Voice Over Internet Protocol (VoIP) WIFI Based calling System

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Abstract- *An innovative application for communication platform based on SIP (VoIP) protocol is presented in this paper.The main theme of this paper is to define the relationship between the number of calls made in an Asterisk server and use of the processor in this server as the processor is one of the major hardware resources. Subsequently, we derived the formulation for the generalization of this relationship or a mathematical model will be obtained. In conclusion of this article, the results of the mathematical model developed are compared with results obtained in the experiment from the graph produced with scilab.*

Keywords- *Voice Over Internet Protocol (VoIP), Session Initiation Protocol (SIP), IPPBX (Internet Protocol Private Branch Exchange), Asterisk, Qos, Proxy, TCP, UDP, Spam, PSTN, DSL, Playout, LAN .*

I. INTRODUCTION

Obviously the Voice over Internet Protocol (VoIP) technology has become the most used Internet in terms of communication. The use of various protocol standards by manufacturers shows the orientation of research towards this technology phone. In addition we are now witnessing a convergence towards "everything over IP". An advance in research around the new information technology and communication specifies those trend and utilities said open sources are a remarkable progress in the technology and especially in the transmission of voice over IP network. The communication circuit to packet communication it is still a current model .

The existence of many standard protocol level leads many researchers worked on the difference between the protocol ,SIP (Session Initiation Protocol) is one of those protocols and open source alone accounts for many many documentation and research .

Our works is therefore in the study and analysis of the existence of a relationship between the numbers of calls and use the server's CPU.

A private branch exchange (PBX) is a telephone routing system that directs all calls from outside lines and routes them to the appropriate phone. This type of system is most commonly used in an office space. PBXs make connections among the internal telephones of a private organization usually a business and also connect them to the public switched telephone network (PSTN) via trunk lines.

IP Communication solutions offer migration at an organization's preferred pace. By integrating with most of the major legacy PBXs and voicemail systems, as well as ot
business applications, most leading IP players empower customers to migrate to full IP based on their business needs, instead of being forced to adopt technologies due to limitations like interoperability of the various business applications. Successful customer migration to IPPBX communications is as much about processes as it is about technologies. Understanding this, leading industry players have developed detailed plans and processes that make migration smoother, faster, and easier for companies of all sizes.

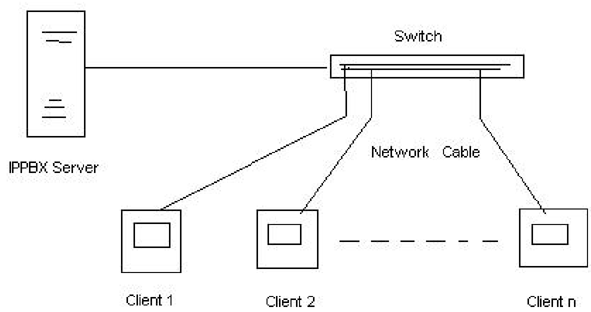

Fig 1.:-IPPBX Connection

In the IP communications world, telephony is just one of the services in the network. And, this service is available from anywhere in the network, independent of location. For example, a multisite business may deploy the call control (IP PBX) software only at the central site, then enable the remote sites to access the service remotely over the network.

II. SYSTEM ANALYSIS

2.1 Existing Systems

Till date different kinds of browsers are being used to browse the web content like Internet explorer, Mozilla, Netscape etc. which needs a typical computer and network connectivity to the web content. A web browser is a Software Application which enables a user to display and interact with text, images, videos, music and other information typically located on a Web Page at a website on the World Wide Web or a Local Area Network. Text and images on a Web page can contain hyperlinks to other Web pages at the same or different website.

2.2 Proposed System

The existing internet protocol network is connecting VOIP (Voice over Internet Protocol) together enabling users to make a call in a hassle free environment. Just a stable WiFi connection is required and a user can make a call from anywhere in the world independent of the location. Proprietary systems are easy to outgrow: Adding more phone lines or extensions often requires expensive hardware modules. In some cases an entirely new phone system is required. Not so with an IP PBX: a standard computer can easily handle a large number of phone lines and extensions just add more phones to your network to expand.

III. TECHNOLOGY OVERVIEW

3.1 Voice Over Internet Protocol (VoIP)

In just a few years, the old circuit-switched voice-centric communications network will give way to a data-centric, packet-oriented network that seamlessly supports data, voice, and video with a high quality of service. The switching equipment, protocols, and links are already being put into place. A transition network is currently in place that joins the packet data world with the circuit-switched world. Integrated access solutions are being installed that support integrated data, voice, and other media into the Internet or the PSTN.

Voice over Internet Protocol (VoIP) is a protocol optimized for the transmission of voice through the Internet or other packet switched networks. VoIP is often used abstractly to refer to the actual transmission of voice (rather than the protocol implementing it). VoIP is also known as IP Telephony, Internet telephony, Broadband telephony, Broadband Phone and Voice over Broadband. "VoIP" is pronounced voyp.

Despite a number of technological issues, real-time multimedia transmission (voice and video) over IP networks and the Internet has largely been worked out. Advanced compression techniques have reduced voice data transfer rates from 64 Kbits/sec to as little as 6 Kbits/sec. Voice over IP or VoIP can potentially allow users to call worldwide at no charge (except for the fee paid to service providers for Internet access). A user's IP address basically becomes a phone number. Additionally, computer-based phone systems can be linked to servers that run a variety of interesting telephony applications, including PBX services and voice messaging.

One of the best reasons to support packet telephony can be seen in the service limitations of the traditional telephone system. The switches are mostly proprietary with embedded call control functions and service logic. That makes it difficult to add new services. In addition, the end devices-telephones-are limited in functionality to a 12-key pad! In contrast, new services are easy to add in the IP telephony world because users simply add new telephony applications on their computers and communicate with other users who are running the same telephony applications.

3.2 Session Initiation Protocol(SIP)

There are many applications of the Internet that require the creation and management of a session, where a session is considered an exchange of data between an association of participant. The implementation of these applications is complicated by the practices of participants: users may move between endpoints, they may be addressable by multiple names, and they may communicate in several different media sometimes simultaneously. Numerousprotocols have been authored that carry various forms of real-time multimedia session data such as voice, video, or text messages.

The Session Initiation Protocol (SIP) works in concert with these protocols by enabling Internet endpoints (called user agents) to discover one another and to agree on a characterization of a session they would like to share. For locating prospective session participants, and for other functions, SIP enables the creation of an infrastructure of network hosts (called proxy servers) to which user agents can send registrations, invitations to sessions, and other requests. SIP is an agile, general-purpose tool for creating, modifying, and terminating sessions that works independently of underlying transport protocols and without dependency on the type of session that is being established. SIP is generic protocol for every IP capable access networks.

The Session Initiation Protocol (SIP) is an application-layer control (signaling) protocol for creating, modifying, and terminating sessions with one or more participants. It can be used to create two-party, multiparty, or multicast sessions that include Internet telephone calls, multimedia distribution, and multimedia conferences. SIP is designed to be independent of the underlying transport layer; it can run on TCP, UDP. It was originally designed by Henning Schulzrinne (Columbia University) and Mark Handley (UCL) starting in 1996. It is a 3GPP (Third Generation Partnership Project) signaling protocol. It is one of the major signaling protocols used in Voice over IP (VoIP).

SIP handles the signaling part of a communication session.

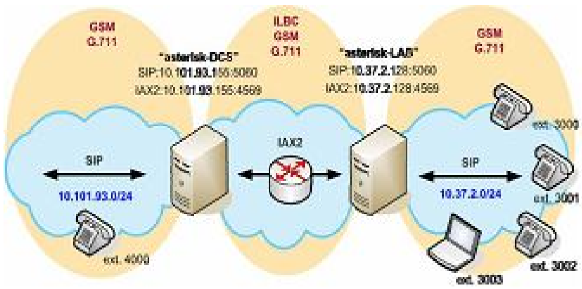

Fig. Physical Design

SIP handles the signaling part of a communication session. It serves as a carrier for the Session Description Protocol (SDP). SDP handles the media portion of the session. The transmission of voice and video content are done by the Real-time Transport Protocol (RTP). A SIP session thus involves packet streams of RTP. SIP is a part of the protocols involved in a multimedia session. The latest version of the specification is RFC 3261 from the IETF SIP Working Group.

3.3 Internet Protocol Private Branch Exchange (IPPBX)

An IPPBX or VOIP phone system replaces a traditional PBX or phone system and gives employees an extension number, the ability to conference, transfer and dial other colleagues. All calls are sent via data packets overa data network instead of the traditional phone network.

An IP PBX is a complete telephony system that provides telephone calls over IP data networks. Typically an IP PBX system is a piece of software running on a server. Depending on the workload, that server can also be performing other tasks, but usually it is dedicated and also acts as the VoIP system's connection to the internet.

An IP PBX is a telephone switching system inside an enterprise that switches calls between Voice over IP (VoIP) users on local lines and lets all users share a certain number of external telephone lines. The typical IP PBX canalso switch calls between a VoIP user and a traditional telephone user, or between two traditional telephone users much like a conventional PBX does. The IP PBX 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

Internet protocol private branch exchange (IP PBX) market offers a ray of hope in the otherwise depressed European telecommunications industry. Encouraging developments in this market have seen enterprises beginning to replace their time division multiplexing (TDM) voice networks with IP enabled/converged voice data networks.

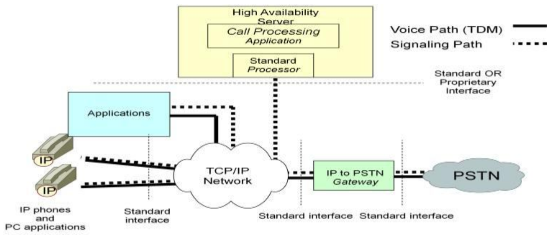

Fig 4: IPPBX Architecture

An IP PBX or IP Telephone System consists of one or more SIP phones, an IP PBX server and optionally a VOIP Gateway to connect to existing PSTN

SIP is the IETF standard for VoIP. Its call control syntax resembles SMTP, HTTP, FTP and other IETF protocols. The back-end runs on Real Time Protocol (RTP). SIP is widely regarded as the up-and-coming standard in VoIP due to its relative simplicity in comparison with H.323 and human-readability. As of the time of this writing, the Asterisk SIP stack has interoperated with a multiple vendors including SNOM and Cisco.

3.4 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 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 total abstraction for all those concepts. Asterisk supports a variety of hardware interfaces for bringing in telephony channels to a Linux box.

3.4 Raspberry pi 3

The Raspberry Pi 3 Model B looks identical to the Pi 2 B at first glance. It is the same size and has much of the same components on board. So what is the difference? The new Pi 3 brings more processing power and on-board connectivity, saving you time with the development of your applications. Perfect for your Internet of Things (IoT) designs.
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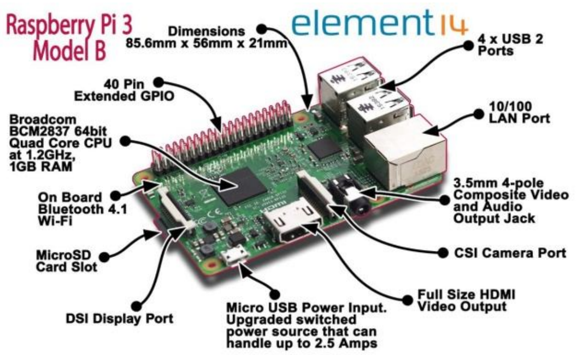

Fig 5: Raspberry pi3

From the implementation of the proposed solution the effect of the performance results, its leads to the following conclusion:

IV. CONCLUSION

1. The Voice over IP is used to communicate with less cost than the normal Telephone line.
2. The open source VolP server (Asterisk) also has some advantages such as it is controlled by the server itself.
3. The Asterisk server ensured the security as it will establish connection for the clients that have been assigned in the server.
4. The Asterisk server can be used for many types of protocol such as Session Initiation Protocol (SIP), H.323, and also IAX, where in this paper only focused in Session Initiation Protocol (SIP)

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