based correspondence framework called IP-PBX is utilized for both internal as well as external calling for the business organizations.

The IP-PBX will deal with the internal calling on the wireless network or local area network of the organization likewise due to the fact the outside business known as outbound. The autodialer is that the system performs that automatic appeal the phone mobile number and also the business agent get connected once decision person picks the call. This can save the time to dial the number manually likewise as number of calls per day is additional. This autodialer is connected to the GSM or FXO gateway for outbound calling. As IP-PBX is VoIP business and as per the govt. rules we tend to cannot create calls from VoIP to PSTN directly. GSM gateways square measure within the most demands because it is user friendly and that we will amendment the SIM card if needed.

This gateway gets the call from VoIP and transfer to the PSTN number to determine call connectivity.

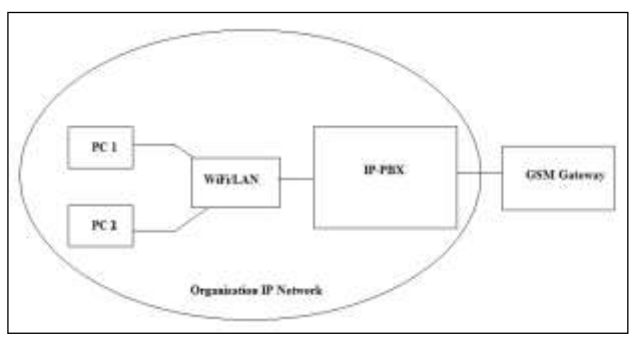

Fig.1. IPPBX System architecture with GSM gateway

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II. LITERATURE SURVEY

As per the products available in the market EPBX is still in demand due to the low cost. IPPBX and Dialers are costly but have lots of advance features than EPBX. Some of the papers and product study shows that there is the need of low cost system with maximum features. The literature shows that the GSM gateway has to be designed as per the market requirement with cost effective and auto dialer features.

VoIP organize assaults, for example, Voice Over Internet Protocol Denial of Service and SPAM.

Our responsibility extended feasibility of Voice Over Internet Protocol area using an authentic learning-based Voice Over Internet Protocol DoS discovery plot and reduced the fake bogus positive pace of SPAM acknowledgment using overhauled assessment factors.

Voice over internet protocols (VOIP) which has now become the most valuable innovation to convey for significant distance calling.

VoIP is a quickly developing innovation in IP network, which requires constant help as it is time sensitive application.

VoIP in IP network is intended for data communication, yet to accomplish dependable high-quality voice over the IP network is a building challenge.

For structuring a decent quality VoIP execution utilizing Asterisk PBX system incorporates picking the best codec and applying perfect technique. The coming of Voice over IP (VoIP) communication altered the universe of media transmission. It has opened a few prospects of broadening the conventional communication ideas. Progression in programming has made it conceivable executed different communication equipment capacities in programming. Asterisk, a Linux based usage of PBX considered as a key expansion to the upset in the cutting edge communication. In this paper, build up a grounds - wide VoIP based system. The Asterisk designed with the capacity to help Auto-specialist, call stopping, call conferencing, approach hold, phone message, music on hold and email warning. The undertaking includes the utilization of different softphones and their setup. The framework tried utilizing different call loads for various voice codecs. The consequences of various codecs were contrasted with one another to choose the best codec for Asterisk PBX. A similar system reenacted in OPNET (organize test system) and the codecs consequence of Asterisk contrasted and that of OPNET result.

In this paper, codec u-law is the best among all these codec's on the grounds that its nature of voice is better among all the above codec's. The outcome might be distinctive if there should arise an occurrence of changing equipment or change kind of system. At that point we contrast these outcomes and OPNET and we find that GSM codec is progressively shut to genuine condition.

Asterisk server gives availability to PSTN and VoIP systems. By utilizing a PC or server and communication card like GSM dongle asterisk mark can assemble GSM passage. There are so many companies in India who provide hosted system for organizations including the IP PBX server and GSM gateway. This system plays virtual receptionist role called IVR and automatic dialing for marketing or customer care support. As per the market survey the minimum system consist for 20 user extensions with 1 channel GSM gateway costs more than 50000 INR and so many organizations can't afford this. Even though the small business organizations require the system for marketing purpose due the cost concern they can't buy this. The plan and execution of our model empowers the SIP customers to speak with one another through asterisk server utilizing free voice and video calls. The usage of GSM gateway empowers the customers having just SIP telephones to call to any PSTN number. This permits parcel changing to circuit exchanging calling. The model can be effectively actualized in scholarly establishments, lofts, corporate associations, research associations, and so on. The gateway permits these associations to get associated with different associations or people as required. Having a gateway among various work areas inside the association would assist association with saving foundation cost. As SIP telephones are introduced on the work areas, representatives can speak with one another at free expense.

The IP telephone benefits more efficient than PSTN and the wired PBX strategy. Rather than utilizing conventional PSTN and PBX strategy we use IP call are directed through LAN port utilizing raspberry pi and supplanting PBX with asterisk. This acquaints a minimal effort arrangement with interface with wanted client. Expenses incorporate equipment prerequisite, preparing cost, which over expense for telephone utilities dependent on whether they are taking a shot at global or neighborhood level. The goal is to call can be utilized web and intranet work might be arrangement in firms.

III. METHODOLOGY

As per the literature survey the proposed system will have almost the same features like IVR and Autodialer using GSM. The system will be cost effective and affordable to the small business organizations. The system server is

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Raspberry Pi ARM based mini computer system. This system runs the Linux based operating system and Asterisk is easy to install and users, dial plan can be easily programmed. The GSM gateway used in this is USB GSM Dongle which has a SIM card in it. The Raspberry pi is connected in the LAN / Wifi network of the organization and users' extensions are configured in it, the system becomes IP PBX. The GSM dongle use to establish the communication between IP PBX and PSTN. Both inbound and outbound communication is possible in it. This system also has a feature of Interactive voice response called IVR means when the PSTN wants to call on the IPPBX system then initially the call will land on the SIM card of the GSM gateway then the call is transferred to the voice response playback like "welcome to our organization.Press 1 for sales, 2 for support" then after pressing the number by the calling person the call will transfer to the respective VoIP extension and communication established for calling. Inbound goes to IVR and Outbound may use autodialer as well as direct VoIP to PSTN calling. The extension numbers are configured in laptop, mobile phone which is in the same network of the organization. The internal extensions can call each other on the same network and no internet is required for this.

Asterisk server and the GSM dongle connect together to form the GSM gateway for the outbound automatic dialing system. The system architecture is shown in the Fig.2 below.

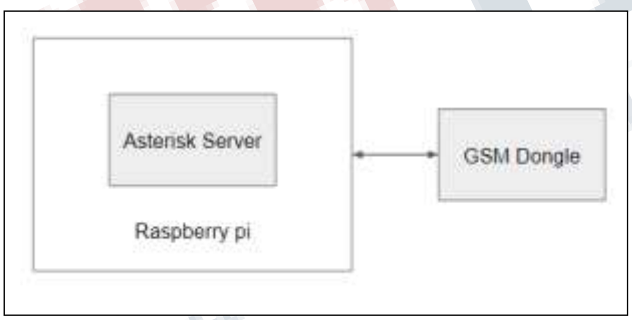

Fig.2. Asterisk System with GSM

3.1 Raspberry pi System

The Raspberry pi is the framework need to the microcomputer framework made by Raspberry pi establishment USA for training to investigation of fundamental software engineering. Raspberry Pi 3 Model B has a 1.2 GHz 64-piece quad center processor, on-board 802.11n Wi-Fi, Bluetooth and USBport. In 2018 the Raspberry Pi 3 Model B+ was propelled with a 1.4 GHz center processor and a gigabit Ethernet.

The Operating system is linux based and installed in the micro SD card which acts like the hard disk for this computer. Min 16 GB card is required to install the OS as well as install the Asterisk Server packages in it. The framework has Ethernet port just as Wifi to associate on the association's system. The system has HDMI port to connect the monitor as well as a USB port to connect the keyboard and mouse. The system runs on 5V 2amp supply and the mobile charger is use to power it on. Such a low power consumption device can run on batteries also to avoid the power interruption.

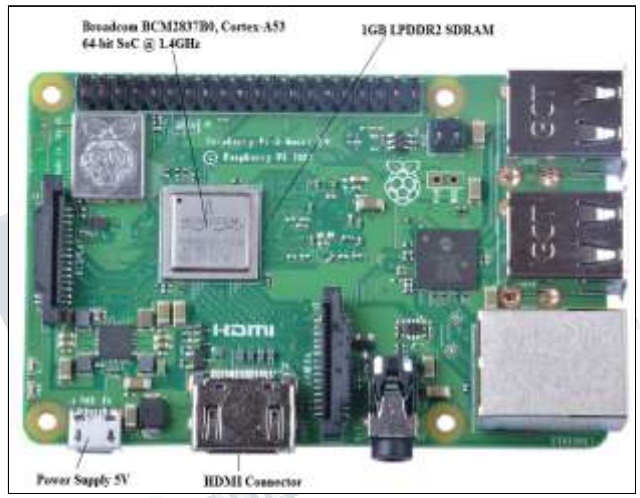

Fig.3. Raspberry pi 3 B+ Model

3.2 VoIP & SIP protocol

A Voice Over Internet Protocol (VoIP) meeting is set up by Session Initiation Protocol (SIP) demand exchange and reaction messages like INVITE, 200 OK, and ACK. In a call condition, initially the REGISTER method is used to register the extension number on the softphone with the Asterisk server. Example, when the extension number which is created in the Asterisk server wants to connect via mobile softphone on the same network of the server then initially the number get registered on the server and then activated to call.

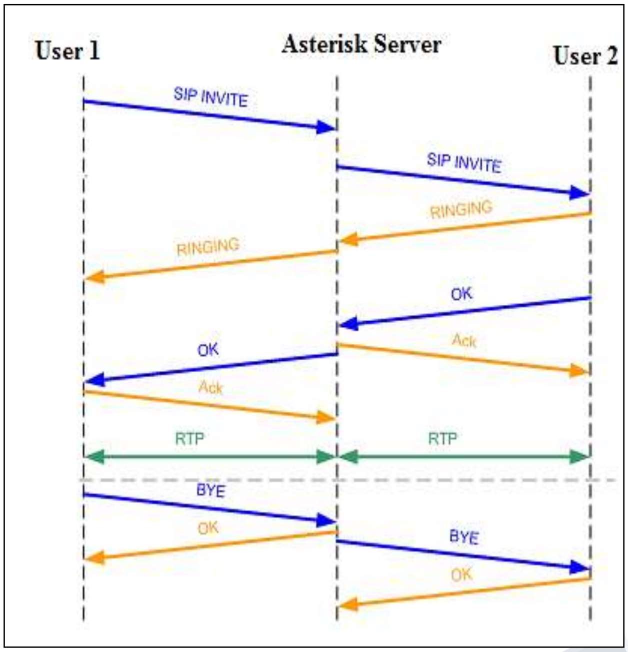

Fig.5. The connection establishing method from user extension to asterisk server

3.3 VoIP to VoIP calling

The SIP user extensions are created in the asterisk server using asterisk programming and after get registered all the extensions are ready to call each other on the same network called VoIP to VoIP calling. Initially when the calling party called the request is transferred to the asterisk server. The server check for the user extension live or not. If it is registered and live then signal goes to the called number and ring the bell. Then the ACK sent back to the calling number via server that is ringing tone. After receiving the call the packet switching network established between the both caller via server and the same network till the Hangup i.e. termination BYE signal.Fig.6. shows the calling on VoIP

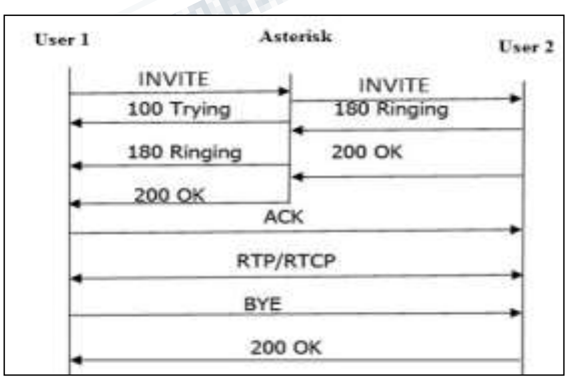

Fig.6. VoIP calling

3.4 GSM gateway connectivity

The GSM dongle is utilized to associate with the raspberry pi for inbound and outbound calling. The dongle is with the SIM card which is active and connects to the PSTN network. If user SIP extension wants to call on PSTN network then call is transfer via SIM card to the PSTN number and the called number displays the SIM card number as caller ID and not the SIP extension number. Similarly when any PSTN number call on the SIM card number then initially the IVR plays and then as per the number pressed the call is transferred to the respective SIP number.

This GSM gateway also use for automatic dialing purpose. The PHP programming is use to upload the mobile number excel sheet in the asterisk server and the number get called one by one with the pre recorded message played when the receiver pick up the call. It has a feature of call detail record to monitor the number of call done successfully.

IV. OPERATING SYSTEM INSTALLATION

The Linux based operating system called Raspbian image is to install on 16 Gb SD card using software called diskimgr. The operating system installed on the SD card and takes 1 hr for the process. After that we can login to the Raspberry pi from Windows remote login software called Putty and we see the login screen.

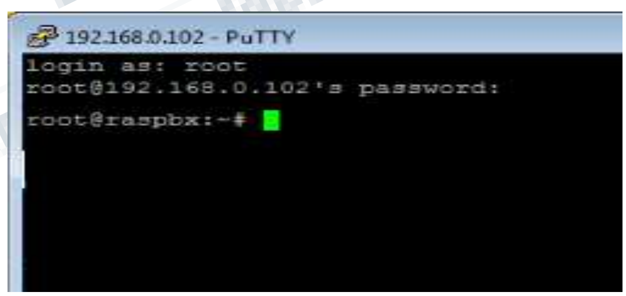

Fig.7. Login to Raspberry pi

To install asterisk on this os download the asterisk package file on server and install it by using the make and configure commands of the linux. For this the root login is mandatory. Now to create the SIP users the configuration file is sip.conf inside the asterisk setup folder. Here the user creation program is as follow

```
[8000]
username = ashwini
secret = 123456
host=dynamic
qualify=yes
allow=all
context = internal_call
```

save this sip.conf file and then restart the asterisk server by using “asterisk - rvv” command. The created user is added in the Asterisk server and can see by using “sip show peers” command. Here 8000 is user extension mobile number, username and password secret 123456 is for security purpose for the app login, host is to allow the number to register from mobile application as well as desktop/laptop application, qualify indicates to get the status of extension number on server, allow indicates to allow all audio codec for calling (ex: G.279) and context is used for the dialplan. Similarly, create 2 more users and then using mobile phone app called SIP softphone app (Zoiper or Xlite) add this user configuration like number, username and password in the app. Fig. 8 shows the results of the 2 extension numbers get registered in the server.

Fig.8 Result of SIP registered on server

If the mobile wifi is connected in the same network of the asterisk server then app shows the message that the user is activated and we can see on the server the status of the user. All the users can now call each other using the given extension number in the same wifi network and no need of SIM card in this phone for this calling of VoIP.

This setup is done and successfully call done between the 2 users 8000 and 8001.

Fig. 9 shows the call from android app called C SIP Simple that user 8001 dialing to 8000 and call is established in the network.

Fig.9. Call establish between 2 SIP users

V. CONCLUSIONS

As per the requirement from the market the VoIP system for the autodial and IVR purpose with GSM gateway to be in the process of implementation. The Asterisk server is act like the softswitch and internal wifi voice communication is done. There is the wifi range limit for the internal calling but it increases by using wifi booster or repeater if required. The small business organization or any educational institution can have the internal calling system on the wifi network using this server to save money on the telephone bills as well as if the mobile PSTN coverage is an issue then this system will work without the PSTN network.

According to the necessity from the market the VoIP framework for the autodial and IVR reason with GSM gateway to be currently execution. The Asterisk server is act like the softswitch and internal wifi voice communication is finished. There is the wifi range limit for the internal calling yet it increments by utilizing wifi booster or repeater whenever required. The private venture association or any instructive organization can have the internal calling framework on the wifi network utilizing this server to get a good deal on the phone charges just as in the event that the portable PSTN inclusion is an issue, at that point this framework will work without the PSTN network.

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