Tribhuvan University
Institute of Science and Technology

Quality of Service (QoS) Analysis of IEEE 802.11e Networks on the basis of EDCA parameter: TXOP

Dissertation
Submitted to

Central Department of Computer Science and Information Technology
Kirtipur, Kathmandu, Nepal

In partial fulfillment of the requirements
for the Master's Degree in Computer Science and Information Technology

by
Yamuna Ghimire
July, 2009

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Supervisor
Associate Prof. Dr. Subarna Shakya

Tribhuvan University
Institute of Science and Technology
Central Department of Computer Science and Information Technology

Student's Declaration

I hereby declare that I am the only author of this work and that no sources other than the listed here have been used in this work.

… … … … … … …
Yamuna Ghimire
Date: … … … …

Supervisor's Recommendation

I hereby recommend that this dissertation prepared under my supervision by **Miss. Yamuna Ghimire** entitled "Quality of Service(QoS) Analysis of IEEE 802.11e Networks on the basis of EDCA parameter: TXOP" in partial fulfillment of the requirements for the degree of M. Sc. in Computer Science and Information Technology be processed for the evaluation.

… … … … … … … … …
Dr. Subarna Shakya
Associate Professor
Department of Electronics and Computer Engineering
Pulchwok Campus, IOE, Tribhuvan University
Date: … … …. … … …

Tribhuvan University
Institute of Science and Technology
Central Department of Computer Science and Information Technology

LETTER OF APPROVAL

We certify that we have read this dissertation and in our opinion it is satisfactory in the scope and quality as a dissertation in the partial fulfillment for the requirement of Masters Degree in Computer Science and Information Technology.

Evaluation Committee

… … …. … … … … … … …. … … …

Dr. Tanka Nath Dhamala
Central Department of Computer
Science and Information Technology
Tribhuvan University, Nepal
(Head)

. … … … … … … …. … … … …

Dr Subarna Shakya
Associate Professor
Department of electronics and
Computer engineering
Pulchowk Campus, IOE,
Tribhuvan University
(Supervisor)

………………………………………..
(External Examiner)

Date: … … …. … … …

..……………………………..
(**Internal Examiner**)

Date: … … …. … … …

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Yamuna Ghimire

Abstract

Wireless communication has become a very important and rapidly evolving technology. It allows users to transmit data from one remote location to another remote locations or fixed location. Quality of Service (QoS) is the key factor to be considered for its better performance.

In this dissertation QoS in the Wireless Local Area Network (WLAN) has been analyzed. One of the medium access mechanisms of IEEE 802.11e is the Enhanced Distributed Channel Access (EDCA) which has been studied. EDCA has mainly three parameters, Contention Window (CW), Transmission Opportunity (TXOP) and Arbitration Inter-Frame Space (AIFS), which are responsible for maintaining the QoS. It has been analyzed that these parameters are assigned with the static value which decreases significantly the throughput performance and increases the collision rate specifically at high load condition

In this dissertation TXOP limit has been studied in detail. The constant value of this TXOP limit has been replaced by its dynamic value by adding one of the user defined functions in its medium Access Control (MAC) file. The dynamic value has been adjusted according to the applications used on that specific environment.

The whole performance of the WLAN has been analyzed in the popular simulator called Network Simulator (NS-2). The data from the simulator are tabulated and compared in the graphs.

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List of Abbreviations

AC	Access Category
ACK	Acknowledgement
AIFS	Arbitration Inter-Frame spacing
AP	Access Point
CBR	Constant Bit Rate
CFB	Contention free Bursting
CFP	Contention Free Period
CP	Contention Period
CSMA/CA	Carrier Sense Multiple Access / Collision Avoidance
CW	Contention Window
CWmax	Contention Window Maximum
CWmin	Contention Window Minimum
DCF	Distributed Coordinate Function
DIFS	DCF InterFrame Space
DSSS	Direct Sequence Spread Spectrum
EDCA	Enhanced Distributed Channel Access
EDCAF	Enhanced Distributed Channel Access Function
ESS	Extended Service Set
FHSS	Frequency Hopping Spread Spectrum
HCCA	HCF Controlled Channel Access
HCF	Hybrid Coordination Function
IEEE	Institute of Electrical and Electronics Engineers
IP	Internet Protocol
IR	Infrared
ISM	Industrial, Scientific and Medical
MAC	Medium Access Control

NAV	Network Allocation Vector
OFDM	Orthogonal Frequency Division Multiplexing
OSI	Open System Interconnection
PC	Point Coordinator
PCF	Point Coordination Function
PIFS	PCF InterFrame Space
QAP	Quality of Service Access Point
QSTA	Quality of Service Station
QBSS	Quality of Service Basic Service Set
QoS	Quality of Service
RTS/CTS	Request To Send/ Clear To Send
SIFS	Short InterFrame Space
STA	Station
TCP	Transmission Control Protocol
TID	Traffic Identifier
TXOP	Transmission Opportunity
UDP	User Datagram Protocol
UP	User Priority
VoIP	Voice over Internet Protocol
WLAN	Wireless Local Area Network

Chapter 1

INTRODUCTION

1.1 Wireless Communication

Wireless communication is one of the fastest growing technologies in the field of communication. The demand for connecting devices without use of cable is increasing everywhere. Mobility, portability, and instant access (via the Internet) to unlimited information have become the need of businesses and individuals in day to day life. The impact of this technology on our lives will be tremendous and allow us to do things we never imagined. Wireless LANs can be found on college, office, residential buildings and, in many public areas. It provides a wide range of flexibility to mobile users that cannot be solved by traditional Wired LANs. One of the leading wireless technologies for LAN is IEEE 802.11.

Lots of research is going on and different improvements on the quality of service of WLANs are being published day by day. As this technology is dominating other wired communication, reliability of the network and its quality of service are the major tasks to be maintained. This dissertation has also gone through for the improvement of the quality of service of WLANs. From the study of different documents released it has been found that the parameters which are responsible for maintaining the QoS in WLANs are provided with the constant value. And also these static values don't always give the optimum result. At high load condition it results increased packetloss and also the decreased throughput. For the completion of this work lots of research has undergone and finally got the significant improvement on its performance by replacing the constant value of one of the parameters with the dynamic value including a function.

1.2 Introduction to IEEE 802.11

IEEE 802.11[1] is the standard for Wireless Local Area Networks (WLANs) released by IEEE in 1997, and is the most widely used standard now a days. It controls the two layers i.e. Physical Layer and MAC, which is a sub layer of Data Link Layer of the OSI model. It has defined three different physical layer specifications Infra-Red (IR) baseband PHY, Frequency Hopping Spread Spectrum (FHSS) and Direct Sequence Spread Spectrum (DSSS). Both FHSS and DSSS operate at 2.4 GHz and supporting 1 and 2 Mbps data rates. Due to the popularity of WLANs, IEEE has also released its other standards such as 802.11a, 802.11b, 802.11g, 802.11e, 802.11i, etc. The details of these topics are discussed in the following chapters.

There are basically two network architectures in 802.11 legacies, Basic Service Set (BSS) and Extended Service Set (ESS). BSS without an Access Point (AP) is called Independent Service Set and with an AP is called Infrastructure Service Set. Different BSS connected through an AP over a distributed system is called Extended Service Set (ESS).

IEEE 802.11 has defined two Medium Access mechanisms: DCF (Distributed Coordinate Function) and PCF (Point Coordination Function). DCF is Contention based medium access which is based on CSMA/CA mechanism and can be used for both infrastructure and Ad-hoc networks. PCF is Contention free medium access based on polling technique and requires an AP (Access Point) to control the station while accessing the medium. So, PCF is used only in Ad-hoc networks. DCF is highly used than the PCF.

DCF can support only the best effort services. All traffic are handled in the same way, there is no service differentiations mechanism. All the stations get the access to the medium with the same priority, whether it is the time sensitive multimedia or the text file. Thus, DCF cannot guarantee Quality of Service.

1.3 Introduction of IEEE 802.11e

Though wireless networks are better than wired networks regarding the ease of installation cost and flexibility, they suffer from lower bandwidth, higher delays and higher bit error. Thus providing QOS over such networks is the challenging and requires additional measures.

IEEE 802.11e [2][3] is the standard released by IEEE for the QOS enhancement in 802.11 networks. In this standard there is service differentiation mechanism i.e. the high priority traffic gets better services. To support service differentiation, it assigns different priorities for each data traffic. Furthermore, four different Access Categories (AC) queues are used with different priority.

IEEE 802.11e has also defined two medium access mechanisms: EDCF (Enhanced Distributed Coordination Function) and HCF (Hybrid Coordination Function). The EDCF manages the medium access in the contention period while the HCF is responsible for both the Contention Free Period (CFP) and the Contention Period (CP).

1.4 Organization of the Dissertation

The first chapter of this dissertation gives the brief introduction to new technology wireless communication. This is followed by the brief introduction of wireless Local Area Networks and spreads the concept of IEEE 802.11 and IEEE 802.11e.

The second chapter presents the problem statements and also the objective of this dissertation. This chapter also contains the literature review related to this dissertation.

The third chapter details about the IEEE 802.11 networks with all its standards. Different wireless network architecture and all its medium access mechanism are also presented.

Chapter four introduces about the Quality of Service (QoS) in networks. This chapter explains on detail of QoS with its mechanism and all its parameters. At the end of this chapter, limitation of IEEE 802.11 is also discussed.

Chapter five gives the detail about IEEE802.11e and its access mechanics EDCA and HCF. This chapter also explains about the priority of application and its QoS. Different EDCA parameters such as AFSN, TXOP limit and CW window are also explained.

Chapter six contains the implementation and the result of this dissertation work. For the implementation purpose, a topology file is designed and run in the network simulator. The output obtained from the implementations is tabulated. Also results comparing the default values with the improved ones are shown.

The last chapter concludes the research work. It also contains the recommendations for the future work for better improvement of QoS of the WLAN.

Chapter 2

BACKGROUND AND PROBLEM FORMULATION

2.1 Background

IEEE 802.11 wireless LAN (WLAN) [1] is gaining its popularity and is being largely used all over the world. Due to its many characteristics like, simplicity, flexibility and low cost Wireless technology plays a major role in the next generation wireless communication. This technology provides ubiquitous communication and computing environment in offices, hospitals, campuses, factories, airports etc. Now a days, people demand for wireless high speed data communication like VoIP, Multimedia Communications, High Definition Television (HDTV) even when they are moving around the areas. To fulfill these demands, multimedia applications require some quality of service (QoS) support. To provide these qualities of services, different functions of medium access control (MAC) layer and variable physical (PHY) layer characteristics are used.

Lots of research has been going on to provide the better QoS support in 802.11. IEEE 802.11 Working Group is currently focusing on enhancement of QoS support which is known as 802.11e. IEEE 802.11e is in its standardization process and its final draft has been released. IEEE 802.11e has defined two medium access mechanisms which are basically the improved version of DCF and PCF. The basic MAC (Medium Access Control) mechanism of 802.11 known as Distributed Coordination Function (DCF) is based on distributed channel access and employs CSMA/CA (Carrier Sense Multiple Access / Collision Avoidance) protocol for the medium access. Another access mechanism is centralized Point Coordination Function (PCF) which requires the AP as a point coordinator (PC). Today most of the wireless installations use DCF, whereas PCF is hardly implemented because of its complexity in design and inefficiency in access mechanism. Though IEEE 802.11 has become more popular, widely deployed and cost effective, it lacks to provide quality of service (QoS) support. Here, different applications demand different QoS guarantees, for example Voice over IP, or audio/video conferencing and Internet telephony require specified bandwidth, delay and jitter, but can tolerate some losses whereas text data can tolerate some delay but no packet loss. Here, all types of data traffic are treated equally in both DCF and PCF, regardless of the QoS requirements of the traffic. Hence, it cannot provide quality of service support. As different applications require different traffic specification, some mechanisms must be provided for service differentiation to give higher priority data traffic a better service. Due to these problems 802.11 MAC mechanisms face a big hurdle in adaptation of multimedia data transmission in wireless.

IEEE 802.11 task group has been working to provide quality of service, which is known as IEEE 802.11e. It provides a distributed access mechanism to support Quality of Service by introducing service differentiation. Here, different types of traffic are assigned with different priorities based on their requirements and service differentiation is introduced by using a different set of medium access parameters for each priority.

2.2 Problem Definition

Though wireless networks have many advantages over wired networks in the ease of installation and flexibility, there is more chance of service degradation i.e., low bandwidth, higher packet loss, etc. due to different factors like weather, noise and other environmental factors. So, maintaining the QoS is more challenging.

This dissertation focuses on the analysis of QoS in the IEEE 802.11 networks. The detail of the implementation of QoS in IEEE 802.11e networks will be presented. It includes the definition of various medium access mechanisms of IEEE 802.11 networks. Different EDCA parameters like Contention Window (min and max), AIFS and TXOP limits are being studied.

The main problem of EDCA is the static reset of contention window and TXOP limit which decreases significantly the throughput performance and increases the collision rate especially at high load condition. This dissertation would suggest the better approach for resetting these static values as dynamic. Just making these parameters dynamic would not solve the network problem. It results better in some aspect, let's say throughput and again degrades in another aspect. To overcome this problem, an application environment [10] would be defined. Analyzing the traffic on that specified environment, the EDCA parameters would be set accordingly. The popular network simulator NS2 would be used to analyze the parameters.

2.3 Objective

The main objective of this dissertation is to study and analyze the performance of WLAN. For this, a network environment is defined i.e., remote village school. Different traffic load occurring on this environment is analyzed. To provide better network performance, different EDCA parameters would be studied and examined. The performance of the network in terms of throughput, latency and packet loss would be observed and compared with the default network environment. Using NS2 simulator, performance of wireless LAN would be evaluated on the basis of EDCA parameters.

2.4 Literature Review

Wireless technologies are becoming need of every people of anywhere. There has been a lot of research going on in this field. The demand for wireless data services and multimedia application has grown. To provide better service to meet the growing demand, there has been a lot of research in the field of QOS. In this section, a brief summary of current work in this field is presented.

Lamia Romdhani, Qiang Ni, and Thierry Turletti,2002 [4], review the one of the main problem of EDCF i.e. static reset of the Contention Window(CW) which decreases significantly the throughput performance and increases the collision rate specially at high load condition. They proposed the formula to resize the Contention Window (CW) for each traffic class. They became able to increase medium utilization ratio and decrease the collision rate.

Mohammad Malli, Qiang Ni, Thierry Turletti, Chadi Barakat[5], review the limitations of IEEE 802.11e Enhanced DCF (EDCF) and other enhanced MAC schemes that have been proposed to support QoS for 802.11 adhoc networks. Then they describe a new scheme called "adaptive fair EDCF" that extends EDCF, by increasing the contention window when the channel is busy, and by using an adaptive fast backoff mechanism when the channel is idle. Their scheme improves the quality of multimedia application and also increases the overall throughput obtained both in medium and high load cases.

Anni Mtinlauri, 2008,[6] analyzed the txoplimit values. He proposed that to improve fairness while not disturbing high priority traffic, there should be use of large TXOP limit values. First of all lower priority traffic are set to infinite so that low priority queues can send all its packets when it gains access to the channel. The result shows that infinite TXOP limit improves fairness when channel is getting congested. Also infinite TXOP limit doesn't notably weaker high priority traffic performance.

Qiang Ni, lamia Romdhani, Thierry Turletti 2004[7], summarized a large number of 802.11 QOS enhancement schemes. They made a survey of current research activities and analyzed the QOS limitations of IEEE 802.11 wireless MAC layer. They described and classified different QOS enhancement techniques of IEEE 802.11 with their advantages and drawbacks. Finally they introduced the upcoming IEEE 802.11e QOS enhancement standard.

Nabil Tabbane, Sami Tabane, Ahmed Mehaouna 2005[8], presented SEDCF: Seasonal Enhanced Service Differentiation Methods for forecasting resources to meet the QOS requirements for real-time services. Their result showed that SEDCF protocol performs better than conventional EDCF

Feyza Keceli, Inanc Inan, and Ender Ayanoglu 2007[9], presented the unfairness problem between uplink and downlink flows of any access categories (AC) in 802.11e EDCA, when the default setting of EDCA parameters are used. They proposed the simple analytical model to calculate EDCA parameter setting to get the weighted fair resource allocation for both uplink and downlink flows. They also proposed the simple mode-assisted measurement-based dynamic EDCA parameter adaptation algorithm. They showed that their proposed Contention Window(CW) and Transmission Opportunity limit (TXOP) adaptation at AP provides fair UDP and TCP access between uplink and downlink flows of the same AC while preserving prioritization among ACs.

Chapter 3

IEEE 802.11

3.1 Standards

In 1997, IEEE (Institute of Electrical and Electronics Engineers) released the 802.11 Wireless Local Area Network (WLAN) standards [1]. As the name suggests, it belongs to the group of popular IEEE 802.x standards, e.g., IEEE 802.3 Ethernet and IEEE 802.5 Token Ring.

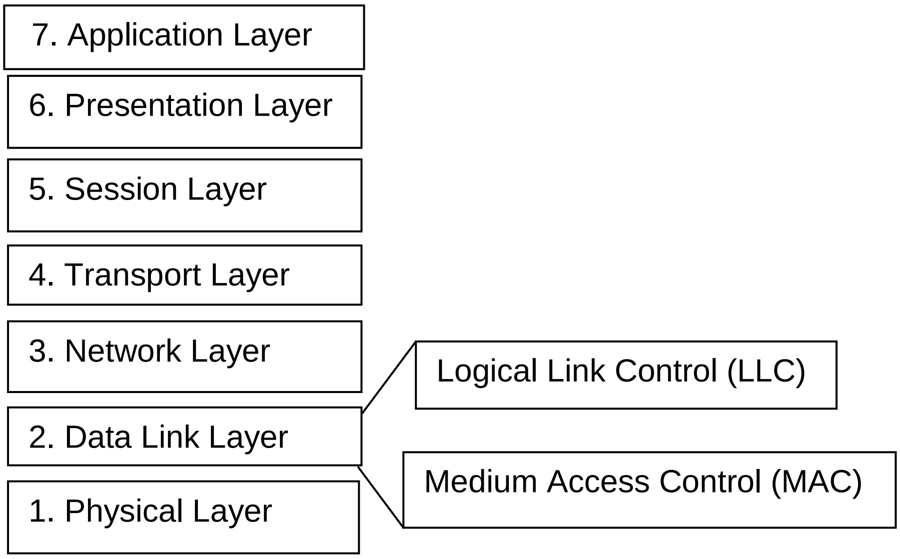


Figure 3.1: Seven layers of OSI Reference Model [10].

IEEE 802.11 controls Media Access Control (MAC) sub-layer and physical (PHY) layer of the OSI network reference model for Wireless LANs. A large variety of PHY layer specifications are defined. The legacy IEEE 802.11 standard provides three different PHY definitions: Infra-Red (IR) baseband PHY, both Frequency Hopping Spread Spectrum (FHSS) and Direct Sequence Spread Spectrum (DSSS) operating at 2.4 GHz, all supporting 1 and 2 Mbps data rates.

In 1999, IEEE defined two high rates extensions: IEEE 802.11a[11], based on Orthogonal Division Frequency Multiplexing (OFDM) in the 5GHz band with data rates up to 54 Mbps, and IEEE 802.11b[12], based on the DSSS technology, in the 2.4 GHz band with data rates up to 11 Mbps. In 2003 IEEE 802.11g [13] was approved. It extends the 802.11b PHY layer to provide data rate as high as 54Mbps in the 2.4GHz band.

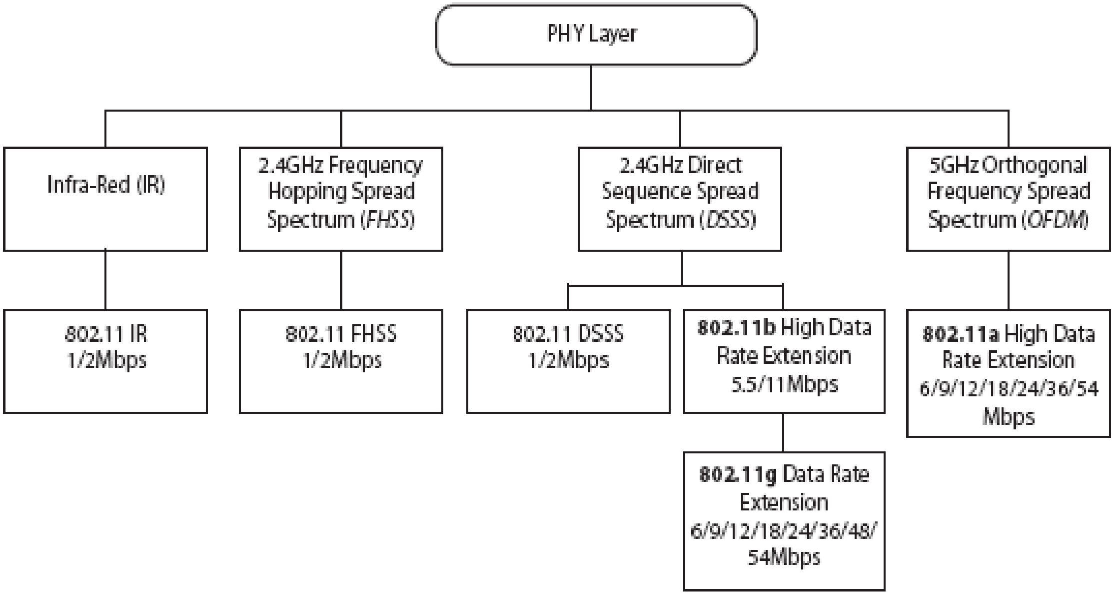


Figure 3.2: Snapshot of the IEEE 802.11 PHY standards [10]

In the MAC layer, the IEEE 802.11e improves the Quality of Service (QoS) performance of IEEE 802.11 and IEEE 802.11i enhances the security and authentication mechanisms

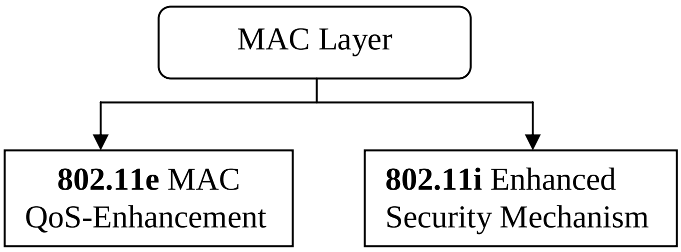


Figure 3.3: Snapshot of the IEEE 802.11 MAC standards [10]

IEEE 802.11e was approved in September 2005 and published by IEEE in November 2005 and IEEE 802.11i was released in June 2004.

3.2 Network Architecture

The standard defines two kinds of services i.e., the Basic Service Set (BSS) and the extended service set (ESS) [14]. A basic service set is made of stationary or mobile wireless stations and an optional central base station known as the access point (AP). The BSS without an AP is a stand-alone network and cannot send data to other BSSs. It is called an Independent BSS just like ad hoc architecture. In this architecture, station can form a network with out the need of AP; they can locate one another and agree to be the part of a BSS. A BSS with an AP is referred as an Infrastructure network. An ESS is made up of two or more BSSs with APs. When BSSs are connected, the stations within the reach of one another can communicate without the use of AP. However, communication between two different BSSs usually occurs via two APs.

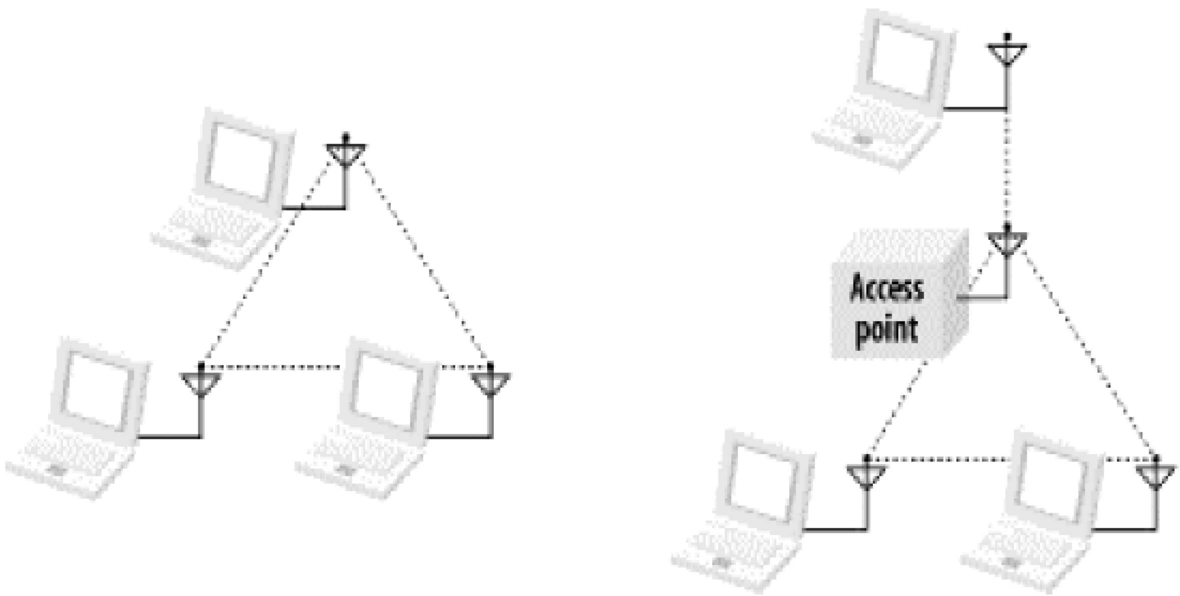


Figure 3.4 Independent BSS and Infrastructure BSS [10], [24]

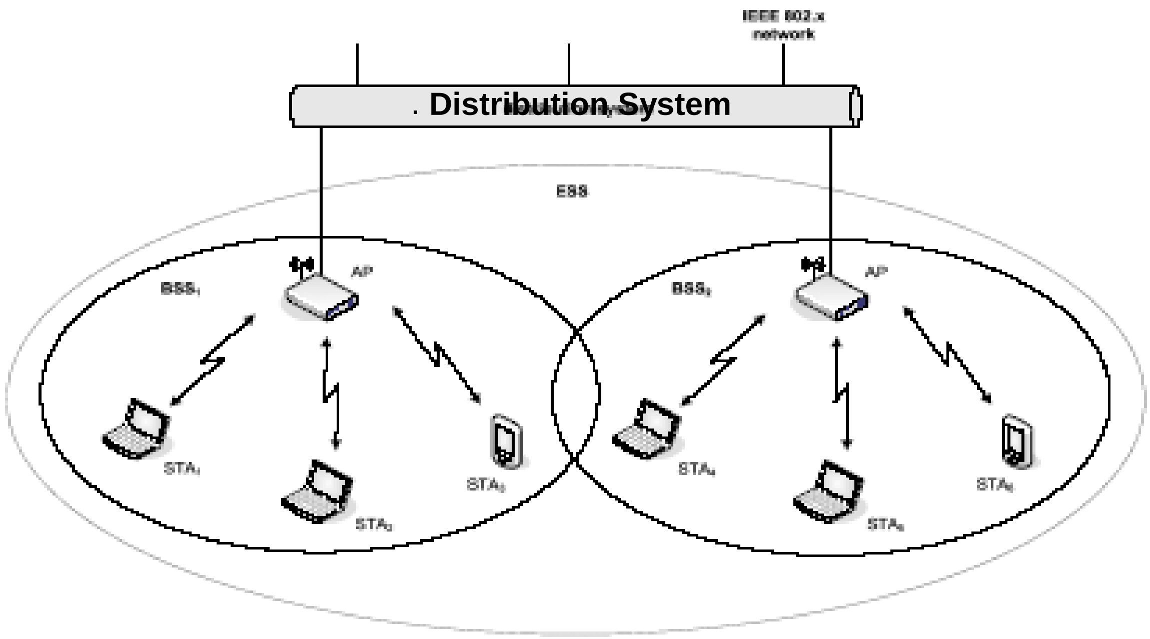


Figure 3.5: Architecture of IEEE 802.11 Networks [24]

3.3 Medium Access

The IEEE 802.11 MAC sub-layer introduces two medium access coordination functions, the mandatory Distributed Coordination Function (DCF) and the optional Point Coordination Function (PCF). DCF is based upon the Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) protocol which can be implemented in all stations for use within both ad-hoc and infrastructure network configuration. PCF is based on polling technique, i.e. nodes are allowed to transmit only when a central coordinator gives them permission to transmit. PCF can only be used in an Infrastructure BSS since it requires an AP as Point Coordinator (PC).

In 802.11 MAC, station delay transmission until the medium becomes idle and this is carried out by using varying interframing spacing. Different interframing spacing creates different levels of priority for different types of traffic. The different levels of priority facilitate so that high-priority traffic doesn't have to wait as long after the medium has become idle thus gets earlier chance to access the medium then lower priority traffic. To be interoperable between different data rates, the interframe space is a fixed amount of time, independent of the transmission speed. Different physical layers, however, can specify different interframe space times.

Interframe spacing plays a significant role in coordinating access to the transmission medium. In 802.11 there are four different interframe space which are described below:

I. Shortest Interframe Space (SIFS): Shortest interframe space is the shortest interframe so it is used for the highest-priority transmission, such as RTS/CTS frames and positive acknowledgements. As soon as SIFS time elapses, high priority transmissions can begin. And once high-priority transmission start, medium becomes busy.

II. PCF Interframe Space (PIFS): It is also called priority interframe space. It is used by PCF during contention free operations. Stations that have data to transmit in the contention free period can transmit after the PIFS elapsed and prevent any contention based traffic.

III. DCF Interframe Space (DIFS): It is minimum medium idle time for contention based services. Station has to wait for DCF interframe time to get access to the medium. After that it may have immediate access to the medium once the medium is free for a period more than DIFS period.

IV. Extended Interframe Space (EIFS): Extended interframe space is not fixed interval. It is used only when there is an error frame transmission. It is not used to control access onto the radio link

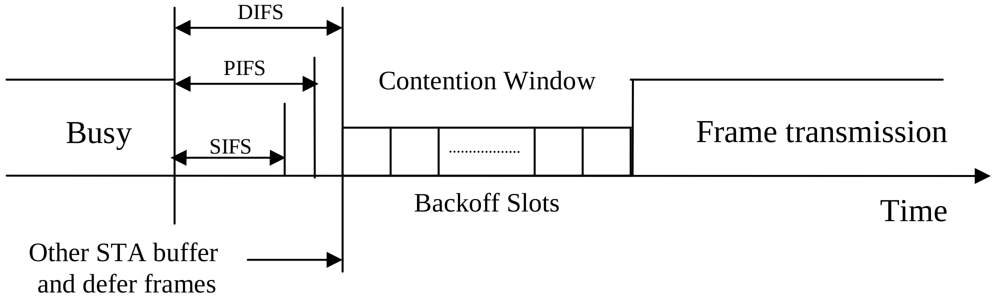


Figure 3.6: Interframe Spacing Relationship [25]

3.3.1 Distributed coordinate function

DCF is the basic medium access mechanism of the 802.11. It uses the CSMA/CA access mechanism. In this mode, a station must sense the medium before sending a packet. If the medium is found idle for DIFS time period, then transmission starts otherwise the transmission is deferred and waits for the medium to be clear. When destination receives frame it acknowledges by sending back ACK frame after SIFS time period. Collision is avoided by assigning different backoff values for each station contending to access medium once it is used by other station. Backoff value is the random value which is drawn between the Contention Window (CW).

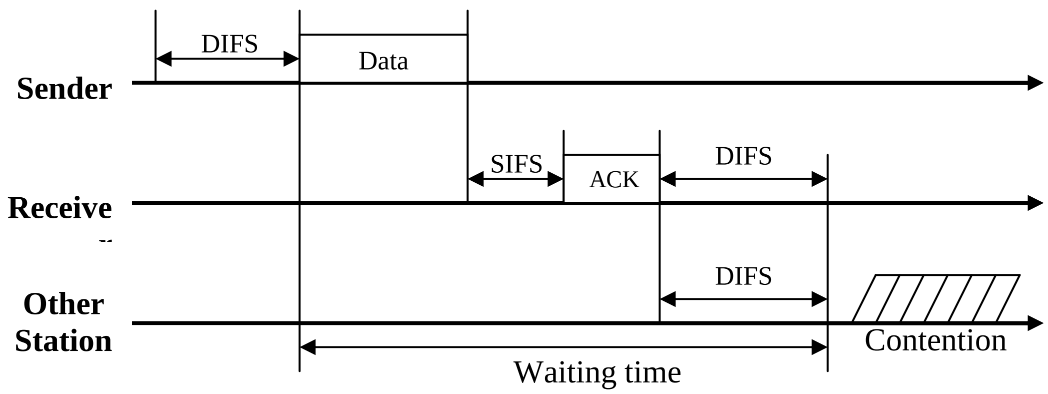


Figure 3.7: DCF basic access Mechanism [7], [25]

Carrier sensing is performed at two stages: Physical (PHY) carrier sensing at the air interface and virtual carrier sensing at the MAC layer via the Network Allocation Vector (NAV). Physical Carrier Sensing function is provided by Physical Layer and depends on the medium and modulation used. It is difficult and expensive to build physical carrier sensing hardware for RF-based media, because transceivers can transmit and receive simultaneously only if they incorporate expensive electronics. Virtual-Carrier Sensing is provided by the Network Allocation Vector (NAV). The NAV is a timer held by each STA, indicating the amount of time the medium will be busy. When a STA sends a data frame, a RequestToSend (RTS) or a ClearToSend (CTS) control frame, it uses the duration field in the MAC header to reserve the medium for a certain time period. All STAs located in the same BSA update their NAVs according to the duration field. When the NAV is non-zero, the virtual carrying function indicates that the medium is busy. When it reaches zero, the function indicates that the medium is idle.

Physical carrier sensing cannot provide all the necessary information for solving hidden node problem. Hidden terminals are STAs out of reach of each other but yet within range of a common receiver. As illustrated in fig below two STAs (STA 1 and STA 3) can be within range of a common receiver (STA 2) but not in range of each other. If STA1 sends a frame to STA2, STA3 may not detect channel activity because it is out of range of STA1 and may initiate a transmission, which results in a collision at STA2.

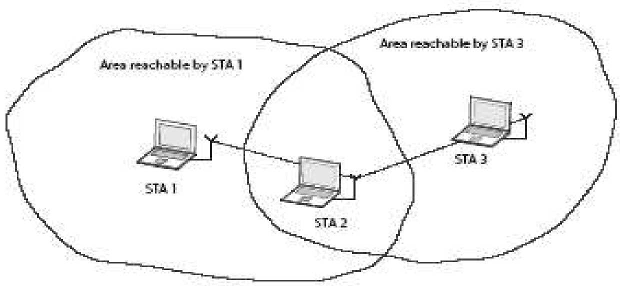


Figure 3.8: Hidden node problem [10]

The RTS/CTS mechanism is used to avoid hidden node problem. In this mechanism sender and receiver exchange RTS and CTS control frames by performing handshake mechanism. The source sends a short RTS frame before the data frame. Then the receiver answers with a CTS frame after a Short InterFrame Space (SIFS) which silences all the STAs within range of the receiver. Afterward, the source can send its data frame. With this scheme all STAs hearing a RTS, a CTS or a data frame can update their NAVs and will not start their transmission before their NAVs reaches zero. The collision of a short RTS or CTS frame is less severe than a collision of a large data frame as less time has elapsed when the collision is detected. The RTS/CTS handshake exchange can therefore improve the performance of DCF considerably. For small data frames, the overhead implied by the transmission of RTS and CTS frames becomes relatively large and the RTS/CTS handshake exchange is not desirable any longer. Therefore, the RTS/CTS exchange will be initiated only for packets larger than RTS threshold.

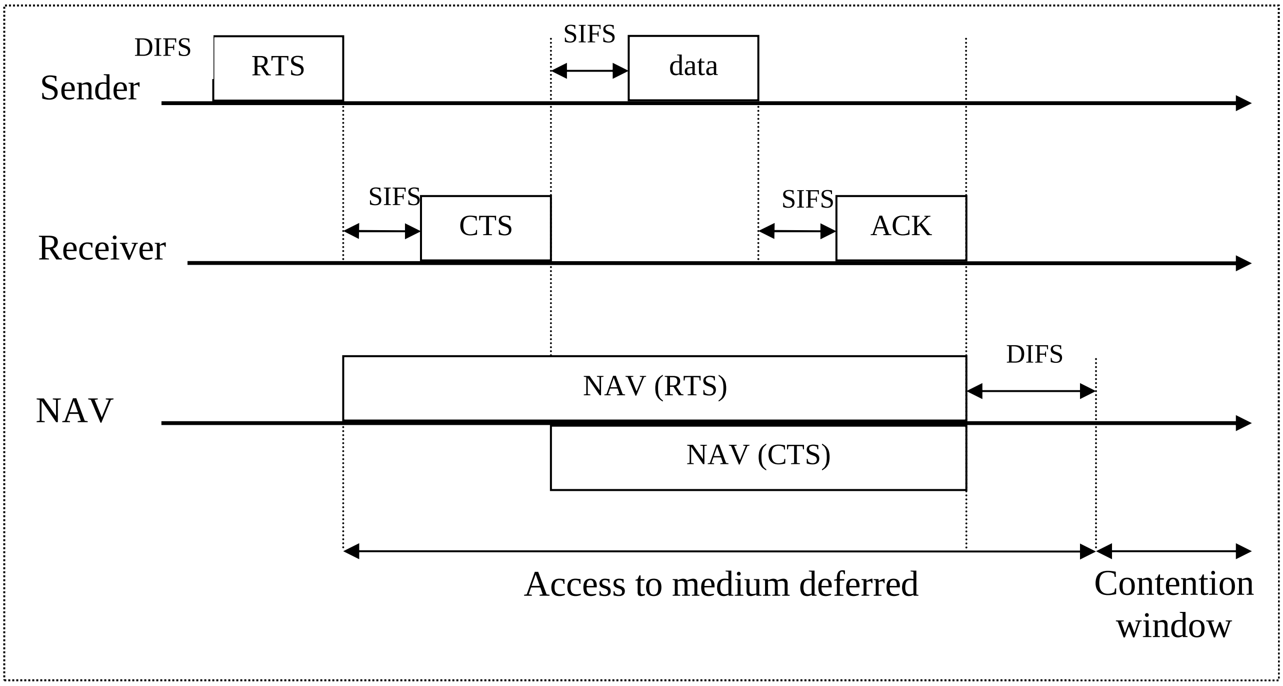


Figure 3.9: Frame exchange sequence with basic RTS/CTS mechanism [7], [25]

Frame fragmentation: Higher-level packets and some large frames may be fragmented to minimize the Bit Error Rate (BER) or to improve the reliability in presence of interference. An uncorrectable error in a larger frame leads to a higher waste of transmission time as compared to an error in a smaller frame. Fragmentation is controlled by the Fragmentation Threshold parameter in MAC. Packets larger than Fragmentation Threshold are broken up into several fragments and sent in a fragmentation burst. Each fragment is transmitted and acknowledgement is received in return and then only another transmission starts. Once the station reserves the medium, it can send multiple fragments of the frame i.e., in each fragment burst period multiple fragments can be transmitted which are separated by SIFS as shown in the following figure.

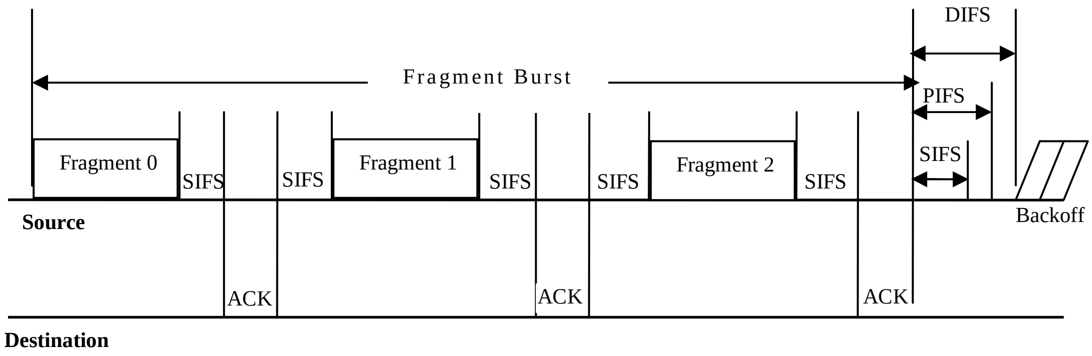


Figure 3.10: Frame Fragmentation [25]

3.3.2 Point Coordination Function (PCF)

The PCF can only be used in an infrastructure-based network because it requires an access point (AP). Usually the Point Coordinator (PC) is installed on this AP. The PC manages the access to the medium in the CFP by polling stations. The CFP, as the first part of a superframe, is being started periodically by a beacon frame sent by the PC after a PIFS-idle medium. Therefore the CFP may be delayed due to a long frame sent in the end of the CP. Because the PCF was developed on top of the DCF all stations have to set their NAV in the beginning of the CFP to the CFPMaxDuration value. After sending a beacon frame the PC has to wait at least SIFS before it can send a poll or a data frame (or both piggybacked). The PC polls all stations sequentially in the CFP. A polled station is allowed to answer with a data ACK frame after SIFS to the PC or to any other station in the network. If a polled station does not answer, the PC polls the next one after PIFS. If neither the PC nor the stations have frames to send, the CFP ends with a CFP-end frame which is sent by the PC. All receiving stations reset their NAV and the CP begins. If the PCF is used for the transmission of time-bounded data the PC should support a polling list. Every station listed there should be polled at least once per CFP. Stations are able to request for a place in the polling list with association management frames. Due to its complexity as well as its overhead the PCF is implemented but mostly not used in current installed WLANs.

Chapter 4

QUALITY OF SERVICE (QOS)

There is no formal definition on Quality of service. In the field of packet switched networks and computer networking, QoS informally refers to the probability of packets succeeding in passing between two points in the network. It measures the the reliability and consistency of a network. There are number of parameters used to measure QOS such as bandwidth delay, jitter, packetloss etc.

4.1 QoS Parameters:

4.1.1 Bandwidth

Bandwidth refers to the amount of data that can be transfer during a given period of time. It is often measured with respect to throughput, which is the data transfer rate, measured as the number of bits transmitted per second. Greater the bandwidth, lager the application receives data packets and vice versa. Several other terms are used for bandwidth such as data rate, transmission rate, bit rate and capacity. Some applications are bandwidth sensitive which requires data transfer at constant rate, such application in the absence of bandwidth results in undesirable delays and data loss. For example, multimedia applications, internet telephony (VoIP) and videoconferencing require dedicated bandwidth. On the other hand some applications like email, file sharing, web and instant messaging does not require bandwidth constraints but require delivery guarantee. To eradicate the bandwidth problem one simply solution is increasing the link capacity to accommodate all applications and user, with some extra bandwidth to spare. Though this solution is simple, but increasing bandwidth is expensive and takes time to implement. Also, there are technological limitations in upgrading the existing system. One another solution can be classifying the traffic into QoS classes and prioritize traffic according to its importance. Thus, voice and video traffic should get higher priority where as background and best effort traffics will get remaining bandwidth.

4.1.2 Delay

Delay refers to the unpredictability longer time for packets to reach the destination due to unavailability of network resources. End to end delay is the total delay from the time packet is generated at the sender side to the time it is received by receiver. It contains all types of delay such as Processing delay, Queuing delay, Serialization delay and Propagation delay.

I. Processing delay: The time a networking devices take the packet from input interface and puts into the output queue of the output interface. The processing delay depends upon CPU speed, CPU utilization etc.

II. Queuing delay: The time a packet resides in the output queue before it is transmitted. Queuing delay depends upon number of packets, size of each packet in the queue, bandwidth of the interface and the queuing mechanism.

III. Serialization delay: The time to place frames on the physical medium for transport.

IV. Propagation delay: The time to travel packets on the physical media interface. Propagation delay depends upon the velocity of propagation of the signal across the transmission media.

4.1.3 Packetloss:

Loss of packet is caused by collision of packets and congestion in the link. Lost of packets result in speech dropouts or a stutter effect. Most of the multimedia applications are loss tolerant but are sensitive to bandwidth and delay. i. e, they require strict bandwidth and delay guarantees but can tolerate certain amount of data losses. Jerks in videos and drop out voices are cause of data loss which reduces the voice and video quality. The effects of such losses on the quality and the amount of tolerable losses depend upon the application and technology used for coding. Whereas data oriented applications such as email, file transfer, instant messaging, web documents can tolerate delay for some amount but require reliable transfer of data

4.1.4 Jitter:

Jitter is variation of delay. Jitter becomes significant in constant bit rate multimedia data transmission. Jitter is difference in the end to end delay values of two voice or video packets. For such data transmission, decoder application at the receiver application is used which decode the received data according to the bit rate it was encoded at the sender station. Here, high variation in delay results problems in decoding, so, most of the multimedia applications use buffer to store the received data before decoding. The mechanism to control the frames in buffer is controlled according to the maximum expected jitter and the bit rate of the data it was sent.

4.2 QoS Mechanism

The Internet Engineering Task Force (IETF) has defined two different frameworks, Integrated Services (IntServ) [15] and Differentiated Services (DiffServ) [16], to support QoS for the traffic over Internet.

4.2.1 Integrated Service (IntServ)

IntServ can provide very high QoS to IP packets. Essentially, applications signal to the network that they will require special QoS for a period of time and that bandwidth is reserved. With IntServ, packet delivery is guaranteed. However, the use of IntServ can severely limit the scalability of a network.

Some applications, such as high-resolution video, require consistent, dedicated bandwidth to provide sufficient quality for viewers. IntServ was introduced to guarantee predictable network behavior for these applications. Because IntServ reserves bandwidth throughout a network, no other traffic can use the reserved bandwidth. Bandwidth that is unused, but reserved, is wasted.

IntServ is similar to a concept known as “hard QoS” With hard QoS, traffic characteristics such as bandwidth, delay, and packet-loss rates are guaranteed end to end. This guarantee ensures both predictable and guaranteed service levels for mission-critical applications. There will be no impact on traffic when guarantees are made, regardless of additional network traffic. Hard QoS is accomplished by negotiating specific QoS requirements upon establishment of a connection and by using Call Admission Controls (CACs) to ensure that no new traffic will violate the guarantee. Such guarantees require an end-to-end QoS approach with both complexity and scalability limitations. Large network environments that contain heavy traffic loads will be extremely challenged to track QoS guarantees for hundreds of thousands of signaled flows. Using IntServ is like having a private courier airplane or truck dedicated to the delivery of your traffic. This model ensures quality and delivery, is expensive, and is not scalable.

IntServ is a multiple-service model that can accommodate multiple QoS requirements. IntServ inherits the connection-oriented approach from telephony network design. Every individual communication must explicitly specify its traffic descriptor and requested resources to the network.

In the IntServ model, the application requests a specific kind of service from the network before sending data. The application informs the network of its traffic profile and requests a particular kind of service that can encompass its bandwidth and delay requirements. The application is expected to send data only after it gets a confirmation from the network. The application is also expected to send data that lies within its described traffic profile.

4.2.2 Differentiated Service (DiffServ)

DiffServ provides the greatest scalability and flexibility in implementing QoS in a network. Network devices recognize traffic classes and provide different levels of QoS to different traffic classes.

The Internet was designed for best-effort, no-guarantee delivery of packets. This behavior is still predominant on the Internet today. If QoS policies are not implemented, traffic is forwarded using the Best-Effort model. All network packets are treated exactly the same an emergency voice message is treated exactly like a digital photograph attached to an e-mail. Without the implementation of QoS, the network cannot tell the difference between packets and, as a result, cannot treat packets preferentially. When a letter is posted in standard postal mail, it uses a Best-Effort model. The letter will be treated exactly the same as every other letter; it will get there when it gets there. With the Best-Effort model, the letter may actually never arrive and, unless it has a separate notification arrangement with the letter recipient, It may never know if the letter does not arrive.

DiffServ was designed to overcome the limitations of IntServ models. DiffServ can provide an “almost guaranteed” QoS, while still being cost-effective and scalable. DiffServ is similar to a concept known as “soft QoS.” With soft QoS, QoS mechanisms are used without prior signaling. In addition, QoS characteristics (bandwidth and delay, for example), are managed on a hop-by-hop basis by policies that are established independently at each intermediate device in the network. The soft QoS approach is not considered an end-to-end QoS strategy because end-to-end guarantees cannot be enforced. However, soft QoS is a more scalable approach to implementing QoS than hard QoS, because many (hundreds or potentially thousands) of applications can be mapped into a small set of classes upon which similar sets of QoS behaviors are implemented. Although QoS mechanisms in this approach are enforced and applied on a hop-by-hop basis, uniformly applying global meaning to each traffic class provides both flexibility and scalability. With DiffServ, network traffic is divided into classes based on business requirements. Each of the classes can then be assigned a different level of service. As the packets traverse a network, each of the network devices identifies the packet class and services the packet according to that class. In this model packet can choose many levels of service. For example, voice traffic from IP Phones is usually given preferential treatment over all other application traffic. E-mail is generally given Best-Effort service. And non business traffic can either be given very poor service or blocked entirely.

Chapter 5

IEEE 802.11e

There has been agreement that the legacy IEEE 802.11 MAC does not meet the QoS requirements in the future advanced multimedia applications well. This is because DCF does not support QoS. All the data traffic is transmitted on a first come first serve, best-effort basis. There is no differentiation between data flows to support traffic with QoS requirements. All stations in the basic service set (BSS) contend for the wireless medium with the same priority. This causes asymmetric throughput between uplink and downlink, as the AP has the same priority as other stations but with much higher throughput requirement. When the number of stations in a BSS increases, the probability of collisions becomes higher and results in frequent retransmissions, which results in QoS decreases as well as overall throughput in the BSS.

In order to support QoS in the legacy IEEE 802.11 MAC, IEEE is working on new standard called IEEE 802.11e. In this standard, there is a provision for service differentiation so that higher priority traffic gets better services. To support service differentiation, it assigns different priorities for each data traffic. Furthermore, four different Access Categories (AC) queues are used with different priority. Access to the medium is then granted based on the priority of the data by mapping the data traffic to specific Access Category.

In IEEE 802.11e, the AP and STA that provides QoS services are referred to as QAP (QoS Access Point) and QSTA (QoS Station) respectively, and the BSS they are operating in is called QBSS (QoS Basic Service Set). IEEE 802.11e introduces a new coordination function, called Hybrid Coordination Function (HCF), to provide QoS support employing prioritized medium access.

5.1 HCF (Hybrid Coordination Function)

Hybrid Coordination Function (HCF) is a new mechanism to provide service differentiation to the different traffic. This new coordination function is backwardly compatible with the legacy DCF and PCF. HCF has two concurrent modes of operation: a contention access, called Enhanced Distributed Channel Access (EDCA)[20] and a controlled access called

HCF Controlled Channel Access (HCCA)[19]. EDCA operates only in the CP while HCCA can operate in the CFP and in the CP as well.

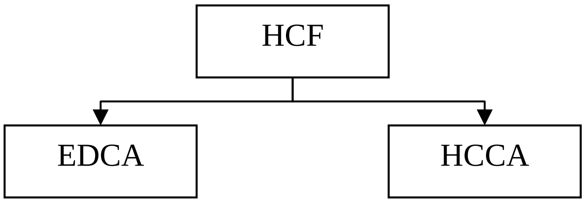


Figure 5.1: Hybrid Coordination Function

In HCF four access categories (AC) queues are used in addition to eight traffic stream (TS) queues at MAC layer. When a data frame arrives at MAC layer, it is marked with a traffic priority identifier (TID) according to the QoS requirement, whose value ranges from 0 to 15. The frames having TID 0 to 7 are mapped into four access categories using EDCF access rule whereas frames with TID 8 to 15 are mapped into eight traffic streams(TS) queues using HCF controlled channel access rule. AC is used to support strict prioritized QoS while TS is used to support parameterized QoS.

5.2 EDCA (Enhanced Distributed Channel Access)

EDCA provides differentiated, distributed access to the medium using different priorities for different types of data traffic. The detailed description of the components and operations of EDCA are as follows:

5.2.1 Access Categories (ACs)

Four Access Categories (ACs) are defined in EDCA for different types of data traffic. Service differentiation is introduced such that for each AC, a different set of parameters are used to contend for the medium. These parameters are referred to as EDCA parameters. Here, data frames from different application profiles are mapped into different ACs in MAC depending on its QoS requirements. The four Access Categories are named AC_BK, AC_BE, AC_VI AND AC_VO, for background, best effort, video and voice data traffic respectively. Here, AC_BK has the lowest priority and AC_VO has highest priority. So, each frame from the higher layer arrives at the MAC layer along with a priority. This priority value is called User Priority (UP) and is assigned according to its service requirement. There are eight different priorities values ranging from 0 to 7.

Priority	User Priority (UP)	Access Category (AC)	Designation
Lowest	1	AC_BK	Background
.	2	AC_BK	Background
.	0	AC_BE	Best Effort
.	3	AC_BE	Best Effort
.	4	AC_VI	Video
.	5	AC_VI	Video
.	6	AC_VO	Voice
Highest	7	AC_VO	Voice

Table 5.1: User Priority (UP) to Access Category (AC) mappings [17]

5.2.2 EDCAF (Enhanced Distributed Channel Access Function)

EDCAF is an enhanced version of DCF, which contends for the medium as in DCF i. e, CSMA/CA mechanism. The EDCF [17] is designed for the contention based prioritized QoS support. Here, each QoS enhanced station (QSTA) has 4 queues called Access Categories (AC) to support 8 user priorities (UPs) as defined in IEEE 802.1D [20]. Since, there are 8 user priorities [18] and only 4 priority queues, so more than one UPs are mapped to the same AC queue as shown in table 5.1. This is because usually eight different applications do not transmit frames simultaneously, and using less ACs than Ups reduces the MAC layer overheads. Here, each AC queue acts as an independent DCF station and uses its own backoff parameters.

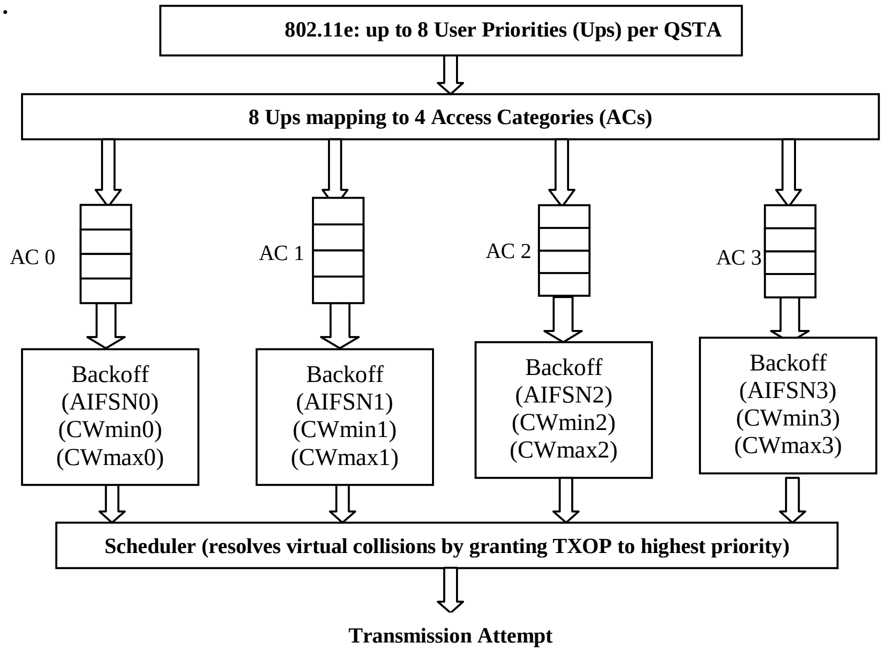


Fig 5.2: Enhanced Distributed Coordinated Function (EDCF) [7], [25]

In EDCF, two methods are introduced to support service differentiation; the fist one is to use different InterFrame Space (IFS) sizes for different ACs. Second one is allocating different CW sizes for different ACs. High priority AC is assign less CW size so that it gets opportunity to use the medium earlier. If two or more stations have backoff counter zero at the same time, a scheduler inside the station will avoid the virtual collision by granting the EDCF-TXOP to the highest priority AC. And other colliding AC will double its CW and starts backoff as if external collision has happened.

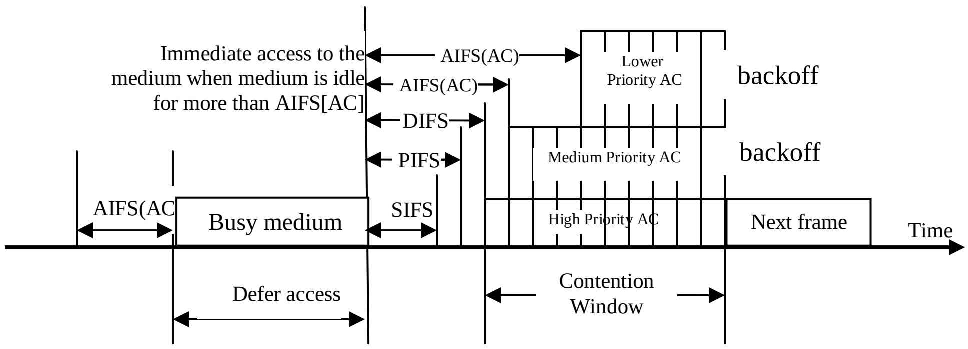


Figure 5.3: EDCA channel access prioritization

5.2.2.1 EDCA parameters:

Following are the parameters associated to Access Category (AC) which are used for EDCF contention.

- AIFS – Time period the medium has to be idle before the transmission start.
- CWmin, CWmax – Minimum and maximum size of Contention Window used for backoff.
- TXOP Limit – The maximum time, during which two stations can use the medium after they have acquired it.

EDCA parameters are specified different for different Access Categories. As shown in figure 5.3, the higher priority AC has to wait less time i.e., AISF time period than lower priority before accessing the medium. Also, the size of Contention window varies for different ACs, i. e. size of contention window is small for higher priority traffic while larger for lower priority traffic since backoff values are drawn from this contention window. On the other hand TXOP limit also varies; it is larger for higher priority so that it can use the medium for longer period of time and shorter for lower priority traffic. In summary we can say that for higher priority

ACs, AIFS and contention window will be small while TXOP will be larger. Since, the EDCA parameters are AC specific, so they are referred as AIFS [AC], CWmin [AC], CWmax [AC] and TXOP limit [AC]. Thus, the main difference between DCF and EDCF is EDCF uses AC specific parameters AIFS [AC], CWmin [AC], CWmax [AC] instead of only one DIFS, CWmin and CWmax.

QAP is scheduled to advertise the EDCA parameters periodically. QAP determines these parameters dynamically by considering the present network condition. Following are the EDCA parameters i. e. AIFS, CW and TXOP, used for service differentiation:

AC	**CWmin**	**CWmax**	**AIFSN**	**TXOP Limit**	
				FHSS	**DSSS**
AC_BK	CWmin	CWmax	7	0	0
AC_BE	CWmin	CWmax	3	0	0
AC_VI	(CWmin+1)/2 – 1	CWmin	2	6.016ms	3.008ms
AC_VO	(CWmin+1)/4 – 1	(CWmin+1)/2 - 1	2	3.264ms	1.504ms

Table 5.2: Default EDCA parameter values

i. AIFS (Arbitration Inter-Frame Space): It is the time the medium should be idle before acquiring the medium or backoff is started. The AIFS [AC] is calculated as

$$AIFS\ [AC] = AIFSN\ [AC] * SlotTime + SIFS$$

The default values of AIFSN are shown in the above table 5.2. AIFSN specifies the number of slot time plus SIFS time period. The minimum value of AIFSN is 2 as the DIFS is equal to 2 * SlotTime + SIFS, it shows that the minimum length of AIFS is equal to DIFS. But in the case of HCCA, the minimum value of AIFSN is 1 as 1 * SlotTime + SIFS equals to PIFS. AIFSN value is directly proportional to delay. So, higher priority traffic is assign low AIFSN value that is 2 as shown in above table so that higher priority traffic will get larger share of bandwidth. Though higher priority data are given preference, low priority may suffer from longer delays but since, these low priority data are delay tolerable, certain amount of delay do not degrade the performance beyond the acceptable level.

ii. CWmin and CWmax: As in the DCF in EDCF the size of CW is also not constant and varies according to AC. Contention Window (CW) is also directly proportional to delay. So, higher priority traffics (AC) are assigned low value of CW so that it is able to access the medium ahead of lower priority traffic (AC). If two ACs try to access the medium at the same time then internal collision will occur. In that case the scheduler inside the QSTA selects higher priority AC to access the medium and other lower priority traffic enter a backoff process with doubling the CW[AC] size as in case of external collision.

	FHSS	DSSS
CW_{min}	15	31
CW_{max}	1023	1023

Table 5.3: Contention window parameters for different physical layers

The CWmin and CWmax values of AC_BK and AC_BE are same as in the legacy 802.11 DCF, but priority is given to AC_BE over AC_BK by assigning it AIFSN value 3 which is less than AIFSN 7 of AC_BK. The values of AC_VI and AC_VO are different and smaller as one half or quarter compare to lower priority ACs. This is to provide smaller backoff values for higher priority ACs and thereby shorter medium access delays. Here, one drawback of smaller contention window value is that, there is more probability that two or more ACs get same random backoff value leading to an internal collision. To minimize this internal collisions CWmax value is set such that it is always less than CWmin of lower priority traffic ACs. So, even though there is collision and CW is doubled, its value never exceeds the CWmin of the lower priority traffic thus it avoids overlapping values facilitating to get different CW value. So, it is confirmed that higher priority traffic ACs get greater share of the bandwidth even in the congested network condition. However, this may lead the lower priority ACs to starvation.

The transmissions is said to be failed or collision is said to occur when two or more ACs or STA tries to access the medium at a same time. For each collision, the value of Contention Window is doubled by following equation:

$$CWmin = 2^m * (CWmin +1) -1,$$

where m is the maximum backoff stage.

$$CW = 2^i * (CWmin + 1) – 1, \text{ if } 0<i<m,$$

$$\text{and } CW = CWmax, \text{ if } m<=i,$$

where, i is the number of unsuccessful attempts. Once it reaches CWmax, its value remains constant i. e. CWmax, after first successful transmission its value will be reset to CWmin.

iii. TXOP limit: Transmission Opportunity limit is the maximum time duration during which multiple packets can be exchanged between two stations acquiring the medium without interferences of other stations. The multiple packets also include ACKs frames, RTS/CTS frames which are separated by SIFS within the TXOP period.

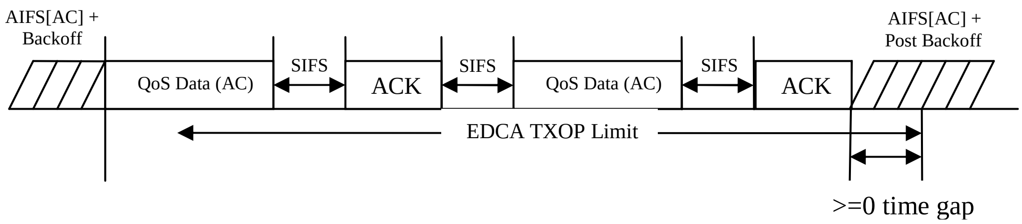


Figure 5.4: Contention Free Bursting (CFB) [17]

The maximum value of TXOP is called TXOP Limit and it is determined by QoS AP. The default value of TXOP is shown in the above table 5.2. The zero value of TXOP for AC_BK and AC_BE indicates that CFB is disabled and only one frame can be exchanged during TXOP. If RTS/CTS is enabled then RTS/CTS frame is also included in the transmission. If the time to transfer first frame exceeds TXOP Limit then the frame should be fragmented. But the TXOP Limit value of AC_VO and AC_VI are 3.264 ms and 6.016 ms in FHSS respectively, so, these AC can transmit multiple frames in TXOP Limit duration provided that the frames belong to the same AC. This period is known as contention free bursting period. In this period the frames are separated by SIFS time period. The multiple frames of same AC are only allowed to transfer for whom the TXOP was obtained during this time. If RTS/CTS mechanism is employ in CFB, then the RTS and CTS frames are exchanged only once during the first time, and later frames can transfer with the gap of SIFS till the TXOP Limit. In the above table, the default values of TXOP limits for the low priority ACs, AC_BK and AC_BE are set to zero indicating that CFB is disabled. But for high priority AC_VO and AC_VI, the CFB allows to access the medium for large duration this provides service differentiation for high priority AC. But this may lead lower priority AC suffer from starvation. When CFB is applied, to let the other stations aware of it, virtual carrier sensing is applied such that the duration field in the frame header is set to remaining duration of the whole TXOP which is transmitted

5.2.2.2 EDCA operation:

EDCA works similar to DCF, only difference is that, it has different AIFS, CWmin, CWmax and TXOP Limit for different ACs. When the medium is sensed free for AIFS time period, ACs draws a random backoff value from contention window interval. This backoff value is decreamented at each slot time and once its value reaches zero, it can start the transmission acquiring the medium.

Considering the following figure, here all the four ACs have frames to transmit so are contending for the medium.

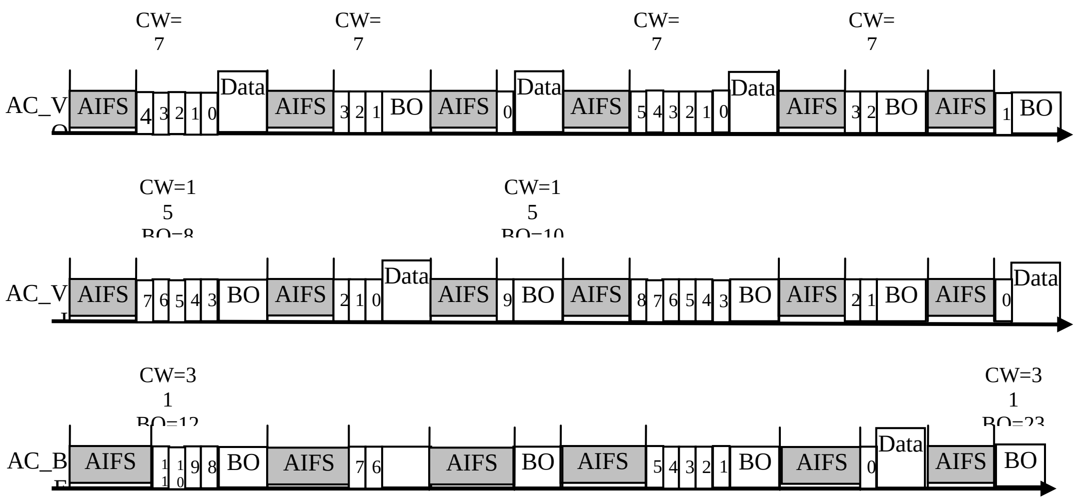


Figure 5.5: EDCA access mechanism [24]

From the table 5.2, the AIFSN values of AC_VO and AC_VI are 2 and that of AC_BE and AC_BK are 3 and 7 respectively. So, AC_BE and AC_BK have to wait for some additional slot time to access the medium. As high priority AC has smaller minimum and maximum contention window limits, it gets smaller backoff values and so has to wait less time contending the medium. Here, the highest priority AC gets access to the medium and other ACs pause their backoff timer until the medium is idle for AIFS time period. Thus at a certain time, lower priority AC has smaller backoff value when higher priority AC chooses a new backoff value for every next frame. Lower priorities ACs just decrement its paused backoff value. This helps to avoid starvation of low priority ACs. In this way higher priority ACs gets larger share of the bandwidth transmitting the frame more frequently then low priority ACs. In the above figure it is clearly seen that AC_VO send 3 frames, AC_VI send 2 frames, and

AC_BE send 1 frame. While AC_BK which is the lowest priority AC could not send single frame till that time since it has to sense the medium to be idle for longest AIFS time period. Actually, it is unable to decrement its backoff value because another AC acquires the medium before its AIFS is finished.

When two of more ACs tries to access the medium at a same time then collision is said to occur. This happens when backoff timer of two or more ACs decrement to zero at a same time. Such collision is called internal collision. To handle such situation, the internal scheduler selects the highest priority ACs and grant access to the medium, while other ACs doubles its CW and draws new backoff value after the medium becomes idle for AIFS time period. The situation is shown in the following figure 5.6.

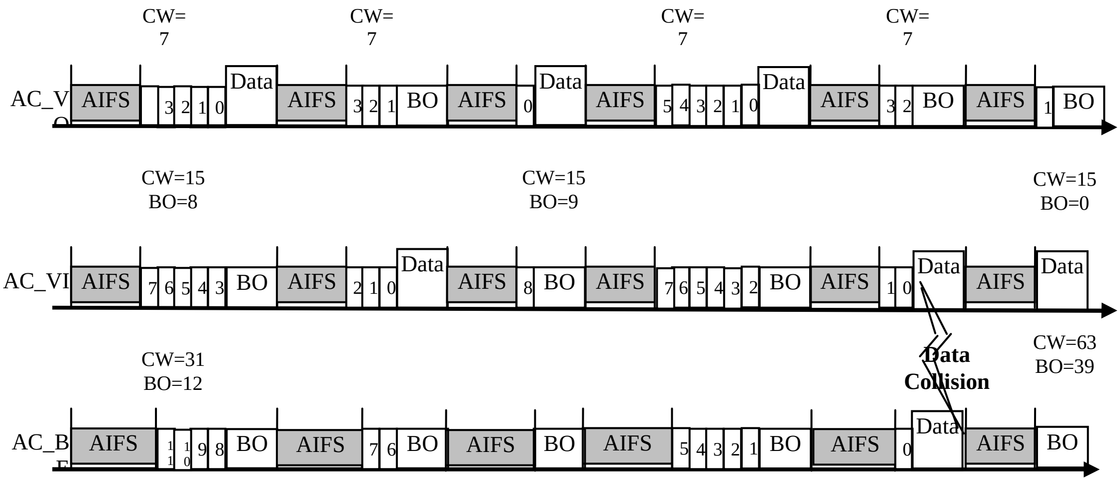


Figure 5.6: EDCA access mechanism and internal collision [24]

In the above case, the only drawback is low priority AC has to wait for longer time. The case gets even worse when EDCF for low priority collides, and the backoff value is drawn from double the last CW size.

When two or more stations tries to access the medium at a same time, then the collision is said to occur. This collision is called external collision and occurs when backoff value of two or more STAs countdowns to zero at a same time. The recovery process is somehow similar to internal collision, only difference is, here station is considered instead of particular AC. Here for external collision, the al the colliding EDCAFs increases their contention window to double and new backoff value is drawn while other stations starts countdown from their last value. In the figure below, Two EDCAFs for AC_VO and AC_VI in two different stations contends for the medium and their backoff timer countdowns to zero at a same time. Both stations try to access the medium and transfer their data. When no ACK frame is received then the stations realize that collision has occurred. Now both colliding EDCAFs double their contention window. Other stations continue decrementing their paused backoff values while colliding stations start from new backoff.

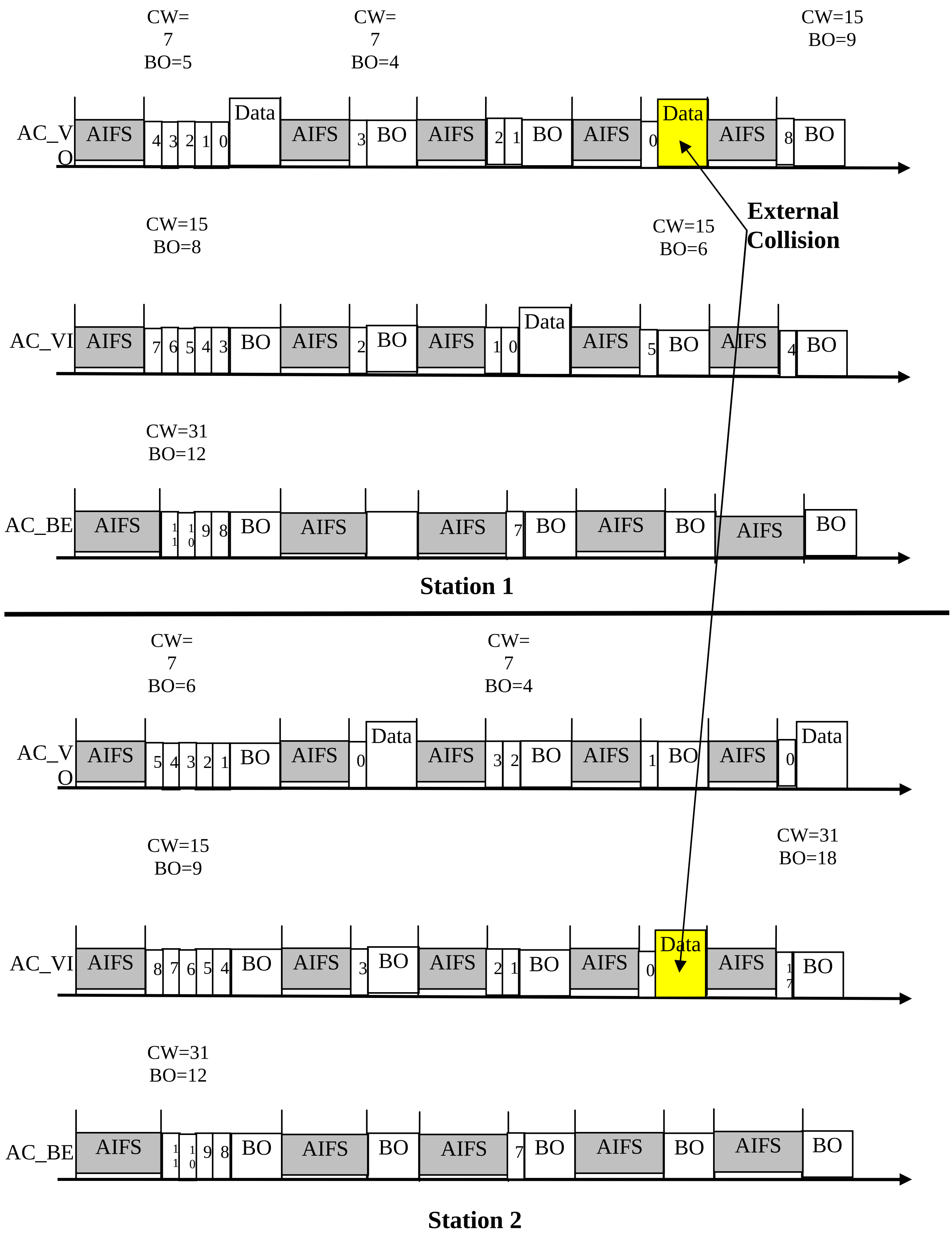


Figure 5.7: EDCA access mechanism and external collision [24]

Chapter 6

IMPLEMENTATION AND RESULTS

6.1 Methodology

The popular network simulator ns2 [21] has been used for the implementation of this dissertation. Performance of the network is evaluated and measured in terms of its parameters such as packetloss, throughput and latency. To get the improved performance of the network the throughput needs to be increased whereas the packetloss and latency should decrease. The whole dissertation is carried out to get better performance of WLANs. First of all only the constant value of the transmission opportunity limit is made dynamic just keeping the general environment. On general environment network though the simulator produce the output as increased throughput on the other hand it increase the packetloss. So just changing the static value to a dynamic value without considering network environment may give better result in some respect but again degrades the performance on another aspect.

To implement 802.11e, four queues have been maintained for different priority data. Each queues has different AIFSN[i] value i=0to 3 and the queue that reaches AIFSN value zero will get the opportunity to transmit the data first. Thus higher priority data gets more opportunity to transmit than lower one. Even the lower priority data does not have to wait longer to be transmitted. In this way the quality of service is maintained.

The EDCA parameters that are responsible to provide quality of service are Arbitration Inter-Frame Space (AIFS), Contention Window (CW) and Transmission Opportunity (TXOP) limit which are described in chapter 5.Among these only the TXOP limit has been considered in this research.

The topology file contains four applications of same traffic type Constant Bit Rate (CBR). On this environment the parameter for the TXOP is adjusted in such a way that if total number of collision is equal to total packet received then the TXOP limit value remains its default value. If total number of collision is greater than the total number of received packets then the new TXOP limit becomes the half of the previous TXOP limit value otherwise the new TXOP limit is increased by two in the previous TXOP limit. It has been defined in a function and added to mac_802.11e.cc file which is shown in appendix B.

In the earlier version of 802.11e the value of transmission opportunity is constant which is described in priority.tcl file in NS2. An example of such file is shown in appendix A.

6.2 Simulations

For the implementation of this work, a topology file has been defined considering all its application and their data traffic. According to the network environment defined in the topology file values of the TXOP has been adjusted in mac802_11e.cc file which resides on the mac/802.11e of ns2. A function named “myTXOP”(shown in Appendix B) has been defined and included in the mac802_11e.cc file. After updating the C++ file, it is compiled using the make command and all its object files are created. When the object files are created the topology or the tcl file is run on the ns2. It creates its output in a trace file (.tr). These trace file contains the raw data all in the columnar form which cannot be easily readable. So to get the required output in terms of packetloss, throughput and latency perl script has been used. These perl script files are shown in appendix C. The simulation steps are shown in figure below:

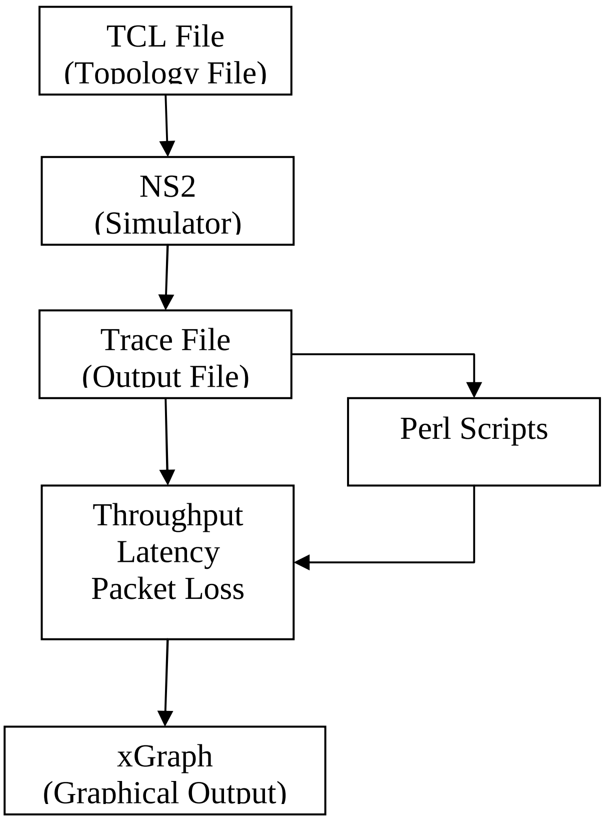


Figure 6.1: Showing the simulation steps

6.3 Results

Simulation of the network performance is carried in NS2 under different scenarios i.e. changing the no of stations from 10 to 50. Results obtained from simulation while using default TXOP limit value and the proposed TXOP limit value are listed in the table below

Results on Default TXOP			
No. of Stations	Packetloss (%)	Throughput(Bytes/sec)	Latency(Sec.)
10	40.9431843700876	27647.7342529175	0.090342478883143
20	70.4437400950872	27255.2446632873	0.292602387778000
30	80.2940092427206	27201.6483526953	0.299429739038681
40	85.3891165659724	26953.5287119350	0.301490043770975
50	88.4134287805240	26703.3108957097	0.306911809250176

Table 6.1: Values obtained on simulating under Default TXOP

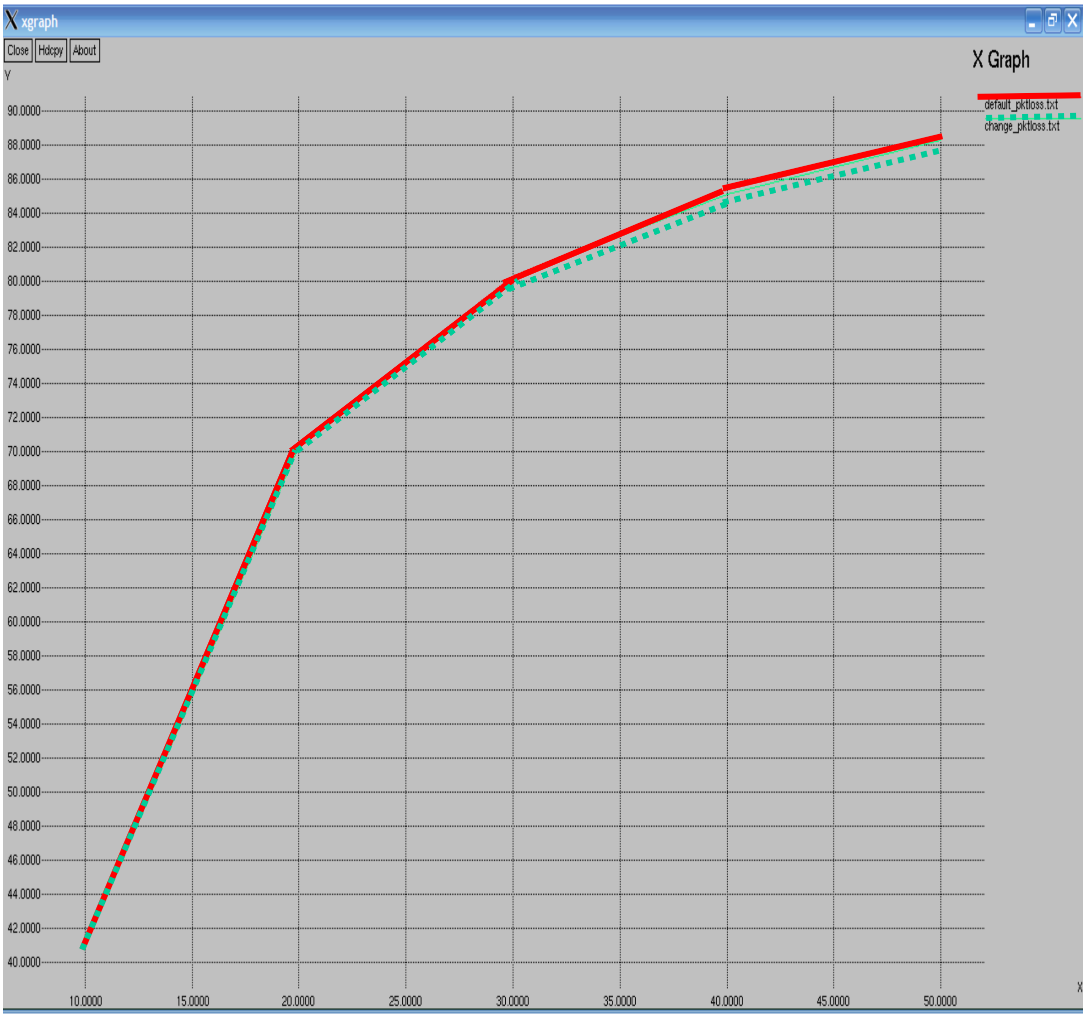
Results on Dynamic TXOP			
N o. of stations	Packetloss (%)	Throughput(Bytes/sec)	Latency(sec)
10	40.8596451830227	27732.5806960529	0.088616006299148
20	70.2895446952499	27333.5536358732	0.214553837073914
30	80.2259646735278	27209.4452194113	0.220519651357750
40	85.0912586083062	27021.1053610089	0.224821892822871
50	88.2594907426879	26867.1220046256	0.235776307280380

Table 6.2: Values obtained on simulating under Dynamic TXOP

The above table shows that the dynamic TXOP limit gives better performance than the default value. The above result can also be shown in the graphical form using Xgraph of NS2 as shown follows:

6.3.1 Comparison between the packetloss under default and dynamic value

Packetloss in percentage



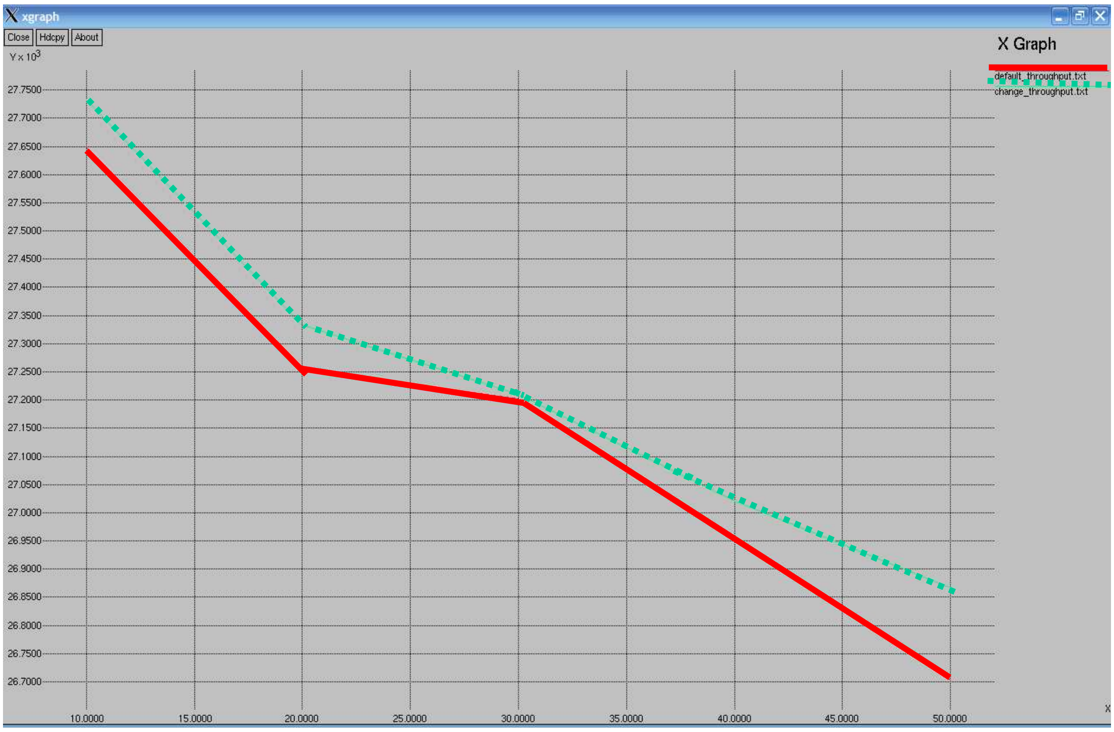
Number of stations

Figure 6.2: Packetloss Comparisons (Number of stations Vs Packetloss in percentage)

Comparison between the packetloss has been shown in the above graph. When the number of stations is 10 then the packetloss is about 40% but when it reaches upto 50 station the packetloss has also increased upto 88%. The above graph also shows that when the number of stations which we want to connect is upto 10 then we get the slight improvement on the performance, the packetloss percentage has been decreased by about 0.22%. In the same way the no of stations has been increased and we get the better performance. In average about 0.20% improvement has been achieved simulating under the Dynamic value of TXOP.

6.3.2 Comparison between the throughput under default and dynamic values

Throughput (Bytes/sec)



Number of Stations

Figure 6.3: Throughput Comparison (Number of stations Vs Throughput)

Comparison between the number of stations and the throughput in terms of Bytes/sec has been shown in the above figure. At fewer numbers of stations, higher throughput has been obtained, as the number of station increases the throughput value has been decreased.

When simulating under 10 stations, about 0.34% improvement has been obtained on Dynamic TXOP limit value. But while simulating under 50 stations about 0.61% improvement has been obtained. In average using the dynamic value of TXOP about 0.48% improvement on the throughput has been obtained.

6.3.3 Comparison between the latency under default and dynamic values

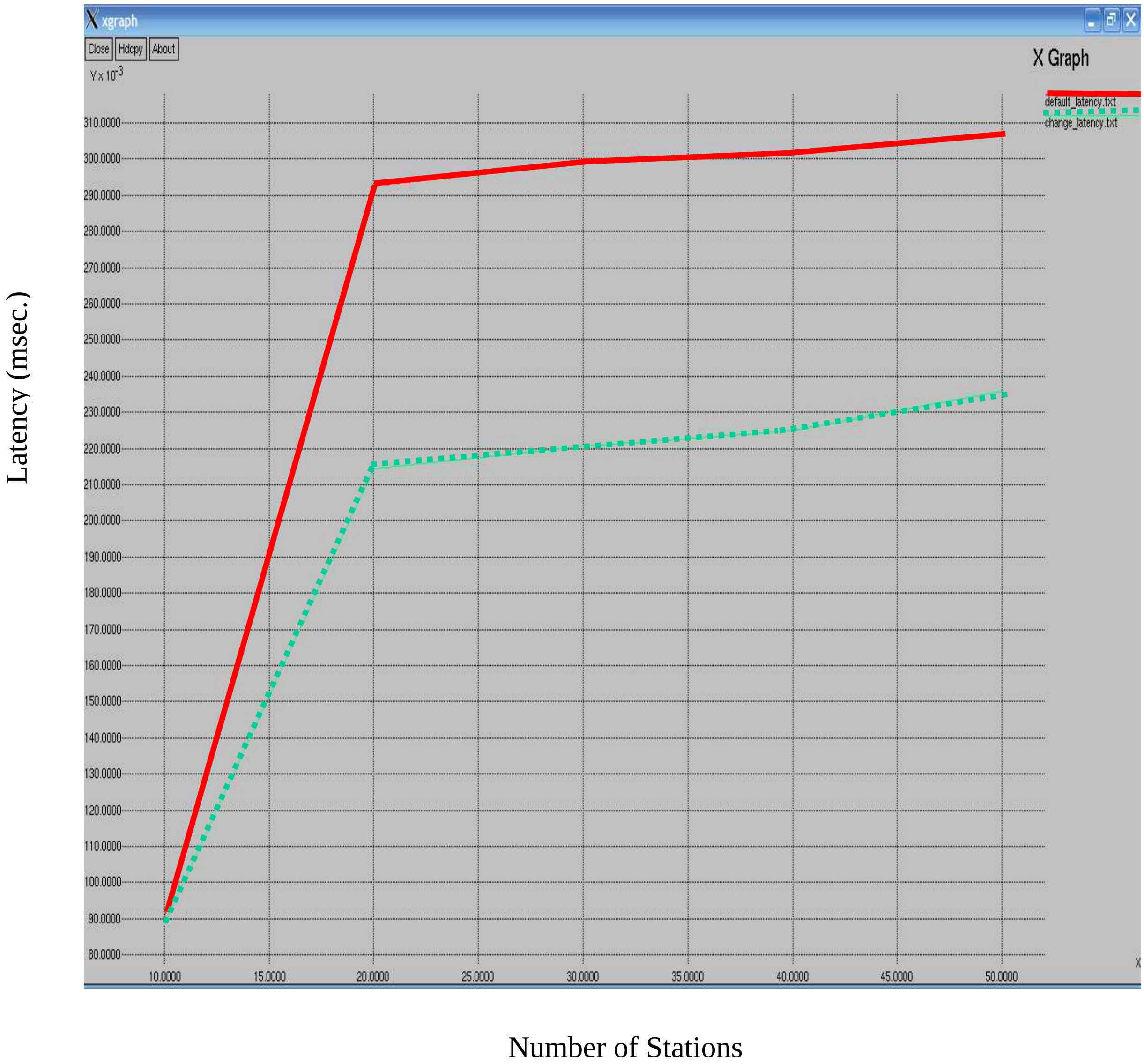


Figure 6.3: Latency Comparison (Number of stations Vs Latency)

Comparison between the number of stations and latency has been shown in the above figure. It has been shown that as number of stations increased latency has also been increased. This graph has also shown that there is exponential increase in the latency upto 20 stations but when the stations increase from 20 the growth of latency is constant.

Simulating the network on default and dynamic values of TXOP under different scenario using different number of stations, it has been found that the about 12.5% improvement on the network performance has been obtained.

Chapter 7

CONCLUSION AND FUTURE WORK

7.1 Conclusion

The main objective of this dissertation is to improve the QoS in IEEE802.11e. One of the EDCA parameter TXOP limit is analyzed. The default or the constant value of the TXOP limit is replaced by the dynamic value which varies on each transmission. The simulation is done in NS2 under different network scenarios having different no of stations ranging from 10 to 50. The result from the simulation shows that there is slight improvement in the network performance. Hence this dissertation concludes that the dynamic TXOP value increases the network performance.

7.2 Future Work

Some of the recommendations has been made for future work. All the applications of same traffic type CBR have been used. In future different applications with different traffic such as Pareto and Exponential can be used. To get the more optimum result, specific environment should be defined. On that specific environment different applications with their priority can be figure out. Performance of the whole network has been evaluated in this dissertation but the performance of each application can be done in future. More complex formula can be drawn to get better result. In this work, only TXOP limit has been changed, on changing the CW and AIFSN better results might be obtained.

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Appendix A

Priority.tcl File used for Default TXOP Limit Value

```
# 802.11b parameters (default EDCA parameter set), aCWmin=31,
aCWmax=1023
proc priority { ifq_name } {
   upvar $ifq_name ifq
        # parameters for Queue 0
    $ifq Prio 0 PF 2
    $ifq Prio 0 AIFS 2
    $ifq Prio 0 CW_MIN 7           ;# (aCWmin+1)/4 - 1
    $ifq Prio 0 CW_MAX 15          ;# (aCWmin+1)/2 - 1
    $ifq Prio 0 TXOPLimit 0.003264      #default value
        #parameters for Queue 1
    $ifq Prio 1 PF 2
    $ifq Prio 1 AIFS 2
    $ifq Prio 1 CW_MIN 15          ;# (aCWmin+1)/2 - 1
    $ifq Prio 1 CW_MAX 31          ;# aCWmin
    $ifq Prio 1 TXOPLimit 0.006016           #default value
       #parameters for Queue 2
    $ifq Prio 2 PF 2
    $ifq Prio 2 AIFS 3
    $ifq Prio 2 CW_MIN 31         ;# aCWmin
    $ifq Prio 2 CW_MAX 1023       ;# aCWmax
    $ifq Prio 2 TXOPLimit 0        #default value
       #parameters for Queue 3
    $ifq Prio 3 PF 2
    $ifq Prio 3 AIFS 7
    $ifq Prio 3 CW_MIN 31         ;# aCWmin
    $ifq Prio 3 CW_MAX 1023       ;# aCWmax
    $ifq Prio 3 TXOPLimit 0   #default value
}
```

Appendix B
Function added in mac802_11e.cc file to calculate dynamic TXOP limit value

```
double Mac802_11e::myTXOP(int i)            //i changed here: implimentation of the process
{
  double prevtxoplimit;
  double nowtxoplimit;
  double colcount;
  double rec_count;

  colcount = col_count[i];
  rec_count = recv_count[i];
  prevtxoplimit = prevTXOP[i];
  printf ("Packet received counted %d\n", rec_count);

        if(colcount>0)
        {
                if(colcount==rec_count)
                {
                        return -1;
                }
                else if(colcount>rec_count)
                {
                        nowtxoplimit=prevtxoplimit/2;
                        return nowtxoplimit;
                }
                else                {
                        nowtxoplimit=prevtxoplimit*2;
                        return nowtxoplimit;
                }
        }
```

```
        col_count[i] =0;                //i changed here:
                                        need to refresh this count
        recv_count[i] =0;               //i changed here: need to refresh this count
    prevTXOP[i] = nowtxoplimit;         //i changed here: remembering current TXOP
limit for later use
return nowtxoplimit;
}
```

Appendix C

Perl files (.pl) used for calculating packetloss, throughput and latency

1. Packetloss.pl

```
#!/usr/local/bin/perl

if (@ARGV < 2)
{
        print "Usage: packetloss.pl <trace file> <cbr//tcp>\n";
        exit;
}
$infile = $ARGV[0];
$kind = $ARGV[1];

$sum_sent = 0;
$num_dropped = 0;

open (DATA, "<$infile") || die "Can't open $infile $!";
while (<DATA>) {
   $line = $_;
   @x = split(' ');
   last if ($x[4] =~ /END/);
   $num_sent++  if ($x[0] eq 's' && $x[6] =~ /$kind/);
   $num_dropped++ if ($x[0] eq 'D' && $line =~ /IFQ/ && $x[6] =~ /$kind/);
 }

 $dropped_ratio = 100*$num_dropped/$num_sent;

 print STDOUT "Percentage of $kind packets that were dropped: ${dropped_ratio}%\n";
 close DATA;
 exit(0);
```

2. Throughput.pl

```
#!/usr/local/bin/perl
if (@ARGV < 3)
{
	print "Usage: throughput.pl <trace file> <cbr\\tcp> <num stations of this kind>\n";
	exit;
}
$infile = $ARGV[0];
$kind = $ARGV[1];
$num_stations = $ARGV[2];
$header = 20;

$sum = 0;
$clock = 0;
$initial_clock = -1;
$final_clock = -1;
open (DATA, "<$infile") || die "Can't open $infile $!";

while (<DATA>) {
   $line = $_;
   @x = split(' ');
   if ($initial_clock<0) {
	$initial_clock = $x[1];
   }
   $final_clock = $x[1];
   last if ($x[4] =~ /END/);
   if ($x[0] eq 'r' && $line =~ /AGT/ &&  $x[6]=~/$kind/)
   {
     $size = $x[7];
     $sum = $sum + $size-$header;
   }

 }
```

```
$delta_t = $final_clock - $initial_clock + 0.000001;
$throughput = $sum/$delta_t;
$throughput = $throughput/$num_stations;
print STDOUT "Average Throughput was: $throughput\n";
close DATA;
exit(0);
```

3. Latency.pl

```
#!/usr/local/bin/perl
#Explanation: The send time is when the packet leaves the Udp (Agent) layer and
#when the channel is busier the time it takes till the mac layer decides to transmit
#the packet is longer. (CW is higher at this stage) So latency gets longer.
if (@ARGV < 2)
{
        print "Usage: latency.pl <trace file> <cbr\\tcp>\n";
        exit;
}
$infile = $ARGV[0];
$kind = $ARGV[1];

$sum_latency = 0;
$num_counted = 0;
%packet_hash=();
open (DATA, "<$infile") || die "Can't open $infile";
while (<DATA>) {
   $line = $_;
   @x = split(' ');
   $id = $x[5];
   last if ($x[4] =~ /END/);
   next if ($line !~ /AGT/ || $x[6] !~ /$kind/);
   if ($x[0] eq 's')
   {
        $packet_hash{$id} = $x[1];
   }
```

```
    elsif ($x[0] eq 'r')
    {
          $latency = $x[1] - $packet_hash{$id};
        $sum_latency = $sum_latency + $latency;
        $num_counted++;
    }
}


$avg = $sum_latency/$num_counted;

print STDOUT "Average $kind latency was: $avg\n";
close DATA;
exit(0);
```