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The Design and Development of Low Cost Voip Device Using Linux

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Abstract
In order to improve the quality of the secure communication we need to require secure system hence the review paper has to be defining the requirement of the secure system as well as the cost effective system. Our Embedded system is to design and develop a low cost feature which is based on embedded platform for VOIP media using ARM architecture. VOICE OVER INTERNET PROTOCAL (VOIP) terminal, dedicatedly used for the encryption/de-encryption of the VoIP signal and the RTP (Real-time Control Protocol) voice packet. The encryption flow of the packet is described when the VoIP protocol is SIP (Session Initiation Protocol) and the encryption algorithm is RC4.
Keyword: mini2240 development board , Linux OS, RTP,SIP,Qt creator

I. Introduction

In present era there are number of communication devices already presented but need of review paper is that no one system can define low cost and highly secure voice call embedded system. Voice over Internet Protocol is solution for that, this system can reduces cost effectiveness of present call rate of different service providers. Previously the International call rate is very much high but by using VoIPprotocol it term to very less. Just by using Internet we can call any one in the world. The development procedure for the low cost VoIP device is that by using ARM9 development board and real time operatingsystem like LINUX we can easily develop the low cost embedded system. Basically the communication is based on the TCP/IP protocol stack, which is used in the system. The network layer is based on Internet protocol (IP)it delivered packet connectionless, in best effort manner. Transmission Control Protocol (TCP) is much reliable connection oriented control protocol above IP but it is not suitable for real time communication as VoIP, because its connection control mechanism that can make delay. Basically UDP is not used in the system because it is connectionless protocol. To make a system much secure TCP/IP can overtake on UDP. In case of UDP once the packet is transmitted it does not take any guarantee to receive all packets at destination because it does not have acknowledgement regarding the transmitted packets, Hence the data loss is not recovered in case of UDP on the contradiction in TCP/IP protocol it takes the acknowledgement every single packet transmitted over a network so that if any packet has been lost then it can retransmit it.

II. Review

Now a day in market different VoIP phone System is present. These systems are much costly as VoIP system can be used by only industries but not normal people. Hence it is necessity to decentralise the VoIP system and make a system available to normal people. Now to develop such a system we try to use open source terminal that doesn’t required licensed hence by using this review we can assured to develop a low cost VoIP system. Traditionally PSTN systems were used. The services provided by PSTN system were costly hence to reduce the call rate VoIP system is introduced which can give rise to negligible processing charge. Hence now a day VoIP is much popular in western countries. Because of it remote calling facility all world are connected with each other.

2.Propose System

2.1 Why VoIP:

- Cheaper telecommunications
- Less phone line rental
- Less wiring require
- Free phone calls in some situations
- Video conferencing possibilities
- Branch offices may not need a PABX

2.2 Comparison

Network Features	PSTN (Voice)	VoIP (Voice)
Switch	Circuit Switched	Packet Switched
Connection	Connection Oriented	Connection Oriented
Bit Rate	Fixed and low<=64kb/s	Standard Bit Rate
Bursts	Nonexistent	
Error tolerance	User error control	Self error Control
Info resending	Can not (real time)	It Can
Delay	Must be low and stable	Very Less Delay

2.3 Cost Saving:

This can be achieved by reusing the devices and wiring for the existing data network as most of the organisations already have their own networks. However, the most attractive reason to adopt VoIP maybe is dramatically reduced phone call cost. Soft phones such as Skype enable PC-to-PC users can bypass traditional long-distance toll calls charge as voice traffic over the Internet, they only need to pay flat monthly Internet-access fee. Soft phones also allow a PC as a VoIP phone to call a mobile phone or a home line phone at a lower rate.

2.4 Advancedmultimedia applications.

Cost effective is only one of the good reasons to use VoIP. VoIP also enables multimedia and multi-service applications that increase productivity and create a more flexible work environment, e.g. real time voice-enabled conferencing systems that may include white boarding, file transferring, etc. which combine both voice and data features.

III. Basic Principle

This section describes the necessity of VoIP protocol and hardware.

3.1 Hardware Design:-

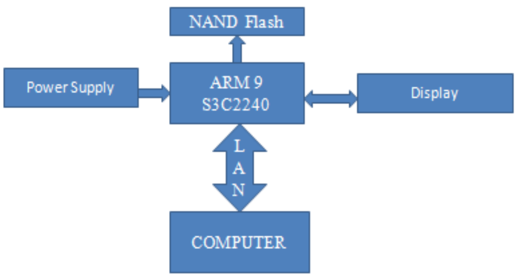

Fig 1: System Diagram

The operating frequency of Samsung Mini S3C2440 is ranges from 400MHz to 533MHz. It provide on board JTAG (Joint Test Action Group) debugger. By using JTAG port which is an international test protocol standard we can simulate software, single step debugger and vivi-boot download can also be done. It is a simple and efficient means of developing and debugging embedded systems. It has inbuilt 64 MB SDRAM, working on is 3.3V voltage, having 32bit data bus,and SDRAM clock frequency can reach up to 100MHz.Flash memory is divided into 64M NAND flash and 2M NOR flash The development board mini S3C2440 CPU chip can provide two kinds of boot mode sliding switch: booting at the NAND flash and booting at the NOR flash. The allocation of the storage space of the chip selections is different in the two boot modes. The boot loading process can be done at NAND flash and execution of an application can be done at NOR flash mode.

3.2 Audio Encoding Process:

The transmitting data is in analog form but the system is in digital mode so first step is to convert the analog data into the digital form. The traditional technique is used that is Pulse Code Modulation (PCM). It is much simplest method to convert analog data into the digital form with the higher accuracy. Most of the VoIP user use PCM 8 bit 8 KHz mono format. In the PCM amplitude of analog data is sampled at fixed rate with regular time interval. The bandwidth is break into quantization level. This is called as linier quantization but this make the error hence the natural logarithmic process is use for quantization.

3.3 Development Tool:-

For the Development of the VoIP communication we need IDE for development of Software application hence Qt creator IDE is used this IDE is open source and more power full by using this IDE we can develop application like java, mobile application , C++, GUI etc. Qt Creator is much used full tool due to open source this terminal can be used and develop a GUI Application like Dial pad for IP address generation this IP address can be communicate with other system this Qt creator can be easily available on Linux OS.

Application development

Fig 2: Dial Pad IP Generation

Once this application is developed then next term is to installed this application to development board generally in OS manufacture are installed his own file so to removed this application file in kernel then put the Qt dial Project in to application. Hence maximum free source date used and try to implement low cost VoIP application project.

IV. Basic Protocol

4.1 SIP Protocol:

In the network system there are different protocol standard present but for VoIP communication SIP protocol is used. To launch the call or message the data is in analog form, so first step is to convert it into the digital form. The mini S2440 board having on board ADC. The SIP is application layer protocol. It used for establishing, developing and terminating multimedia communication system. The basic reason for why we use SIP protocol is that SIP has superior real time communication in critical situation furthermore it translate to control protocol that means network management is easy to deploy. The number of positive quality of SIP protocol makes it easy to use in VoIP embedded application.

4.2 RTP Protocol:

It is an application layer protocol. The main purpose of RTP is to transmit audio as well as video data over a network. It has much reliable services. Its cooperation with RTCP while real time protocol transmit audio packets within the network at the same time RTCP is monitoring and control the packet.

4.3 RC4 Algorithm:

The RC4 algorithm basically design for the Encryption and Decryption process it is key role of the system that the date should be transmitted in the secure manner.

V. Conclusion

In this review paper we tried to make a system which is basically a low cost VoIP Encryption Device that make more feasible to the common people in the real life. Because the traditional call rate of is not affordable to the people hence the low cost VoIP device will help common people to make call at low cost. Due to the versatile use of internet it may have chance of attack on internet but as we have used Encryption technique in low cost VoIP device it can guarantee a secure communication.

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