Voice over IP Based Digital 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 phones, digital Phones and IP phones. This system helps us to build our own Private Branch Exchange System with the use of the most versatile source available to us .i.e. the Internet without the intervention of huge circuits. The only criteria is that the PCs should be equipped with an open source software known as Asterisk, which enables us to switch and route calls between different PCs which has to be under same LAN.Today, PABX use digital technology or IP technology supporting IP terminals as well.**

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.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 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).

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. 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. 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.

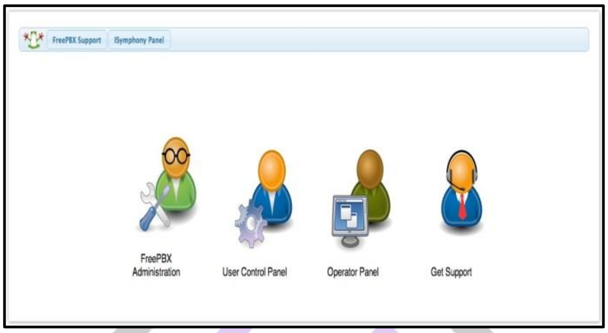

Fig. 1

C. Creating Extensions

On the FreePBX dashboard go to Application and click onExtensions tab. As shown in figure 2. Click on Generic SIPDevice and click on submit button. Now enter the Userextension 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.

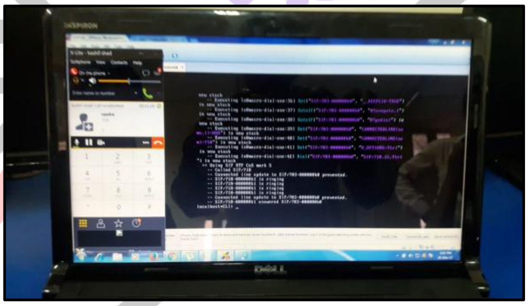

Fig. 2.a Fig. 2.b

As the configuration of extensions is 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.

D. DigitalPhone in PBX System

GRANDSTREAM: The Grandstream GXP1620/GXP1625 IP phones are geared specifically for small to mid-sized businesses. Cost effective and feature rich, these phones offer support for multiple SIP Accounts/Call Appearances, HD Audio & Conference Calling.

CONFIGURATION: The setup is based the **GXP1610**, running on firmware revision **1.0.1.10**; if you are running a different software version or are using the GXP1620 some menus/settings may be slightly different. These instructions are also based on using the GXP1625 in its factory default configuration, which obtains a dynamic IP address automatically from your router using DHCP

STEP 1: Logging into your device

Log into the GXP1610 administrative interface by using the supplied default username-**admin** and default password-**admin.**

Fig. 3

STEP 2: Configuring your call centric account

From the top menu bar, please select the **Accounts** option. After doing so, select the **Accounts 1** >> **General Settings** option; which is located on the side menu bar. Here we enter all the Account details i.e. Account name, SIP server IP address, User ID and Password.

STEP 3: Checking Registration status

To ensure that GXP1610 has successfully registered to our service click on the **Status** option on the top menu bar. Locate the **Account Status** section; you should see that the Account you just configured is now **registered**.

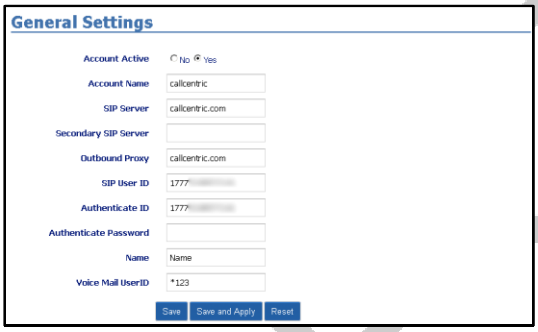

Fig. 4.a

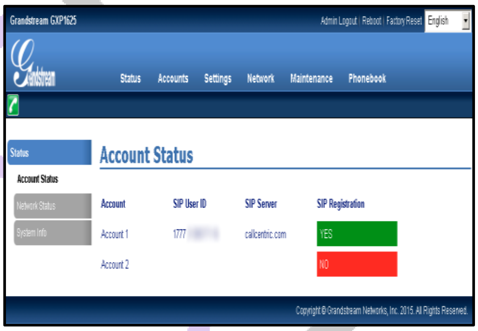

Fig. 4.b

As it shows, the Account 1 has been registered which is indicated by a YES with green indication i.e. Account 1 is active.

SOFT PHONES

Along with IP and Digital phones we are also using a soft phone known as X-lite or Zoiper. The soft phone facilitates PC to PC call under the same LAN. As it is open source this helps us to make calls at a very low cost.

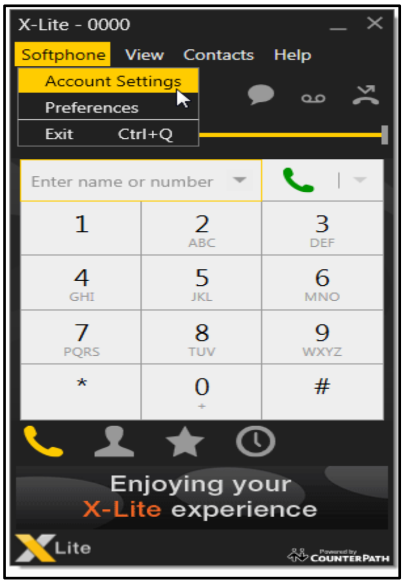

Fig. 5.a

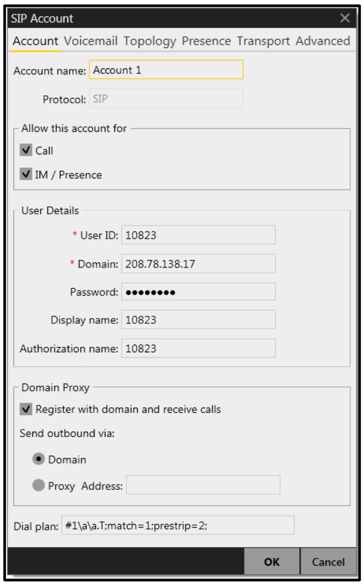

Fig. 5.b

This also comprises of Account settings having same Account name, Sip server IP address, Extension number, user ID

IV. Result and Conclusion

i. EPABX system with IP and digital phone enabled us to organize an internal switching phone system through our open source software AsteriskNOW.

ii. The open source software is made to run a virtual machine known as Vmware. It can simultaneously run software on windows operating system like Ubuntu without requiring a Linux background, in the same way AsteriskNOW is running on windows.

iii. After setting up AsteriskNOW, the interface window of our system gives an overview of our active calls, duration etc.

iv. We have used IP phones and digital phones that were configured using IP address , SIP proxy giving proper extension and changing the corresponding settings in the phone as well. The phone gets activated as the Account is been registered.

v. All in all 255 phones can be connected within a single LAN. The main server has a unique IP address which is entered during configuration.

vi. The soft phone is completely operational as they can initiate a call from one soft phone to anothe4r or from a soft phone to an IP or digital phone.

vii. The performance was tested for long duration calls and voice quality. The voice quality is fairly decent with no call dropping. T

viii. he system is completely reliable for an organization or college which requires internal call routing through LAN connection.

Refrences

1] Alexander A. Kist and Richard J. Harris, "Using Virtual SIP Links to Enable QoS for Signalling," RMIT University Melbourne,Networks, 2003. ICON2003. The 11th IEEE International Conference onepbax.

[2] HarshadaJagtap, Prof. D.G.Gahane, Asterisks Internet Protocol Private Branch Exchange with Smartphone, Vol. 3, Issue 5, May 2015.

[3] Fumikazu Iseki, Yuki Sato, Moo Wan Kim,, "VoIP System based on Asterisk for Enterprise Network,", in IEEE,Advanced Communication Technology (ICACT), 2011 13th International Conference on, 13-16 Feb. 2011.

[4] G. Konstantoulakis, M. Sloman, Morgan Stanley, "Call Management Policy Specification for the Asterisk Telephone Private Branch Exchange," ,in IEEE, Distributed Systems and Networks (Policy 2007), 13-15 June.

[5] AmelChowdhury (Corresponding author), JakiaAfruz, JalalurRahman, "Analysis of Telephone System of a University Campus and Design of a Converged VoIP System," , in CCSE ,Vol .1,NO.4, November 2008.

[6] Mr. Sandeep R. Sonaskar, Dr. M. M. Khanapurkar, "Design & Implementation of IP-PBX for Small Business Organization," , in IJEIR, Volume 1, Issue 3, ISSN : 2277 – 5668.

[7] Prof.S.D.Giripunje, SandeepSonaskar, "Low Cost IP Private Branch Exchange (PBX)," ,in IJCA, Volume 23– No.3, June 2011.

[8] Mohammed A Qadeer, Ale Imran, "Asterisk Voice Exchange: An Alternative to Conventional EPBX," ,in IEEE, Computer and Electrical Engineering, 2008. ICCEE 2008. International Conference on computer and Electrical Engineering.

[9]https://www.callcentric.com/support/device/grandstream/gxp1625