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VOIP BASED ANALOG COMMUNICATION SYSTEM

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Abstract— Private Automatic Branch Exchange (PABX) system is an phone switching system which makes connections with many other internal phones(Analog, Digital & Soft Phones) of a private organization (or an enterprise) and also connects them to the public telecommunication network. PABX can have countless types of extensions such as analogue phone, digital Phone, IP Phone, etc. Currently, almost every private branch exchanges are automatic; the abbreviation "PBX" usually implies a "PABX." The main purpose of a PABX is to save the cost of requiring individual line for each user from PSTN telephone operators. Today, PABX use digital technology or IP technology supporting IP terminals as well. Due to its IP in nature, the PABX industry are rapidly increasing and providing numerous features & facilities based on IP including Analog phone system.

We made use of asterisk software, vm-ware, various digital phones such as the IP phones, soft phones like x-lite & zoipher. In this project we have installed asterisk now version 5 on vm-ware which is a virtual machine it stood as a platform for the asterisk software. We have configured the asterisk for Analog Phones with main line server that's on the asterisk. With the help of that any call can be routed from IP phone to soft phone & Analog Phone. Along with this features such as voice messages, call conferencing and operator console etc. has also been implemented.

Keywords- Aterisk NOW, Vmware, IP, analog & soft phones,PC

I. INTRODUCTION

A PABX (private automatic branch exchange) is a system enterprise that switches calls between client to client on local lines while allowing all other clients to share a certain number of external phone lines. The main purpose of a PABX is to save the cost of requiring a line for each user to the telephone company's central office. Many organizations uses Electronic Private Automatic Branch Exchange System for the communication using extension numbers assigned to the users. It utilizes the man power and extra wiring for the installation as well as it doesn't support the advance facilities like call holding, voicemail, Conferencing, call Transferring etc. Its main disadvantage is that the change of extension is very difficult task. The Private Branch Exchange run on VoIP telephony provides the organization the sophisticated installation and configuration of the user extensions this technology reduces the cost and the time of the installation and configuration, it doesn't require that much manpower as EPBX system. The project aim is the implementation of the VoIP telephony system software and its configuration. The base is the virtual machine called "VMware" which is Linux based VoIP PBX server operating system. These operating systems consist of the telephony package called "Asterisk NOW". This package consists of several features such as Voicemail, Call Waiting, Caller ID, Conference, Call Hold, Call Transfer etc. AsteriskNOW supports audio protocols such as SIP which is Session Initiation Protocol used for the communication which is mainly dependent on audio. The VoIP PBX system for the organization use the backbone of Local Area Network on which all the extensions were configured by using computer system. The "VMware" server is the Linux based and the clients were the windows based or Linux based using the "Softphone" for the communication. Instead of Softphones the VoIP telephone devices such as IP phone and Analog phones can be used.

II. EASE OF USE

A. Developing VoIP based PBX system by using "AsteriskNOW"

Obviously considering development cost it is very much desirable to use the open source software. So we have selected open software that is "AsteriskNOW". As the SIP server's view point, some software are superior to Asterisk in terms of functions, but Asterisk support various protocols (e.g. H.323, MGCP, SCCP,IAX) other than SIP and it also has a lots of additional PBX services that is (Voice Conference, Automatic Call distributor). So we have decided to use "AsteriskNOW" to development of VoIP system for the any enterprise network.

B. Realizing high security by using Open VPN

When we develop the large scale or small scale enterprise network by connecting multiple Asterisk or asterisk NOW servers that are located in different sites or different server based on AsteriskNOW proprietary protocol (i.e. IAX, IAX2), some method are necessary to realize high security because the voice data among sites is not encrypted. For all this purpose we have been introduced a new scheme to establish VPN by using a Open VPN.

III. OVERVIEW OF ASTERISK

Asterisk is an of open source software executed on Linux or Ubuntu to implement IP-PBX system and support various Features of VoIP protocols such as SIP, H.323, IAX, MGCP, SCCP. It can be connected with IP network and also can be connected with the extensions telephone networks via analog/digital/Softphone interfaces. In Channel Section it consist of various logical communication interfaces modules and Application portion consist of the IVR, additional PBX service modules & voicemail function.

A. SIP Using AsteriskNOW

Session Initiation Protocol (SIP) is used in IP-PBX for setting up calls between the peers. Its main purpose is to allow two ends talk to each other but not dealing with media of the call. while making a call between two users. First the users should be registered with AsteriskNOW server; once the all users are registered call can be placed between the all users. While calling an invite request it is first sent to ASTERISKNOW server then via ASTERISKNOW server it is send to the called User. SIP only takes and makes a call while the media session is carried by another protocol RTP. RTP protocol is used to deliver voice over the devices

B. PBX Switching system Using AsteriskNOW

The essence of Asterisk, of course, is a Private Branch Exchange Switching system connecting calls together between various users and automated tasks. The Switching Core transparently connects callers arriving on various hardware and software interfaces.
Application Launcher launches applications which perform services for uses, such as voicemail, file playback, and directory listing.
Codec Translator uses codec modules for the encoding and decoding of various audio compression formats used in the telephony industry. A number of codecs are available to suit diverse needs and arrive at the best balance between audio quality and bandwidth usage.
Scheduler and I/O Manager handles low-level task scheduling and system management for optimal performance under all load conditions.

C. Configuration

Now to create the user extensions assess the server by typing http://192.168.05.116 on the web browser then the following screen will appear with main server options with different panels and support system. This appears on the main screen click on switch to login to free PBX administrative mode by typing username admin and password as asterisk. In this go to the Asterisk tab and select free PBX.

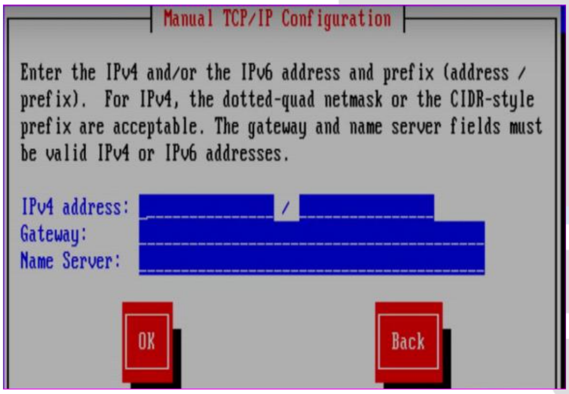

Fig.1 Configuration of AsteriskNOW

Fig.2 Configuration of EPBX

D. Creating Extensions

On the FreePBX dashboard go to Application and click on Extensions tab. As shown in figure 2. Click on Generic SIP Device and click on submit button. Now enter the User extension number (700), display name (EPBAX) and secret password (asterisk.,@1234). Do not edit the default values. Enable the voicemail status on the bottom on the page and enter the voicemail password (12345). Click on submit button and then click on Apply Config. Configuration changes on the top of the page.
Repeat the above steps for adding more extensions.

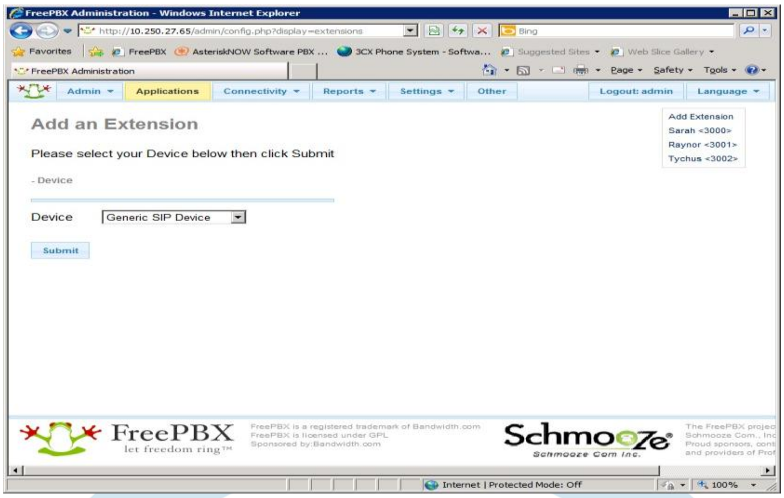

Fig.3 Extension Page in EPBX server

As the configurations of extensions are done on the server now we need to configure the extensions on the analog device handler which will support the analog device in collaboration with ASTERISKNOW.

E. Analog Phone in PBX System (Matrixs)

The Matrix gateway 'SETU VFX88l/VFX44L'is a SIP based VoIP product. Sending voice signals over Internet is called Voice over IP (VOIP), session Initiated Protocol (SIP) is an internationally recognized standard for implementing VoIP. They are fully compatible with SIP industry standard and can interpret with many other SIP compliant devices and software on the market. It supports multiple SIP Accounts so that offers a rich set of functionality and sound quality. The VFX88L can be configured to select SIP trunk out of nine trunks for Incoming Call (IC) or Outgoing Call (OG) call. Maximum 8-calls can be set up simultaneously. The gateway also supports peer to peer calling and Emergency number dialing. In this we are using a Vm ware workstation and assigning one IP address to the analog Extension in which it further gives the
Other 8 extensions to the analog device that is matrixs VFX. In this Maximum 8-calls can be set up simultaneously as it the 8 port device. It also supports peer to peer calling.

IV. Applications

STANDALONE-You can make OG and at the same time configure the gateway using PC connected at Ethernet port. BEHIND THE PBX More extensions of the pbx can also access the IP Networks through the gateway and make calls. LIFE LINE In case of power down VFX88L can be used for OG call from the lifeline port (virtual FXO).
PEER TO PEER CALLING- You can dial IP Address to make call using non-proxy server to call another Gateway, using peer-to-peer calling feature.

V. RESULT:

There are eight analog extensions which has been successfully configured by using matrixs VFX88l. The fig.4 shows that calls has established between analog phone to all other type of phone (soft phone, VoIP phone & digital phone) and vice versa. We have added some features like call conferencing, call transferring & voice mail.

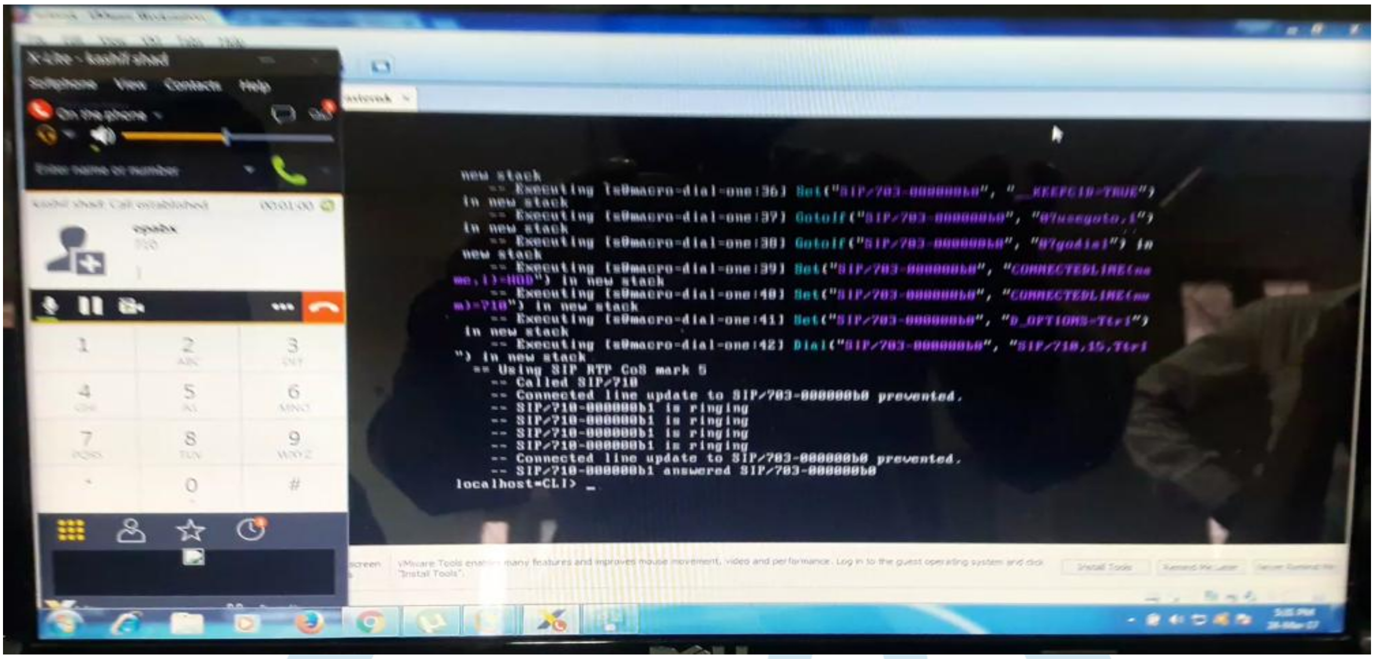

Fig.4 Call establishment

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