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QoS of VoIP in Wireless Networks

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Table of Contents

Table of Contents 0
Acknowledgement 3
Dedication 3
ABSTRACT 4
CHAPTER 1 INTRODUCTION 5
1.1 Overview 6
1.2 Motivation 6
1.2 Problem statement 7
Thesis Structure 7
CHAPTER 2 BACKGROUND 8
2.1 Wireless Networking 9
2.2 Wireless Architecture: 9
2.3 Traffic Routing in Wireless 10
  2.3.1 Circuit Switching 11
  2.3.2 Packet Switching 11
2.4 Real Time Protocols 11
  2.4.1 RTP 11
  2.4.2 RTCP 12
2.5 802.11b and OSI Layers 13
  2.5.1 Transport Layer 13
  2.5.2 Network Layer 13
  2.5.3 Data Link and Physical Layer 14
2.6 Wireless LAN Standards 15
2.7 Wireless LAN Performance 16
  2.7.1 Performance Factors and Issues 16
  2.7.2 Impacts on Performance 17
2.8 Voice over IP: 20
  2.8.1 General overview: 20
    2.8.1.1 VoIP Types: 22
  2.8.2 VoIP Call Equipments: 22
    2.8.1.2 Call Signaling: 23
2.9 Protocols in VoIP: 23
  2.9.1 H.323 Protocol: 23
  2.9.2 Session initiation Protocol (SIP): 26
2.10 Encoding Voice: 27
2.11 VoIP in Wireless LANs: 28
2.12 QoS of VoIP in Wireless: 29
  2.12.1 Things that affect Voice Quality in VoIP: 30
2.13 Performance of VoIP in Wireless: 31
  2.13.1 Conversational Quality: 33
2.14 VoIP Issues: 34
  2.14.1 Security 35
  2.14.2 Issues in Wireless Environment: 35
2.15 How to achieve QoS: 36

CHAPTER 3 ANALYSIS....37
3.1 Approach: 38
3.2 Scenario 1: 38
3.2.1 Test Environment: 38
3.3 Scenario 2: 39
3.3.1 Handover: 39
3.3.2 Handover Delay or Latency: 40
3.3.3 Test Environment: 41
3.4 Scenario 3: 43
3.4.1 Test Environment: 43
3.5 Results Gathered: 44
3.6 Summary: 45
CHAPTER 4 EVALUATION 47
4.1 Specification Background....48
4.2 Introduction 48
4.3 Standards for IEEE 802.11 49
4.4 Physical Layer for IEEE 802.11 50
4.5 MAC Layer and IEEE 802.11 51
4.5.1 Distribution Coordination function (DCF) 51
4.5.2 PCF (Point Coordination Function)....52
4.6 MAC Layer and IEEE 802.11E....53
4.6.1 EDCF (Enhanced Coordination Function) 53
4.6.2 HCF (Hybrid Coordination function)....54
4.7 Channel Access Function 55
4.7.1 Legacy 802.11 MAC DCF Channel Access 55
4.7.2 IEEE 802.11e MAC EDCF Channel Access....56
4.8 Evaluation....57
4.8.1 Simulation Results: 58
4.8.1.1 Simulation: 58
4.8.1.2 Simulation: 61
4.9 Proposed scheme: 64
CHAPTER 5 CONCLUSION 66
5.1 Conclusion: 67
5.2 Future Work:....67
LIST OF FIGURES AND TABELS: 68
ACRONYMS:....70
REFERENCES:....73

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Dedication

“I would like to dedicate my thesis to my mom and dad, to my Nana, Uncles and Cousins and to my friends back home for their love and their endless support.”

Naveed Iqbal.

“I would like to dedicate thesis to my parents for their support.”

Fahad Mumtaz Cheema.

ABSTRACT

In this thesis we have focused in the wireless environment and how to run voice application over it. Conducive environment that makes it possible for the voice services to run in wireless is necessary. As we know this well that wireless is a contemporary technology due to it low cost and its effectiveness, and one major advantage of it is the mobility that is one fell free to move anywhere but have the access to the resource. So this makes wireless networks of great value, we in this thesis have focused on wireless LAN's. In second part of the thesis we have shed some light on the VoIP showing how it works in the wireless environment. Analysis phase is relatively more important phase then the previous section which shows issues or hindrances in carrying voice over wireless environment. This analysis shows that these issues still prevails and should be addresses and the corresponding results are also discussed and by looking at those results we have derived a summery out of it. Next chapter we firstly tried to explain why we have chosen specific protocols and then showing some graphical representation measurements that are to address the problem based on the work done. We tried to evaluate EDCF and DCF as these play important role in handling real time applications like voice. After that we proposed a scheme through which these effects can be minimized and to enhance the method is necessary to avoid the issues still in effect.

CHAPTER 1
INTRODUCTION

1.1 Overview

Wireless technology is currently used due to its advantages one bigger one is mobility. Wireless operates in air interface so performance always is directly dependent on the signal strength and varies from topology to topology. Wireless is also being used in medical field to keep intact with the employee. [5] Now a day's voice over internet protocol is accepted and is adopted in institutions. First of all this application was build for a computer to computer transfer but it got its reputation due to its reliability and cost.[1] VoIP is packet-switched technology so Quality of service is an issue. Data being send through this technology is relatively a risk in a sense that receiver is not assured that he will receive the data. "Higher level of protocols is responsible for reliable data sending". [1] In VoIP delay is taken into consideration it should not cross over 120 ms or 150 ms. In case of mobiles when one node often changes its location due to this the strength of the signals weakens and time so needed to switch to new BS and this will create gap or delay. [4] There are may issue related to the deployment of VoIP over wireless LANs as according to Gartner " a wireless LAN for voice costs about double what a data only one costs"[4] This thesis will emphasis on the state of the art communication.

1.2 Motivation

Wireless Networks are implemented in many of the organizations and are popular these days. The value increases when to discuss voice over wireless networks. [1] There is also organizational demand to provide them with this technology. This will be great sense of achievement for us to study and propose some way to enhance and contribute in this field as this always inspires us every time we happen to come across.

This motivated us to access the limitations and our goals are to suggest ways to overcome these problems. For this we read out materials that are available any analyze the deficits, and so we will be able to present our way of thinking in which it should really be in for the best results.

1.2 Problem statement

Voice quality is the basic issue in the wireless networks or environment and sub problems also attached to them as mobility is involved in it. Security, throughput and other factors affect the performance of VOIP in Wireless Networks.

As far as QoS is concerned delay and packet loss are the major issues and in future it would be a challenge for IP based networks. [2] Consider in the case of BS, hand off becomes a problem. Breakage of the signals as the subscriber moves between different base stations. Security is also a concern in this type of environment. "WEP is used as the security mechanism in networks to authenticate user."[3] With the passage of time newer problem arise so the adaptation techniques are required and that can be possible as to introduce newer ways as a remedy.

Thesis Structure

- In Chapter. 1, how we were tempted to take this thesis, discuss the problems in this environment.
- In Chapter 2, background of the technologies that are part of our thesis. Complete over on wireless Networks and VoIP and how it works.
- In Chapter 3, Analysis if VoIP QoS as per the work done and compare the results.
- In Chapter 4, Evaluation that are small set of graphical representations by us of measurements done already for QoS to evaluate between the protocols of IEEE 802.11 and IEEE 802.11e and the proposed scheme to make it nor effective.
- In Chapter 5, Conclusion and Future work pointers.

CHAPTER 2
BACKGROUND

This section gives an overview of the technologies that are part of this thesis that are Wireless Network and VoIP.

2.1 Wireless Networking

Wireless Networking is primarily used for wireless communication. Wireless Networking is a computer network that is wireless, interconnecting nodes without using wires. It can be of two types wireless LAN or Wireless MAN. IEEE standard for Wireless Network is 802.11 and the band used is 2.5 GHz while using the protocols defined by IEEE 802.11 and IEEE 802.11b standards.[4]

Mobile Network having BS and that BS's, in turn, is connected to central MSC or MTSO. This central hub provides connectivity to PSTN provides global access to the subscriber connecting to BS. Mobile subscriber connects to BS through and radio link is established using CAI known as *handshake*. This specifies four channels namely:

1. FVC voice transmission from BS to mobile device.
2. RVC voice transmission from mobile device to BS.
3. FCC and RCC used for initiating calls.

MSC known as MTSO is connected to BS through landline or microwave links. Here MSC is responsible to establish connection with PSTN as well, and BS seems bridge between mobile and MSC for handshake whenever mobile wants to establish call. MSC moves to unused channels by informing BS and FVC and RVC come into act changing the frequency. Once call is in progress MSC keeps on changing the channel to maintain good quality of voice. [6]

PBX is a conventional phone system and many of the companies are trying to change this. This will in future be replaced by VOIP.

2.2 Wireless Architecture:

Now a day's world is getting much dependent on wireless networks. It has become an integral part of our life as people like to move freely from place to place not remain sitting at one place.

There can be many types of architectures but one of tem is shown in fig 2.1. This architecture is the combination of both wired and wireless which is integrated to perform the functionality. Data is to be delivered between many devices like laptops, mobiles and other devices. Here one of the functionality is to provide VoIP for the devices like mobiles and laptops.

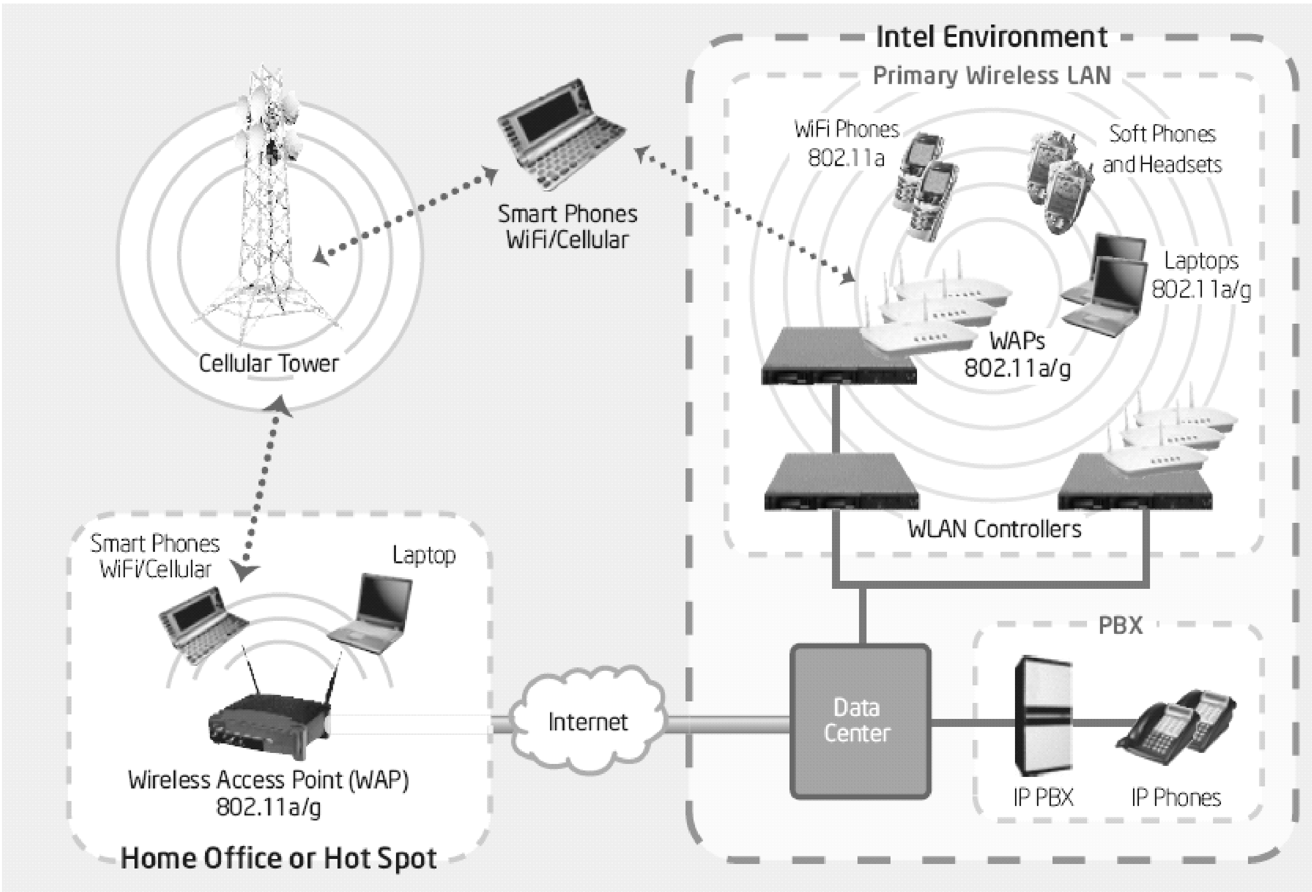

Fig 1 Wireless LAN Architecture [16]

2.3 Traffic Routing in Wireless

In wireless environment knowledge of the traffic type must be on the foremost priority. Voice traffic need dedicated link and delivery should be real time while some of the traffic need not to send real time. There are two types of routing techniques connection-oriented routing and connectionless routing.

In connection oriented routing path is dedicated between the transmitter and the receiver so integrity of the data sent is acknowledged. Error control is maintained and if call breaks, sender has to retransmit the data and repeat the whole procedure again.

Connectionless routing doesn't have fixed channels. Data sent is in the form of packets which in turn forms a message. Packet can be disturbed and are reordered at the receiver end. Retransmission is not done in the type of routing. [17]

2.3.1 Circuit Switching

In circuit switching bandwidth is defined even before the communication parties communicates they will have to communicate on the channel allocated to them.

A voice channel is dedicated by MSC between BS and PSTN throughout the time the call is active. There can be handoff but the link is maintained between MSC and PSTN. Circuit doesn't support wireless communication due to their inactive period that results in breakage of the signals.[8] Circuit switching is reliable and one the call is established communication is error free.

2.3.2 Packet Switching

In packet switching data is transferred in form of packet and that channel is shared be other communication parties also. Different routes can be taken by each packet as per the information and depending on the type of switching. To ensure delay and error removal max length of packet is defined. Extra amount of information is added to provide source and destination information. [13]

At the receiver end data is arranged or assembled. Data transferring doesn't need dedicated link as the data is divided into chunks and these are to be transported to the receiver as strategy adopted is sharing of resource. X.25 and IP are the examples of Packet switching.

2.4 Real Time Protocols

There are two types of real time protocols used for communication.

2.4.1 RTP

Real time transport protocol (RTP) is useful for real time communication and defines Packet format. RTP protocols are used for Audio and Video transmission in multicast and unicast atmosphere. It runs over UDP and is part of application layer. [14]

Encapsulation of data is done. RTP is comely used in IP telephony. RTP has information regarding.

- Contents of message
- Sequence number
- Jitter
- Monitoring timely Delivery

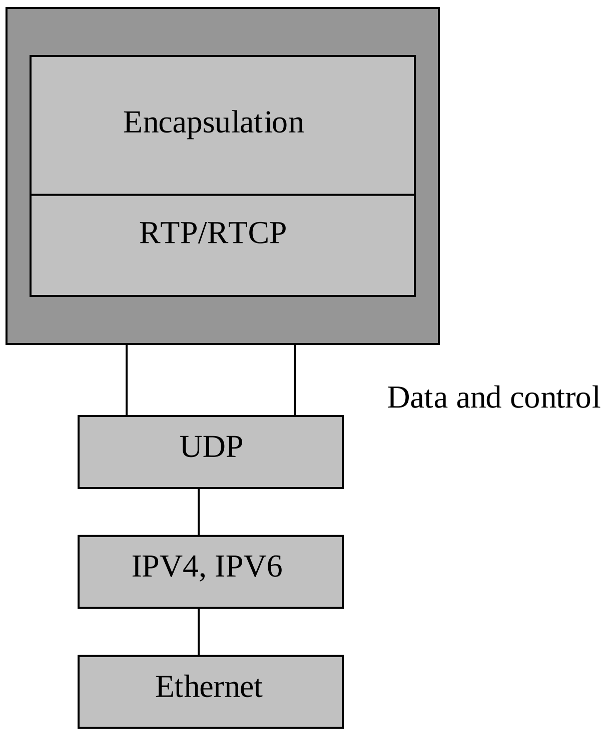

Fig 2 RTP/ RTCP Format [11]

2.4.2 RTCP

RTCP usually functions with RTP protocol and by the combination of both multimedia data is transported between the parties that are sender and the receiver. It is use dot provide feedback on the services like QoS provided by RTP. Packet differentiation being transmitted is done through ports. Higher port is used by RTCP than to RTP. [1]

2.5 802.11b and OSI Layers

802.11b consists of five layers that are application, transport, network, data link and physical layer.

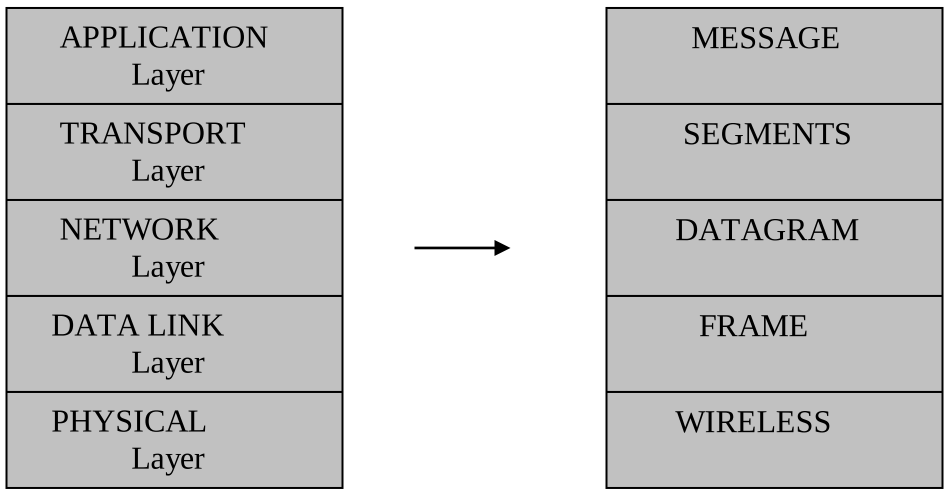

Fig 3 OSI Layers with their purpose. [11]

2.5.1 Transport Layer

Transport Layer provides service to application layer that is subsequent to it using UDP protocol. UDP is needed to send data in VOIP at it minimum threshold and is not reliable in any context as it don't provide mechanism to prevent packet loss. [11] On other side SIP and H.323 protocol are developed for establishing calls. TCP is also used by transport layer for delivery of packets.

2.5.2 Network Layer

Network Layer uses IP (Internet Protocol) which is responsible for Datagram routing between router and the work station. IP Datagram format in IPv4 is:

<table>
<tr><th colspan="5">32bits</th></tr>
<tr><td>Version</td><td>Header</td><td>Type-of Service</td><td colspan="2">Datagram length [bytes)</td></tr>
<tr><td colspan="3">16-bit identifier</td><td>flags</td><td>13 bit fragmentation offset</td></tr>
<tr><td>time-to-live</td><td colspan="2">upper-layer protocol</td><td colspan="2">header checksum</td></tr>
<tr><td colspan="5">32-bit source IP address</td></tr>
<tr><td colspan="5">32-bit destination IP address</td></tr>
<tr><td colspan="5">Options</td></tr>
<tr><td colspan="5">Data if any</td></tr>
</table>

Fig 4 IP Datagram Structure [11]

2.5.3 Data Link and Physical Layer

Data link layer move Datagram and Physical layer moves bits node to node. Data link splits into two LLC and MAC while physical layer is divided into PLCP and PMD.[11] LLC functionally is to multiplexing protocols when receiving from MAC and de multiplexing when transmitting, it often provide with flow control and retransmission. MAC layer defines two modes DCF and PCF. PCF has goon old age and DCF is normally based on CSMA/CA. CSMA/CA detects and works effectively to stop collision. It checks the channel is free if it’s vacant send packet and waits otherwise. Timer is set randomly and channel is checked again if slot is available.

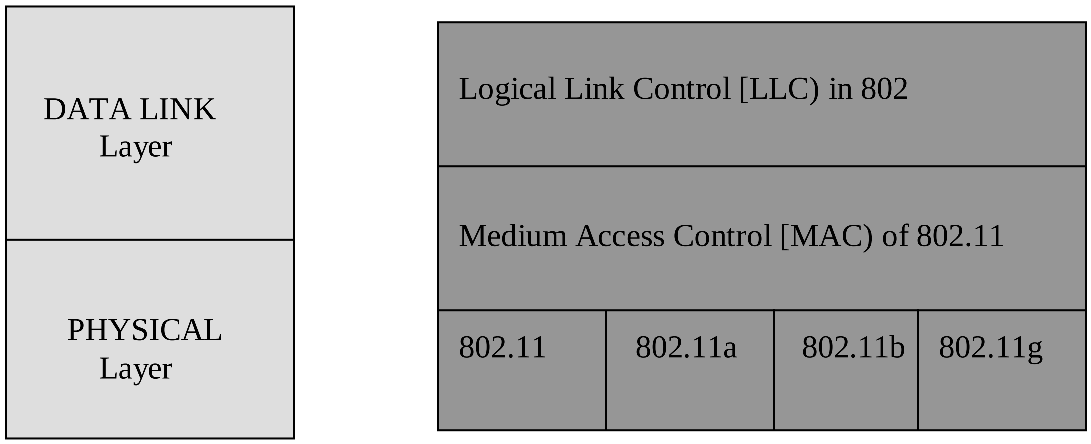

Fig 5 OSI Layers and 802 [12]

2.6 Wireless LAN standards

There are several families. Most common family of IEEE 802.11 standards are:

802.11b: 11 Mb/s bandwidth 2.4 GHz.

- Modulation DSSS- CCK.
- Widely used standard. 14 channels are available.
- Fewer AP’s for large area coverage. [4][7]

802.11a: 54 Mb/s in 5 GHz bandwidth.

- Modulation used is OFDM.
- More AP’s than 802.11b for larger area coverage.
- Lesser Radio Frequency [RF) interference.
- Bandwidth is greater so it supports video applications as well. [4]

80211 g: Up to 54 Mb/s in 2.4 GHz bandwidth.

- Modulation OFDM or DSS [4][7]

Wireless Protocols	Number of Non-Interfering	Modulation	Maximum Link Rate	Maximum TCP Rate	Maximum UDP Rate
802.11b	3	CCK	11Mbps	5.9 Mbps	7.1 Mbps
802.11g [with 802.11b)	3	OFDM/CCK	54 Mbps	14.4 Mbps	19.5 Mbps
802.11g [11g-only mode)	3	OFSM/CCK	54 Mbps	24.4 Mbps	30.5 Mbps
802.11a	19	OFDM	54 Mbps	24.4 Mbps	30.5 Mbps
802.11a Atheros Turbo Mode	6	OFDM	108 Mbps	42.9 Mbps	54.8 Mbps

Table 1 Characteristics of various forms in wireless LANs. [15]

Security mechanism in all of them is WPA and WEP. WPA is not reliable so WEP is the security mechanism that is used for encryption. [7]

IEEE 802.16 is standard of WiMax bandwidth is 75 Mb/s, used in Wireless MAN, security mechanism used is AES and modulation both OFDM and DES3.[4]

Bluetooth is also wireless technology using bandwidth 2.4 GHz, modulation FHSS, Security SSL or VPN. [4][9]

There are many others but 802.11c, 802.11d, and 802.11e etc. Most commonly used is the standard 802.11b.

2.7 Wireless LAN Performance

Wireless LAN performance is affected by several issues. This section will put light on these.

2.7.1 Performance Factors and Issues

Wireless LAN performance is directly affected by signal strength. To improve signal strength placement of AP’s is required. This can be achieved by the experts or by automated loops.[4] We consider in case of out door AP’s are placed in direct line of sight with the client while in the case of open offices or closed offices there is no line of sight as there are many obstacles in between like buildings. For large area coverage more and more BS must be deployed which are directly connected to MSC. [8] Mobility and the interference destroy the versatility of Wireless Networks. There should be no packet loss in this type of communication to avoid this encoded algorithm is employed which selects lower and higher transmission rate. [4]

Through this maximum throughput is achieved, and in case of VOIP it has a significant impact and performance level is decrease. A unique and hostile problem is the channel capacity. Node wants to be facilitated from long, continuously and this should be quick, here switching of BS is required by MSC and it has limited spectrum or bandwidth. [8] Modulation plays a vital role here.

Security has clear and undoubted impact on the performance. WEP is used as to encrypt data but it can be cracked also. While SSL and VPN encryption is also used to secure data. If security is not achieved it will create performance deficits. Wireless MAC has some problems and delay and packet loss still pertains and in case of VOIP these factors these has serious influence on throughput and demand. [4] These factors have influenced performance of Wireless LAN.

2.7.2 Impacts on Performance

Coverage radius is very important factor in throughput. Coverage radius is the distance between the AP's and the area covered by them, here obstacles are also encountered. Performance is affected by these factors Interference, scattering and fading. Load on network and location. [10]

Interference: Device occupies the same spectrum which is already in use. This impact's SNR creating noise. [10]

Scattering and Fading: Obstacle creates fading and scattering of signals. This produces errors in transmission which end in a delay.

Load on Network: Two factors are involved in this type of problem one is number of users on the same medium. Sharing of a medium is a big issue.

Second is ''traffic mix Applications may be more demanding in terms of bandwidth as compared to others''. [10]

Location: distanced AP's also create performance deficits client increase which decreases rate of data. [10] It is affected by client frequently changing location.

OSI Layers: Each layer in OSI has it own functionality by processing data it passes it to the next layer. Packets are encapsulated and to avoid overheads these are emphasized. Due to occurrence of the overhead significant impact on voice is observed. [11]

Threshold is to obtain certain bit rate which can be the minimum level or higher level if this decrease or increase it impacts performance. Good transmission rate guarantees good performance and if transmit power decreases it impacts performance. Switching between receiver and transmitter requires time and if switching period increases results in bad performances. Wired and wireless Networks have their own impacts while considering different types of matrices. This is shown by the table below.

Metric Category	Wire-Network Metrics	Wireless-Network Metrics	Impact of an unstable wireless environ ment
Packet forwarding	Loss	Loss	**High**
	Forwarding rate	Impact of rate adaptation on loss	**High**
		Impact of roaming on loss	**High**
		Impact of overlapping BSS on loss	**High**
		Impact of RTS/CTS on loss	**High**
		Impact of power management on loss	**High**
		Impact of MAC layer fragmentation on loss	**High**
		Impact of encryption on loss	**High**
Security		Association performance	**High**
		Authentication performance	**High**
		Association and authentication capacity	**High**
QoS	Delay	Delay	**High**
	Jitter	Jitter	**High**
		Impact of rate adaptation on delay & jitter	**High**
		Impact of roaming on delay & jitter	**High**
		Impact of overlapping BSS on delay & jitter	**High**
		Impact of RTS/CTS impact on delay & jitter	**High**
		Impact of power management on delay & jitter	**High**
		Impact of MAC layer fragmentation on delay & jitter	**High**
		Impact of MAC layer fragmentation on & jitter	**High**
		Impact of MAC layer fragmentation on delay & jitter	**High**
		WME relative priority forwarding rate	**High**
		WSM stream bandwidth allocation	**High**
Behavioral	Head-of-line blocking	Forwarding in presence of congestion	Medium
	Error analysis	Security counter measures	Low
		Power save	Medium
Rate adaptation		Rate adaptation time	**High**
Rate adaptation		Rate adaptation hysteresis	**High**
		Rate vs. range	**High**
Roaming		Roaming time	Medium
		Roaming session continuity	Medium
		Roaming hysteresis	**High**

Table 2 Wired and Wireless Networks and impacts on them [4]

2.8 Voice over IP:

VoIP is modern technology being used due to it advantages that it provides over conventional systems. In this chapter it will be discussed in detail.

2.8.1 General overview:

Internet has become a major part our daily life and for that matter demand of different applications have also increased. To fulfill the user demand newer applications are being built. VoIP is among them and is supplement or an alternative to PSTN which is lesser beneficial then VoIP in terms of cost and flexibility. First experiment on this technology was done in 1970 on then ARPNET. At that time nothing was done as the equipments were didn't had the capacity to compress voice but in start 90s first Internet Telephony application launched. [18] There are advantages and disadvantages using VoIP. Major Advantages are

1. PC to PC calls are very cheep so VoIP has low cost.
2. In Wireless connections, VoIP is portable.
3. Call forwarding, voice mailing etc are also provided by VoIP.

Disadvantages:

1. Emergency calls are sometimes unreliable.
2. In case of no electric supply you are not able to make calls.
3. Jitters, Echo, Data loss are the pertaining problems.

VoIP is full duplex communication between two parties and uses packet switching for communication. In this way of communication there is no channel reservation, if any other party wants to send or to use the channel at the same time it can. Queuing takes place and the packet previously transmitted stops and suffer delay. On the other hand PSTN use circuit switching and link is dedicated between hosts. Resource is always reserved even when the system is idol. [18]

Concept	Voice over PSTN	Voice over IP
Switching	Circuit Switching (end to end dedicated link)	Packet Switching
Bit Rate	64kbps per 32kbps	14 kbps with overheads (only when talking)
Latency	Lesser than 100ms	200-700ms depending on total traffic on IP network.
Bandwidth	Dedicated	Dynamical allocated
Cost of access/billing	Business customer. Monthly charge for line, plus per minute charge.	Business customer. Cost of IP infra-structure, Hybrid IP/PBX and IP Phones.
Equipment	Dump terminal (Less expensive) intelligence in network	Integrated smart programmable terminals(expensive) intelligence not in network
Quality of service	High(extremely low loss)	Low and variable, but traffic is sensitive depending on packet loss and delay experienced.
Network availability	99.999% up time	Level of reliability not known.
Security	High level of security because of dedicated link.	Possible eavesdropping at router.

Table 3 Comparison of quality of Voice over PSTN and over IP. [18]

2.8.1.1 VoIP Types:

Computer to computer, IP Phones, Computer to phone, ATA are the types of configuration that can be found in internet telephony.

- **Computer to Computer**: This is the easiest way of communication without paying for the call. MSN or Skype are type of it, through which people can communicate. Distance is not an issue in this and user needs a headset and a DSL connection.
- **IP Phones:** It is Voice over Internet protocol (VoIP) communication and works on internet instead of PSTN. RJ-45 connector is used instead of RJ-11 and phone looks like a normal phone. [18] It can also be a Soft phone installed in computer. SIP to SIP call can be made as both parties have the software installed or SIP hardware. The call will be free if either one of them is used.
- **Computer to Phone:** In this way on communication the charges and money making starts when a call is made through a computer to a land line phone. It works in the same way as Computer to computer call, VoIP software installed in computer. But these calls are cheaper as compared to landline to landline calls.

ATA:

Easiest way of using VoIP technology is through ATA. This device is used to convert analog signals to digital signals. Cable through your telephone is connected to ATA and then plugs cable out of ATA to computer and computer to internet. Some of ATA has software installed n them then one don’t need to connect the computer.

2.8.2 VoIP Call Equipments:

VoIP looks same like the traditional telephony system, but in real it has some fundamental differences. Major difference between PSTN and VoIP is the later functions on Internet or IP. VoIP calls being made through computer using headphones and microphones are common but in modern technologies IP Phones are being used for the communication. They are very much identical to the normal phones. Newer technologies are being introduced by the manufactures in the market nowadays, the technology using Wi-Fi, Bluetooth etc for the communication and selecting the network that has best performance. [19] Most common of the devices used in this type of systems are:

- *End to end systems* are the devices that clients have that can be the IP phones.
- *Gateways* help the call to be placed and handled between multiple users. [18]
- *Servers* In VoIP important role is of the call server that manages the call that is being handled at the same time. H.323 can be used for the purpose.[19]

2.8.1.2 Call Signaling:

Call signaling includes many things like setting up calls, interpretability, user identification and registration [22] and many others. Simple example in setting up calls firstly all of the equipments are configured either by using computer or IP phone. Call contents must be received at the receiver end as they were transferred. Additional protocols are needed for that purpose and H.323 and SIP serves the best. SIP is the most reliable of all the protocols that transports the data to the receiver. [21]

2.9 Protocols in VoIP:

Protocols are used for transmission of data/voice that enables the call to take place. Two common protocols used in VoIP are H.323 and SIP.

2.9.1 H.323 Protocol:

H.323 uses many other protocols to performs it functionality like ‘H.245 for call setting up for data conferencing T.120 for data conferencing G 711/712 for codec specification RTP/RTCP for sequencing audio and video packets [22]

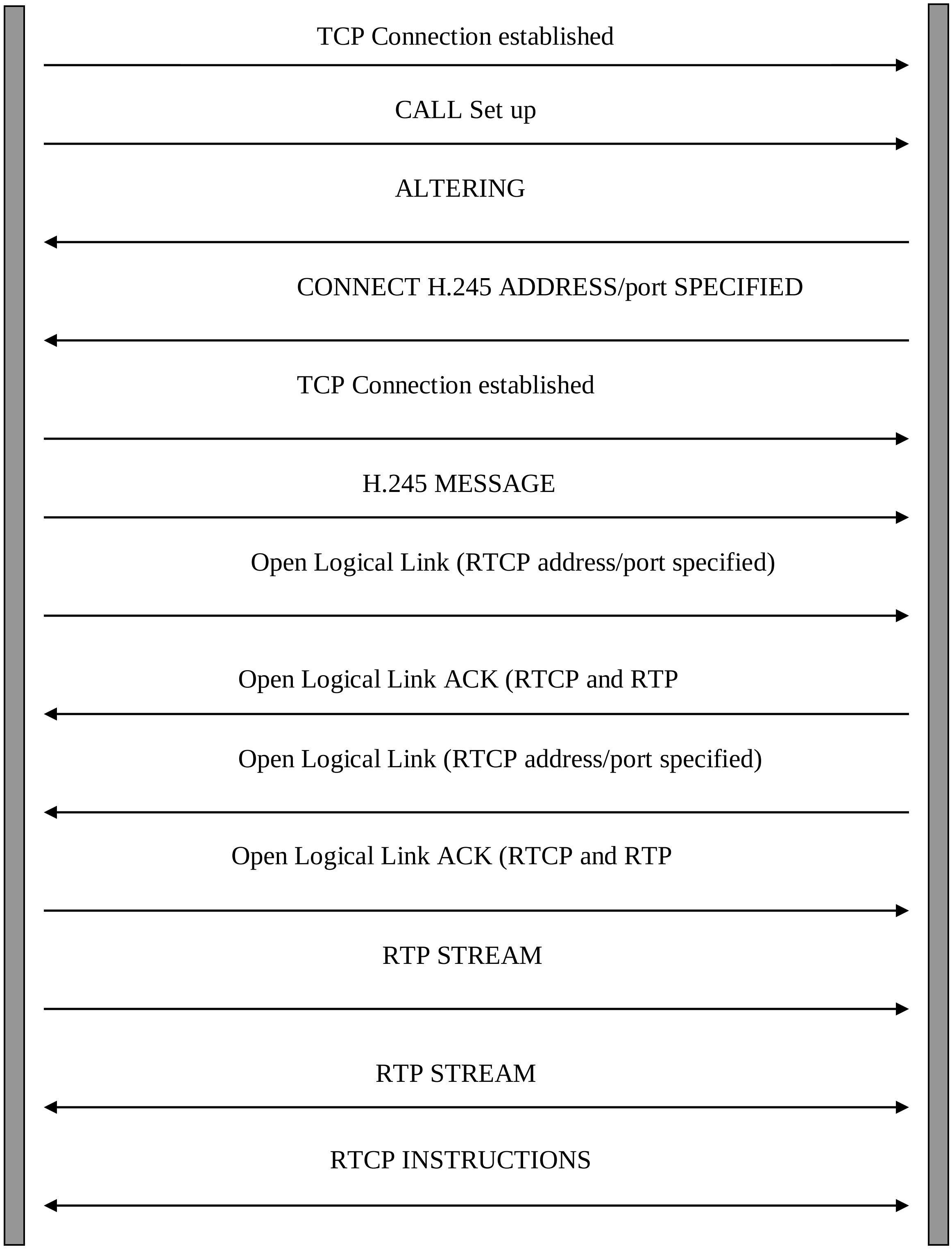

Fig 6 H.323 Call Set Up Process [36]

Protocol used in IP based networks for audio and video transmission. Entities are **Terminals** the clients, **Gateways** for the connectivity between different networks like PSTN, **Gatekeepers ‘‘**responsible for call control, address translation and bandwidth management ’’ [22] and **MCU** for multi-conferencing.

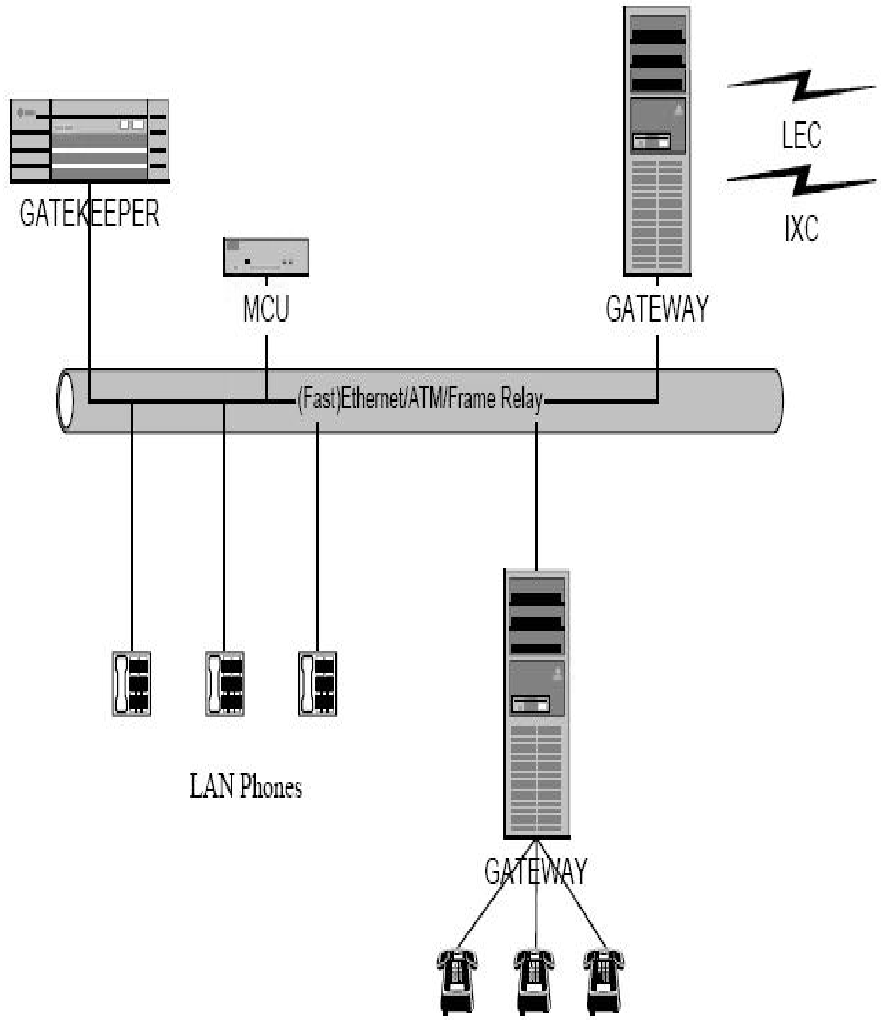

Fig 7 VoIP Structure [23]

2.9.2 Session initiation protocol (SIP):

Developed by IETF There are two major components of SIP. SIP has the same functionality as of HTTP and can be carried by UDP or TCP. SIP networks are made up of Proxies, Location register, End points and servers (location servers). [24] SIP has following functions to perform namely. [34]

- Translation of names and reach out to the location to which call is placed.
- Groups having calls they can be more then one can have negotiation on features. These features can be changed as new part involves.
- Parties in call can make and cancel calls in SIP.
- Codec's selection for good quality

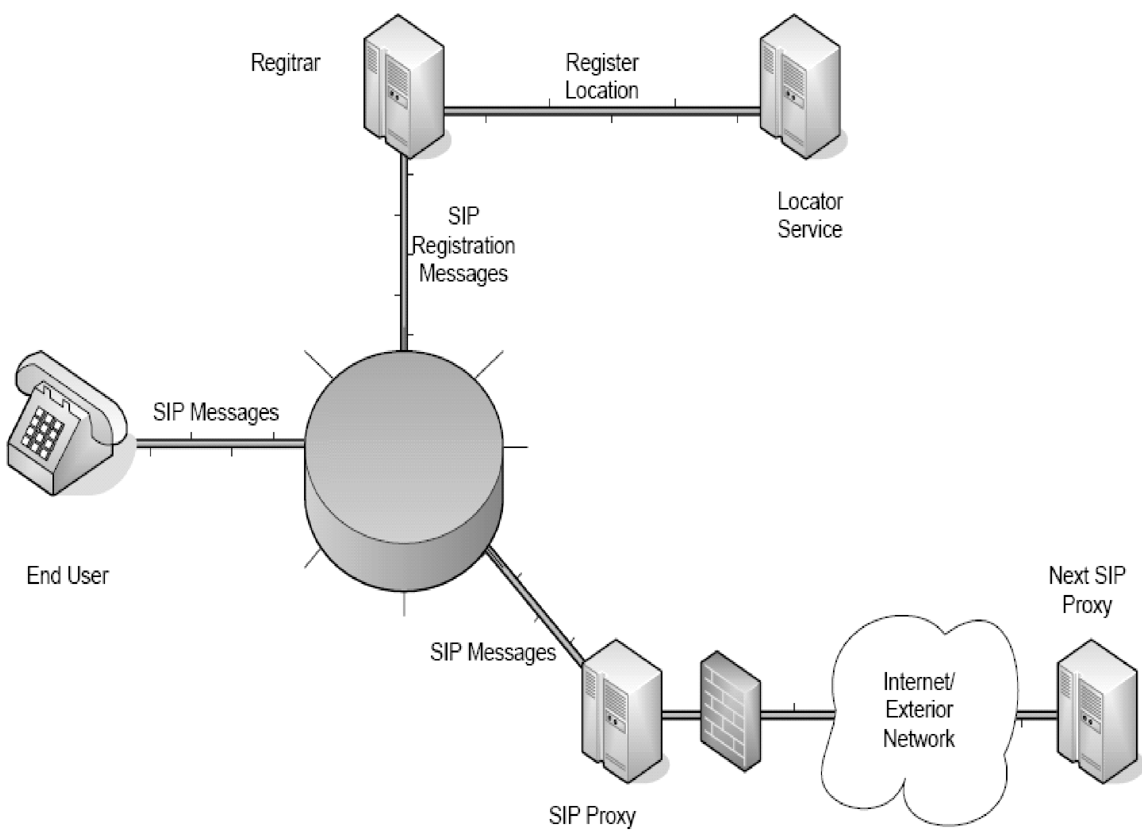

Fig 8 SIP Architecture [24]

2.10 Encoding Voice:

VoIP act differently in many aspects from conventional PSTN. Signals are converted to digital before sending and are done by A/D converter. Reverse procedure is applied on the receiver to gather data done by D/A converter. There are many intermediate devices which serve the purpose as shown in the Fig.

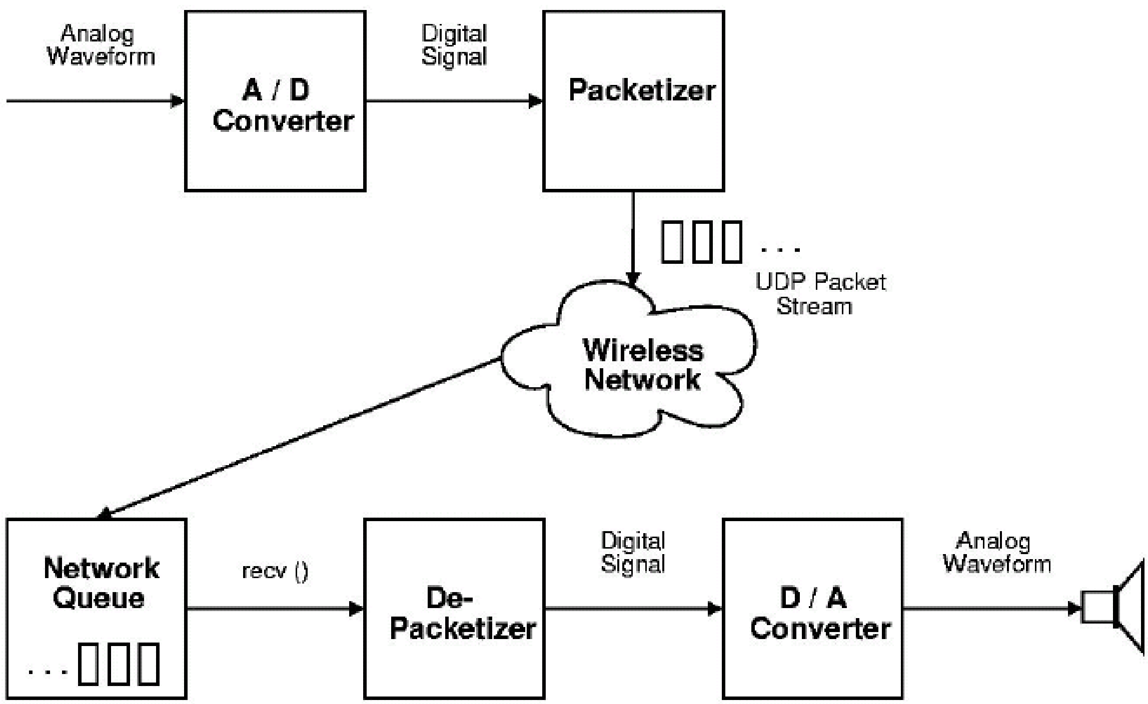

Fig 9 VoIP process [21]

Codec’s serves the purpose of encoding voices. And it task is to compress/decompress, Encrypt/decrypt. Some of the commonly used codec’s are listed with their performance

Codec’s	Performance
G.711	200 PPS(5 ms packet duration) 98 bytes packet size (40 byte payload+ 58 byte overhead) Bandwidth required: 156.8 kbps per direction
G.729	100 PPS(10 ms packet duration) 968 bytes packet size (10 byte payload+ 58 byte overhead) Bandwidth required: 54.4 kbps per direction
G.723r63	32 PPS(30 ms packet duration) 82 bytes packet size (24 byte payload+ 58 byte overhead) Bandwidth required: 21.64 kbps per direction
G.711r53	33 PPS(30 ms packet duration) 78 bytes packet size (20 byte payload+ 58 byte overhead) Bandwidth required: 20.592 kbps per direction

Table 4 VoIP Codec’s characteristics [31]

2.11 VoIP in Wireless LANs:

Voice Signals are sent / transmitted in the form of data between two or multiple mediums in VoIP. Both Wireless and VoIP is getting popular day by day. Mobility is one good advantage that VoIP in Wireless LANs has over VoIP in Wired but there are also so disadvantages as well for that it is not that much popular in current era. There are many other hindrances in these technologies and among them is QoS. [35]

Multiple access points are needed in wireless environment as mobility is the key feature and this feature introduces some problems. User often moves out of the range of the AP so to overcome this problem Multiple APs are required not to degrade the quality of voice. If user moves form one AP to another roaming takes place. Roaming has an effect on call quality that is gap in receiving voice (Delay).

2.12 QoS of VoIP in Wireless:

When ever VoIP goes wireless it faces hazard like packet loss, jitter, and latency. VoIP is normally used for data transferring and preferred to be used for calling purpose. VoIP performs very well and encounter no problem when numbers of active calls are less. Problem arises when the caller increases example is ten callers are simultaneously handled there will be an obvious degradation of voice quality and many of the calls will be dropped as well. If some other data (Background Data) is added in VoIP environment there is a serious damage to the voice quality. This shows that there is a lack of QoS that has control over data flowing through the system and data can of different type as well. [19] Real time applications thus are not supported by VoIP. 54 Mb/s in wireless networks is more then enough as utilization is up to 1%. Different applications with different requirements in Wireless environment are to have QoS implied on them for better performance.

If we see other way around QoS has effect, applications like voice calls has significant effect on them without decreased quality and handling many calls at the same time. If background data is added effect is significant and performance is decreased. [19] Bandwidth management is also an integral part of QoS. Bandwidth is allocated to the work group on which data is transferred and data rate is considered but higher priority data should be considered. [19]So, there is a lot to be done in this technology.

In this we have to take care of following that are addressed before namely Delay, Jitter and Packet loss.

- Good perceiving voice is when **Delay** is less tan 80ms and far more if we talk about acceptable voice that is 150 to 180ms. this is directly associated with poor bandwidth and the congestion in the system. To avoid this there should be guarantee of the bandwidth allocation that is needed and with no congestion.
- In simpler terms Variation of delay is **Jitter**. In terms of range, the range that is acceptable is up to 30ms.This is also associated with congestion when as when the data sending streams is affected and thus variation in delay occurs. It can be improved by the jitter buffer that store the data streams and smoothly delivers the data afterwards. For the prevention from jitter management of bandwidth is necessary.

- Packet loss as clearly stated by the word losing information that is required. In terms of acceptable that is 1% and beyond this and when reaching to 3% or more one will notice breakup of voice and thus ends up with conversation breakup and call are dropped.

2.12.1 Things that affect Voice Quality in VoIP:

User always wants to have a reliable and qualitative data that he receives otherwise this can be assumed that there is still Dark Age on the technology. VoIP is among the technologies which suffer many drawbacks still to be overcome. People still are hesitant to use the technology as they are used to Landline Phones which serves great to their interest. To maximize the Voice Quality following things are to de considered.

Bandwidth:

Voice quality is affected by the connection you possess, if connection support communication in this environment quality will increase. Dial up connection don't support the technology but broadband do but also has an effect is shared by other applications to. [26]

Router:

For Routers ‘‘Compression technology supported, Echo cancelation, Security ’’ are to be considered. [26]

Equipment:

Good quality equipment plays an important role in quality of voice as compared with degraded and devalued equipments. These degraded equipments are the cheep equipments and in turn budget has an impact on the voice quality as well. Routers, IP Phones specs are to read carefully and then choice is to be made what to go for.

Frequency of Phone:

Frequency of IP phone can in conflict with other VoIP equipments like Routers or ATA. 5.8 GHz phone in some of the environment counter problems and many techniques fails other than changing the phone having lower frequency 2.4 GHz. [26]

Location of Equipments and Weather conditions can be the problem in voice quality as in case if ATA are Routers placed to close to each other quality will be effected due to electrical feedback. [26] Top VoIP Providers are: [27]

- Vonage
- AT&T
- Voip.com
- Packet8
- Broad Voice

2.13 Performance of VoIP in Wireless:

Performance is the key to success of the system otherwise user will opt for other technologies as user always need good performance and he pays for it. To make it possible it is important to have good assessment of UPQ. User Datagram quality in telecom networks plays an important role as many new transmission systems are used so the characteristics also changes of these system which must be kept in view. In UPQ test are done and signals are sent between two communicating parties. Results are computed and on the bases of these gathered results, damages are checked. [19]VoIP performance has:

Lost/Damage Packets: Overloading at Routers, AP's results in transmission error, Packet lost and damaging of packets. [25] VoIP in wireless LANs use UDP protocols for transmission which is not reliable.

Delay: Delay is receiving the data packets late and due to this voice quality is affected. So the reliability is not achieved as packets are sent to the receiver machine and they are received late or not received. If delay is constant so it is acceptable but most of the time it is not so this inconsistency of data results in jitter. PSTN has 10ms delay and VoIP has 200 to 400 ms delay.

End to End Delay: [25] Types are Queuing delay, Packet delay, Delay by decoder, Transmission delay, Serialization delay.

Out of Order Packet Delay: This happens normally in IP networks and is overcome by RTP sequencing. [25]

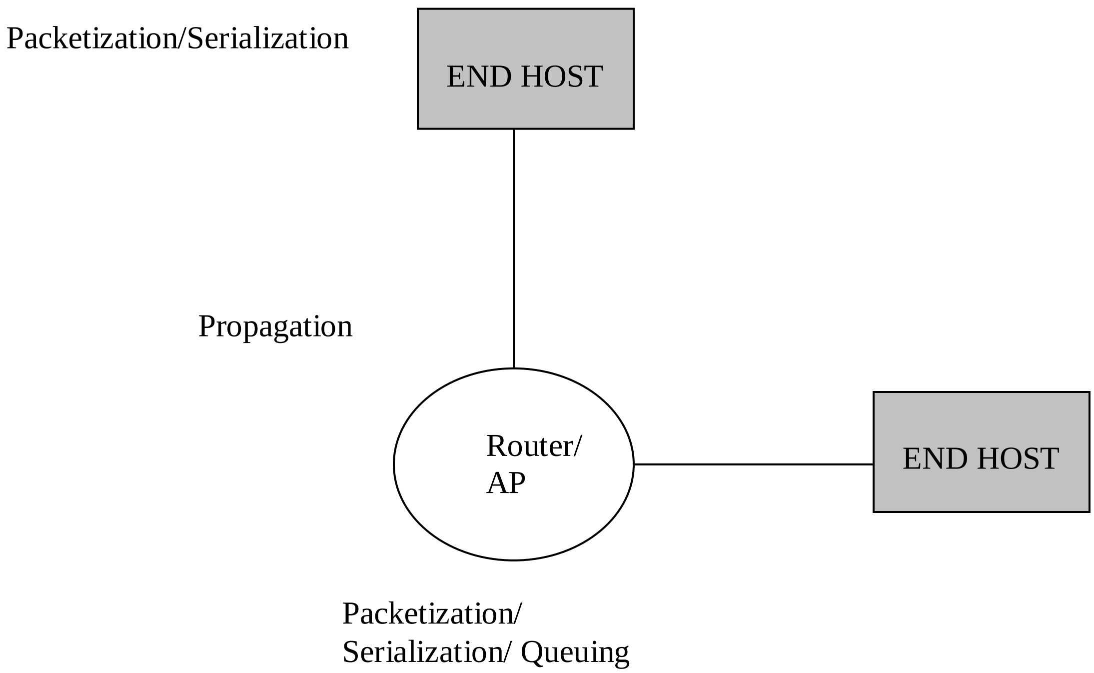

Fig 9a Delays [30]

2.13.1 Conversational Quality:

Convergence of communication is nearly to happen in VoIP. As an example to understand this, person has a budget to purchase a one device but methods are devised so that he can buy two of them in single price. Like giving options in one device and one can enjoy the facility of different devices in a device. As the services are the most important thing and people opt for and go for certain things after having it features in view. [29] As in IP based networks video, voice, audio are combined together and is provided onto a same network.[29] In VoIP limited resources are used and they perform huge functionality which one can never imagine before. This is Convergence of communication. But there are some problems that affect the quality like limited bandwidth, security and applications. Conversational quality can be enhanced by selection good Codec's which is used to compress data. Each compression has its own functionality to perform. Codec's not only compress data but encode data as well. *Loose compression* is in which some data is lost. Codec's are selected while keeping in view delay factors like G.723 always adds latency of 30ms and efficiency like payload efficiency. Some codec's performance are shown in the table

Packet Duration	5ms	20ms	30ms	40ms
PPS Per call	400 PPS	100 PPS	66.7PPS	50PPS
Packet Size	98 bytes	218 bytes	298 bytes	378 bytes
Required Bandwidth	313.6 Kbps (156.8 Kbps per direction)	174.4 Kbps (87.2 Kbps per direction	158.9 Kbps (79.5 Kbps per direction	151.2 Kbps (75.6 Kbps per direction

Table 5 G.711 Codec [31]

Packet Duration	10ms	20ms	30ms	40ms
PPS Per call	200 PPS	100 PPS	66.7PPS	50PPS
Packet Size	68 bytes	78 bytes	88 bytes	98 bytes
Required Bandwidth	108.8 Kbps (54.4 Kbps per direction)	174.4 Kbps (31.2 Kbps per direction	158.9 Kbps (23.4 Kbps per direction	151.2 Kbps (19.6 Kbps per direction

Table 6 G.729 Codec [31]

Some other issues have relation with bad voice quality like delay, jitter and packet loss. *Delay* always produces Echo in VoIP calls and overlapping of voices. [22] Echo is the repetition of voice to the speaker. Voice overlapping is when two simultaneous speakers talk at the same time interrupting each other. ‘‘ ITU recommends round trip delay up to 300 ms that is 150 ms one side’’. [22]

Loss of Packets happens due to the overloading of link, when excessive data is being sent on to the link. Codec’s are to be held responsible for this situation. One reason is when packet loss is more than 5% and other is grouping of packet losses forming an accumulative power as one. In such cases packet loss seems to be unnoticed by the codec’s. [32]

Jitter is the delivering of packets in the wrong odder. Jitter depends on the codec’s equipments. ‘‘If jitter exceeds the levels for which codec’s can buffer, the call begins to clip and dropped’’. [33] 15 ms is the jitter threshold.

2.14 VoIP issues:

Some of the key issues with this technology still prevail for that commission like Federal communication commission (type of a regulatory authority) works on them to minimize the issues related tot VoIP. Some of them are.

2.14.1 Security

Introduction of VOIP has increased the need for security. Our data and voice needs to be confined. Both personal and monetary data needs to be private and must be protected. Not many organizations worry to encrypt voice traffic over conventional telephone lines. The same theory is not true for Internet-based connections. In simpler term, when putting an order over the phone, many individuals will read their personal data on credit cards e.g. credit card number. In distinction, the jeopardy of transferring data across internet is more significant. The packets sent from user's PC to any online vendor may pass through several systems that are uncontrolled by the user's ISP or the vendor. [36] Since the sent information is digital in nature, which can be accessed by software's scanning packets. To safeguard user's information encryption is used to hide the information from the person with malicious intent. It implies that security measures need to be implemented for transmitting such information over the internet. ''The existing internet structural design does not provide the same security as the conventional phone lines. The solution to secure VOIP is to use the security. The key to securing VOIP is to use the security mechanisms like those of in the data networks, firewalls and encryption '' [36] to follow the security standards that conventional phone line users have.

2.14.2 Issues in Wireless Environment:

The main issues in setting up VoIP over WLAN restrain mainly from access point congestion this result in collision. Resultantly Wireless LANs experience higher delay with more jitter and packet loss, in comparison to wired LANs. Whenever VoIP goes wireless these problems arises, so one have to keep these problems in view. The latency and the congestion should be reduced to minimal value to ensure the QoS. ''Usually this value should be less than 150ms''. [35] Only a lightly loaded AP may produce the best results. Priority queuing or traffic priority is not done in present 802.11x protocols. Another important issue in wireless medium is the 'security' and 'privacy'. Sniffing of data in wireless environment is more likely to take place than in wired. Simultaneous data transfers are another issue. If QoS is not implemented for VoIP the number of carried conversations drops even more. Distance between the user and the access point should be minimal in order to receive the good signal strength. RF signal loses strength with increasing distance. Typically the problems in WLAN environment are as follows: [20]

- CSMA/CA — Collision detection is not easy to execute in wireless networks, as it needs instantaneous transmission and reception while on the same band.
- Small coverage area and high bandwidth has clear effect on the delays as this lowers it down.
- ‘‘Typical frame loss rates are less than 2.5 percent (at maximum frame size)’’. [20]
- Power control is not present.

2.15 How to achieve QoS:

Quality in VoIP means listening or speaking without any interruption and voice should be continuous without loss of voice and quality has these factors. [28]

1. Packet drop
2. Jitter(Variation in delay)
3. Through put
4. Distortion

Service is the communication leverages and the facilities that are provided to the client. There may be lack of resources like considering a machine with many process are running simultaneously and to run all of these process necessary bandwidth is not provided of VoIP communication .This degrades voice quality. QoS is achieved by applying the policy of QoS on to your system. Devices having QoS implemented on them like Router having QoS software installed in it can perform well. Normally your service provider ISP provides his user with QoS applications so in this case VoIP is supported by the system [28]

CHAPTER 3
ANALYSIS

3.1 Approach:

We have worked on the technologies involved in the report that are VoIP and wireless LAN’s so when combining these two often creates some of the issues that are to be addresses to enhance conversational quality. This chapter discusses the approach we took firstly we will study the works done and the information available on these problems before. This will be study part of the report that will suggest or trigger to what extant this problem has been resolved, after having a deep look into them we will show simulation results already done and try to suggest some techniques through which these issues can be eliminated or lessen their effect on the technology. In this study we will focus on the issues still pertaining. Codec selection is one of the important phases of this environment best codec selection results in good conversational quality and less packet loss.

3.2 Scenario 1:

In this analysis three different VoIP aspects are considered call signaling, Network Environment (in this case it is Wireless) and the VPN (PPTP and IPSec). This will be the work environment of the analysis.

- Call signaling protocols used are H.323 and SIP.
- Networking environment is Non- Ideal Wireless Environment.
- Secure Data transferring is also considered ad to achieve this VPN technology is introduced.
- Here RTP packets are taken into account structure having IP Header, UDP Header, RTP Header and RTP Payload.

3.2.1 Test Environment:

Test Environment consists of two machines. One of them is connected to a gateway wirelessly as AP acting as an intermediate and in turn this gateway is connected to another gateway. Second machine is connected to this gateway. Histogram approach is used to show the results and effects that produce jitter and delay in RTP packet scheme.

Values of Packet Loss can be determined by the percentages of packets being arrived at the receiver end or by the packet transmitted.

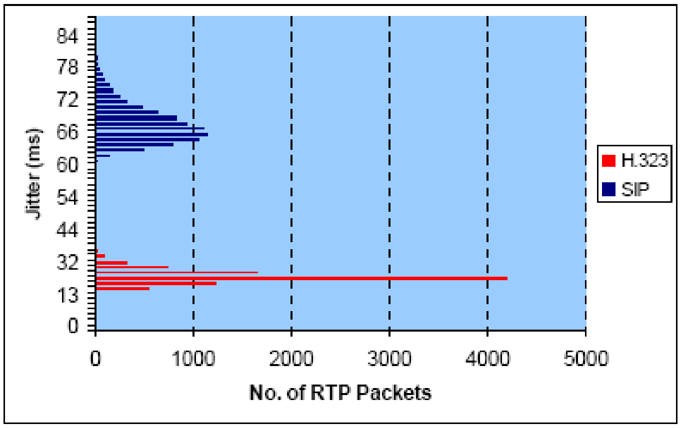

Fig 10 Jitter value for RTP packets for both H.323 and SIP [37]

3.3 Scenario 2:

This study has an emphasis on the effects of handover on VoIP and focus on delay in transmission, jitter and quality of voice.

3.3.1 Handover:

Mobile devices are always changing location and due to this voice quality decreases and not to compromise the voice quality multiple AP's are placed. Call is transferred to the AP in range not to disrupt the voice. This changing of AP's is Handover.

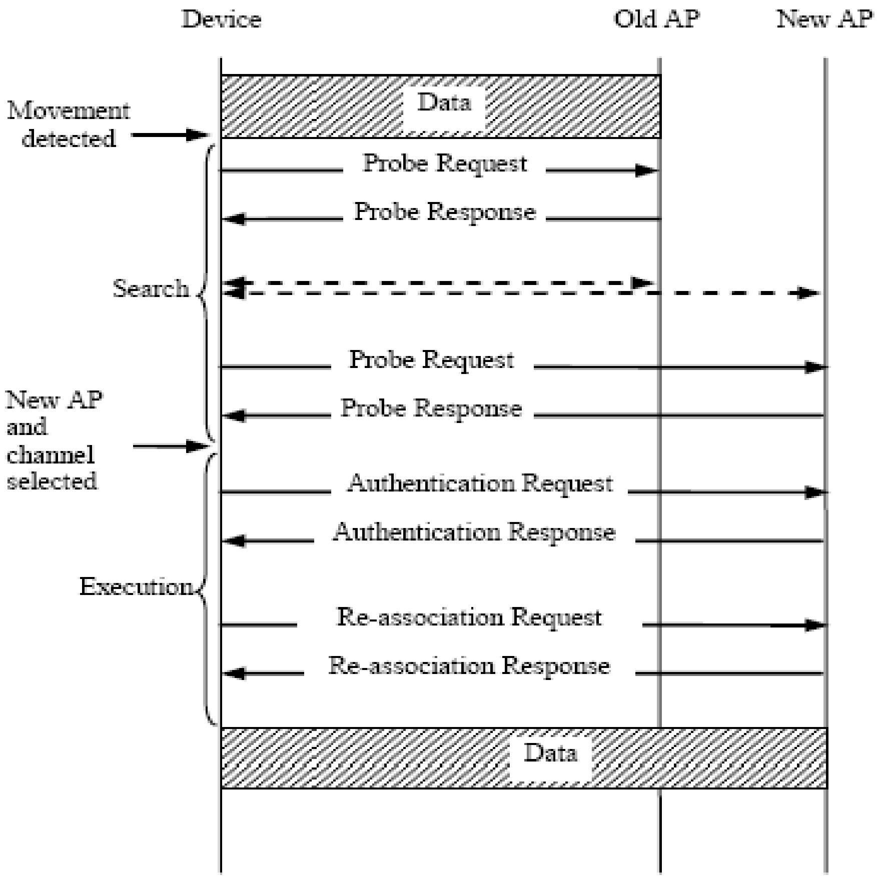

Fig 11 Handover Procedure [38]

3.3.2 Handover Delay or Latency:

Communication can be interrupted while changing AP's by mobile devices. Mobile device can use one AP at a time. The time when Mobile is unable to send or receive by old or new AP is Handover Latency. [38] As 802.11 usually carry data not voice so data needs to be prioritized and without implementing QoS on it will result into jitter, Packet Loss or delay.

3.3.3 Test Environment:

This is taken by [38] and this shows.

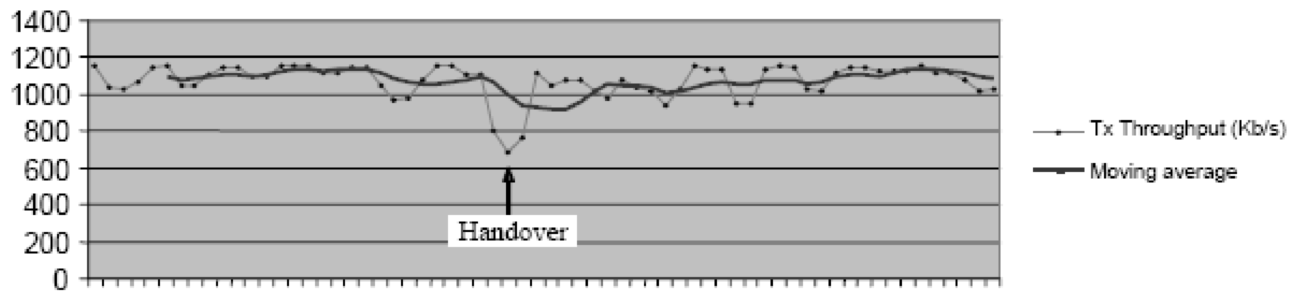

Fig 12 Handover's effect on Throughput [38]

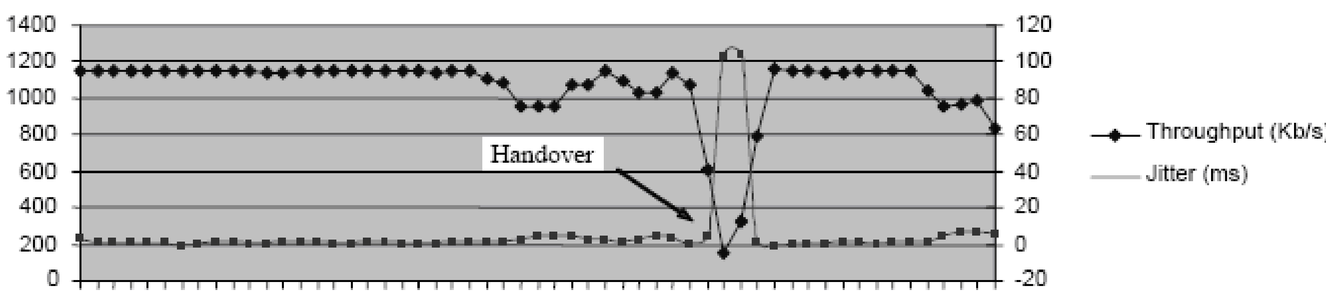

Fig 13 Handover Effect on Jitter [38]

Fig 12 shows that there is sudden dip in the through put while handover takes place this is the latency that occurs when mobile is switching AP's. **Fig 12** shows that when handover takes place jitter value rise to 100 ms for approximately 5-6 ms.

Codec's: These are used to digitize voice data that is encoding and decoding.

Codec's	MOS(Mean opinion score)
G.723	3.6
G.729	3.9
G.728	4
G.711	4.1

Table 7 Codec's and Mean opinion Score [38]

Mean opinion score is the point allotted by the user after using the technology this shows the quality of the communication that they had.

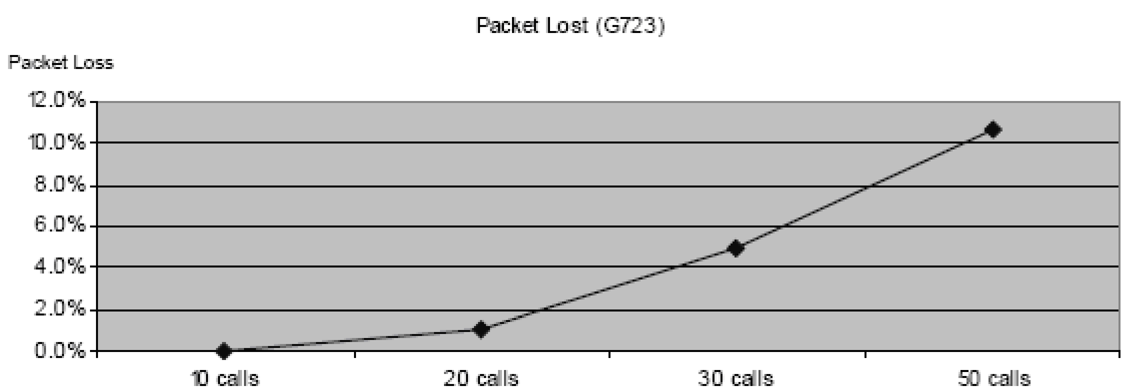

Fig 14 Packet loss in G.723 codec [38]

Calls Number	Packet Loss
10	0 %
20	1 %
30	5 %
40	10.7 %

Table 8 Packet Loss [38]

3.4 Scenario 3:

This analysis depicts a newer picture regarding delay, throughput and the frame loss having multiple stations.

3.4.1 Test Environment:

By [40] test scenario has a BSS with multiple VoIP MS and one Access point. Connection is Bidirectional. After simulating result produced are

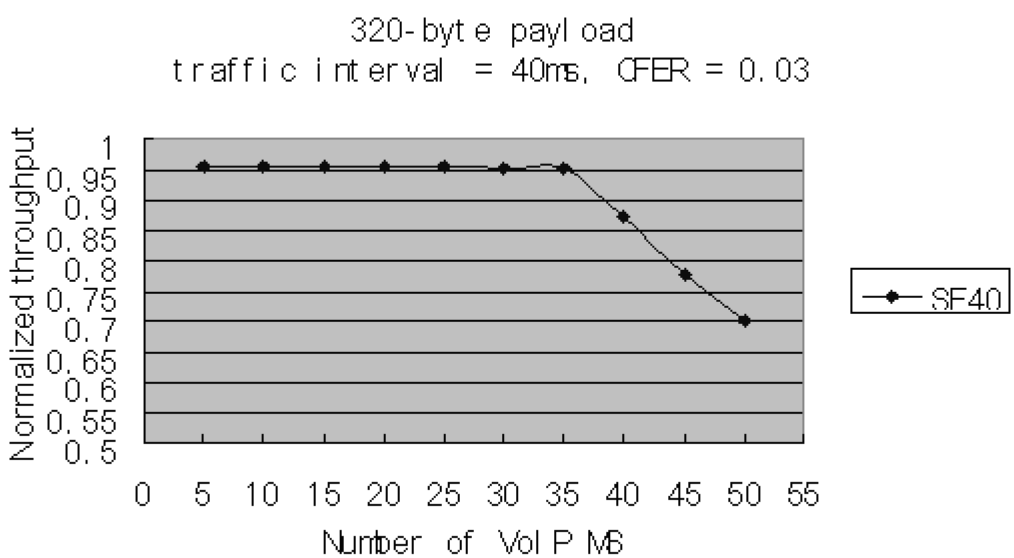

Fig 15 Max number of MS for 320 byte payload [40]

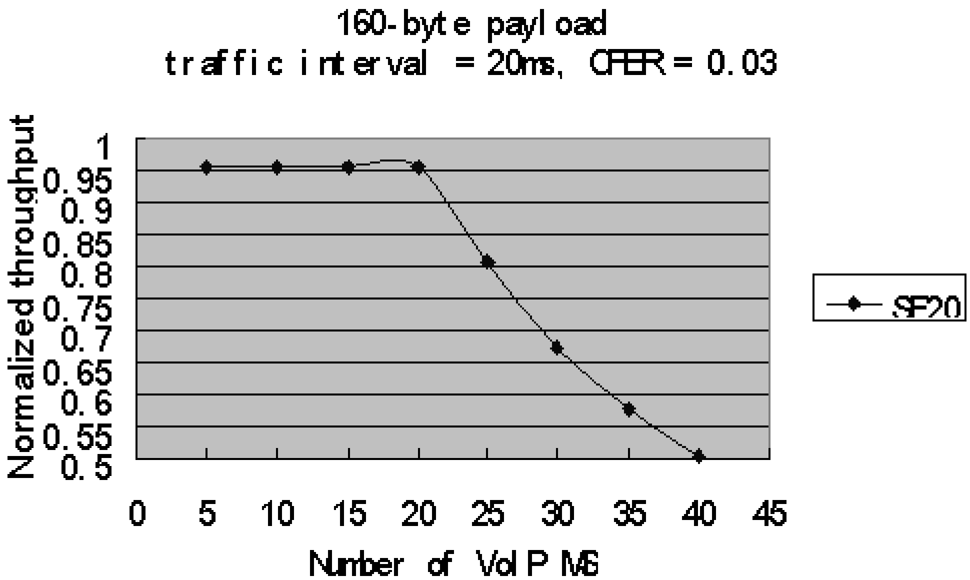

Fig 16 Max number of MS for 160 byte payload [40]

3.5 Results Gathered:

1. Talking of **Scenario 1**, QoS is primarily influenced by three parameters call signaling, Network Environment and VPN Protocols. Due to these parameters jitter and delay are produced and both in Wireless and Wired Environment. Packet loss also affects QoS of VoIP due to congestion in network. Lower jitter by H.323 and higher by SIP for RTP packets. Packet loss and delay are near to same by SIP and H.323. This is also seen that by implementing VPN delay and jitter increases. [37]
2. In the **Test case 2** we can say that during handovers degradation of voice takes place, jitter is produced. Latency also takes place due to hand over. Codec's plays an important role as well. In **Fig 14** we can see that as the number of calls increases packet loss increases simultaneously.
3. **Fig 15** and **Fig 16** in **Scenario 3** show that increasing the payload can handle multiple calls as seen that 160 byte payload has a sudden dip after having 20 simultaneous call while on the other hand 320 byte payload can handle up to 40 calls at a time will it encounter a dip. Normal throughput and less loss of frames can be achieved by the longer payload but the delay is more. One thing is also

seen that as the no of caller is increased it ultimately ends in degradation of voice one after less number of call and other little more numbers of calls. 160 byte payload can handle 20 and 320 byte payload 40 calls simultaneously.

3.6 Summary:

Voice over Wireless LAN's is the contemporary technology and people are trying to adopt it as the combination of VoIP is cost effective and the mobility of Wireless come together to be one . The conclusion from the analysis done above is voice over WLAN needs some of the special things that must be deployed to remove **jitter, delay, packet loss, and throughput**. Strong uplink is required to neglect delay and jitter. Simultaneous call drops the packets being transferred, so remedy can be increasing the capacity of channel. Best codec's must be used and can be selected on the bases of MOS. MOS is a rating scheme in which user is directly involved as he is asked to rate the technology he is using. Convergence and strong security is needed which identifies that user is authorized and call is confidential, there should be no interruption in calls while handoff. Lower transmission power is also an issue in Voice over Wireless LAN's, so for this AP's must be deployed in shorter distances so those users always remain in range. Another problem is that if the newer technology is developed can it be integrated with the existing one. Effectively implementing VPN protocols can help enabling and enhancing VoIP communication in some of the case but one can see that there is a valued increase in Latency and the jitter whenever VPN is implemented. M-M Multiplex- Multicast scheme [39] is introduced that sufficiently decreases delay and jitter. This scheme works by combining data from various streams that are in coming turning then into a single packet. These packets are at last multicast at the destination. Data stream are passed through MUX and it responsibility is to replace RTP, IP, UDP header with a mini header. This header has an ID which shows the session of VoIP packets. These packets are Multicast and then DMUX to retrieve RTP header. [41]This scheme propose that there is very less that can be negligible packet loss up to 1% and delay can be confined up to 32 ms. On the other hand when asked in terms of blocking of traffic due to the increase number of users or due to various schemes M-M schemes overrules them all as less blockage of voice. In next section we will try to show how these prevailing issues can be resolved through some results shown. Addressing these issues is necessary as this is the contemporary technology. We will try to propose effective schemes/methods that will help understanding and getting rid of the prevailing issues. .

CHAPTER 4
EVALUATION

4.1 Specification Background

Previously we have discussed in detail the technologies involved in our thesis that are VoIP and Wireless then in analysis we talked about the impact of delay, packet loss, through put on the Quality of Voice (QoV). In this section we will be specific on the issue being worked upon in this project. MAC have the techniques namely DCF and PCF for IEEE 802.11, is responsible for carrying data though physical layer. We will emphasize on this layer talking little of physical layer as well. IEEE 802.11e is the contemporary technology and helping for QOS of Voice packets. EDCF and HCF are the protocols of it. Comparison will be done to show which one will be better to use among legacy 802.11 and IEEE 802.11e.

4.2 Introduction

There has been an increase of Wireless LANS due to its low cost and effectiveness. This has introduced real-time applications (VoIP) for instance, and IEEE 802.11 tends to provide simpler solution. VoIP traffic is real-time for that matter delay, throughput, packet loss is considered and these variants touch the maximum limits when using wireless technology. VoIP over Wireless has distributed access, they are time-varying and also contention based characteristics. QoS requirement are very necessary to make the user content. IEEE 802.11 is working on this for quit a long time to provide MAC layer supplement, and the outcome was 802.11e. IEEE 802.11e has expanded the domain of legacy 802.11 by providing the applications like video and audio voice) to run over it. We in this paper are focused on QoS support for voice in wireless environment. Communication between various 802.11 stations is maintained by MAC layer. 802.11 has two mandatory parts that are Distributed Coordination Function (DCF) and Point Coordination Function (PCF). When talking about newer component that is 802.11e Enhanced DCF (EDCF) and Hybrid Coordination Function (HCF) and HCF is enhancement of DCF and PCF.

4.3 Standards for IEEE 802.11

This table gives the insight to the standards used nowadays and shows which one handles out addressed problem well.

IEEE Standard	Description	Remarks	Available
802.11a	Operates in 5 GHz. Eight available radio channels.54 Mbps per channel link rate. Data rate decreases as the distance between the user and the AP increases	Higher rate of performance. I most office environment will be greater than the others.	Completed in 1999. Available now.
802.11b	Operates in 2.4 GHz. Eight available radio channels.11 Mbps per channel link rate. Data rate decreases as the distance between the user and the AP increases	Good Performance. As the number of the user's active increases installation speed id affected. Non- overlapping channel limit is three.	Completed in 1999. Available now.
802.11d	This is the supplementary to the MAC layer in and IEEE 802.11 is promoted. It allows the communication between the AP's through allocation of channels, and on the power acceptable by the user. Legal permission is required.	Promote global usage. Countries don't want to develop their own products that are country specific. User has to carry WLAN cards to have access to them.	Completed in 2001. Available now.
802.11e	Supplementary to the MAC layer and provides QoS on wireless LAN's. Purpose of this is to provide state of the art real-time services	Quality of service is it primary Goal and differentiation between the traffics is also done	Completed in 2005. Available now.

Table 9 IEEE 802.11 Standards [42]

4.4 Physical Layer for IEEE 802.11

Physical layer is the first layer among the seven layers in OSI model. The communication request from Data Link Layer to send signal over to other devices is done through this layer. MAC Layer of 802.11 uses the physical layer of 802.11 to perform this task of sending and receiving the signals or 802.11 data frames.

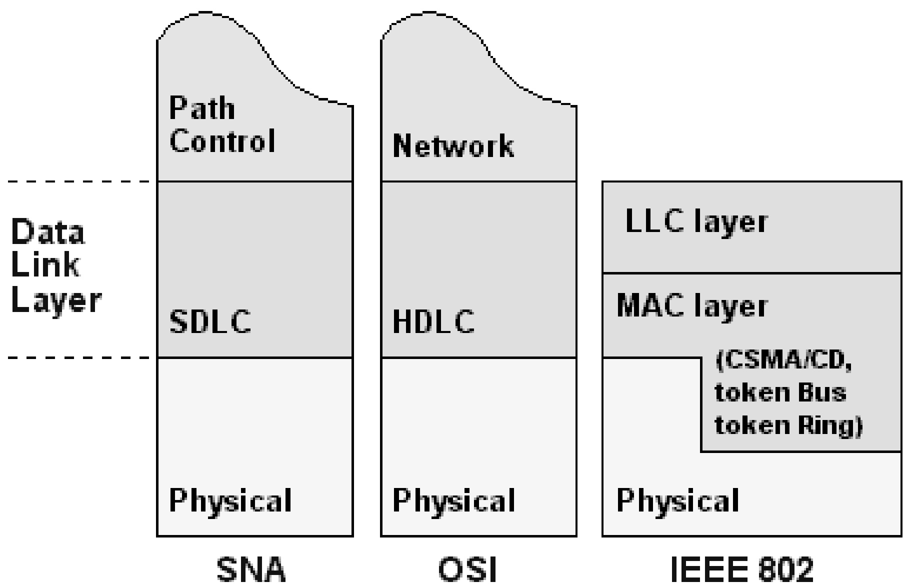

Fig 17 Physical and MAC Layer [43]

While talking of physical layer in wireless environment it plays a vital role and the efficiency of the system is directly related to it as it serves as a bottleneck. Channels are affected due to noise in them which in turn affects overall functionality of the system. Due to congestion produced by noise packets are lost and through put of the system demolishes. While talking of this and considering QoS as well, this affects QoS. If constant demand to gain access onto a same resource cause collision. All this is handled by physical layer.

Function of Physical Layer (PHY) is as follows:

1. Convergence function, Physical Layer Convergence Procedure (PLCP) is put into accord defining the procedure of mapping MAC Sub-layer Protocol Data Unit (MPUD) into a state in which they can be sent and received. [44]
2. PMD system, enabling the function in which system can receive and transmit in Wireless Medium (WM). [44]

Modulation schemes are widely used by PHY to transmit data like BPSK or QPSK are used in IEEE 802.11. These schemes are used to transmit data over physical channel in form of signals. Here one thing is considerable that is Bit to Error Ratio (BER) and it should be below certain level to achieve quality Voice.

4.5 MAC Layer and IEEE 802.11

MAC Layer is one of two sub layers that are LLC and MAC that resides in Data Link layer. Mac layer is responsible for moving data from one STA to another onto a shared link. When talking of legacy IEEE 802.11 it has two techniques DCF and PCF.

4.5.1 Distribution Coordination function (DCF)

DCF is CSMA/CA based technology. In this function station first listens and then starts to transmit, to avoid collision. Station always transmits on the allocated time. Distributed Inter Frame spacing (DIFS) and Short Inter Frame spacing (SIFS) are two modes of DCF and prioritizing the traffic is done through this. In case of multiple stations each station sense DFIS medium time distribution and if medium is free station try to send the packet. If station sees the channel to be busy back-off counter is added or entered. Randomly back-off value is chosen and this is known as Contention Window (CW). Contention Window is the collection of these values. One can see the problem when two or more stations like to send packet at the same time, in this both of these will back-off for certain time by collection of values. There is min and max value for CW and it is 1 less then the power of 2. [45] Now station looks for an idle period of DIFS, for that it listen the medium and after each time slot decrements the counter. When this counter becomes zero data is sent. [35] DCF functionality is well judged when the number of stations increase and they all try to send data over the channel or medium at the same time. When this number increases simultaneously delay increases thus degraded voice is produced other problem that should be mentioned is Traffic Prioritization is not done by DCF this means differentiation between data and voice packets are not acknowledged and are treated as same.

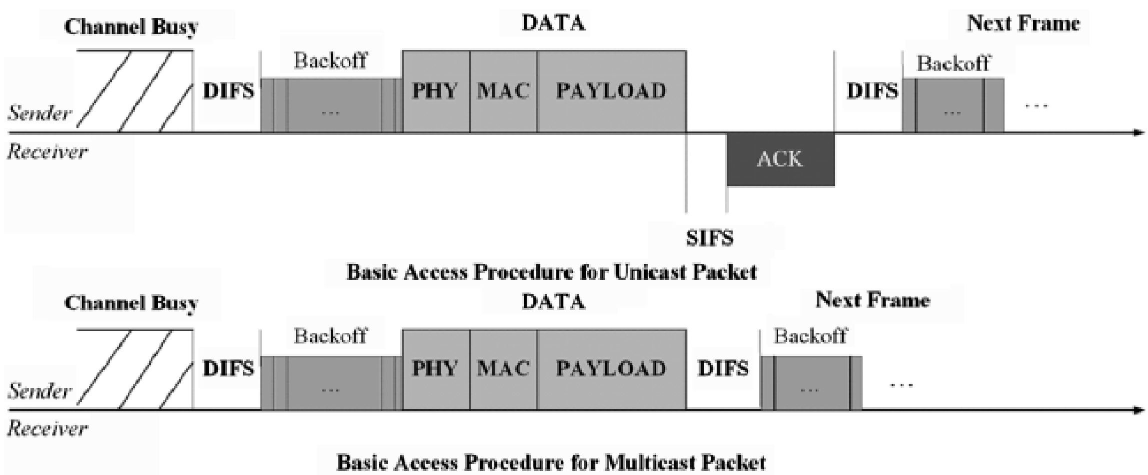

Fig 18 Basic operation of IEEE 802.11 DCF [46]

4.5.2 PCF (Point Coordination Function)

PCF handles Real-Time applications and provide contention free services. Point Coordinator (PC) is a function that is used for the allocation of medium for communication and it is implemented in AP. Method of pooling is adopted by PCF. Contention Free Period (CFP) and Contention Period (CP) form frames (super frames) in PCF and minimal length of CP is required for DCF as for the acknowledgment for station or during DCF. Inter frame spacing used during PCF period is PCF inter frame spacing (PIFS).

Polling starts at PCF and station receive data from PC, and after send PC waits for the acknowledgment and if it doesn’t arrive at certain time defined in PIFS it send request to another station and this goes on. This is done till CFP value expires and updating the list is done during CP. [35] this type of system is not widely encouraged and thus DCF is used as it serves the purpose well.

4.6 MAC Layer and IEEE 802.11E

We have made a choice to work on IEEE 802.11e MAC layer as by the standards shown in **Table 9** IEEE 802.11e is the contemporary technology developed in 2005 specifically for voice QoS. Two of the protocols discussed earlier that are Enhanced Distributive Function (EDCF) and Hybrid coordination Function (HCF). EDCF is primarily used in CP and Hybrid use both of CP and CFP.

4.6.1 EDCF (Enhanced Coordination Function)

Prioritization and differentiating between data frames is not done by legacy 802.11e MAC.DCF provide channel access to all stations without prioritizing them and all the station in the system have the equal probability to receive data. However, talking of the real-time applications this is not a good procedure and is not desired. EDCF has emerged to provide solution to this. In this eight priorities that are from 0 (Lowest) to 7(Highest) are given to the frames and the channel access is distributed. EDCF does not act separately but is the part of Hybrid Coordination Function (HCF). Priority value is assigned to the frames approaching MAC and this value is given to MAC header frame. These values in turn are mapped with Access Categories (AC) that is four 0 to 3.

Priority	Access Category (AC)	Designation
lowest	0	Best effort (BE)
lowest	0	Best effort (BE)
.	0	Best effort (BE)
.	1	Back Ground
.	2	Video (VI)
.	2	Video (VI)
highest	3	Voice (VO)
highest	3	Voice (VO)

Table 10 Mapping of Priority with AC

4.6.2 HCF (Hybrid Coordination function)

Hybrid Coordination Function includes feature for both distributed access when talking of DCF and PCF centrally controlled access of medium, this combination is to achieve a level of QoS by using the features of both of them it makes this function Hybrid. [48] We can say this is the Enhancement of both DCF and PCF.

1. Distributed contention based medium access i.e. Enhanced Distributed Channel Access (EDCA).

2. Controlled centrally contention-free medium access i.e. HCF controlled Channel Access (HCCA).

Here AP has Hybrid Coordinator (HC), and each of the station by the procedure of pooling is allotted TXOP. In HCF CP and CPF rule are applied as discussed earlier. Stations in Contention Free Period (CFP) are pooled considering the priority and this priority is done by HC. EDCF rules are applied in Contention Period and the TXOP is initiated. Two terminologies are used here one is CFP-Pool and CF-End, these are for QoS. During CFP, CFP-Poll initiates and the communication is through by allocation of the channel. CF-End is the end of CFP. Method of granting medium and ending the conversation is done by HC. [49] HC listen the request and without contending the stations can send the request of allocation of TXOP.

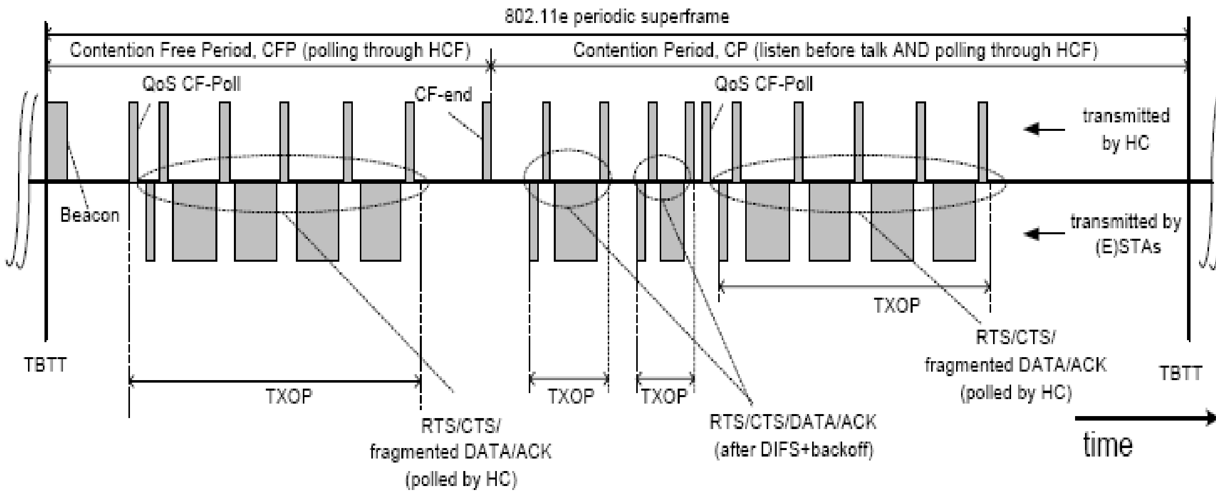

Fig 19 Hybrid coordination Function pooling [49]

4.7 Channel Access Function

IEEE 802.11e defines set of enhanced features of it ancestor Legacy 802.11. QoS enhanced stations are differentiated and distinguished form the legacy 802.11 non Quality of Service enhanced AP. Channel access and traffic specs are the functionalities of IEEE 802.11e. Here we will look into the channel access function of DCF and it enhanced version EDCF.

4.7.1 Legacy 802.11 MAC DCF Channel Access

Acknowledgment process is stronger between the stations. When frame is received by the receiving station it sends in ACK packet. Back-Off procedure is resumed after SIFS ACK frames are transmitted. While talking of Contention Window (CW) its initial value is set as CWmin and values keep on adding up whenever there is failed attempt of transmission. In other words this can be said as packet is not acknowledged. In this type of scenario Back-Off occurs and the value is updated 2*(CW+1)-1, considering the CWmax upper bound.[47] For collision avoidance this is necessary and in case of successful transmission CW value is the value of CWmin as it is reset. Post Back-Off is done after transmission ‘‘this is to ensure minimum one back off interval between two MSDU transmission .... dependent on underlying PHY’’ [47]

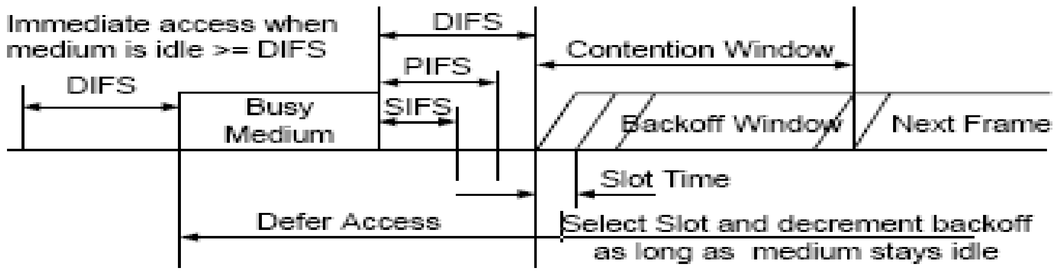

Fig 20 Legacy 802.11 DCF [47]

4.7.2 IEEE 802.11e MAC EDCF Channel Access

As defined earlier DCF use CWmin, CWmax, and DIFS and wile talking of EDCF use CWmin [AC], CWmax [AC], AFISD [AC]. AIFSD value is greater than 0 and is an integer.

Formulation,

AIFSD [AC] = SIFS+AFIS [AC]* Timeslot

Back-Off counter is [1, 1+CW [AC]] in EDCF, Back-Off counter is [0, CW] in DCF

These values are given by AP and this is done by PC present in AP which generates Bacon frames. Depending upon the network Condition, values are taken by AP.

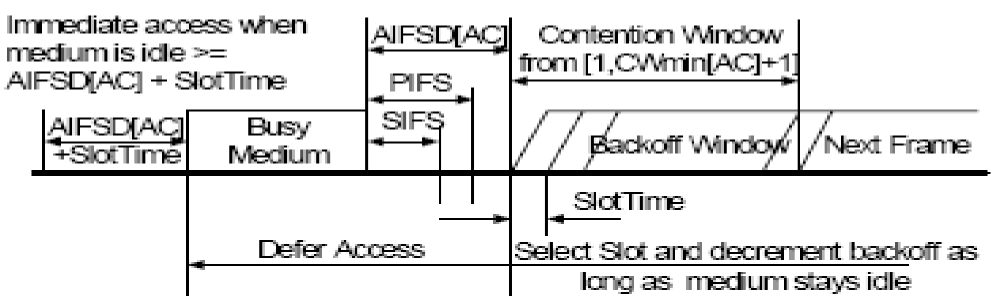

Fig 21 IEEE 802.11e EDCF [47]

When talking of contention Period (CP), Transmission opportunity (TXOP) is a newer term talked of in which station strive for it and check for the free channels. This is done for AIFS and Back-Off counter is initiated immediately. Each station has to wait for AIFS, and Back-Off counter starts ticking and this done by TC. TXOP defines the time in which station can transmit, and TC contends for it. This is a modification done by 802.11e. One problem here to notify, Traffic Category TC or priority setter can collide as

TC always contends for the value and in the case similar values is assigned, this type of problem is named as Virtual Collision and here scheduler comes into play.

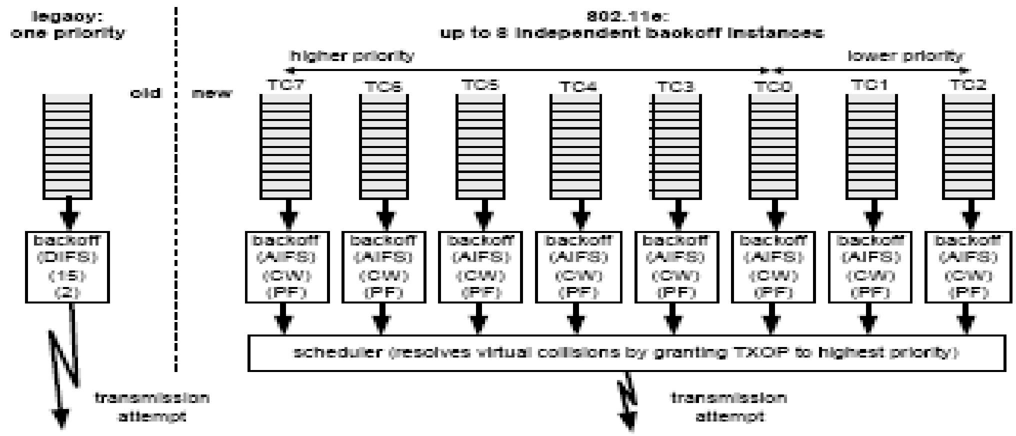

Fig 22 Virtual Back-Off for Eight traffic categories [49]

4.8 Evaluation

This evaluation is done by providing initially by defining the environment we are working on which is done previously by explaining in detail MAC of Legacy 802.11 and IEEE 802.11e and on the basis of these information we will try to sort out the issues pertaining. Legacy has DCF and PCF, while 802.11e has EDCF and HCF. In this evaluation we will try to compare MAC of Legacy 802.11 and IEEE 802.11e, which of them support best QoS. To be more specifies we will compare the results of EDCF and DCF with introducing HCF and PCF at some points. We will take the real time scenarios from the papers and try to evaluate performance of EDCA and DCF. QoS parameters are as follows

1. Delay: Time at which data packets are sent and the time at which they are received.
2. Through put: Data delivered at a specific time. Represented by bit per second.
3. Packet Drop: Rate of loss of data frames that are to be reached at the receiving end.

4.8.1 Simulation Results:

Set of simulation are taken into account to show the comparison between two of the protocols and suggest which one has batter performance and which supports QoS among IEEE 802.11e and legacy IEEE 802.11.

4.8.1.1 Simulation:

Three types of traffic are handled in this simulation VoIP, Busty data and video. In this paper we are working on Voice so we stick with voice only.

Here, audio source generates 60-byte message. Bit rate of 24 kbps as it is generated after 20 ms. Encoding scheme used here is G.729A with an overhead of RTP/UDP/IP.[50] Parameters of the network for the simulation of IEEE 802.11 and IEEE 802.11e are put into accord as mentioned in the earlier section in standards.

Poisson distribution is the method on which busty data is based upon, data messages are generated through this. Generated traffic is (64,0.6) , (128,0.06) , (256,0.04) , (512,0.02) , (1024,0.25) , (1518,0.03).[50]

For necessary results scheduling algorithm is focused as it has a significant affect upon the performance of EDCF. EDCF here works on the priority basis *these are given by the adjustment of CW* and TC accomplishes this well and each TC has it own stack/queue in a particular station. Overlapping avoidance of CW or for no compromise on QoS it is importance to have limited CW. This is also introduced and both CW and queue has it own Back-Off Counter. While AFIS, traffic with high priority has less valued CW when compared with high valued low priority traffic. But AIFS value cannot be very high, this affects through put of the system

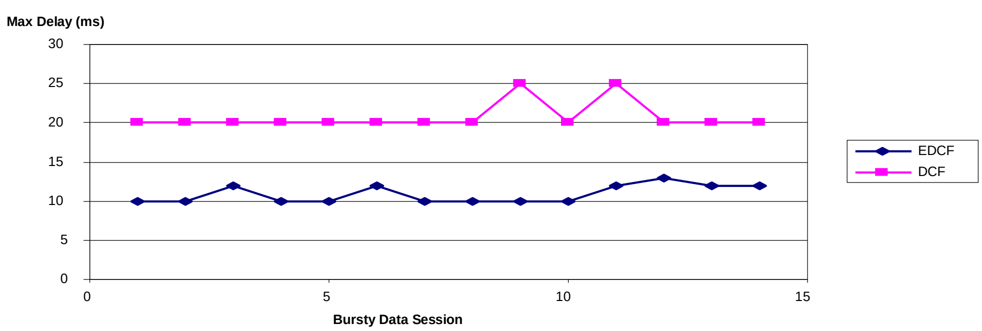

Fig 23 Maximum Packet delay in VoIP

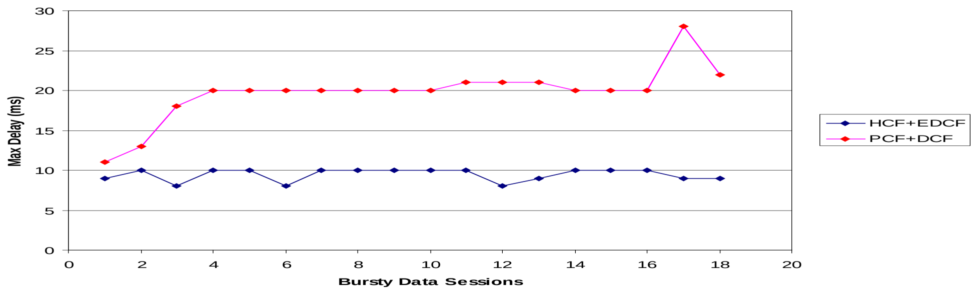

Fig 24 Maximum Packet delay in VoIP with PCF and HCF introduced

This simulation study shows that IEEE 802.11e EDCF performs well when compared with DCF of Legacy 802.11. It has been discussed earlier that due to non provision of the prioritization mechanism Voice application are not well supported and they encounter contention while there flow. EDCF keeps the delay below 13 ms which is acceptable. There is an associated problem with EDCF while using it alone as one station want to send before others can set it priority high. So here many stations are contending for the same network resource, but problem can be avoided by making TC well in function as it controls the priority settings. [50] There is a significant lacking of QoS while combining PCF and DCF and the maximum delay reaches to 28 ms. On the other hand we see that more the busty sessions lesser come the delay in case of combining EDCF and HCF. For VoIP we can also observe that when introducing PCF and HCF, HCF seems to be performing well than PCF.

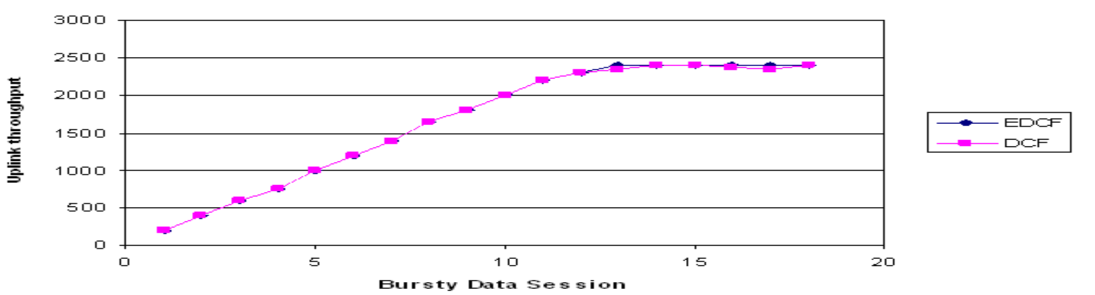

Fig 25 Uplink Throughput of data in Kbps

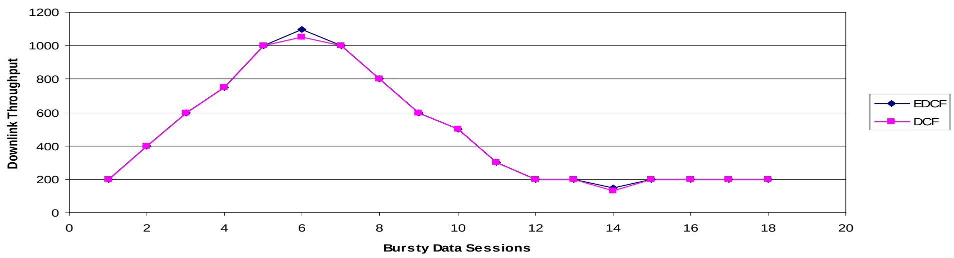

Fig 26 Down Link Throughput of data in Kbps

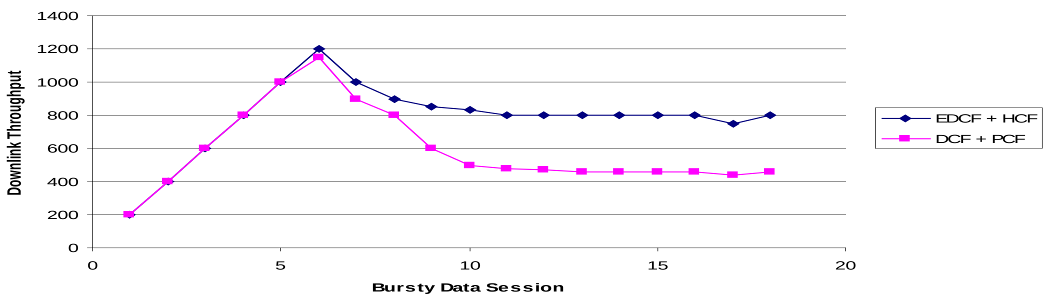

Fig 27 Better Downlink Throughput of data by EDCF+HCF

Through put can be taken in combination of uplink through put and the downlink through put. Here considering data streams are 6 and can be seen clearly that uplink through put is higher when EDCF and DCF are used alone. Downlink through put is always less when these are used alone. Uplink and downlink difference is approximately 2.2 Mbps. [50] It is good to use EDCF and DCF alone for uplink traffic but not in the case of downlink. This downlink problem can be explained in such a way, there is one CSMA for each AC, in an AP. In a scenario of x terminals and for every AC it has x uplink CSMA and one downlink CSMA this creates the problem. This can simply be resolve by increasing HCF or PCF values. As we have seen by the above scenario that HCF has performed well when compared with PCF. It can be clearly seen and can be stated that QoS of Voice is well supported by IEEE 802.11e then by legacy 802.11. EDCF has proved itself to be more viable and reliable then its ancestor DCF. EDCF no doubt has an advantage when talking of uplink but when see downlink it don’t perform well alone. But by combining HCF with Enhanced DCF serves the purpose well. HCF when compared with PCF it seems that HCF is more flexible. HCF has low transmission delay and on the other side when looking into CFP in PCF, QoS is demolished as frames have to wait for CP period to finish. [50]

4.8.1.2 Simulation:

In this we will discuss how Access Category (AC) mechanism is effective. This priority mechanism is achieved by giving AFIS, CWmax, CWmin, TXOP values. Here four working stations are attached to an AP wirelessly and in turn connected to a Network (Correspondent Node) CN through a backbone. Three traffics are considered as to show the traffic priority.

Voice has Priority 7, Access Category 3, AFISD is PIFS, CWmin 7, CWmax 15 and TXOP limit 3 milliseconds.

Video has Priority 5, Access Category 2, AFISD is PIFS, CWmin 15, CWmax 31and TXOP limit 5 milliseconds.

Data has Priority 7, Access Category 0, AFISD is DIFS, CWmin 31, CWmax 1023 and TXOP limit 30 milliseconds.

AC 3 has priority 7 as the Back-Off counter has the lowest values as this is obvious form the values of CWmin and CWmax. [51] NS-2 simulator is used here with Network Load of 200Kbps, 400Kbps and 500Kbps.

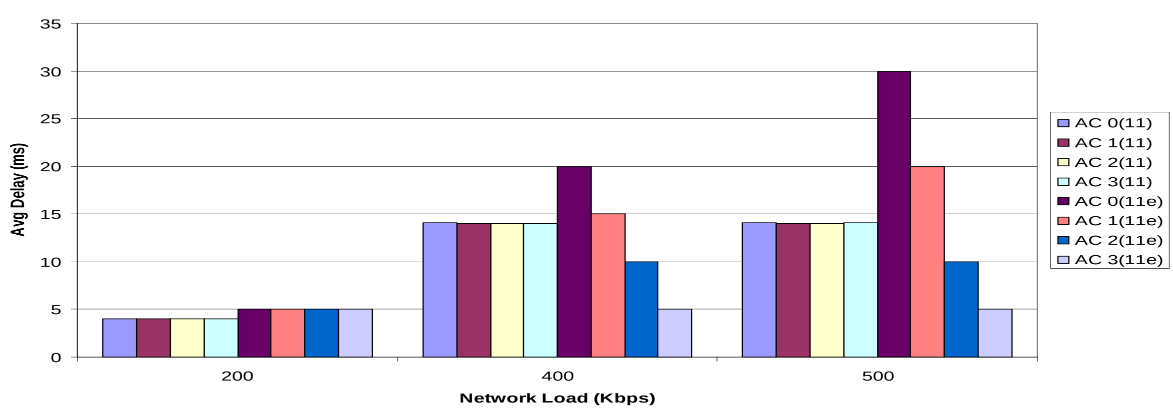

Fig 28 Comparison of Delay when increasing Network load

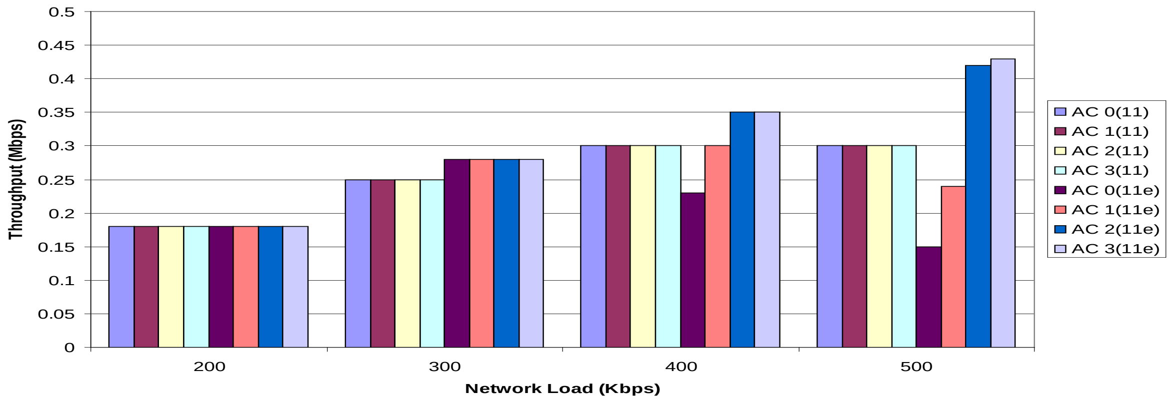

Fig 29 Comparison of Through put when increasing Network load

These diagrams are basically to show the priority mechanism in IEEE 802.11e and how is this effective. When talking of delay it is obvious that all traffic in 802.11 is treated to be the same and encounters same delay as well. While on the other hand IEEE 802.11e deals traffic on priority basis. We observe that when data rate is small delay in both 11 and 11e is same but the performance is enhanced when data rate is increased for 802.11e and delay is kept less than 5ms. In IEEE 802.11 DCF we observe that if the delay for the higher priority traffic is the same as low priority traffic and in this case lower priority traffic has to wait more for there turn, so total delay is increased. When talking of through put as we know that for lower rate of traffic there is enough resource available for the traffic to move, so in 802.11 and 802.11e for the traffic with less network load through put is same and there is not significant difference. When load is increased 802.11 manages to have same throughput as there is no traffic category while on the other hand 802.11e supporting traffic category highest priority traffic has good through put. Now when talking of the lowest priority traffic they have to wait to occupy the resource. This mechanism will have delay in case if the lowest priority traffic, but problem are well addressed and can be removed by having good control scheme and by HCF and EDCF tuned according to the type of application being used.

Now we will talk of Packet loss in this type of system and the performance of EDCF. One thing should be kept in mind that is audio and video are time bounded applications and they require delay and through put guaranties but they can afford little bit of loss of packets and it will always be there, we can't run out of it. Video applications are considered here only to show the mechanism of the Access categories but only description here would be about audio. The topology is the same and also the traffic pattern for voice, MPEG (video), WWW and FTP. [51] Traffic load is to be increased form 1 to 4 and priority of the traffic is AC3 for voice, AC2 for video. We will consider on packet loss here. Traffic is movement is divided in to uplink and downlink.

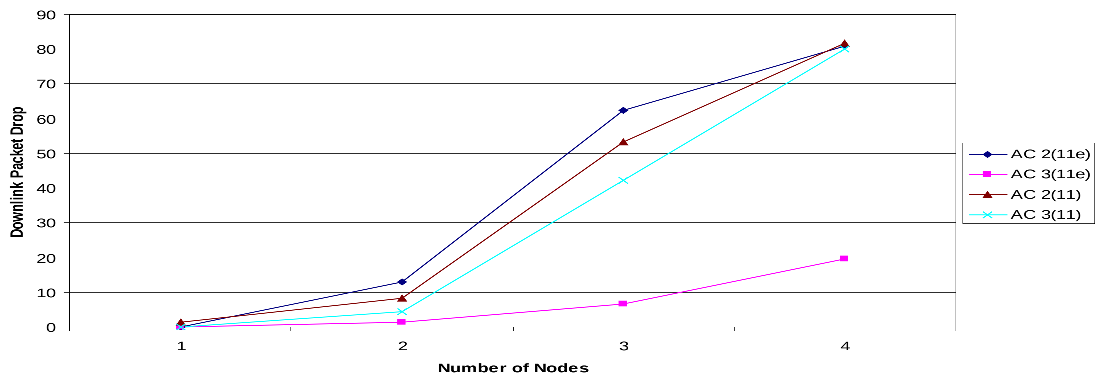

Fig 30 Downlink packet drop in (%)

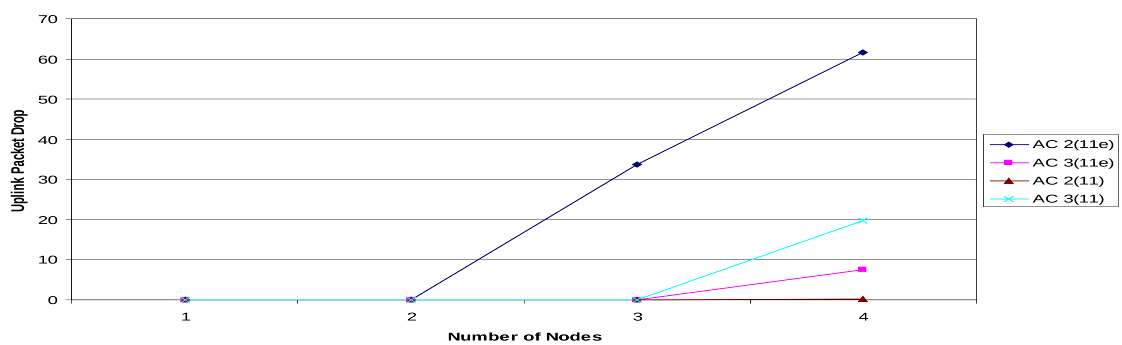

Fig 31 Uplink packet drop in (%)

This is also seen that packet loss is directly proportional to the number of nodes into function. EDCF handles voice for uplink well and keeps it below 8% when maximum number of nodes is added. This problem is more with the contention based transmission that is DCF as number of contestants for the transmission increase collision takes its place and packet are lost. Till 3 nodes there is no packet loss as resource can handle 3 simultaneous calls but when it is increased from 3 graph rises suddenly. All of the stations are sending at the same time as there is no AC mechanism. Here on thing is noticed that is downlink has very bad performance when compared with uplink packet drop. This is because of one reason that is sharing of one buffer by same AC number in base station and access mechanism for both stations and BS is same.

4.9 Proposed scheme:

This proposed scheme is the hint to what we think should be done to avoid little bit of the flaws in EDCF. When talked all of this, one method to improve efficiency come into mind, EDCF has contention based nature and provides priority scheme. This priority schemes allows the frames to be prioritized but on the other hand we see that there is a bit of a compromise on Quality of Service. It is quit fair to say that EDCF is superior to DCF and function well but compromise on Quality of Service is due to its contention based nature. These real time applications are time bounded and QoS measures are to be taken to avoid delay, packet loss and increase through put. Hindrance to achieve this is due to collision and which occurs often in contention based services. EDCF has a big problem in it that is it not adaptive to the condition and nature of the network which changes frequently. It is noticed that in case of collision and the link is shared that is in contention based, contended stations wants the resources which are allocated through priority mechanism. Problem here is that collision may occur firstly this should be addressed but it isn’t so the demand to send the packets of the sender increases significantly, and due to this throughput is affected. As we have seen above that throughput of the stations are the same when talking of EDCF, so serious measures are to be taken to null void this issue of collision. In this we have to focus of the contention window (CW) and back-off timer adjustments. Contention window can be adjusted in a way that, a value can be assigned to it after collision and after data is sent. One thing is noted in EDCF is that when ever data is delivered successfully value of the contention window is set to it minimum value that may not be correct. In EDCF sender is not able to know that collision has happen or not, it knows ultimately when for the next time he get the turn in between there can be many collisions. EDCF handles high priority traffic well but does not handles lower priorities traffic efficiently, and we have to look all into it as many of the services are on the local wireless area network. Knowing condition of the network is very important when a collision has occurred after sending the packet when back off time is 0 so to send packet after that cause more collisions. So to avoid this value to the CW can be decremented gradually rather then assign CWmin to it directly or can be adjusted in such a way that minimum collision occurs, keeping in mind that EDCF maintains priority mechanism and this should be discriminated form others as well. Keeping in view one thing that Contention window value is to be lower then the others for the highest priority traffics so as it has to back off for the minimum time to send the data.

CHAPTER 5
CONCLUSION

5.1 Conclusion:

Major objective of this thesis was to look into the current technologies influencing the market like wireless LANs due to the less cost and effectiveness and how are they helping real time application especially voice to run over them. Addressing these problems prevailing in the technology and subjecting a way to remove them through evaluation phase which help us to know the problem occurrence and which of the factors are involved in them to occur. We have tried to address the problems that are delay, jitter, increasing through put and packet loss. We have gathered by study this technology in detail that using de-jitter buffer algorithm can solve the problem easily that is to store data into a buffer firstly and the play out according to its priority. Minimum delay and increased through put can be achieved through this by adjusting the protocols in the MAC that are EDCF. You will always going to have some of the packet loss it cannot be removed 100%. But data drop can also be minimized by adjusting Contention Window minimum the collision minimum will be the delay and packet loss. Conversational quality is not a big issue, particularly if Codec G.711 is used.

- As seen through the examples above it is seen clearly the EDCF is more effective then the conventional DCF due to is Traffic Category Mechanism (TCM) which in turn increases the through put and lessen the delay and data drop.
- As address above that in WLAN there can be may types of traffics working simultaneously that are voice, video, data so traffic have low priorities can starve for access to the channel. To make voice quality good this aspect must be looked into to avoid these problems.

5.2 Future Work:

Future work to this can be on the Contention window adjustment so as to achieve good results. Back off procedure should be working effectively to avoid the collision and overall performance of the system depends upon it. So through simulation one can show how this will be implemented and working on MAC EDCF of IEEE 802.11e. I propose its name to be Contemporary EDCF (C-EDCF) which focuses on the EDCF problems and how to over come on them by resetting the values to the contention window.

LIST OF FIGURES AND TABELS:

CHAPTER 2

List of Figures

Figure 1 Wireless LAN Architecture [16]**..........10**

Figure 2 RTP/ RTCP Format [11]**..........12**

Figure 3 OSI Layers with their purpose. [11]**..........13**

Figure 4 IP Datagram Structures [11]**..........14**

Figure 5 OSI Layers and 802 [12]**..........15**

Figure 6 H.323 Call Set Up Process [36]**..........24**

Figure 7 VoIP Structure [23]**..........25**

Figure 8 SIP Architecture**..........26**

Figure 9 VoIP Process [21]**..........27**

Figure 9a Delays [30]**..........32**

List of Tables

Table 1 Characteristics of various forms in wireless LANs. [15]**..........16**

Table 2 Wired and Wireless Networks and impacts on them [4]**..........19**

Table 3 Comparison of quality of Voice over PSTN and over IP. [18]**..........21**

Table 4 VoIP Codec's characteristics [18]**..........28**

Table 5 G.711 Codec [31]**..........33**

Table 6 G.729 Codec [31]**..........34**

CHAPTER 3

List of Figures

Figure 10 Jitter value for RTP packets for both H.323 and SIP [37]**..........39**

Figure 11 Handover Procedure [38]**..........40**

Figure 12 Handover's effect on Throughput [38]**..........41**

Figure 13 Handover Effect on Jitter [38]**..........41**

Figure 14 Packet losses in G.723 codec [38]**..42**
Figure 15 Max number of MS for 320 byte payload [40]**...43**
Figure 16 Max number of MS for 160 byte payload [40]**..44**

List of Tables

Table 7 Codec’s and Mean opinion Score [38]**...42**
Table 8 Packet Loss [38]**...43**

CHAPTER 4

List of Figures

Figure 17 Physical and MAC Layer [43]**..50**
Figure 18 Basic operation of IEEE 802.11 DCF [46]**...52**
Figure 19 Hybrid coordination Function pooling [49]**...55**
Figure 20 Legacy 802.11 DCF [47]**...56**
Figure 21 IEEE 802.11e EDCF [47]**..56**
Figure 22 Virtual Back-Off for Eight traffic categories [49]**..57**
Figure 23 Maximum Packet delay in VoIP [50]**..58**
Figure 24 Maximum Packet delay in VoIP with PCF and HCF introduced [50]**...........59**
Figure 25 Uplink Throughput of data in Kbps [50]**...59**
Figure 26 Down Link Throughput of data in Kbps [50]**...60**
Figure 27 Better Downlink Throughput of data by EDCF+HCF [50]**...........................60**
Figure 28 Comparison of Delay when increasing Network load [51]**..........................61**
Figure 29 Comparison of Through put when increasing Network load [51]**.................62**
Figure 30 Downlink packet drop in (%) [51]**...63**
Figure 31 Uplink packet drop in (%) [51]**..63**

List of Tables

Table 9 IEEE 802.11 Standards [42]**..49**
Table 10 Mapping of Priority with AC**..54**

ACRONYMS:

QoS: Quality of service.

VoIP: Voice over Internet Protocol.

BS: Base Station.

WEP: Wired Equivalent Privacy.

LANs: Local Area Networks.

MSC: Mobile Switching Center.

MTSO: Mobile Telephone Switching Office.

PSTN: Public Switched Telephone Network

CAI: Common Air Interface

FVC: Forward Voice Channel

RVC: Reserve Voice channel

FCC: Forward Control Channel

RCC: Reserve Control Channel

DSSS-CCK: Direct Sequence Spread Spectrum with Complementary code Keying.

OFDM: Orthogonal Frequency Division Multiplexing.

WEP: Wired Equivalent Privacy.

WAP: Wi-Fi Protected Access.

AP: Access Point.

RF: Radio Frequency

DSS: Digital Signature Standard.

MAN: Metropolitan Access Networks.

WiMax: Worlds Interoperate Ability for Microwave Access

AES: Advance Encryption Standard

DES3: Triple Data Encryption Standard.

FHSS: Frequency- Hopping Spread Spectrum

SSL: Secure Socket Layer.

VPN: Virtual Private Network.

OSI: Open System Inter-connection

UDP: User Datagram Protocol

TCP: Transfer Control Protocol

SIP: Session Initiation Protocol

LLC: Logical Link Control

MAC: Medium Access Control

PLCP: Physical Layer Convergence Procedure

PMD: Physical Medium Dependent

SNR: Sound to Noise Ratio

PBX: Private Branch Exchange

PCF: Point Coordination Function.

DCF: Distributed Coordination Function

CSMA/CA: Carrier Scenes Multiple Access/ Collision Avoidance

RTCP: Real Time control Protocol

ARPNET: Advance Research Project Agency Network.

PSTN: Public Switched Telephone Network.

VoIP: Voice over Internet Protocol.

SIP: Session Initiation Protocol.

Wi-Fi: Wireless Fidelity

MCU: Multipoint control unit.

IETF: Internet Engineering Task Force

UPQ: User Datagram quality

PPS: Packet /Second

ATA: Analog telephone Adapter

ISP: Internet Service Provider

MOS: Mean Opinion Score.

MS: Mobile Station

DCF: Distributed Coordination Function

PCF: Point Coordination Function

EDCF: Enhanced DCF

HCF: Hybrid Coordination Function

PLCP: Physical Layer Convergence Procedure

MPUD: MAC Sub-layer Protocol Data Unit

PMD: Physical Medium Dependent

BER: Bit to Error Ratio

DIFS: Distributed Inter Frame spacing

SIFS: Short Inter Frame spacing

CW: Contention Window

PC: Point Coordinator

CP: Contention Period

CFP: Contention Free Period

PIFS: PCF inter frame spacing

EDCF: Enhanced Coordination Function

EDCA: Enhanced Distributed Channel Access

HCCA: HCF Controlled Channel Access

HC: Hybrid Coordinator

MSDU: MAC Service Data Unit

TXOP: Transmission opportunity

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