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Implementation of Voice over IP on WiFi Back Bone

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ABSTRACT: Voice over internet protocol which is a communication protocol existing over a network. The IP network makes possible to users to make telephone call using the VoIP technology. These new phone services are based on the transmission of voice over packet switched IP networks. VoIP can be realized on any data network that uses IP like internet and local area network (LAN).The purpose of this is to design and implement a telephony program that uses WIFI. VoIP has the potential to provide interactive communication service like video and video conferencing. VoIP helps to transfer data which are difficult in transferring over circuit switched networks. There are several methodologies which allow a mobile handset to be integrated into a VoIP network. One implementation which turns the mobile device in to a standard SIP client and uses a data network to send and receive SIP messaging, and to send and receive RTP for the voice path. This methodology of turning a mobile handset into a standard SIP client requires a mobile handset support, at minimum high speed IP communications.SIP is protocol designed to initialize, control and manage session over VoIP on internet.

KEY WORDS: Voice over Internet Protocol(VoIP), Session Initiation Protocol(SIP),Local Area Network (LAN), WI-FI

I. INTRODUCTION

Voice over internet protocol is form of communication that allows us to make phone calls over a broad band internet connection. Voice over IP (VoIP) also known as IP telephony, which is delivery of voice information over Internet Protocol (IP) packet switched network. Voice over internet protocol (VoIP) is a technology for communicating using “Internet Protocol”. This technology results in the introduction of VoIP which is meant to transfer the voice over the internet. WIFI phones are the next generation phones for VoIP. A major advantage of VoIP is that it can avoid the tolls charged by normal telephone service by utilising fixed charge IP network services such as broadband. VoIP allows calling others who are also receiving calls over internet. Some VoIP services need regular phone connection, while other while others allows us to make telephone calls using an internet connection instead. The VoIP allows to do free calling over the internet in whole world. Some VoIP service may allows to call using same service some other may allow to call any telephone number including local, long distance, wireless and international numbers. For VoIP to be a realistic the replacement for standard public switched telephone network (PSTN) telephony services, customers need to receive the same quality of voice transmission they receive with basic telephone services with consistently high-quality voice transmissions.

SIP (Session Initiation Protocol) is a protocol used in VoIP communications allowing users to make voice and video calls, mostly for free and it is a major signaling protocol used in voice over IP. SIP is a protocol that initiates and manages interactive user sessions involving voice, video, instant messaging, and other such multimedia sessions. SIP works with voice over IP to integrate voice and data on a single network .SIP which is used to initiate, manage, and terminate interactive sessions between one or more users over the Internet. SIP is increasingly used for Internet telephony signalling..The Real-time Transport Protocol(RTP) is one of the technical foundations of Voice over IP and this is often used in conjunction with a signaling protocol such as the Session Initiation Protocol (SIP) which establishes connections across the network. The Real-time Transport Protocol (RTP) is used which is a network protocol for delivering audio and video over IP networks, where as Real time Control Protocol (RTCP) is used to

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monitor transmission statistics and quality of service (QoS)
There are two major reasons to use VoIP
1. Lower Cost
2. Increased functionality
1. Lower Cost:
In general phone service via VoIP it costs less than equivalent service from traditional sources. There are also some cost savings due to using a single network to carry voice and data. when users are under-utilized network capacity it is particularly true that they can use VoIP without any additional costs. In the most extreme case, users see VoIP phone calls (even international) as FREE. Generally it costs for internet services but using VoIP it may not involve any extra charges so the users view the calls as free.
2. Increased Functionality:
VoIP makes easy some things that are difficult to impossible with traditional phone networks. Incoming phone calls are automatically routed to VoIP phone where ever we plug it into the network. Take VoIP phone on a trip, and anywhere we connect it to the Internet, we can receive incoming calls. Call centre agents using VoIP phones can easily work from anywhere with a good Internet connection.
The following figure depicts how VoIP call take place.

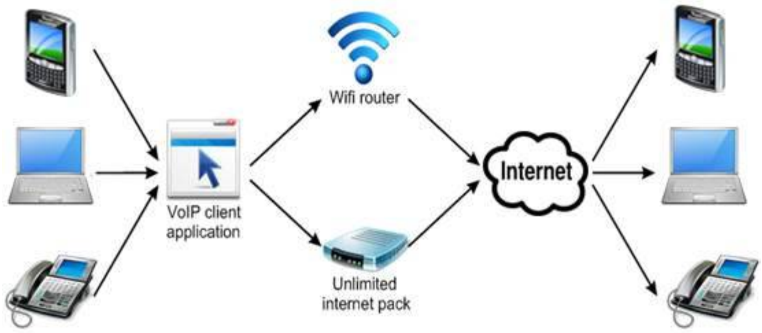

Fig 1: VoIP call

II. LITERATURE SURVEY

In the past, the goal of telecom engineers is to provide better services at any cost. The costs were being imposed on a customer. To this end, only the rich could afford these services.There have been changes to this situation over the years. The industry is driving to the positive direction where better services are being provided at very low cost to customers. In addition, telecom companies have experienced a significant increase in number, which has led to a high level of competition among them. At the same time, the number of customers has also grown tremendously. Thus, there is need for better management of resources such as optimization of the quality of the services. Trade-offs need to be made between costs, quality, and priorities. There are currently systems like Skype, Gtalk, which are useful for low cost communication. For using these services we need to have access to net connection. It could be a costly affair for small companies. Installation and maintenance of wired LAN is long and costly affair. Comparatively installation of WLAN is simple and quicker. Maintenance required is also less. It is also easier to troubleshoot. Hence we propose a wireless system for audio and video calls.The motive behind system is to enable the cost effective voice and video communication. We have designed a client server model based system to implement it. Also there is no need for internet connection for working of this system.Communication has been of prime importance to man since past. Various methods have been make use of communication. In early days of voice transfer PSTN networks were used. These consisted of Private Branch Exchange office owned by service providers.
Wired LAN was later employed to transfer voice and video over local area network. As it was wired systems connected though it lacked mobility. Also configuring LAN required time. Wires need to be setup to individualPC's. Troubleshooting and maintaining this type of network was great trouble. Also topology like star could bring whole network down if central hub fails. In the LAN applications fiber lines mostly serve the backbone to interconnect servers and other high-speed elements of the local networks.Voice over Internet Protocol (VoIP) has been popular in the telecommunications world since its emergence in the late 90s, as a new technology transporting multimedia over the IP

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network. In this , the multimedia (or rich media) includes not only voice, but also video, instant message,presence data, and fax data over the IP network. Today people commonly make phone calls with IP phones or client software on their computer, or send instant messages to their friends. This gives them convenience and cost savings. Many telecommunications companies and other organizations have been switching their legacy phone infrastructure to a VoIP network, which reduces costs for lines, equipment, manpower, and maintenance. Wireless LAN was employed to remove some of shortcoming of wired LAN. Setting up WLAN was easy and less time consuming. Also systems connected through wireless LAN can be mobile. There is no need to draw costly copper cable to each PC. It is easy to maintain and troubleshoot this system. The main advantage of using wireless LAN is that it provides the ability to change the network infrastructure of an organization easily and without the need for expensive re-routing of cable or the installation of new cable runs. A WLAN can be configured in two basic ways: Peer to peer (ad-hoc mode) and Client-server (infrastructure networking) The ad-hoc mode consists of two or more PCs equipped with wireless adapter cards, but with no connection to wired network. It can be used to quickly and easily setup a WLAN where no wired infrastructure is available, such as at a conference center or off-site meeting location. The client-server configuration typically consists of multiple PCs using wireless links to communicate with a central access point that is itself connected by cable to the backbone of the wired network.

III. PROPOSED WORK

The application on implementing sip based VoIP applications on smart phone OS such as android mobile. The purpose of this is to implement a telephony program that uses WIFI or WLAN (Wireless Local area network) as a means of communication between mobile phones at no cost. VoIP generally used for communicating two persons by sending voice packets in real time fashion. Generally various protocols are involved in implementing VoIP. The major task is to establish a session between two communicating parties. The protocols involved in establishing the session are called control plane protocols. The signalling protocol SIP which controls and manages the call .The session initiation protocol which has many advantages majorly adapted by telecommunication industry. The main advantage is it is human readable and less complex.SIP is scalable, easy to implement, and requires less setup time than its predecessor protocols. Being text based, it is easy to program.SIP is based on Hypertext Transfer Protocol (HTTP). It basically deals with embedding call setup and signaling features in networking elements such as user agents and proxy servers.SIP enables a VoIP application to have a kind of high quality and reliability that traditional telephone systems can deliver. The signaling and call setup can be used by IP-based communication system to support the call processing functions.SIP allows people around the world to communicate using their computers and mobile devices over the Internet. It is an important part of Internet Telephony and allows us to take the benefits of VoIP (voice over IP) and have a rich communication experience. But the most interesting benefit from SIP is the reducing the communication costs. Calls (voice or video) between SIP users are free, worldwide. There are no boundaries and no restrictive laws or charges. Even the SIP apps and SIP addresses are obtained free.

USE OF SIP PROTOCOL IN VoIP: The basic idea of our approach is to make a voice call. In the beginning, the mobile. A and the mobile register itself to the server for the service. Both A and B mobile has same IP domain. When mobile A calls to call B, mobile A sends the request to server where server check the IP address of mobile B and sends the IP address of mobile B to mobile A for peer to peer connection. After receiving the IP address of mobile B. Practically, here it goes. You get a SIP address, you get a SIP client on your computer of mobile device, then you need to configure your SIP client. There are a number of technical stuff to set, but the configuration wizards nowadays make things really easy. We can use many android apps like c sip simple, zoiper, voip by anti sip etc but in this project we used c sip simple app. CSIP simple is a free open source app that offers many features, including filtering, call recording, and optimized codecs. The call quality is good. Now you can enjoy crystal-clear phone calls over the Internet using any Android phone. Android SIP client application CSIP Simple enables customers to make free phone calls to other VoIP .VoIP users or very cheap phone calls to anyone else in the world from your mobile phone. Android VoIP phones works wherever you have access to the internet via Wi-Fi or over 3G / 4G.

IV. PROJECT ARCHITECTURE

In this project by using cent OS operating system. As in windows7 we can’t install asterisk directly, we use a cent OS which is flexible in installing asterisk. This product is largely composed of software packages distributed under free

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software license and source code. To open asterisk Linux commands are necessary and useful. To copy files or to change directories etc., we use linux commands in asterisk. We have to install cent OS operating system first, which can be flexible and act as a operating system for installing of asterisk. Asterisk can run a number of operation systems. Linux is the only officially supported OS. Asterisk is a complete VoIP Soft switch, designed to reproduce the features of standard office PBX system. Asterisk is also a Voice over IP toolkit which allows interaction between these PBX features and IP-based networks (local and remote.) Asterisk is hardware independent, and is designed to run on numerous operating systems. Asterisk is Open Source PBX (Private Branch Exchange). A complete PBX in software. It runs virtually on any OS. It supports for most of VoIP protocols. In asterisk the most full-featured PBX (Public Branch Exchange) features already built in.

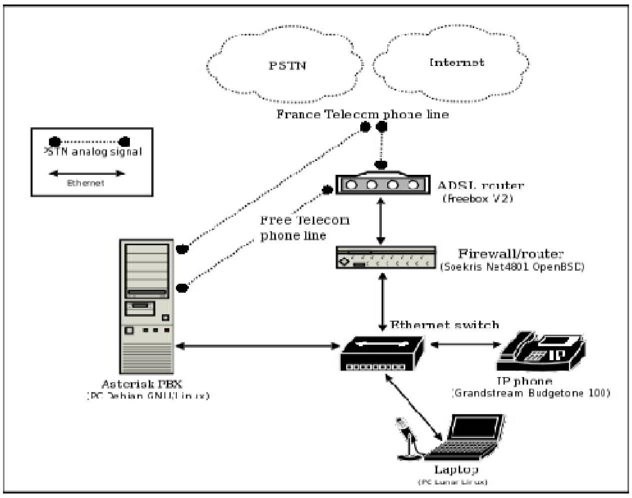

Fig 2:Asterisk overview

Asterisk supports most SIP telephones, acting both as registrar and back-to-back user agent, and can serve as a gateway between IP phones and the public switched telephone network. So after installing asterisk we have to login to linux machine as the root user.

Asterisk File Locations (debian)

• /etc/asterisk/ - Asterisk configuration files
• /var/lib/asterisk/ - contains the astdb, firmware and keys
• /var/spool/asterisk/ - temporary files and voicemail files
• /var/log/asterisk/ - Asterisk log files
• /var/log/asterisk/cdr-csv/ - Asterisk call detail records

The first thing to do is to create a configuration file in sip.conf which can register a device with asterisk. In sip.conf we can create very simple configuration file that allows the SIP phones to connect with asterisk. Dial plan applications are used in extension. conf to define the various actions that can be applied to call. Calls come in on channels and are then handed to the “extensions.conf” file.Dial plan contains logical sections of matches called ‘Contexts,’ and each channel sends a call into the dial plan with a context name and a dialed number. The dial plan then matches the number being dialed, and runs applications accordingly. Each match on the dialed number has an order of steps called ‘Priorities’, and are indicated with an integral incrementing number.Asterisk.conf configuration shows how asterisk runs as a whole. After creating all the above information now we are ready to make a call, we have to open the root .firstly, make the system to connect to the Wireless LAN. To see whether it is pinging or not .we has to give ifconfig and we can see the server IP at the same time. We are going to give in the mobile same IP to connect asteris server. Connect the server as well as clients to the access point.Give the following commands to start the call.

Safe_ asterisk: To start the asterisk server this command is used.

Ps –ax : To see whether asterisk running or not.

Asterisk –r: To start the asterisk related command line interface.

Sip show peers command which lists all configured sip devices and then by registering same as in another mobile with

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same user name and IP address we can connect two mobiles and the call is routed. To see how many active channels the command used is core show channels. In CDR (Call Data Record) it is designed to facilitate as many methods of storing call detail records, the core stop now command which immediately stops asterisk and to exit and returned to LINUX prompt.

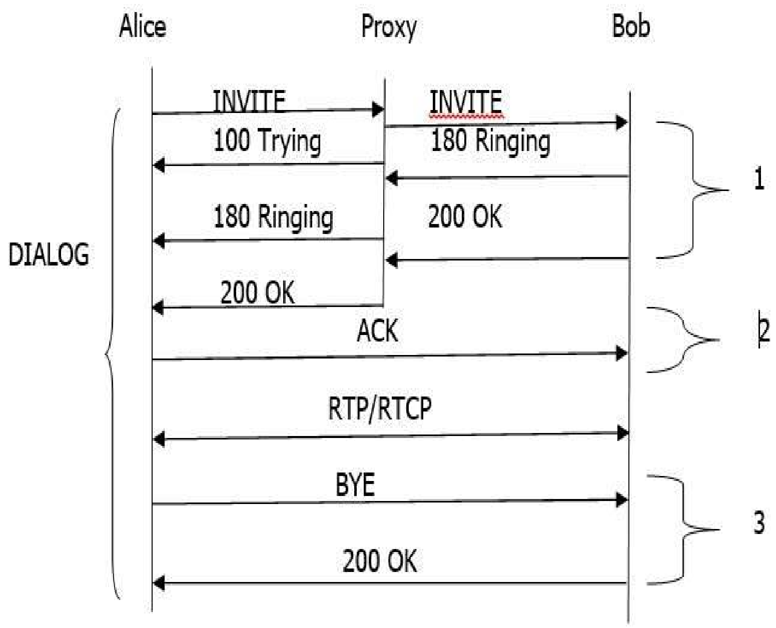

Fig 3: Call Flow

Given below is a step-by-step explanation of the above call flow:

- An INVITE request that is sent to proxy server is responsible for initiating a session.
- The proxy server sends a **100 Trying** response immediately to the caller (Alice) to stop the re-transmissions of the INVITE request.
- The proxy server searches the address of Bob in the location server. After getting the address, it forwards the INVITE request further.
- Thereafter, **180 Ringing** (Provisional responses) generated by Bob is returned back to Alice.
- A **200 OK** response is generated soon after Bob picks the phone up.
- Bob receives an **ACK** from the Alice, once it gets **200 OK**.
- At the same time, the session gets established and RTP packets (conversations) start flowing from both the ends.
- After the conversation, any participant (Alice or Bob) can send a **BYE**request to terminate or end the session.
- **BYE** reaches directly from Alice to Bob bypassing the proxy server.
- Finally Bob sends a **200 OK** response to confirm the BYE and the session is terminated.
- In the above basic call flow, three **transactions** are (marked as 1, 2, 3) available.
- The complete call (from INVITE to 200 OK) is known as a **Dialog**.

To see the routing of calls and how particular call is routed wireshark is used. Wireshark is a network packet analyzer. A network packet analyzer will try to capture network packets and tries to display that packet data as detailed as possible. Wire shark can open and save packets captured from a large number of other capture programs.

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V. SIMULATION RESULTS

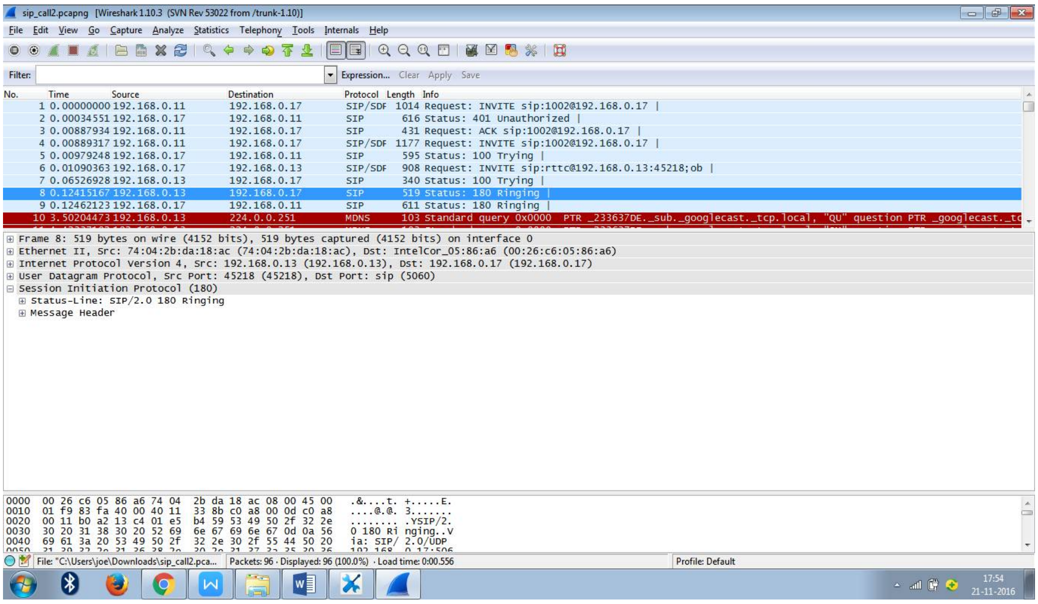

Fig 4: SIP Call Flow

The above figure shows how the routing of call takes place through SIP. The single call analysis is shown in above figure. SIP request message INVITE which is used to invite a user to participate in multimedia session.ACK to confirm the final message has received. 100 trying and 180 ringing are generally response codes. 401 unauthorized are client errors.BYE is a method used to terminate an established session. This is a SIP request that can be sent by either the caller or the called to end a session. The total call status from initial to final can be known in the above figure. As in same way we can also see the RTP and RTCP packets where it hasbeen involved when call routing takes place from source to destination.

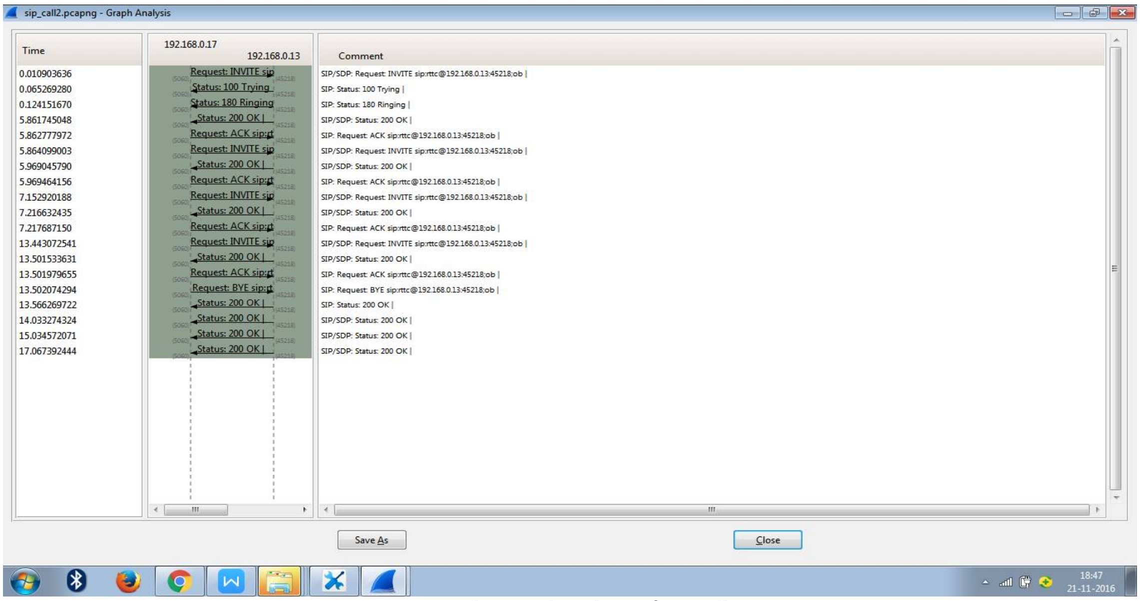

Fig 5:Flow graph analysis of SIP call

The above flow graph which shows the flow graph analysis of the call which is routed in fig.4., This flow which shows

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totally the flow sequence of sip call from source to destination. In the above figure which shows in detail the timing of the call and also the call routing from UAC to UAS. The User Agent Client (UAC) which sends a request and receives a response and the User Agent Server(UAS) it gets requests, processes those requests and generate responses. The routing of the call which can be clearly notifies from starting of the call to termination of the call.

VI. CONCLUSION

Voice over internet protocol phones is the wave of future. As the technology improves VoIP phone systems will continue to thrive as a more efficient, cost effective way to terminate phone calls. The main effect of using VoIP is cost efficiency. It is the main use for long distance communication. Before VoIP , internet users expecting the risk associated with sending data over internet. The issue of the VoIP technology is it is not secured. To defeat VoIP security threats, a well secured plan needs to be designed. The plan should include voice encryption, authentication, voice specific firewalls and separation of data and voice traffic. Using intrusion prevention system (IPS) and specific firewalls changing default passwords on different components of VoIP , system used for securing VoIP. VoIP is not traceable as normal telephony services and so will not work with emergency numbers. Steps to rectify this should have sufficient Bandwidth so that information moves quickly. The main things to consider in VoIP are Jitter which is received in wrong order to receiver and can be overcome by using jitter buffer and Latency which is delay in amount of time to reach destination this can be recovered by having quality network management and bandwidth reservation. The Session Initiation Protocol used here to carry VoIP traffic.SIP trunking which means use of voice over IP to facilitate the connection of private branch exchange to internet.SIP trunking with VoIP provides a new foundation for advanced services. SIP holds lot of promise in today and tomorrow communication world. It has been established as a standard for call control and signaling on 3G cell phone networks by the Third Generation Partnership Project (3GPP). This means that all multimedia and IP voice call signaling will be done through SIP. New services involving fixed network IP services can thus introduced with ease.

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