Interactive Voice Response System by Using Asterisk

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ABSTRACT: Electronic Private Branch Exchange System is an existing system for the communication using assigned extension numbers to the receiver. It requires more manpower and introduces difficulties at the time of installation. The main disadvantage of using EPBX that to shift that setup is difficult. If we are choosing an EPBX for organization then it is a difficult task and requires deep knowledge as well as observation for requirements of the office. The proposed system is Interactive Voice Response system (IVRs) utilizes prerecorded voice or messages. The main advantage of this system is that it helps to make system independent of human intervention. IVR system provides appropriate responses in the form of voice. In IVRs we use GSM Gateway having facility of inserting SIM. The number provided by SIM we can make calls. This approach help to improve traditional telephone system to makes it easier and flexible.

KEYWORDS: Interactive Voice Response, EPBX, GSM Gateway, Extension, IVRs.

I.   INTRODUCTION

Interactive Voice Response (IVR) system is a complete telephony system that provides calls over IP data network which utilizes Voice over IP (VoIP). It will help to automate the customer support and very efficient as compare to traditional system. Proposed system required to create new users and assign number to each   client. Using that number call is diverted to appropriate destination. IVR system provides features like call waiting, voicemail, conferencing to every user. Proposed system is independent of human intervention. The proposed system first uses an IP-PBX server and number of clients (desktop, Smartphone, etc) within same network. GSM Gateway which connects GSM with VoIP. IVR systems have evolved to address basic customer service and communications problems associated with scheduling and managing inspections, freeing up resources to undertake more complex tasks, such as staff positions that previously were tied to answering such routine calls.

II.   RELATED WORK

According to Sandeep Ramesh Sonaskar [1], in 2011, the major drawback of the current exchange system is it is inefficient and that it required extra wiring for the installation. This system is not flexible in order to shift the extension to different location of a particular user. It neither has the caller ID facility nor the "voicemail" facility when user is offline. To find the solution to overcome the problem the Voice over Internet Protocol (VoIP) technology is used. This research includes an architectural solution to integrate the voice over IP (VoIP) services in Infrastructure management network of an organization as a Private Branch Exchange (PBX) instead of   EAPBX system. IVR-White paper focused on various strategies for implementing tools to provide better customer services [7]. In 1998, WashingtonCounty received national award for their IVR System as a national "Streamlining Best Practice" from the National Conference of States on Building Codes and Standards, an association affiliated with the National Governors Association.

Shelby County also benefited by applying its IVR system to include automated notification of utilities as to when services (lights, water, gas) should be activated and by including a Spanish language module to all of its systems[8]. Seema Mishra, described the IVR system uses pre-recorded or computer generated voice responses to provide information in response to an input from a telephone caller. In this they captures voice by touch-tone and Dual Tone Multi Frequency (DTMF) signal. This system simply designed by using simple microcontroller and chips interfaced to computer [3]. Author discusses the challenges and opportunities introduces while creating scalable voice forums. In this a new open-source system, IVR Junction, that provides free services and commercial tools to simplify the difficult

process of creating a voice forum [6]. This paper gives information while choosing between Manual Testing or Automation Testing to test when we have to test an IVRs, also illustrates reasons Why automation is difficult in testing IVR applications and uses automation tools for monitoring purposes [4].

III.   PROPOSED SYSTEM

Customer calls in your company. Call is answered, not by conventional receptionist but by an electronic receptionist, an automatic attendant. The answer your customer gets is, "Welcome to XYZ Company. Dial the extension number of the department or person you want to reach" by a recorded voice. And thus it goes on giving you options. This e-receptionist directs you towards getting your query resolved. Who is this electronic receptionist? This Electronic Receptionist is **IVR** / **IVRS** or **Interactive Voice Response System**. With IVRS, the caller to your company gets directions to receive the answer for their query via pre-recorded messages. Interactive Voice Response (IVR) is a software application that accepts a combination of voice telephone input. Provides appropriate responses in the form of voice, fax, call back, e-mail and perhaps other media. Interactive Voice Response (IVR) systems allow callers to get access to information without human intervention. IVR systems have evolved to address basic customer service and communications problems associated with scheduling and managing inspections, freeing up resources to undertake more complex tasks, such as staff positions that previously were tied to answering such routine calls.

A. NETWORK ARCHITECTURE OF IVR SYSTEM:

In IVR system firstly, all clients are connected in LAN with server. When any client wanted to establish a connection then first request goes to server then it will divert call to appropriate client.
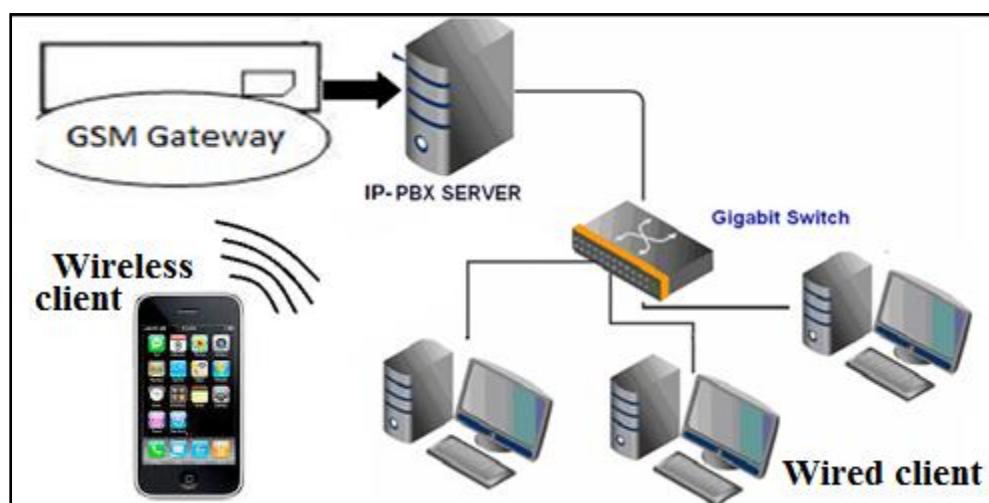

Fig 1: Network Architecture of IVR system

As shown in above figure there is a LAN network, in which IP-PBX server and desktop clients. All clients are connected using router or switch. In desktop clients (**Wired Client**) X-Lite is installed and mobile (android Smartphone) clients (**Wireless Client**) having C-SIP Simple softphone (Android app). We have GSM Gateway which is used mainly for incoming calls routing to organization for which an IVR system is designed. Sim-card is put into GSM-gateways (or SIM-bank connected to GSM-gateway) in order to register it in GSM network, at the same time the gateway is connected with VoIP through program switch. Accordingly the traffic can be converted in and out between GSM and VoIP channels.

An IVRs is a complete telephony system that provides telephone calls over IP data networks. All conversations are sent as data packets over the network. The technology includes advanced communication features but also provides a significant dose of worry-free scalability and robustness that all enterprises seek. IVRs is also able to connect to traditional PSTN lines via an optional gateway – so upgrading day-to-day business communication to this most

advanced voice and data network is a breeze! Enterprises don't need to disrupt their current external communication infrastructure and operations. With IVRs deployed, an enterprise can even keep its regular telephone numbers. This way, the IP PBX switches local calls over the data network inside the enterprise and allows all users to share the same external phone lines.

B. WORKING OF IVR:

- When customer call on mobile number of the organization first call goes to GSM gateway.
- Interactive voice response activated and greeting message played like

    "Welcome to our organization,

    Press 1 for -------,
    Press 2 for -------
    Press 3 for ------- "

- As per the customer press the number from his mobile the call is diverted to Software client phone of department.

Fig 2: Levels of working

As shown in above fig there is a 3 level IVR system, in which when caller makes call to organization, an interactive voice response introduces **'Menu1'** in which 4 choices are there, **"press1 for DEPT, press2 for Office, press3 for Library"** after customer gives runtime input i.e. **choice 1** then another **'Menu2'** is activated like **"press1 for CSE, press2 for ETC, press3 for ELEC"** similarly Menu3 is given. According to the choices of caller, the call will be automatically divert to appropriate person.

III. METHODOLOGY

A. IP-PBX Server:

An IVR System consists of one or more SIP phones, an IP PBX server and optionally a VOIP Gateway to connect to existing PSTN lines. The IP PBX server functions in a similar manner to a proxy server: SIP clients, being either soft phones or hardware-based phones, register with the IP PBX server, and when they wish to make a call they ask the IP PBX to establish the connection. The IP PBX has a directory of all phones/users and their corresponding SIP address and thus is able to connect an internal call or route an external call via either a VOIP gateway or a VOIP service provider.

B.   PROTOCOLS USED IN IVR:

- VOIP
- SIP
- TCP/IP protocols

[1] VOIP:

Voice over IP (VoIP) is one of the widely used protocols for transfer voice data over Internet. More specifically, Voice over Internet Protocol (also called VoIP, IP Telephony, Internet telephony, Network Telephony and Digital Phone) is the routing of voice conversations over the Internet or any other IP-based network. The voice data flows over a general-purpose packet-switched network, instead of traditional dedicated, circuit-switched voice transmission lines. While Voice over IP is often thought as little more than a method of obtaining free long-distance calling, the real value of VoIP is that it allows voice to become nothing more than another application in the data network. Thus allows all internet users to simple use VoIP for communicating each other without any additional charge. On the other hand VoIP does not only offer free long-distance call, but also a number of other services. VoIP makes easy some things that are difficult with traditional phone networks.

[2] SIP:

SIP stands for Session Initiation Protocol. It is an application-layer control protocol which has been developed and designed within the IETF. The protocol has been designed with easy implementation, good scalability, and flexibility in mind.SIP has been designed in conformance with the Internet model. It is an end-to-end oriented signalling protocol which means, that all the logic is stored in end devices (except routing of SIP messages). State is also stored in end-devices only, there is no single point of failure and networks designed this way scale well. The price that we have to pay for the distributiveness and scalability is higher message overhead, caused by the messages being sent end-to-end.

[3] TCP/IP protocols:

Protocols are sets of rules for message formats and procedures that allow machines and application programs to exchange information. TCP/IP carefully defines how information moves from sender to receiver. First, application programs send messages or streams of data to one of the Internet Transport Layer Protocols, either the **User Datagram Protocol** (**UDP**) or the **Transmission Control Protocol** (**TCP**). These protocols receive the data from the application, divide it into smaller pieces called *packets*, add a destination address, and then pass the packets along to the next protocol layer, the Internet Network layer. The Internet Network layer encloses the packet in an **Internet Protocol** (**IP**) datagram, puts in the datagram header and trailer, decides where to send the datagram (either directly to a destination or else to a gateway), and passes the datagram on to the Network Interface layer.

IV. SIMULATION RESULT

The system has proved economic feasible as it reduced the cost in many different ways. This system offers savings made on hardware, separate networks, maintenance and specialist resources. With a traditional system, the capacity for that system needs to be scoped in advance, ensuring it will meet the needs of that business in years to come. This is inefficient and forces the business into paying for capacity they will not use for potentially years. Traditional systems also often have hidden costs when it does come time to upgrade. IVRs offers a clear upgrade path with fixed, 'pay as you grow' cost increases enabling the solution to scale with the business meaning you only pay for what you need, when you need it. In addition, as all sites are linked over a single IP network, there are no charges for calls made between sites.
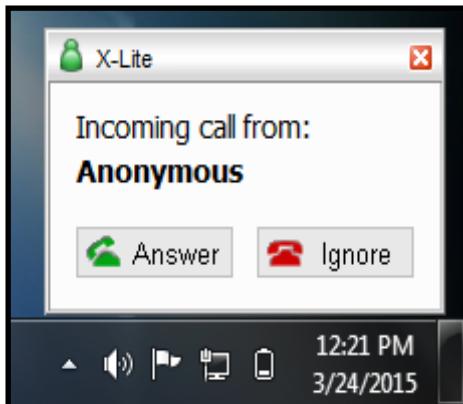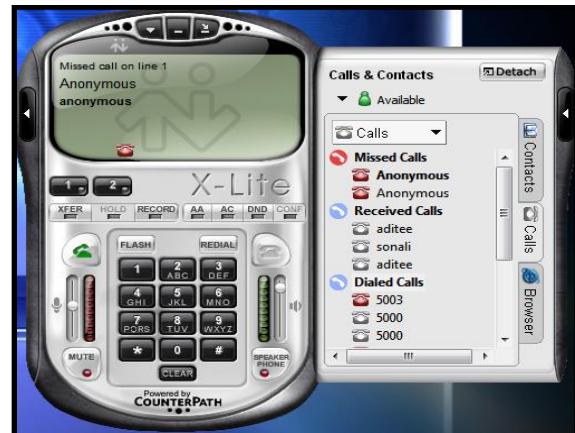
Fig 3:Call notification

Fig 4:Call Log in X-Lite

As shown in above figure, results will be displayed after working of IVRs, in which **Fig 3** shows incoming call notification at **desktop or wired client**. In which caller's name, extension number along with answer and rejection option is available for recipient of call. **Fig 4** shows Call log in which call history, caller's details, contact list and again gives answer-rejection option.

In below **Fig 5**, call log is shown in which date, time, name, extension number,server IP address of caller is given. **Fig 6** shows scenario at **smartphone client or wireless client** having two accounts **etc** and **ankita.** This fig shows smartphone side scenario at the time of Account registeration.
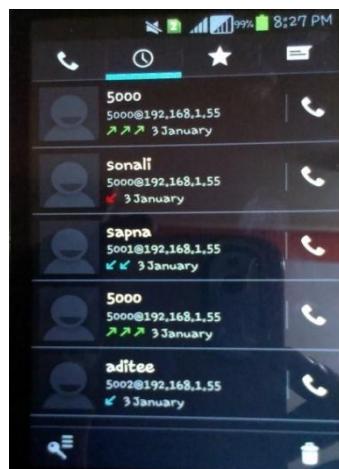

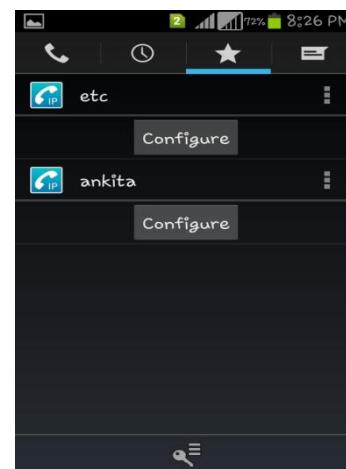
Fig 5: Smartphone client's Call log

Fig 6: Account registration at wireless client

We can define our system's economic feasibility in following simple way:
- Reduced network costs
- Reduced call costs
- Free calls between sites
- No longer require specialist PBX and voice network resources.
- Savings on hardware costs, and the wiring cost.
- Reduce maintenance cost.

V. CONCLUSION AND FUTURE SCOPE

Traditional telephone system have limited no of extensions as well   as have complex wiring. So we required more convenient system, our proposed outcome provide Interactive Voice Response which makes system independent of human intervention and helps to provide better productivity and customer support. VoIP is revolutionizing the way our telephone system works. Instead of having fixed circuit switches as the medium for voice transmission, which is very costly, digitizing voice and video and using the widespread IP network to transmit the real time data not only works, but is very cost efficient. Since VoIP is becoming the new age of telephones.

Asterisk is publishing as open source software which gave others the freedom to use, change and improve the software. Our approach is using the widespread IP network to transmit the real time data to achieve scalability and efficiency. VoIP is revolutionizing the way our telephone system works which achieves Cost efficiency and System independent of human intervention. In future each and every individual employee of the organization will get an Extension number and is always in communication via LAN or Wi-Fi network so as to improve the quality and the work productivity. IVRs is future of telephony system, will be used everywhere to automate communication system. IVR system can be merged with voice recognition system to solve customer queries and make system completely independent of human.

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