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VoIP without SIM calling system using Raspberry Pi for the small business organization

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ABSTRACT

This paper highlights the design & implementation aspects of a VoIP based Asterisk voice exchange, developing a fully functional voice exchange requires to set up a server based on Asterisk, connecting clients to that server with the help of softphones and then configuring the softphones with the help of a server.

Keywords: *VoIP, PBX, IAX, SIP, Asterisk, Trunk.*

1. INTRODUCTION

Although voice over IP (VoIP) has existed for several years, it has only recently begun to take off as a viable alternative to traditional voice systems. Interest in VoIP has grown in part because the technology can help the enterprises to reduce costs by using a single IP network for both data and voice applications. VoIP provides a means of transmitting voice communication over an IP based network. VoIP can use a variety of types of protocols, by far the most common types are SIP i.e. session initiation protocol.

The IPPBX system is a phone switch serving a business or organization. The PBX provides phone services including internal calling, auto-attendant, voice-mail, and automatic call distribution services for the organization.

In the traditional circuit-switched telephony world, people were connected by dedicated circuits that were designed over 100 years ago.

The introduction of IP (Internet Protocol) has changed the telephony market on many levels. IPPBX offers an alternative to EAPBX for voice, and with it, IP based solutions for telephony systems. One of the reasons large enterprises are drawn to IP telephony is the potential efficiency gained from combining the voice and data functions in one single organization.

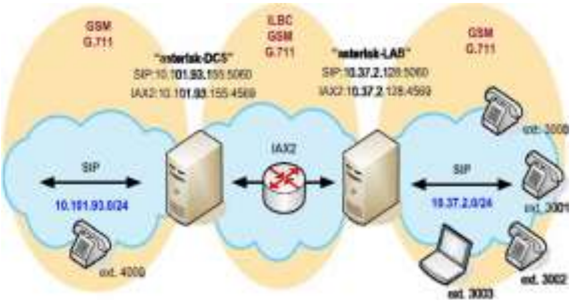

Fig 1: Physical Design

A) INTRODUCTION TO ASTERISK

Asterisk is a fully Open Source, hybrid TDM and packet voice PBX and IVR platform. Asterisk is and has been Open Source under GNU GPL (with an exception permitted for linking with the OpenH323 project, in order to provide H.323 support). Commercial licensing is available from Linux Support Services, Inc. (http://www.linux-support.net) for applications in which the GPL is inappropriate. Unlike many modern "soft switches", Asterisk can use both traditional TDM technology and packet voice (Voice

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over IP and Voice over Frame Relay) protocols. Calls switched on TDM interfaces provide lag-less TDM call quality, while retaining interoperability with VoIP packetized protocols. Asterisk acts as a full-featured PBX, supporting virtually all conventional call features on station interfaces, such as Caller*ID, Call Waiting, Caller*ID on Call Waiting, Call Forward/Busy, Call Forward/No Answer, Call Forward Variable, Stutter Dialtone, Three-way Calling, Supervised Transfer, Unsupervised Transfer, ADSI enhancements, Voicemail, Meet-me Conferencing, Least Cost Routing, VoIP gatewaying, Call Detail Records, etc. At the same time, Asterisk provides full IVR capability, programmable at several layers, from a low-level C interface to high-level AGI scripting (analogous to CGI) and extension logic interfaces. Asterisk IVR applications run seamlessly from one interface to another and need not know anything about the physical interface, protocol, or codec of the call they are working with since Asterisk provides a total abstraction for all those concepts.

B) LITERATURE SURVEY

A large portion of the literature on Asterisk reveals that it is different for many reasons, the most important being it's all software approach. Instead of switching analogue lines in hardware, it routes and manipulates Voice over Internet Protocols (VoIP) Packets in software. The backbone of the system generally becomes an IP enabled network, and phones can be hooked into that. However, it also supports old analogue phones using gateway devices (4). Asterisk provides more than what one would except a conventional PBX. Users get a variety of features such as paging, (which may be from one-to-one or many-to-one, depending on the usage requirements), Interactive voice responses (IVR), Conferencing, Voicemail, Music on hold to name a few.

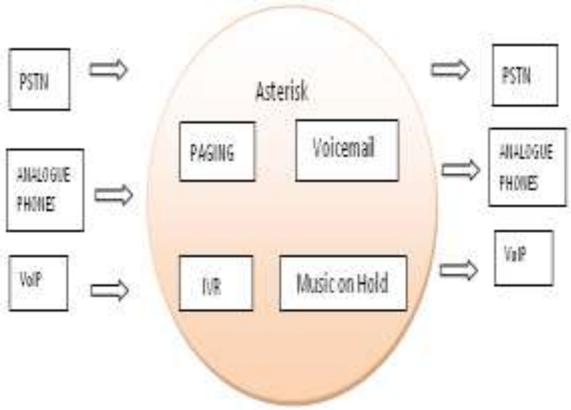

Fig 2: Overview of ASTERISK based System

On top of that users can get interfaces to the operating system and programming languages for the extreme in power, optional web-based administration interfaces, a configuration in SQL databases or flat files, detailed call logging into a database and many more features (3). Hence to summarize it up, it can be said that that with Asterisk user can:

- Provide basic service to Analogue and Digital phones.
- Develop a call routing logic in order to choose a least expensive way to route a particular call.
- Route Incoming and Outgoing voice call over standard voice lines or the internet.
- Provide voicemail and teleconferencing services
- Develop complex or simple interactive menus
- Operate small or large queues for call centers
- Announcing the estimated hold time to the callers
- Call other programs on the system.

2. PROBLEM DEFINITION

a) Objective

- Setup an IPPBX Raspberypi based on Linux operating system (Asterisk)
- To reduce the cost of the organization by using existing Network infrastructure for the IPPBX system
- Create and test SIP accounts using softphones
- Link accounts to voicemail

b) Scope of problem

In Electronics Private Branch Exchange system the cost of wiring for the extensions was increased as well as it is unable to increase the extensions and not flexible for users.

c) Reasons for selecting the problem

- To reduce the cost of the private branch exchange system
- To made flexible for user as well as administrator to install and maintain the PBX system.
- To provide the extra features of the private branch exchange system.

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3. PROPOSED PLAN OF WORK

- To install the Linux based Raspberypi Operating system called 'Asterisk'
- To configure the graphical user interface for the operating system
- To create the extensions
- To link the extensions for voicemail
- To configure the softphone on client

4. METHODOLOGY

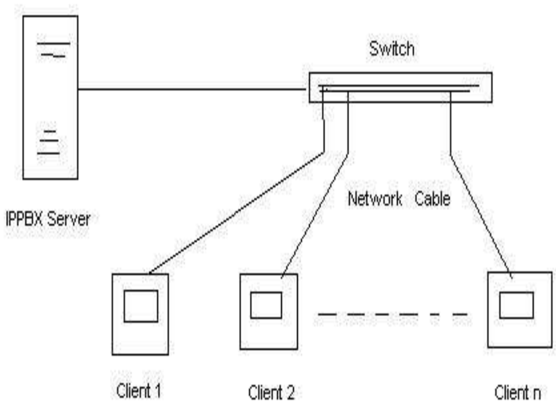

Fig 3: Network Architecture

Here the Raspberry Pi provides the VoIP service to the client operating system based softphone. IP is flexible and has enabled the development of a wide variety of innovative telephony solutions less costly IP PBX systems to help their installed base migrate to IP, as well as bring PBX functionality to a new set of customers they could not previously reach with a TDM-based PBX

The base of the system is the called "Asterisk" operating system which is Linux based open source operating system. Asterisk consist of the telephony platform called "Asterisk".Asterisk is software that turns the computer into a voice communications Raspberry Pi. It is used by small businesses, large businesses, call centers, carriers, and governments worldwide. Asterisk is open source software.

The client extension phone is having two type's softphone and a Hardphone. Softphones are the SIP-based client phone device which runs on Windows or Linux operating system. A soft phone is a computer application that allows users to make telephone calls directly from their computer. The soft phone is part of a category of technology known as computer telephony integration.

A softphone can be cheaper to use than a regular telephone since the technology requires no hardware beyond a network-connected PC. Softphones offer the same features as traditional standalone phones. In fact, most softphone programs offer more features than most standard telephones. Among the features available to softphone users are call forwarding, conference calling, Caller ID, and voice mail.

5. RESEARCH METHODOLOGY TO BE EMPLOYED

Hardware Setup:

- Intel Pentium Processor with chipset motherboard
- 160Gb SATA Hard drive
- ATX cabinet with 400W SMPS and cooling fan
- 10/100 Dlink Ethernet card
- CD Drive for installation of the Operating system
- Existing Network infrastructure
- Softphone setup and configuration

b) Software

- "Asterisk" operating system
- Softphone

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6. EXPECTED RESULT

The IPPBX system employees the features of the organization such as call transfer, music on hold, voice mail and audio conference. It uses the existing network infrastructure so the cost of wiring reduces. It is easy to configure the soft phone. It is also flexible for the users as well as an administrator. It is easy to create and delete the extensions.

Essentiality of the communication in the global world is the core area of concern since people increasingly becoming relay on the Internet. This is so with the voice over internet protocols (VOIP) which has now become the most useful technology to communicate for long distance calling. VoIP is a fast growing technology in IP network, which requires real-time support as it is time sensitive application. VoIP in IP network is designed for data communication, but to achieve reliable, high-quality voice over the IP network is an engineering challenge. For designing a good quality VoIP implementation using Asterisk PBX system includes choosing the best codec and applying the perfect technique.

7. REFERENCES

[1] Voice Over Internet Protocol (VoIP) By BUR GOODE, SENIOR MEMBER, IEEE
[2] J. Rosenberg, H. Schulzrinne, Camarillo, Johnston, Peterson Sparks, Handley, and Schooler, "SIP: Session initiation protocol v.2.0," IETF RFC 3261, 2002
[3] J. Rosenberg and H. Schulzrinne, "A Framework for Telephony Routing over IP," IETF RFC 2871, 2000.
[4] www.voip-info.org
[5] www.Asterisk.org
[6] www.linuxjournels.com

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