Design and implementation of wifi based intercom system using Arm11

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Abstract - *The new technology has the ability to transmit a voice over Internet protocol process networks by using an Asterisk PBX.In present day companies are implement to produce a different VoIP products to many features in market.This paper aim is introduce of VoIP and implementation of wifi based intercom system using ARM11 to based on Asterisk PBX.In this paper ,first introduce theory of VOIP[1].Next step is using ARM11 on Asterisk PBX .And Final step is the live project live experimental set up how to connect SIP voice traffic on Arm11.This paper is show the ability of voice calls to initiate using SIP and best stability and accuracy using Asterisk PBX.*

***Keywords: ARM11, GSM, ISDN, PBX, RTP, SIP, TCP, VoIP*1.**

1.INTRODUCTION

In many year ago a old telephone system was improved a new substitute known as Private Branch Exchange (PBX). PBX system performs communication tasks such as inbound calls and outbound calls. PBX system is performs VoIP algorithm. Voice over Internet Protocol,is the VoIP.It is used to transmit voice data over internet, using the same type data those are we send and receive e-mail. PSTN is like a internet or broadband connection that changes to VoIP services. It offers to make a calls.By converting consumer's voice into packets we use VoIP technology is use to transmits voice signals over the internet using packet switching technique. These packets trasmited through the internet from one user to another user and collected another user of these packets gets converted voice. In VOIP introducing a latest advantageous features. Because in pervious VOIP so many QoS questions and security [3]. Main aim of VoIP service to a good voice quality and security.In this VOIP PBX some issue was obtained after that we used a new development of Asterisk based on Voice Exchange.It is best voice exchange based on Asterisk,it is work on VOIP and it is good solution for flexibility[4].In 1995 VoIP technology was developed.

1.1 Aim and Objective

Aims and Objectives of different part describe in stages concerning to compleat of this project.

1. To develop VOIP server by using asterisk software.
2. To create configuration files.
3. To install Linux on ARM11.
4. To install asterisk on Linux.
5. To program the user registration.
6. To program the dial plan for calling.
7. To test the calling on LAN network using softphone.

2.LITERATURE SURVEY

VoIP
The VoIP is the voice over internet protocol. In industry IP is the fit image in the voice as compair to old voice telephony technology.In next generation some other technology has come that is VOIP[2]. VoIP technology is low cost and flexiable to user.VoIP send a packet on the internet and converting it into voice signal.VoIP technology is used in small,large and middle industry application for communication. It consist of many added features like multi-carriers to save call charges and good quality service.In old telephone it make only calls or audio signal not video signal but in VOIP is the process of converting audio or video into small signal, and that signal converting in vioce over an IP network and two people can communicate using audio and video.
VoIP principal is very simple.VoIP has a good feature to make a calls between PC to PC and PC to mobile,mobile to PC. VoIP service given offers to service at lower rates for make the calls than perious telephony companies. VoIP technology has to provide to transport the Voice packets by using in Internet protocol. VoIP technology ability to select a required software and hardware which are available. [5]VoIP technology used to convert packets data and transport that packets in Internet.All service user is demanded VoIP services because it is grown fastely. Microsoft Netmeeting and Skype type application. are use VoIP services. VoIP costs is so less to the customer and they took a VoIP services because it have a single set-up for both data and

voice.So It profitable for both customers and service provider.

3. PROJECT OVERVIEW

The Basic block diagram of the system is as shown in fig.1. In fig ARM11 is the server. ARM11 is connected to client1 through LAN using wifi and give IP address to the client1 and client2.Connect client1 and client2 by rining and client2 also connect to ARM11 using wifi.

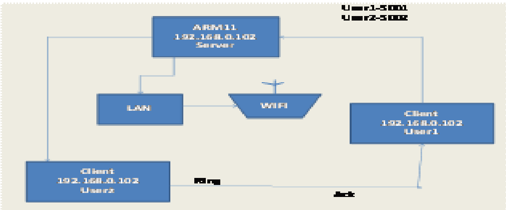

Fig -1: Block diagram of the system

4. DESIGN STEPS FOR IMPLEMENTATION

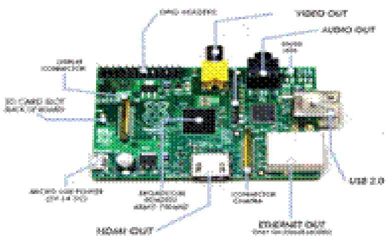

FIG -2: Raspberry pi ARM 11 kit architecture

1.To Implement IP PBX System Using ARM 11.

2.To install Linux on ARM 11.

3.To install Asterisk on Linux.

4.To program the user registration.

5.To program the dial plan for calling.

6.To test the calling on LAN network using SOftPhone.

Hardware Resources

1.Raspberry Pi : The Raspberry Pi is low cost ARM based palm-size computer. The Raspberry Pi has microprocessor ARM1176JZF-S which is a member of ARM11 familyand has ARMv6 architecture. ARM processor operates at 700 megabytes and it has 512 megabytes RAM[6].It consumes 5V electricity at 1A current due to which power consumption of raspberry pi is less. It has many peripherals such as USB port, 10/100 ethernet, GPIO, HDMI and composite video outputs and SD card slot.SD card slot is used to connect the SD card which consist of raspberry linux operating system.

2. Ethernet: Ethernet is the computer networking technology which allows computers to communicate and share resources over the internet. Ethernet was standardized as IEEE 802. . it is one of the most widely implemented LAN standard originally developed by Xerox. Different Ethernet networks also connect to a router that provides access to the internet

5. EXPERIMENTAL SETUP

Fig -3:Experimental setup

Steps for calling the users
1.On the Wi-Fi access soft phones registers its fixed IP [7].Where the Wi-Fi will update this soft phone being active.

2.Each phone is identified by a user name. updating the IP address with a corresponding username.

3. If you Call any user name and the information is available to all users logged in the network.

4.When the Wi-Fi range is not avaliable ,then call handoff.

6. RESULT AND CONCLUSION

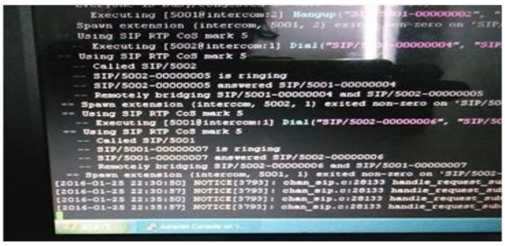

Fig -4: Output result

Conclusion

VoIP technology is one of the most widely using technologies which support to deal with communication from anywhere in the world. VoIP engineering is necessarily varying telephony industry, enabling not just less expensive calls but also providing more advantageous and rich features and more flexible services. Increasing number of service provider is one of the reasons of VoIP technology to be cheaper comparatively with others. Although, challenges stay behind, VoIP technology already plays a key function in businesses communications and is rapidly varying the residential and consumer landscape of domestic and international communication affair. In this dissertation a network is designed and optimised in VMware operating system to evaluate QoS parameters .

Future Scope

1.To set up this network in large organization.

2.TO increase the number of calls and find the performance parameters.

3.TO make a conference call adding some software.

4.It runs on solar panel.

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