Efficient Two Windows MAC Algorithm in Wireless Ad-hoc Networks

XIWEI SHEN

A Thesis

In

The department

Of

Electrical and Computer Engineering

Presented in Partial Fulfillment of Requirements

for the Degree of Master of Applied Science at

CORCONDIA UNIVERSITY

Montreal, Quebec, Canada

March 2004-3-4

Your file Votre référence
ISBN: 0-612-91112-8
Our file Notre référence
ISBN: 0-612-91112-8

ABSTRACT

Efficient Two Windows MAC Algorithm in Wireless Ad-hoc Networks

Xiwei Shen

In this thesis, a Two Windows MAC Algorithm (TWMA) is proposed and extensive simulation studies compare the proposed medium access algorithm and IEEE 802.11 standard. TWMA is introduced to solve the fairness problem in medium access and improve the utilization of the channel i.e. throughput, traffic delay, and fairness of medium access algorithm mainly in wireless ad hoc networks using Carrier-Sense Multiple Access with Collision Avoidance (CSMA/CA). The media access control (MAC) protocol plays a critical role in providing fairness, efficiency, and robustness in wireless networks. IEEE 802.11 MAC using exponential random backoff algorithm can not solve such fairness problem, i.e. stations cannot gain fair access to the shared wireless medium, due to non-homogeneous traffic load distribution, location dependent contention for medium access, and lack of central administration in ad hoc networks. The proposed MAC algorithm uses a variable Channel Status Indicator (CSI) to represent adjacent traffic status and two windows contention algorithm in order to achieve better throughput, latency, and a degree of fairness. Simulation results show the performance characteristics and functionalities of the algorithm that includes throughput, latency, buffer overflow, and etc. The simulation results reveal that the proposed TWMA achieve higher throughput and lower latency than that of the IEEE 802.11 standards MAC algorithm.

Acknowledgements

I am very pleased to express my great gratitude and respect to my supervisor, Dr. Ahmed K. Elhakeem, for his valuable guidance, constant support, precious advice and comments during my research in the wireless fields. My research cannot be done without his encouragement and support.

I give my special thanks to my parents, my brothers, my wife and son yiming for their great loves and supports.

Table of Contents

Chapter 1 Introduction 1

1.1 Introduction to IEEE 802.11 Wireless LAN.. 1

1.2 IEEE 802.11 Network Structures.. 4

1.2.1 BSS.. 4

1.2.2 IBSS Networks (Wireless Ad hoc Network).. 5

1.2.3 ESS.. 6

1.2.4 DCF and PCF.. 7

1.3 IEEE 802.11 Physical Layer.. 9

1.3.1 Direct Sequence Spread Spectrum (DSSS).. 9

1.3.2 Orthogonal Frequency Division Multiplexing (OFDM).. 9

1.3.3 Frequency Hopping Spread Spectrum (FHSS).. 11

1.3.4 IR.. 11

1.4 IEEE 802.11 MAC Sublayer Protocol.. 12

1.4.1 Introduction.. 12

1.4.2 Fragmentation and Reassembly.. 13

1.4.3 IFS.. 15

1.4.4 Binary Slotted Exponential Back off Procedure.. 16

1.4.5 Handshake Mechanism.. 18

1.4.6 Fairness Problem Statement.. 22

Chapter 2 Related WLAN MAC Protocols..23

2.1 Introduction..23

2.2 MACAW (Multiple Access with Collision Avoidance for Wireless)..23

2.3 Balanced Media Access Methods..25

2.4 Generalized Resource Sharing Algorithm (GRS)..28

2.5 Fast Collision Resolution MAC Algorithm (FCR)..29

2.6 Dual Stages Contention Resolution (DSCR)..31

Chapter 3 Two Windows MAC Algorithm (TWMA)..34

3.1 Introduction..34

3.2 Description of TWMA..35

3.2.1 Detailed First and Second Contention Windows Procedures..38

3.3 Simulation Procedure of TWMA..41

3.3.1 Simulation Environment..41

3.3.2 Performance Criteria..45

3.3.3 TWMA Simulation Model and Control Rules..50

3.4 Performance Results..64

3.4.1 Average End-to-end Traffic Delay..64

3.4.1.1 Delay Comparison Between TWMA and 802.11 MAC..66

3.4.1.1.1 Delay Comparison for Different Buffer Size and Packet Generation Probability..66

3.4.1.1.2 Delay Distribution Comparison..73

3.4.2 Average End-to-end Queuing Delay..77

3.4.3 Average Buffer Overflow..78

3.4.4 Average Number of Dropped Packets..80

3.4.5 Average Traffic Throughput..82

3.4.5.1 Throughput Comparison 1..82

3.4.5.2 Throughput Comparison 2..89

Chapter 4 Conclusions and Future Consideration..95

4.1 Conclusion..95

4.2 Future Consideration..97

References..98

List of Figures

Figure 1.1 An Ad-hoc Network (Independent Basic Service Set).........................5

Figure 1.2 Handshake Mechanisms...6

Figure 1.3 ESS with a Wired Distribution System...6

Figure 1.4 MAC Architecture...7

Figure 1.5 Coexistence of PCF and DCF..8

Figure 1.6 Transmission under PCF...9

Figure 1.7 IEEE 802.11 MAC Frames Format..13

Figure 1.8 Segmentation and Reassembly...13

Figure 1.9 Transmission of Fragmented MPDU...14

Figure 1.10 Interframe Relationship..15

Figure 1.11 Example of Hidden Stations...17

Figure 1.12 IEEE 802.11 MAC Backoff Procedure...18

Figure 1.13 Transmission of MPDU using RTS/CTS..20

Figure 2.1 MACAW Backoff Scheme..25

Figure 2.2 Pij Persistent Algorithm..28

Figure 2.3 FCR Backoff Procedure...31

Figure 2.4 Simplified DSCR Flow Chart..33

Figure 3.1 Two Windows of TWMA...35

Figure 3.2 Two Contention Window Medium Access..39

Figure 3.3 An Example of Routing Pattern...43

Figure 3.4 TWMA Simulation Flow Diagrams..51

Figure 3.5 MAC control rules 1-Packet Generation and Distance Check....................53

Figure 3.6 MAC control rules 2-Routing Procedure..54

Figure 3.7 MAC control rules 3-States Transitions...55

Figure 3.8 MAC Control Rules 4- Wait for Contention to Backoff Procedure.............56

Figure 3.9 MAC control rules 5-Backoff Contention Procedure.................................57

Figure 3.10 MAC control rules 6-Packet Transmission and Receiving.......................58

Figure 3.11 MAC control rules 7-States Transitions..59

Figure 3.12 MAC control rules 8- States Transitions..60

Figure 3.13 MAC control rules 10-States Transitions..61

Figure 3.14 MAC control rules 9-ACK Check Procedure...62

Figure 3.15 Average Traffic Delay E[D] Versus Probability of Packet Generation Pg....65

Figure 3.16 Traffic Delay Variance δ^2_{DT} Versus Pg Probability of Packet Generation....66

Figure 3.17 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Pg = 0.1..67

Figure 3.18 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Pg = 0.2..67

Figure 3.19 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Pg = 0.3..68

Figure 3.20 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Pg = 0.4..68

Figure 3.21 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Pg = 0.5..68

Figure 3.22 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Pg = 0.6..69

Figure 3.23 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Pg = 0.7..69

Figure 3.24 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Pg = 0.8..69

Figure 3.25 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Pg = 0.9..70

Figure 3.26 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Pg = 1...70

Figure 3.27 Average End-to-end Traffic Delay Comparison at Buffer Size =4...........71

Figure 3.28 Average End-to-end Traffic Delay Comparison at Buffer Size =8...........71

Figure 3.29 Average End-to-end Traffic Delay Comparison at Buffer Size =16..........71

Figure 3.30 Average End-to-end Traffic Delay Comparison at Buffer Size =32..........72

Figure 3.31 Average End-to-end Traffic Delay Comparison at Buffer Size =64..........72

Figure 3.32 Average End-to-end Traffic Delay Comparison at Buffer Size =128.........72

Figure 3.33 Average End-to-end Traffic Delay Comparison at Buffer Size =256.........73

Figure 3.34 Distribution of Delay at Pg=0.1..74

Figure 3.35 Distribution of Delay at Pg=0.2..74

Figure 3.36 Distribution of Delay at Pg=0.3..74

Figure 3.37 Distribution of Delay at Pg=0.4..75

Figure 3.38 Distribution of Delay at Pg=0.5..75

Figure 3.39 Distribution of Delay at Pg=0.6....75

Figure 3.40 Distribution of Delay at Pg=0.7....76

Figure 3.41 Distribution of Delay at Pg=0.8....76

Figure 3.42 Distribution of Delay at Pg=0.9....76

Figure 3.43 Distribution of Delay at Pg=1....77

Figure 3.44 Average End-to-end Queuing Delay E[QD] Versus Probability of Packet Generation Pg....78

Figure 3.45 Queuing Delay Variance δ_{DQ}^2 Versus Probability of Packet Generation Pg....78

Figure 3.46 Average Buffer Overflow E[BO] VS Probability of Packet Generation Pg....79

Figure 3.47 Buffer Overflow Variance δ_{BO}^2 Versus Probability of Packet Generation Pg....80

Figure 3.48 Number of Dropped Packet Over Total Packet Number Versus Probability of Packet Generation Pg....81

Figure 3.49 Dropped packets variance δ_{DP}^2 Versus Probability of Packet Generation Pg....81

Figure 3.50 Average Traffic Throughput E[TH] Versus Probability Of Packet Generation Pg....84

Figure 3.51 Average Traffic Throughput Variance δ_{η}^2 Versus Probability of Packet Generation Pg....85

Figure 3.52 Throughput Comparison between TWMA and IEEE 802.11 MAC with

Different Pg in the range 0.1 to 1 at Buffer Size is 4..85
Figure 3.53 Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Pg in the range 0.1 to 1 at Buffer Size is 8..86
Figure 3.54 Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Pg in the range 0.1 to 1 at Buffer Size is16..86
Figure 3.55 Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Pg in the range 0.1 to 1 at Buffer Size is 32..87
Figure 3.56 Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Pg in the range 0.1 to 1 at Buffer Size is 64..87
Figure 3.57 Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Pg in the range 0.1 to 1 at Buffer Size is 128..88
Figure 3.58 Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Pg in the range 0.1 to 1 at Buffer Size is 256..88
Figure 3.59 Average Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Buffer Size in the range 4 to 256 at Pg=0.1..89
Figure 3.60 Average Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Buffer Size in the range 4 to 256 at Pg=0.2..90
Figure 3.61 Average Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Buffer Size in the range 4 to 256 at Pg=0.3..90
Figure 3.62 Average Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Buffer Size in the range 4 to 256 at Pg=0.4..91
Figure 3.63 Average Throughput Comparison between TWMA and IEEE 802.11 MAC

with Different Buffer Size in the range 4 to 256 at Pg=0.5....................................91

Figure 3.64 Average Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Buffer Size in the range 4 to 256 at Pg=0.6....................................92

Figure 3.65 Average Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Buffer Size in the range 4 to 256 at Pg=0.7....................................92

Figure 3.66 Average Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Buffer Size in the range 4 to 256 at Pg=0.8....................................93

Figure 3.67 Average Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Buffer Size in the range 4 to 256 at Pg=0.9....................................93

Figure 3.68 Average Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Buffer Size in the range 4 to 256 at Pg=1....................................94

Lists of Abbreviations

ACK	Acknowledgement (frame)
AP	Access Point
BSA	Basic Service Area
CCK	Complementary code keying
CSMA/CA	Carrier-sense multiple access with collision avoidance
CSMA/CD	Carrier-sense multiple access with collision detection
CTS	Clear to send
CW	Contend window
DBPSK	Differential Binary Phase Shift Keying
DCF	Distributed coordination function
DIFS	Distributed IFS
DQPSK	The 2Mbps is done using Differential Quadrature Phase Shift Keying
DSSS	Direct sequence spread spectrum
ESS	(802.11) Extended service set
FCR	Fast Collision Resolution MAC algorithm
FHSS	Frequency hopping spread spectrum
GFSK	Gaussian Frequency Shift Keying
GRS	Generalized Resource Sharing algorithm
IBSS	(802.11) Independent basic service set
IEEE	Institute of electrical and electronics engineers
IFS	(802.11) Interframe space

ISM	Industrial, scientific, and medical
LAN	Local area network
LLC	Logic Link Control
MAC	Medium access control
MACAW	Multiple Access with Collision Avoidance for Wireless
MPDU	Medium access control protocol data unit
MSDU	Medium access control service data unit
NAV	(802.11) Network allocation vector
PCF	(802.11) Point coordination function
PDU	Protocol data unit
PHY	Physical layer
PIFS	Priority Interframe space
PPM	Pulse Position Modulation
RTS	Ready to send
SIFS	Smallest IFS
TWMA	Two Windows MAC Algorithm
WLAN	Wireless local area network
WMAC	Wireless medium access control

Chapter 1 Introduction

1.1 Introduction to IEEE 802.11 Wireless LAN

Wireless LAN is a network that can connect portable computing devices over radio or infrared wireless links in an area where the wired network is not convenient to be deployed. With the rapid development of wireless technology and prosperity of the Internet, the needs for portable and mobile computer or equipment increase dramatically. Wireless LANs have rapidly become a significant hot spot again in the communication market recently due to the fact of emergence of large amount of various mobile devices. IEEE 802.11 [11] wireless LAN standard developed by the IEEE 802.11 committee is to cover wireless networks for portable, moving or fixed stations. IEEE 802.11 standard mainly describes the demand for wireless connectivity to automatic machinery and equipment or stations that need rapid deployment and mobility which may be portable, handheld, or mounted on moving vehicles within a local area. IEEE 802.11 standard applies functionally to either totally wireless ad hoc networks or connected to wired infrastructure networks through access points. The IEEE 802.11 WLAN standard concentrates on the MAC and physical layers for the Access-Point based infrastructure and ad hoc networks. The MAC and physical layer specifications define protocols required to support networking in local areas. Because of the scarce media shared by all stations involved in a certain area, MAC protocol is crucial for multiple access in wireless networks. Lots of protocols such as CSMA, polling, and TDMA have been proposed. However, MAC protocols such as Carrier Sense Multiple Access/ Collision

Detection (CSMA/CD) designed for wired networks cannot be used in wireless networks because wireless transceivers are normally half-duplex and not able to detect collisions. The most popular Carrier Sense Multiple Access/Collide Avoidance (CSMA/CA) became the basis for the IEEE 802.11 MAC protocol. The IEEE 802.11 standard employs CSMA/CA scheme to control access to transmission medium. CSMA/CA first checks if the transmission medium is clear before transmission. Carrier sensing medium access can be applied on both physical and MAC layers. On the physical layer, carrier sensing means that the stations detect signal activity in the shared channel, whereas, on the MAC layer, virtual carrier sensing means network allocation vector (NAV) setting procedure that is used to reserve the channel for transmission period from interference. The value on the duration field (see Figure 1.7) of frames updates other stations' NAV to the latest reserved length of ongoing transmission. IEEE 802.11 standard then uses slotted binary exponential random backoff procedure, RTS/CTS (Request To Send/Clear To Send) interacting scheme, and positive acknowledgement to further reduce collisions and improve reliability [2].

However, wireless LANs have suffered some disadvantages of limited resources such as narrow bandwidth, which result in low data rates, interference etc. It is safe to say that the limited bandwidth is one of the main disadvantages of wireless networks compared to wired networks. Due to the interferences among wireless LANs, it is not practical to increase the bandwidth by placing a second LAN alongside as in wired networks. Therefore, the bandwidths of wireless LANs offered in the current market are lower than those of wired LAN. Also because of the limited channel bandwidth, multiple users in a wireless LAN have to share a common channel, which makes it important to develop a protocol for wireless networks in order to fully utilize the scarce media efficiently. Fairness issues will soon become noticeable issue since IEEE 802.11 standard has been deployed in case of WLAN where multiple users are not able to contend for accessing a shared channel equally or fairly according to the current IEEE 802.11 MAC protocols [3]. This so called fairness problem has seen much attention lately. Obviously, the MAC layer has been playing a critical role in the fairness and robustness problems of wireless LAN. The most popular MAC sub layer functions that IEEE 802.11 deploys are the distributed coordination function (DCF) and the point coordination function (PCF) [16]. DCF is a protocol using distributed random access that uses random backoff time to resolve the channel contention whereas PCF uses a polling scheme that needs the coordination of access point (AP) to determine which user has right to access the shared channel. DCF and PCF can share the same medium bandwidth in a time-multiplexed manner, see Figure 1.5 and 1.6. In this thesis, we mainly focus on the issue of the fairness problems associated with the deployment of DCF in ad hoc networks for users in contending to access the shared channel. A new contention algorithm called Two Windows MAC Algorithm (TWMA) is proposed and the utilization of the shared channel for both TWMA and IEEE 802.11 standard are also investigated and evaluated. IEEE 802.11 MAC increases or decreases a station's contention window size using binary exponential backoff process after its failed or successful transmission to avoid collisions and reduce idle slots i.e. increase utilization of the shared channel. However, the waste of backoff procedure, idle, and collision states reduces the efficiency of channel utilization in IEEE

802.11 MAC. The proposed MAC algorithm tries to improve the fairness, throughput, and latency performance by modifying the IEEE 802.11 MAC backoff algorithm. The performances of IEEE 802.11 protocol enhanced with the proposed TWMA algorithm are studied and compared through simulation. The thesis explores some of the basic performances and features inherent in wireless MAC as well.

1.2 IEEE 802.11 Wireless LAN Structures

Wireless LANs have two configurations: infrastructure and ad-hoc wireless LANs. In infrastructure wireless LANs, wireless LANs set up the communication between stations under the control of centralized access point (AP) coordination with the help of an infrastructure like wired networks. The ad-hoc wireless networks establish peer-to-peer communication with each other independently within the communication reach. To reach distant stations beyond the radio transmission range, it needs the routing help of hops between them. The IEEE 802.11 topology consists of two types of network structures: ad hoc network and infrastructure. Basic Service Set (BSS) is a fundamental unit of IEEE 802.11 architecture. A BSS can be defined as a group of stations that are located in the same geographical area and under the direct control of single coordination function: DCF for contention frame transfer or optional PCF for contention free frame transfer.

1.2.1 Basic Service Set (BSS)

BSS is a network of multiple stations under the control of a DCF or a PCF. BSS consists of two types of networks: Independent Basic Service Set (IBSS) and Extended Service Set (ESS) networks. In a BSS, stations and access points can either work in contention mode exclusively using DCF, or in contention free mode using PCF.

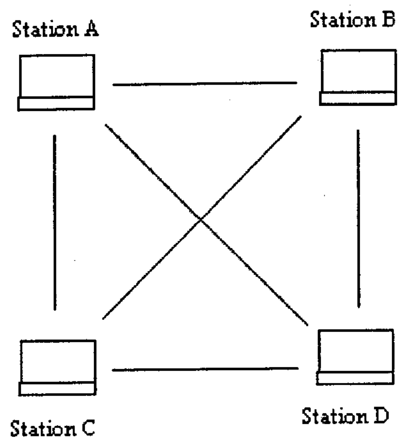

Figure1.1 An Ad-hoc Network (Independent Basic Service Set)

1.2.2 IBSS Networks (Wireless Ad hoc Network)

Ad-hoc network is defined as a network where a station within a local area can communicate directly with any other stations in the same coverage area under the control of a DCF by deploying IEEE 802.11 standard without the help of any infrastructure network, i.e. without the need to channeling all traffic through a centralized access point. The geographical area covered is called Basic Service Area (BSA). IEEE 802.11 standard uses four-way handshake (RTS, CTS, data, ACK) to ensure the reservation of the shared media and reliable transfer of the data packet. RTS/CTS pair exchange is to reserve the channel to prevent stations from collisions caused by potential hidden stations. ACK is used to make sure that the data packet is received correctly. See Figure 1.2 Handshake Mechanism.

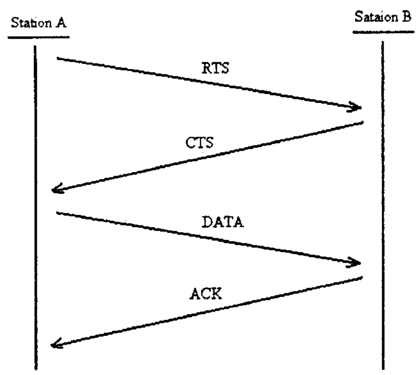

Figure 1.2 Handshake Mechanisms

1.2.3 Extended Service Set (ESS)

ESS is defined as a network where the stations in a local area can extend their range of services to other wireless networks and wired networks in different areas through Access Point bridges i.e. with the help of infrastructure networks. Access point provides a bridge function that connects multiple BSS networks. DCF and PCF can work together in infrastructure networks, however DCF can also work alone.

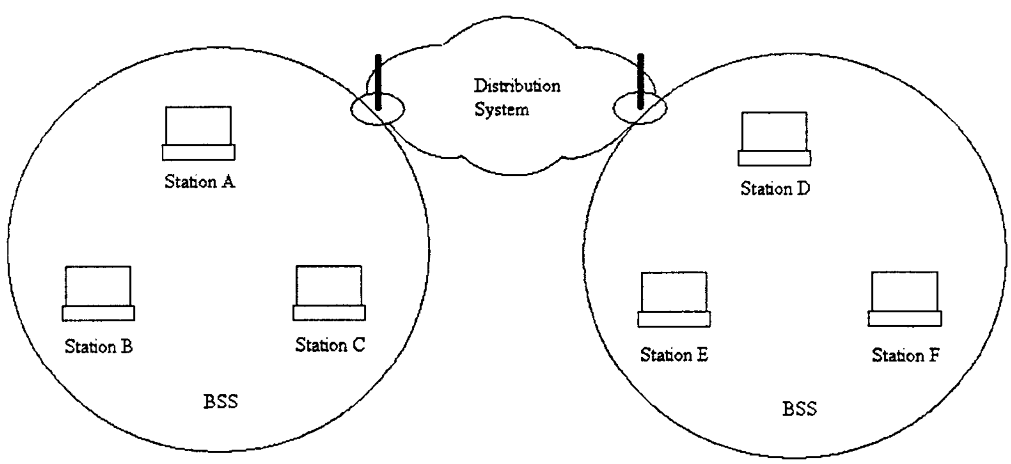

Figure 1.3 ESS with a Wired Distribution System

1.2.4 DCF and PCF

IEEE 802.11 provides two classes of MAC Service Data Unit (MSDU) delivery service: one is for an asynchronous data delivery such as email and File Transfer Protocol (ftp) using Distributed Coordination Function (DCF), another is for a synchronous, time-bounded data delivery such as realtime voice and video data using Point Coordination Function (PCF).

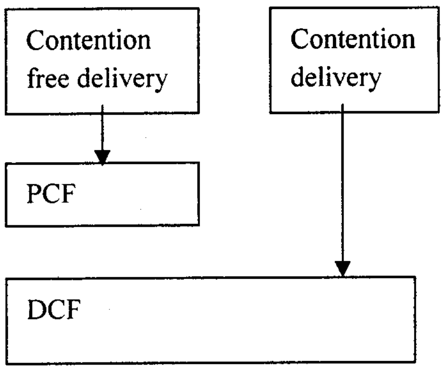

Figure1.4 MAC Architecture

DCF is the fundamental access method used to support asynchronous data transfer on a best effort basis. DCF has no mechanism to guarantee a fixed delay to support time critical data services. Whereas, PCF uses a Point Coordinator (PC) to perform polling, thus enables the polled station to transmit without contention for channel access. All stations must support DCF as specified in IEEE 802.11 standard. In fact, most traffics use DCF, DCF provides contention-based service that allows multiple independent stations to communicate with each other without the help of central control, so it can be used in either ad hoc networks or infrastructure networks. In ad hoc networks, DCF is the only choice. In infrastructure networks, DCF can work exclusively or with PCF. In the contention mode, stations contend for use of the shared channel for data transfer. In the contention free mode, the medium usage is controlled by the access point, which polls stations to access the channel. Therefore, there is no need for stations to contend for channel access under PCF. The channel can be alternated between contention mode and contention free mode respectively or the channel uses contention mode exclusively. As illustrated in Figure 1.5 and 1.6, in the coexistence of DCF and PCF, a certain portion of time is allotted to contention free traffic and the remainder is for contention traffic.

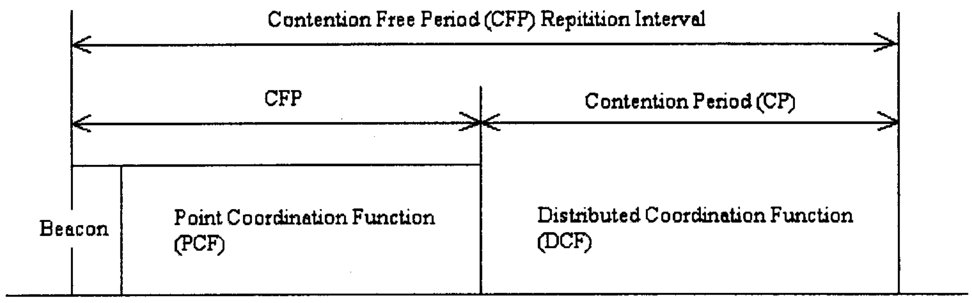

Figure 1. 5 Coexistence of PCF and DCF

DCF and PCF are distinguished by applying different interframe spaces (IFS). Accessing the medium under control of PCF has higher priority than that of DCF because PCF has shorter PCF Interframe Space (PIFS) in a network with Access Point (AP) than DCF Interframe Space (DIFS) used by DCF as depicted in Figure 1.9. In Contention Free Period (CFP), if the shared channel is idle for a PIFS interval, the AP sends a beacon to initiate the start of CFP. The data, polling, and ACK can be combined together as one frame to be sent as illustrated in Figure 1.6. This combination of frames is to improve efficiency of channel utilization. After Point Coordinator (PC) issues CF-End frame to terminate CFP, Contention Period (CP) may start.

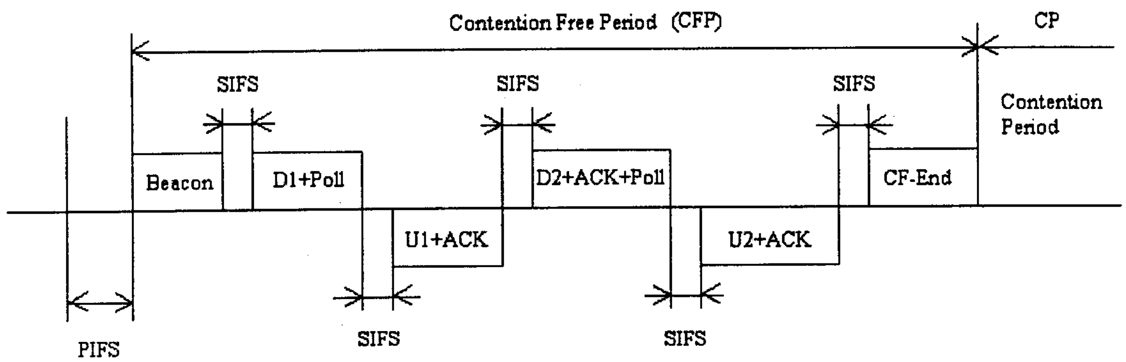

Figure 1.6 Transmission under PCF

This thesis solely focuses on the MAC aspects of ad hoc networks under the control of DCF where transmission medium solely operates in the contention-based mode.

1.3 IEEE 802.11 Physical Layer [2], [11]

IEEE 802.11 and 802.11a, 802.11b standards specify the following implementations of physical layers: Direct Sequence Spread Spectrum (DSSS), Orthogonal Frequency Division Multiplexing (OFDM), Frequency Hopping Spread Spectrum (FHSS), and IR (Infrared).

1.3.1 Direct Sequence Spread Spectrum (DSSS)

DSSS operates in 2.4GHz ISM band. DSSS signal symbol is spread with a sequence in wide bandwidth so that it has less power density. Basic DSSS has data rates of 1 and 2 Mbps. The basic rate 1 Mbps is encoded using Differential Binary Phase Shift Keying

(DBPSK). The 2Mbps is done using Differential Quadrature Phase Shift Keying (DQPSK). To realize spectrum spreading, the total bandwidth is divided into 11 sub channels of each 11MHz wide and 11 chips Barker sequence is used to spread each data symbol. The overlapping and adjacent BSSs can be accommodated by ensuring that the central frequencies of BSSs are separated by at least 30 MHz. This rigid requirement will enable only two overlapping or adjacent BSSs to operate without interference [2]. The IEEE 802.11b uses CCK scheme that is based on the same chip rate of DSSS channelization scheme to extend IEEE 802.11 DSSS by giving a higher rates of 5.5 and 11 Mbps.

1.3.2 Orthogonal Frequency Division Multiplexing (OFDM)

IEEE 802.11a standard has been developed to extend the IEEE 802.11 standard in the 5 GHz band. In the 5-GHz band, OFDM modulation schemes are used to reach data rates ranging from 6Mbps up to 54Mbps. The main idea of OFDM is to spread a high rate data stream over a number of low rate streams and those low rate streams are transmitted through a number of subcarriers. OFDM uses inverse fast Fourier transform (IFFT) to generate the sum of a large number of subcarriers and make sure to maintain the orthogonality between the different subcarriers at the receiver side. BPSK, QPSK, 16QAM, 64QAM are available for OFDM's modulation and data rate can go up to 54Mbps. The guard interval is a key parameter to choose to reduce the intersymbol and intercarrier interference. With variable coding rates and different error corrections together, it makes the modulation robust enough to be used in any indoor and outdoor environment where directional antennas may be used. IEEE 802.11a physical layer still uses the existing IEEE 802.11 MAC protocol.

1.3.3 Frequency Hopping Spread Spectrum (FHSS)

IEEE 802.11 FHSS operates in 2.4GHz unlicensed ISM band with bandwidth of 1MHz. It uses 79 non-overlapping hopping frequency channels with 1MHz channel spacing. The first channel centers at 2.4GHz and other subsequent channels space 1 MHz apart. Three different hopping sequence sets are designed with 26 hopping sequences per set. The choice of a certain channel is done through a pseudo random hopping pattern. Therefore, it enables the coexistence of multiple BSS up to 26 BSS networks in the same geographical area. As a result, maximum throughput is achieved and congestion is reduced significantly. The minimum hop rate is 2.5 hops/s. The basic rate is at least 1Mbps using 2 levels Gaussian Frequency Shift Keying (GFSK), where a logical 1 is encoded as frequency F_c+f and 0 is F_c-f. The enhanced 2Mbps access rate uses 4 level GFSK, 4 frequencies are used to encode 2 bit at a time. FHSS is also a kind of resistance to multipath fading through the inherent frequency diversity mechanism.

1.3.4 IR (Infrared)

Diffuse Infrared is designed for no-directed transmissions for indoor use. It operates in wavelength ranging from 850 to 950 nm. The IR specification was designed to enable stations to receive line-of-site and reflected transmissions [2]. It has a basic access rate of 1Mbps or enhanced access 2Mbps according to the deployed Pulse Position Modulation (PPM). 1Mbps access rate uses 16PPM, whereas, 2Mbps uses 4PPM [1].

1.4 IEEE 802.11 MAC Sublayer Protocol

1.4.1 Introduction

The basic medium access functionality of IEEE 802.11 MAC layer protocol provides asynchronous, time-bounded, and contention or contention free access control on a variety of physical layers through the use of the Carrier Sense Multiple Access/Collide Avoidance (CSMA/CA) protocol, a fundamental method of DCF with acknowledgment (ACK) and a random back-off time procedure following a busy medium condition. In addition to DCF, IEEE 802.11 standard also employs an optional PCF as an alternative access method as described earlier. IEEE 802.11 standard provides MAC and PHY specification for wireless connectivity to fixed, portable or moving stations within a local area. Figure 1. 7 illustrates the frame format of IEEE 802.11 standard. Besides the data frame body (MSDU) and 4 octets frame check sequence (FCS) for error checking, 6 octets MAC address are used to identify source and destination stations. The two octets for duration ID field indicate the time (NAV value) the channel is reserved for successful transmission of an MPDU. The two octets frame control field indicates types of frame as for control (RTS, CTS, and ACK), data or management. The control and data frames work together to ensure reliable delivery of data. Management frames perform supervisory functions that are used to join and leave WLAN and move association from access point to access point. MAC has functions of channel allocation procedures, Protocol Data Unit (PDU) addressing, frame formatting, error checking, and fragmentation & reassembly [2]. It supports multiple physical layers, power management, security (including registration and authentication) and association. The purpose of IEEE 802.11 CSMA/CA is to solve the problem of collisions or reduce the probability of collision between multiple stations while they access the medium.

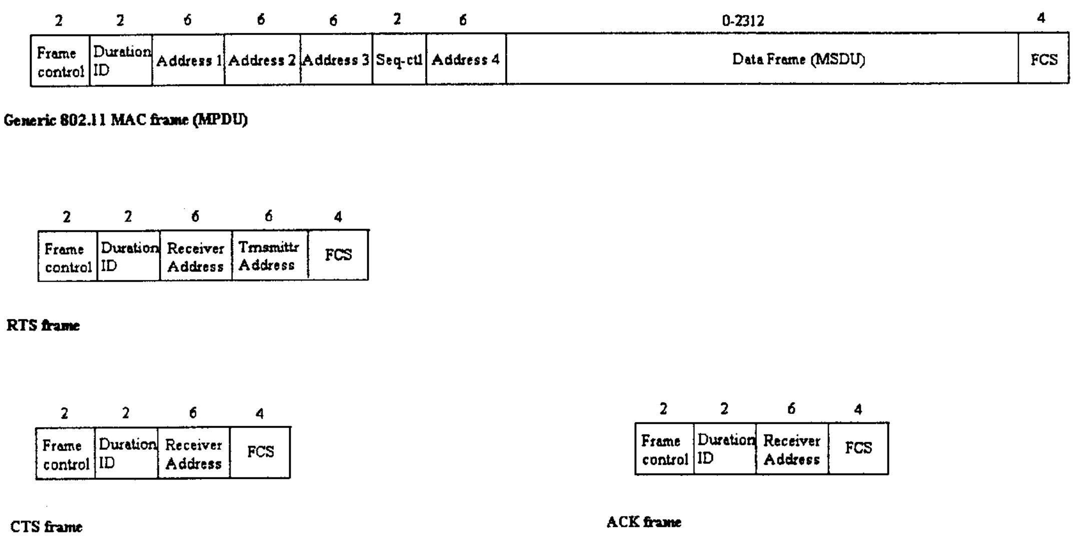

Figure 1. 7 IEEE 802.11 MAC Frames Format

1.4.2 Fragmentation and Reassembly

For the reliability of transmission, a large Medium access control Service Data Units

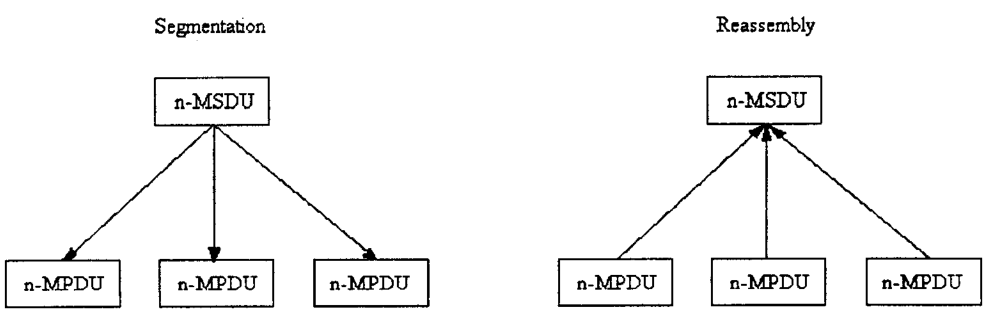

Figure 1.8 Segmentation and Reassembly

(MSDU) from Logic Link Control (LLC) needs to be fragmented into multiple fragments if it exceeds the fragmentation threshold value. Figure 1.8 and 1.9 illustrate segmentation and reassembly of MSDU and transmission of fragmented MPDU respectively. It is obvious that when an uncorrectable error in a large MSDU waste more bandwidth and transmission time than an error in a smaller fragment. Therefore, fragmentation is helpful in a noisy environment but can cause overhead in a good medium condition. All the fragments of MSDU are transmitted sequentially with ACK back to the sending station after each successful transmission of MPDU. If the acknowledgement is not received for its corresponding fragment by waiting only SIFS period, the sending station will stop the transmission and contend for channel access again for the unacknowledged fragment. RTS/CTS exchanges are only used before the transmission of the first fragment and no longer used during the rest of transmissions of sequenced fragments. The sequenced fragments have the duration information (duration field) needed for NAV updating of other stations in the same BSS.

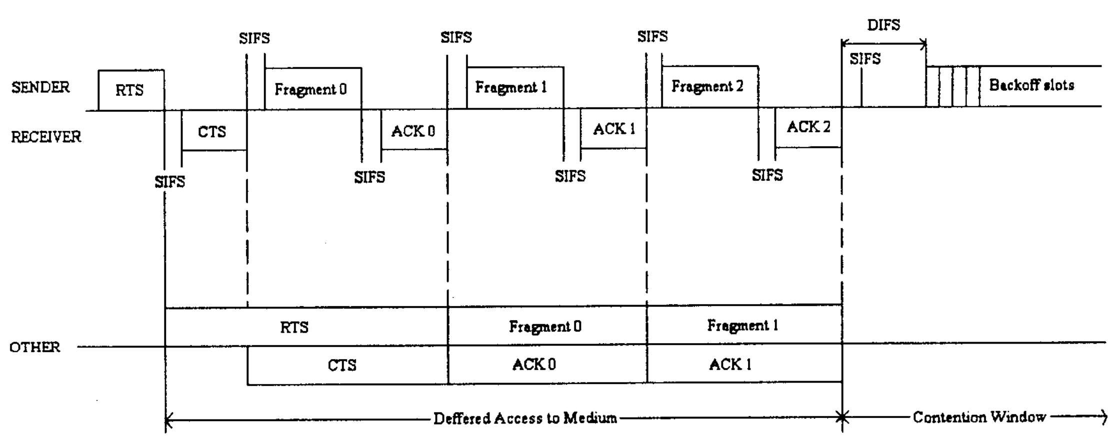

Figure 1. 9 Transmission of Fragmented MPDU

When all the MPDUs are received successfully, the transmission channel will be released. The destination station is responsible for reassembling the fragments to its original MSDU according to their fragment numbers in sequence control fields.

1.4.3 Inter Frame Space (IFS)

Three mandatory idle periods IFS (InterFrame Spacing): smallest IFS (SIFS), priority IFS (PIFS), distributed IFS (DIFS) illustrated in Figure 1.9 between transmission of frames are used to control the priority of accessing the wireless transmission medium.

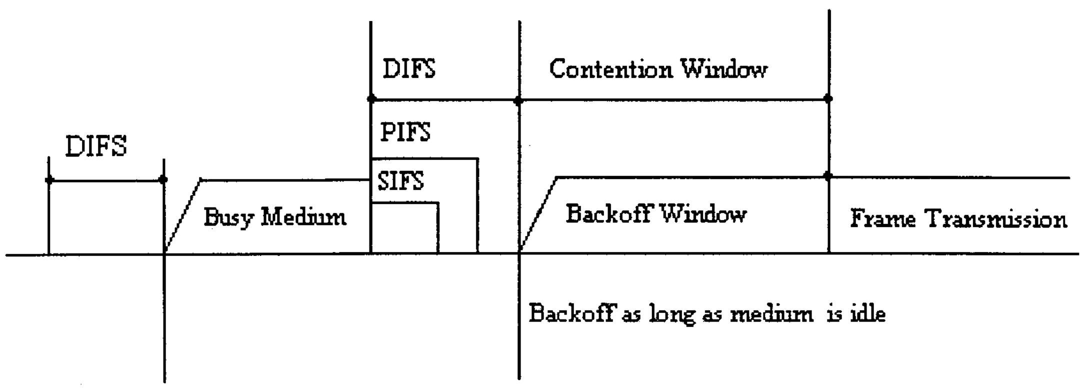

Figure1.10 Interframe Relationship

Different interframe spaces lead to different priority levels for different types of traffic. SIFS has highest priority to access the medium, followed by PIFS and DIFS respectively; SIFS is used by the control frames such as RTS/CTS, ACK, and data frames in multiple frames sequence. PIFS is used by PCF during contention free operation with priority over standard contention based service, so it may be used for time-bounded data frames. The station required to wait SIFS used by control packet has priority access to the medium over those required to wait PIFS or DIFS before transmission. Under DCF, the station starts transmission after the medium is sensed idle for DIFS, once the station gains access to the channel, it can have the channel for as long as needed by using SIFS and NAV reservation.

1.4.4 Binary Slotted Exponential Back off Procedure

When multiple stations wait for accessing the medium, the collisions will most likely happen because of hidden stations. Thus, a random backoff procedure is deployed to reduce the conflicts and collisions in the medium contention. Figure 1.12 illustrates flow chart of IEEE 802.11 MAC backoff procedure. Multiple stations have different random backoff numbers; these numbers refer to numbers of time slots. A station with the smallest random backoff number wins the contention and gains access to the transmission channel. A station that has packets to send needs to sense the channel status using CSMA/CA first until it finds the channel idle for a DIFS period, then it chooses its own random backoff time uniformly in the range of contention window interval [0,CW], where Contention Window (CW) is the size of contention window. If the medium is busy, they have to wait until the channel idles continuously for DIFS again. When a station senses that the channel is idle, the station starts to decrement its random backoff timer. The station decrements its backoff timer one by one solely after it senses that channel is idle until the random backoff timer reach zero. If the channel is busy before the backoff timer counts down to zero, the backoff timer will be frozen. The backoff timer is unfrozen and continues to decrement when the channel is sensed idle for more than DIFS period again. When the backoff timer eventually reach zero and at the same time the channel is idle, the station gains access to the channel and transmits its RTS control frame to its receiver. When more than one station's backoff timer reach zero i.e. more stations are ready to transmit packet simultaneously, a collision happens. Hidden stations shown in Figure 1.11 may cause a collision as well. Station A and C are hidden stations for each other, i.e. they cannot hear each other. For example, assume that station A starts to send RTS to B and ask all the stations in its cluster to defer their contention for channel access. However, station C cannot hear the RTS so it is possible that station C still transmits its packet to station B at the same time and then causes collision at the station B. At every first access attempt, $CW=CW_{min}$, and $CW_{min}=8$ in the simulation. The CW doubles its value until it reaches its maximum CW_{max} (0-1023 in DSSS, 0-255 in FHSS) for every reattempt to access channel after failed transmissions. When CW reaches CW_{max}, CW is reset to CW_{min} for next attempt to access the channel only after a successful transmission of a data packet.

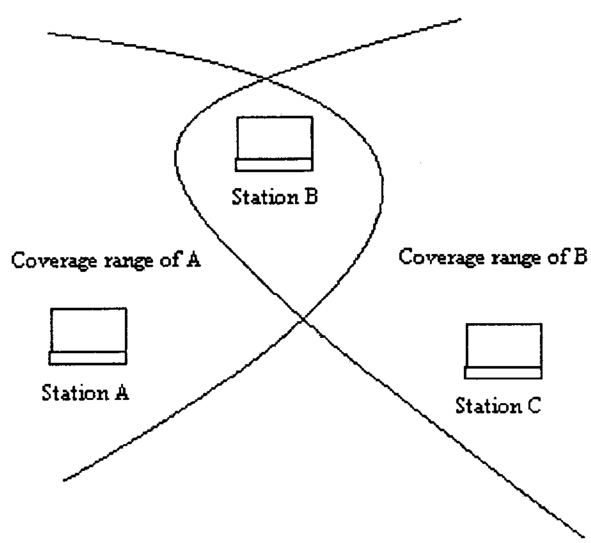

Figure 1.11Example of Hidden Stations

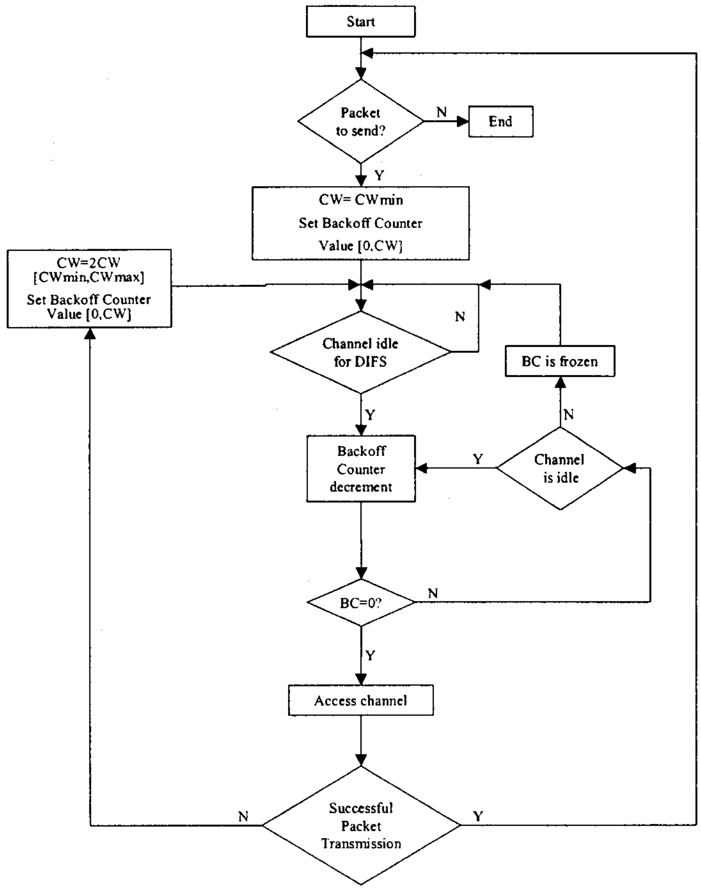

Figure 1.12 IEEE 802.11 MAC Backoff Procedure

1.4.5 Handshake Mechanism

IEEE 802.11 standard [16] deploys a pair of RTS/CTS exchange as an option when the data exceeds the value of RTS threshold. See Figure 1.2 for handshake mechanism depiction. RTS/CTS exchanges are used to make channel reservation, minimize the collision period, and deal with "hidden nodes" which is the most common reason causing collisions. Figure 1.11 shows an example of hidden nodes, station B is in transmission range of both A and C, while station A and C can not hear from each other, so A and C are hidden nodes for each other. Station A and C may send data packets simultaneously to station B, it leads to a collision on their common intended receiver. Unlike wired LAN, a source station cannot hear its own transmission when collision occurs, so CSMA with collision avoidance is used to solve the problem. In order to save bandwidth, in other words, to minimize the collision time, RTS/CTS handshake mechanism is deployed. RTS/CTS pair exchanges minimize the collision time period because the lengths of RTS/CTS frames are much shorter than the length of a packet of MPDU usually. If a collision happens, only a period of RTS/CTS is wasted instead of the whole MPDU. It saves bandwidth and minimizes the waste a lot especially when MPDU (maximum data frame size up to 2346 octets) is relatively much larger than RTS/CTS control frames. RTS and CTS are 20 and 14 octets respectively. Therefore, it is helpful to use RTS/CTS when MPDU exceeds the value of RTS threshold in highly loaded environment with overlapping networks though the usage of RTS/CTS may lead to extra delay caused by RTS/CTS overhead in a lightly loaded channel. Another control frame ACK is used to ensure reliability of the data packet delivery especially for unicast data transmission. In ad hoc networks, if the transmission medium operates in contention mode, all the stations that have packets to send shall have fair opportunities to contend for channel access for all frames. However, there is unfairness in fact existing in IEEE 802.11 standard MAC random backoff procedure that based on binary exponential backoff algorithm. The details of fairness problem will be discussed later. DCF uses CSMA/CA to find activity of transmission in channel and all the stations therefore can get the duration information of how long they must wait until the current transmission session is over if they have packets queued to send. See duration field in Figure 1.7. Whenever a station hears RTS, CTS, and data frame, it reads the duration information in the headers of the frames and updates its NAV value accordingly. NAV is used for virtual carrier sensing and is stored in the duration field of both data and control frames. NAV is actually a timer indicating the length of time that the transmission medium is reserved to complete the successful transmission of the data. When a station's NAV counts down to zero, the medium can be sampled again for idle status.

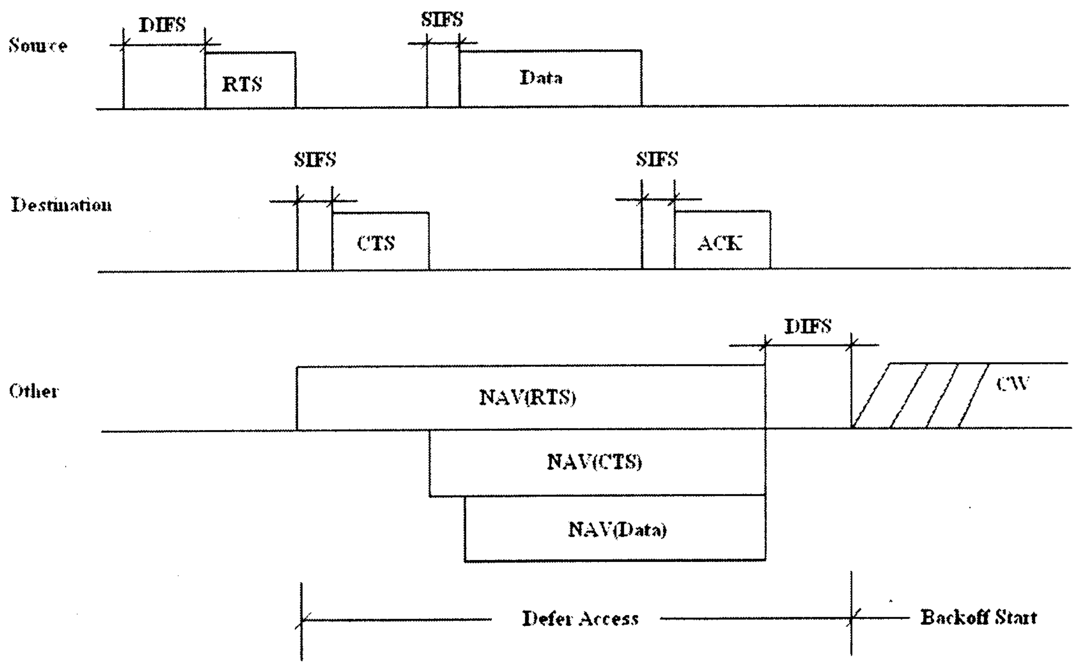

Figure 1.13 Transmission of MPDU using RTS/CTS

Figure1.13 shows the basic access method with RTS/CTS exchanges. As explained earlier, RTS/CTS pair interaction is used to clear out an area and furthermore to avoid collision caused by the "hidden nodes". When a station accesses the shared channel, it sends RTS control frame to its target receiver and other stations that hear the RTS in the same coverage area adjust their NAV value accordingly and maintain silence for NAV period. Then, when target receiver receives the RTS sent by the source station, it sends back CTS control frame when it is ready to receive and the channel is reserved for data packet duration. RTS and CTS tell other stations in both sender's and receiver's coverage areas to update stations' NAV values if they are required to reserve the medium longer than the current NAV values and defer their access to the medium until NAVs elapse. The setting of NAV in multiple frames sequences based on frame by frame. The RTS/CTS interacting procedure makes it possible for hidden nodes in the coverage areas of sending and receiving stations to maintain silent. As a result, RTS/CTS procedures ensure exclusive access to the medium, free from the collisions caused by hidden nodes. After the source station receives CTS successfully, the channel is reserved for the transmission of the whole protocol data unit (MPDU). The source sends its data frame and expects positive acknowledgement (ACK) from the receiver. The receiver checks the checksum of the packet and determines if the packet is correctly received. The receiver waits an SIFS interval and sends back ACK to the source station on the condition of successful reception of the packet. If the data transmission is not successful and the sending frame is considered lost, that means at least one part of transfer fails from the beginning of gaining access to the end of receiving of the ACK. The source station is responsible for contending for channel access again and resending its unacknowledged data until it is successful. If ACK is received successfully, the source may or may not contend for channel access depends on whether it has more data packets to send or not after the medium idles for more than DIFS. See Figure 1.12.

1.4.6 Fairness Problem

It is claimed that IEEE 802.11 standard provides fairness to all stations by giving equal probability to gain access to the wireless channel upon contention. Fairness means that all stations have an equal chance to gain access to the shared media. Actually, it is not so straightforward in IEEE 802.11 MAC protocol. Hidden stations in a scarce-shared media environment cause the fairness problem. The IEEE 802.11 standard may also results in "capture" effects meaning that some stations grab the shared channel and other stations suffer from starvation [3]. For example, assume that a station doubles its CW to maximum value 255(FHSS) slots time after a few failed transmissions, it starts again to compete for gaining access with a station that is just newly entering the same BSS with its initial CW=8 slots. From the former station's perspective, this is unfair because the probability of accessing the channel for the new comer is much higher than the old one due to the length difference between the two contention windows 255 and 8. There is a possibility that a station with a larger CW value is less likely to access the channel for a period of time when there are lot of stations with smaller CW values waiting to contend. Therefore, the binary exponential backoff mechanism favors the latest successful station over the others. The reason why the link layer fairness models used for wired networks cannot be employed for the shared media wireless networks is the unique characteristics of wireless media such as location-dependent contention, inherent conflict between optimizing the shared channel utilization and achieving fairness, non-homogeneous traffic distribution, and under no centralized control.

Chapter 2 Related Wireless LAN MAC Protocols

2.1 Introduction

CSMA is so important that many MAC protocols for wireless LAN have been proposed based on CSMA. For example, MACAW [4], Pij-persistent CSMAC [6], [8], GRS [9], FCR [10], and DSCR [17] related works have been done to deal with the fairness problem and provide robustness to the network. MACAW uses a different backoff algorithm that is called multiplicative increase and linear decrease with a backoff copy scheme to balance the resource allocation. Pij-persistent CSMAC employs a pre-calculated link access probability and adjustment of increasing and decreasing factors of contention windows size to deal with fairness access problem and balance the traffic load among stations. As [7] indicates that there are many unique characteristics of ad hoc network that prevent people from using wired network's fairness models to achieve fairness of multiple wireless media access. The specific properties are: 1. Spatial (location-dependent) contention for the wireless shared channel. 2. Compromise between channel utilization and fairness. 3. Inaccurate state of contention and decentralized control.

2.2 Multiple Accesses with Collision Avoidance for Wireless (MACAW)

MACAW [4] was first proposed to investigate fairness problem by V. Bharghavan in 1994. In order to increase throughput and alleviate fairness problem, MACAW introduced additional control frames and a different backoff algorithm that is called multiplicative increase and linear decrease with a backoff copy scheme. The backoff scheme did not work well in ad hoc network environment. The MACAW originally came from a MAC that uses RTS/CTS exchanges and binary exponential backoff algorithm. The latter became the foundation of IEEE 802.11 standard later on. Though, the packet exchange procedure is quite similar to that of IEEE 802.11 standard, MACAW used a significantly different backoff algorithm. In MACAW, two functions F_{inc} and F_{dec} are used to adjust the backoff counter range as illustrated in Figure 2.1. Retransmission occurs if and only if a station does not receive the CTS in response to its RTS. A station randomly chooses, with uniform distribution, an integer between 1 and backoff counter range. Once a CTS is received after an RTS, the backoff counter range is adjusted by F_{dec}. Whenever CTS is not received after an RTS, the backoff counter range is adjusted via F_{inc}. The MACAW uses a backoff copy scheme to improve fairness in contending for channel access. Backoff copy scheme uses a field in the header to contain the current value of backoff contention window (CW). Whenever a station hears a packet, it copies the value as its own backoff counter CW, thus, in the sender's vicinity every station has same backoff counter CW. Therefore, every station has same probability to contend for media access, as a result, to produce a fair allocation of the shared resource. To further improve the efficiency of the protocol, upon a collision, the CW is increased by a multiple factor 1.5; and upon success, it is decreased by one. However, in ad hoc networks, a single backoff number is not appropriate except when the congestion is uniform and homogeneous. For the reason that the backoff window size increases faster than it decreases, the CW size eventually increases to a very high value. When the copying scheme is used, eventually all the stations have high CW value no matter whether the vicinity channel contention is light or heavy. It is possible to use the scheme in single cell but not multiple cells. There maybe heavy contention in some cells and light contention in others. For the station in a light contention cell, it is a waste of time to use a high backoff CW value. Similarly, for the station in a heavy contention cell, using a low backoff counter value causes more collisions. Therefore using a single number to reflect the ambient congestion level for the channel is not practical and even made worse by the copying algorithm. The proposed MACAW simulation model was merely tried in single cell environment [4] where all stations are in range of each other without considering multiple cells, handoff, and routing, so it is far from realistic. Moreover, it was not tested in a whole ad hoc network environment.

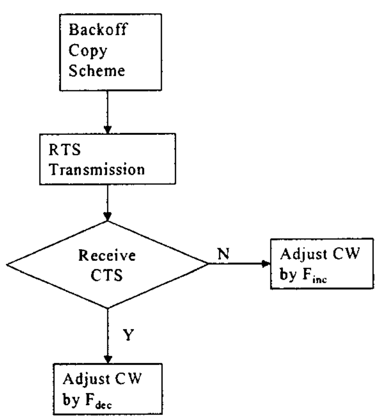

Figure 2.1 MACAW Backoff Scheme

2.3 Balanced Media Access Methods (Probability *p*-persistent Carrier Sense Media Access) [6], [8]

The P_{ij}-persistent carrier sense multiple access (CSMA) based algorithm was proposed to achieve fair wireless access by using a pre-calculated link access probability. Link access probabilities are calculated at the source station in two ways using connection-based and time-based media access methods. It is to say that the proposed methods are based on the exchange of information about the number of connections or the average contention period. For example, in the proposed algorithm [6], every station calculates its own link access probability P_{ij} from station i to station j in two ways, according to either the number of connections between itself and its neighbors, or the average contention period of its and other stations' individual links. When a backoff procedure ends, station i will start to send control frame RTS with probability P_{ij} or backoff again with probability of $1-P_{ij}$ using the same CW size, see Figure 2.2. As stated earlier, the link access probability P_{ij} is calculated at the source station in two methods either with a connection-based or a time-based media access method. To exchange the information, each active station broadcasts information of either the number of logical connections or the average contention time to the stations within the radio transmission range. Therefore link access probability changes and gets updated from time to time after the new information exchanges due to the changes in network topology. It is important to get the exchanging information of each station reflecting the status of the shared media as accurately as possible. However, it is not an easy task to exchange information accurately and timely. In this proposed method, each station reserves a specific priority for each of its links to gain access to the shared medium based on the exchanged information [6]. It uses window-exchange algorithm to find a minimum value for backoff CW. The transmitting or receiving station inserts the information of its last backoff window size into the RTS or CTS packet respectively. Any other station receiving RTS or CTS calculates its new backoff window using function *min*{current CW, received CW} to get a smaller CW.

In the connection-based method, the link access probabilities of stations are calculated based on the exchanged information of the number of connections between each node and neighboring stations in radio transmission range. The information is broadcast whenever stations realize changes in the network topology. A parameter of connection value is introduced in proportion to the priority of links. A higher priority is given to the link that has a higher connection value. The station with higher connection value means that it has more link connections.

As described earlier, link access probabilities are calculated based on the exchanged information of average contention period in the time-based method. An average contention period is a time interval between a packet's arrival at the MAC layer and transmission to the destination. The exchanged information is broadcast in a periodic basis. According to the contention time consumed by stations, a higher priority of link access is given to the station that is blocked and has longer contention period. A lower priority is given to a station that is dominant in the medium over the others and has a shorter contention period. A weight factor is used to control the increase pace of the link access probability according to the average contention period, and furthermore to balance the load among links.

It is claimed that the connection-based method is effective in a fully loaded system and time-based method is good in both fully loaded system and non-homogeneous load distributed system [8]. The performance of the time-based method is better when the network load differ from link to link. The results show neither always provides the best fair access in every scenario. In the [8] simulation, too few, only 6 stations are studied, which is far from reality. Some degree of fairness was achieved by using connection-based method with window exchange algorithm if all 6 links have a similar load. Therefore, whether this technique is effective or not in heavy traffic of large amount of stations in an ad-hoc network needs further investigation.

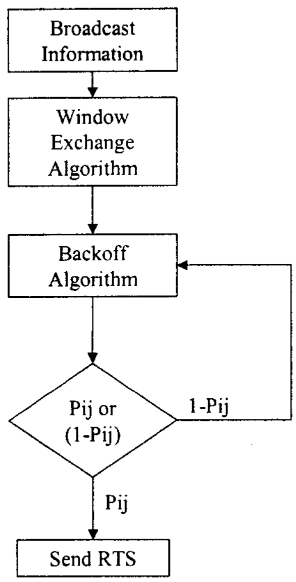

Figure2.2 Pij Persistent Algorithm

2.4 Generalized Resource Sharing algorithm (GRS)[9]

The proposed GRS algorithm is to provide fair distribution of bandwidth amongst multiple nodes in wireless networks including ad-hoc network. Various approaches are discussed to schedule packet transmissions by allocating the bandwidth in proportion to weights of flows. The protocol differs from IEEE 802.11 standard in the way in which the backoff interval is calculated and updated before and after collision of data packets. For the backoff interval calculating methods, two approaches of linear or non-linear increasing and decreasing backoff interval are suggested without giving further investigation. A readjustment to the backoff counter of long pending packet for a station with small weight is also suggested. In order to increase the probability of gaining access for a station that experienced a collision, IEEE 802.11 standard is modified like that: after a station senses a idle time for IFS period, it send a "resolution burst" for one time slot to signal the start of collision resolution phase. Any station that has not experienced the collision would then backoff and allows the colliding stations to contend for channel access. Collision resolutions can be repeated until all colliding stations gain access to the shared media. Two approaches defined to weight are evaluated: constant predefined weight of each flow and dynamically determined weight according to the recent demand of each flow, in other word, recent arrival rate of data on each flow.

2.5 Fast Collision Resolution MAC algorithm (FCR)[10]

This proposed algorithm is claimed to improve throughput performance of MAC by reducing the overheads such as packet collisions and idle slots in every contention cycle. It focuses on reducing the packet collisions and the wasted idle slots resulting from backoff procedure in each contention cycle. The retransmission collision rates increase when the number of active stations increases in the IEEE 802.11 MAC protocol. The algorithms base on the idea that small backoff timer is given to the station which has successfully transmitted a packet at current contention period to decrease the average number of idle slots for each contention period. On the contrary, large one is given to the station that is deferring its packet transmission at current contention period to reduce the probability of collision at subsequent contention period. The difference between FCR algorithm and IEEE 802.11 MAC is illustrated in Figure 2.3: in FCR, if a station senses the channel idle for a slot, it decreases its backoff time by one slot time. If a consecutive idle slot is sensed, the backoff timer decreases much faster by a half. If a station experiences a failure because of a collision, the CW size will be increased and a random backoff time will be chosen uniformly in the range of [0,CW]. If a station has a successful transmission of packet, it reduces its CW size to CW_{min}, same as IEEE 802.11 standard binary exponential backoff procedure. Therefore, in FCR, wasted idle slots are reduced rapidly according to its simulation results, this leads to a increasing of channel utilization to some extend. However, it is not helpful to the fairness among multiple media access and even exacerbates the situation. The reason is that the station which has a successful transmission previously will have a minimum CW size and a smaller backoff timer, hence, it will have a higher probability to gain access to the shared channel, whereas, the other stations with larger CW sizes and larger backoff timers will have lower probabilities to gain channel accesses. These scenarios cause a certain station to capture the shared channel for a long time where others experience the starvation for packets transfer. It is noted that often it is easy to achieve high throughput in a shared media by reducing the sharing with others, to the extreme, only one station exclusively captures the shared media would have a highest throughput.

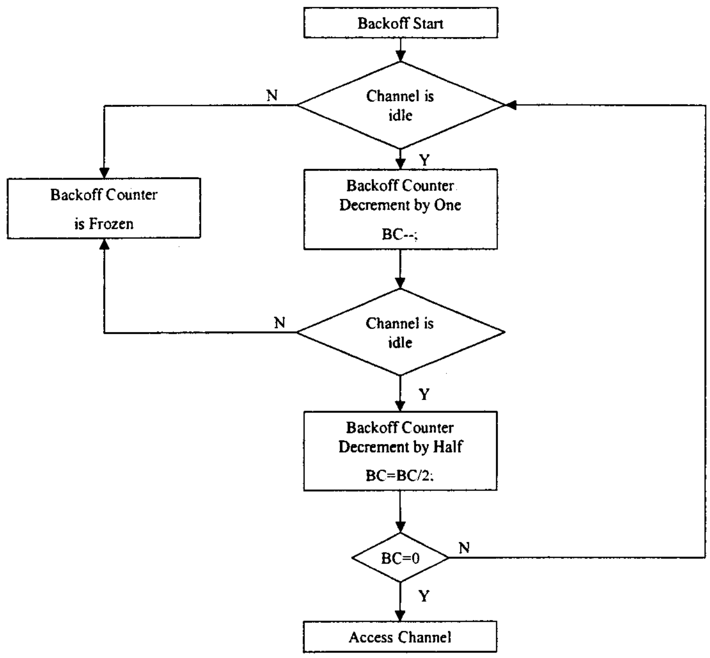

Figure 2.3 FCR Backoff Procedures

2.6 Dual Stages Contention Resolution (DSCR) [17]

Contention resolution of two "virtual" stages proposed in [17] tries to achieve better and more stable performance in channel utilization than IEEE 802.11standard DCF. In IEEE 802.11 standard, the CW is very important because a smaller contention window will reduce the idle channel time and achieve better utilization of channel bandwidth but may cause collision. On the other hand, a large contention window will be more likely to reduce the probability of collision resulting from heavy traffic, but sometimes wastes time in light traffic. It is suggested that an appropriate choice of contention window can optimize the performance of IEEE 802.11 standard. However, it is difficult to achieve the optimum contention window size due to the fact that there is no fixed network pattern like wired networks. The proposed algorithm tries to improve the efficiency of channel usage using two virtual stages contention resolution. As illustrated in Figure 2.4, the first stage is used to reduce the number of stations entering the second stage, whereas, only stations in the second stage have the rights to contend for channel access and then transmit their packets if they gain the channel. The number of stations in the second stage is kept relatively small compared to total number of active stations. Therefore, the MAC protocol can exhibit high stability in the sense that the throughput is less sensitive to the network size. It is reasonable to try to reduce the time that is wasted in idle or collision states to achieve a better channel utilization. It is argued in [17] that an optimum point for contention window size can be found to reduce the time wasted on idle and collision to minimum. The idea in [17] is to reduce the number of stations participating in contention for channel access because efficient resolution of contention among small number of stations is easy to achieve. DSCR is different from IEEE 802.11 DCF only in the contention resolution procedure and the other functions remain the same. Stations in both the first and second stages have the same backoff mechanism as in IEEE 802.11 standard DCF. The stations in the first stage have their own backoff counters. When they have packets to transmit, they start to decrement their backoff counter1 till they reach zero on the condition of idle channel. The first station that gets its backoff counter to zero enters the second stage and is set an initial $CW2_{min}$, which is minimum contention window size in the second stage, the rest will remain in the first stage with unchanged CW1. For all the stations entering the second stage, only one eventually gains the access of channel, whereas, the rest of stations with their contention windows doubled will return back to the first stage. Those stations with double CW1 will have low probabilities to enter the second stage again, furthermore reduce the number of stations in the second stage. The winning station will return to the first stage as well after it finishes its packet transmission successfully with its contention window CW1 reset to $CW1_{min}$, where $CW1_{min}$ is minimum contention window size, and its CW2 reducing to half. When there is a collision, collision stations will double their contention window CW2 and obtain new backoff counter values from [0,CW2]. It is obvious that a winning station returning from the second stage has more probability to enter the second stage again and then has more opportunity to gain access to the shared channel. Though DSCR increases throughput by giving more opportunity to a few stations, it is unfair for all stations in the network. As described earlier, it is easy to achieve high throughput when the shared channel is used solely by single station. Although, DSCR is further modified to increase the probabilities for stations that are hungry for entering the second stage by adopting non-linear backoff counter decrement, the result of performance is not effective.

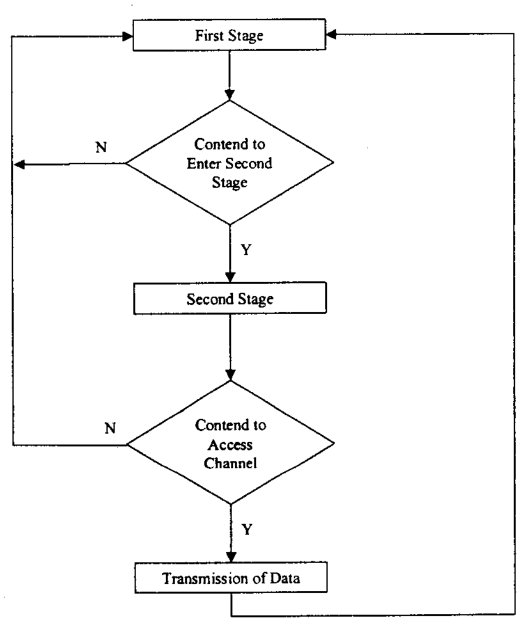

Figure 2.4 Simplified DSCR Flow Chart

The similarity between DSCR and TWMA is that they both have two windows or stages. The difference is: DSCR uses two stages for stations' contention for channel access, a station needs to win the contention in stage 1 in order to go into the stage 2. The stations in stage 2 then contend for channel access. Two stages are used to limit the number of stations contending for channel access. The random backoff algorithms in both stages are the same as that in IEEE 802.11 MAC.

TWMA uses two windows to give priority to the stations that has low probability to access the medium in the 802.11 standard random backoff procedure. The random backoff algorithm is modified. The changes of contention window are related to the traffic situation around.

CHAPTER 3. The Two Windows MAC Algorithm (TWMA)

3.1 Introduction

Due to the fact that stations in wireless LAN share the scarce resource, it is crucial for MAC to provide fairness and robustness to the wireless network traffic. Many MAC protocols related researches have been done and some improvements over IEEE 802.11 DCF backoff algorithm have been proposed as described in chapter 2. In this thesis, an alternative way to improve fairness, throughput, and delay performances is proposed and evaluated. As stated earlier, fairly allocating the shared resource cannot be achieved efficiently through IEEE 802.11 MAC partly because the binary exponential random backoff algorithm does not reflect the status or load situation of the shared media. For example, a station that is newly entering the WLAN with initial minimum contention window size (CW_{min}) has higher probability to win channel access in exponential backoff mechanism over a station that has experienced one or more failed transmissions with its doubled contention window (CW). This is the fairness problem. Another problem is that the random exponential backoff algorithm leads to inefficient utilization of the shared media. The shared channel bandwidth is wasted in idle slots and collisions resulting from the exponential backoff algorithm. Contention window variation merely depends on whether a station has a collision or not and has nothing to do with status of surrounding traffic situation. Stations may have the same contention window size no matter either in a heavy load or a light load environment, this cause much waste from idle slots. Therefore TWMA is proposed and tries to improve the throughput, fair utilization of shared media, and minimize delays emerging from idle slots and collisions during contending period in ad hoc networks.

3.2 Description of TWMA

Based on the description above, in order to solve or reduce the fairness problem and improve utilization of the shared channel, TWMA is introduced. TWMA uses two windows contending algorithm to balance the newly entering stations and old stations experiencing previous failures, i.e. gives priority to stations that have been waiting for access already or for a longer time, in order to reduce or overcome the fairness problem. Figure 3.1 depicts the relationship between two windows CW1 and CW2. The second window CW2 starts W_{th} second after the first one and it is used for lower priority traffic as described below.

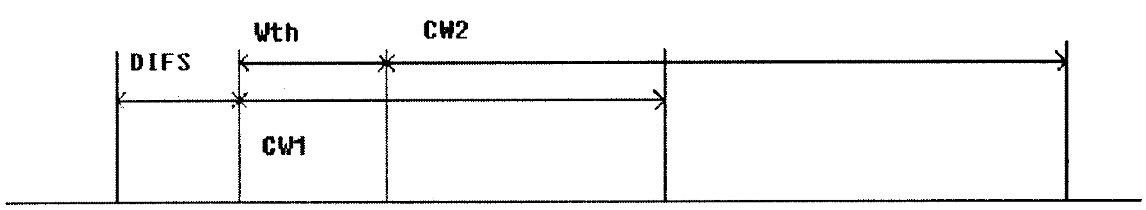

Figure 3.1 Two Windows of TWMA

Another metric Channel Status Indicator CSI (i) for each station i is introduced to reflect ambient traffic status in order to minimize waste of idle slots. CSI (i) is the one to be used in TWMA to reflect the ambient level of congestion in the vicinity of each station i from individual station's perspective. To estimate channel status of congestion, each station calculates its own CSI (i) and relies on its own direct experience i.e. according to its own success, failed transmission, and idle history. Every station has its own CSI (i) value i.e. its own view of channel status and its CSI (i) is used to adjust the station's threshold window size W_{th} (i) using the formula W_{th} (i)=CSI(i)*W, W is an input parameter, which can be set as 4, 6, 8, 10 etc. The threshold window W_{th}(i) is introduced for stations entering the second window.

CSI (i) is defined as the channel status indicator for station i:

CSI (i)=4 indicates heavy load traffic

CSI (i)=2 indicates medium load traffic

CSI (i)=1 indicates light load traffic.

CSI (i) is calculated by formula CSI (i)= L_3 (i)/(L_1 (i)+ L_2 (i)).

L_1 (i), L_2 (i), and L_3 (i) are load parameters for station i:

L_1 (i) is defined as the number of idle time that a station i has had up to the time when L_1 (i) is computing (one idle time lasts for a packet length period).

L_2 (i) is defined as the number of success transmissions that a station has had up to the time when L_2 (i) is computing. See Figure 3.2, L_2 (i) is increased by one after station i has a successful transmission.

L_3 (i) is the number of failures that a station has had up to the time when L_3 (i) is computing. See Figure 3.2, L_3 (i) is increased by one after station i has a failure.

The CSI(i) is calculated before a station attempts to contend to gain access to the shared channel. The basic operations are shown in Figure 3.2. There are two contention windows for stations' backoff counter to choose to contention for accessing the shared channel. A station's packet transmission cycle begins by continually sensing the idle time for DIFS period using the carrier sensing mechanism. If the shared channel is found not busy, then all the *new* active stations with packets to send enter the first window by initially setting their contention window size to $CW1_{min}$, where $CW1_{min}$ is the minimum window size of the first window. Then, they start to select their backoff counter values randomly in the range $CW1= [0, CW1_{min}]$. The backoff procedure begins to decrement the backoff counter by one slot time that equals SIFS length, which will be explained later. If a station continues to sense the idle channel for consequent SIFS slots, proceed to count down until the backoff counter reaches zero. The station whose backoff counter reaches zero first will gain access to the shared channel; others' backoff counters are frozen. If a station senses that the channel is busy, its backoff counter is frozen and the station defers until the channel is sensed idle again for DIFS period. Then, backoff procedure will be invoked again and the backoff counter will continue to decrement from the frozen value. When a station gains access to the shared channel, it sends RTS to its intended receiver and reserves the channel for its data packet transmission, other stations hearing RTS set their Network allocation vectors (NAV) values and defer their contentions for the channel accordingly. When the intended receiver gets the RTS, it sends back CTS to the sender. Stations hearing the CTS set or update their NAV values and defer their contentions for the channel accordingly. This RTS/CTS handshake mechanism makes sure the reservation of the channel from the interference of potential hidden stations. After the sender receives the CTS, it is sure that the shared channel has been already reserved for its use. Then, it sends data packet and waits for the ACK to corresponding data from the receiver. When the sender finally gets the ACK correctly, it means that the data packet transmission is successful and one transmission cycle is finished. The station with successful previous transmission enters the second contention window if it has more data packets to send. This reduces the chances to access the channel for stations that have previous successful transmissions because the second window is W_{th} slots lagged behind the first window. W_{th} depends on the channel traffic status CSI, W_{th} is large in heavy traffic and small in light traffic situation. If the data packet transmission is not successful because one or more parts of cycle failures such as RTS/CTS handshake fail, data error, ACK lost and etc, the station with previous failed transmission will remain operating within the first contention window. A new station also starts its transmission in the first window. The new station is defined as a station that has a packet to send after at least one main iteration time period (one packet length) of non-active status. Active means a station with one or more packets queued to send, where non-active means no packet queued to send.

3.2.1 Detailed First and Second Contention Windows Procedures

Figure 3.2 shows the basic logic of TWMA. Stations in WFCONTND_S state enter the first contention window; states will be described in later section. In the first contention window, the backoff counter's initial value is chosen to be a random value uniformly distributed in the range of $[0, CW1_{min}]$ ($CW1_{min}$ =8 slots time). If a station gains access in the first contention window and successfully complete the data transmission, the station will go into the second contention window with $CW2=\Delta*W$ if it checks its buffer and find one or more packets queued to be sent. At the same time, the number of packets in the station's buffer decrements by one and station's number of successful transmissions is incremented by one. If the station has no packet in its buffer, it will set its state to idle (IDLE_S). If a station gaining access to the channel in the second window successfully transmits a packet, it will remain in the second window. The reason is to give the next successful transmission chance to an old station that experiences previous failure or a new station that just enters the cluster.

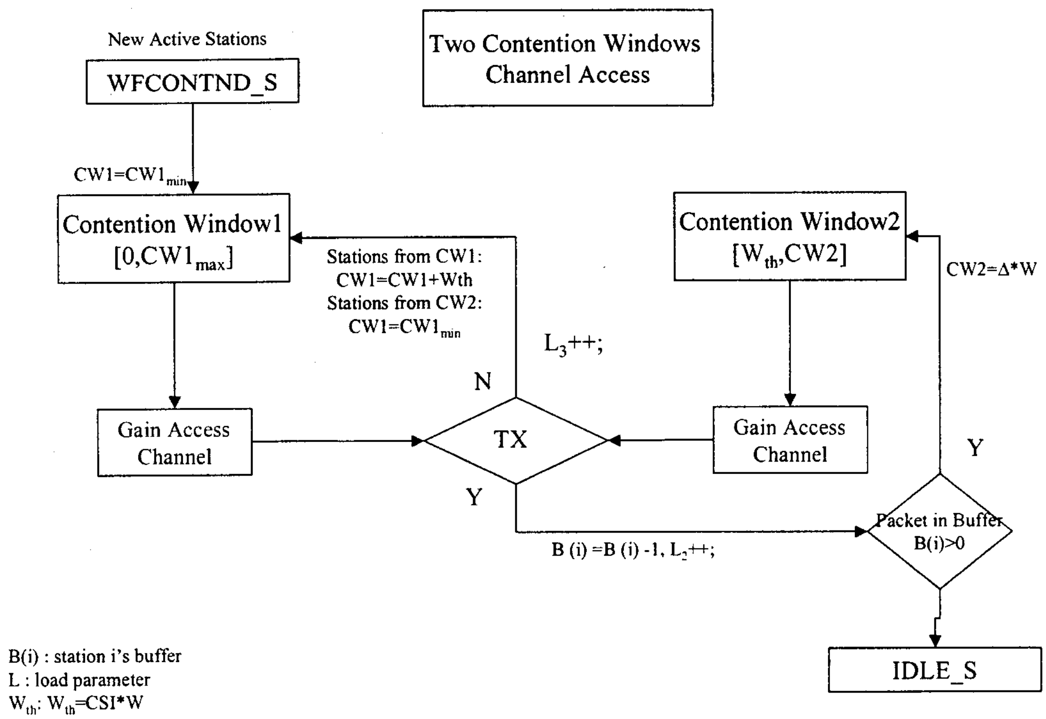

Figure 3.2 Two Contention Window Medium Access

To ensure that stations in the first contention window has higher probability to access the shared channel than stations in the second contention window, W_{th}+CW2 should be higher than or equal to the value of CW1. From the setting of input parameters later, we can see that the minimum (W_{th}+CW2) value is 20 where the value of $CW1_{min}$ is 8. The maximum (W_{th}+CW2) value is 140 where the value of $CW1_{max}$ is 64. These numbers guarantee that W_{th}+CW2$\geqslant$CW1 most the time. If a station fails the transmission in the first window because of collision, it will increase its CW1 using CW1=CW1+W_{th} up to a maximum value $CW1_{max}$ ($CW1_{max}$=64 slots). See Figure 3.2. This is similar to IEEE 802.11 MAC backoff algorithm in order to reduce the chance of future collisions in the first window. However, the difference between TWMA and IEEE 802.11 is that the increase of a station's CW is in proportion to its ambient channel traffic situation (W_{th} is related to CSI) instead of doubling its CW after a failure. If a station in the second window fails to transmit a packet because of collision or other reason, it will go to the first window with $CW1_{min}$. Each station has its W_{th} value and updates it every time the station starts to contend for channel access. The station in the second contention window starts backoff procedure W_{th} time slots later than ones in the first contention window. This gives priority to stations in the first window for channel access contention. Backoff counters of stations in the second window choose their random values in the range [W_{th}, CW2], where CW2=Δ*W, Δ and W are input parameters that can be chosen from 4, 6, 8, 10... In the heavy traffic load situation, stations' CW1 values are likely to be increased due to the increasing of collisions, so the probability of collisions among stations that have previous collisions in the first window will be reduced. W_{th} (W_{th} =CSI*W) is increased as well because CSI increases for each station. Because of the larger W_{th} value, stations in the second window, those who had successful transmissions previously, will have less probability to gain next access to the shared media. As a result, stations in the first contention window have more opportunities to access the channel than those stations lagged behind in the second window. This helps to achieve the fairness access between these stations that have gained access successfully and those stations that have experienced collisions. Under light traffic load circumstances, the situation is quite similar to the above. Access priority is given to stations that have previous failure, i.e. stations in the first contention window.

3.3 Simulation Procedure of TWMA

3.3.1 Simulation Environment:

IEEE 802.11 MAC and TWMA simulations are presented in an ad hoc network environment where all stations can communicate with all other stations directly or via intermediate routing hops under the control of DCF.

A total of 200 stations are confined to an area of 1000 meters wide and 1000 meters long. The station in the simulation has position coordinates (x, y) in the range of (0, 1000) meters, moving direction angle θ in the range (0, 360) degrees, and moving speed V in the range (0, 30) m/s as shown in Figure 3.3. Stations move in a randomly chosen direction from 0 to 360 degree and at a randomly chosen speed in the range from 0 to 30 meters per second. All stations have same radio transmission coverage range of radius from 100 to 300 meters that can be adjustable. In the simulation, radius is chosen to be 150 meters. Assuming that all stations move inside the 1000X1000 m^2 area, when they reach the boarder, they will go back into the area. Therefore, during the entire simulation period, the number of stations remains the same. All stations in the ad hoc network are assumed as asynchronous data users. The end-to-end delay depends on the transmission and queuing delays without considering propagation delay. The queuing delay includes queuing delays in sender, intermediate routing nodes, and the intended receiver. The basic channel rate of 1Mb/s is used in the simulation. The probability of successful transmission Pc is used to represent the physical layer and indicate the channel status without considering the interference and fading effects in details to simplify the channel status. Backoff timer is expressed in terms of the number of SIFS (10us) time slots, the thesis uses single frame sequence, i.e. assume that every station has random probability form 0 to 1 to generate only one packet during the main iteration period which is the MAC Service Data Unit (MSDU) of size 1000 octets, there is no fragmentation and reassembly involved. The channel reuse or the throughput in circumstance that most of stations having large size MSDUs is higher than that with most stations having smaller MSDUs. The reason is that more contentions in the shared channel unavoidably reduce the channel utilization. Users contend during each of its packets' transfer according to TWMA. It is assumed that MSDU packet length is larger than the RTS threshold, so RTS/CTS is necessary. RTS and CTS are set to be 20 and 14 octets respectively. ACK has the same size as CTS. In the simulation, RTS/CTS handshaking mechanism is simulated to reserve the shared medium from interference of hidden nodes. Mobility of stations across clusters is simplified if one assumes that all the transmission can be finished successfully before a station moves across the cluster border. Failed transmission, failed gaining access or losing ACK increment the retry counter associated with the packet. If the retry limit is reached, the packet is dropped. The retry limit as an input parameter can be set to any number preferred, in the simulation, retry limit is 3. That is to say, if a station cannot transmit its packet successfully after 3 attempts, the packet is aborted. Another metric time to live TTL is added to decide whether a packet is dropped or not by giving a maximum lifetime to a packet. If the lifetime limit is reached, the packet is discarded as well. The evaluation criterion is that the MAC should deliver high network utilization, in other words. high overall throughput, and provide fairness to media access, i.e. fair allocation of throughput to stations. IEEE 802.11 backoff algorithm resets CW to CW_{min} whenever a station transmits a packet successfully and double it's CW every time it fails to transmit. Thus, CW range varies widely and doesn't reflect channel contention status. In IEEE 802.11 DCF, resetting CW to CW_{min} after a successful transmission will cause more collisions in a heavy traffic situation because surely one success transmission doesn't necessarily indicate that channel contention is light enough for CW to go back to its minimum.

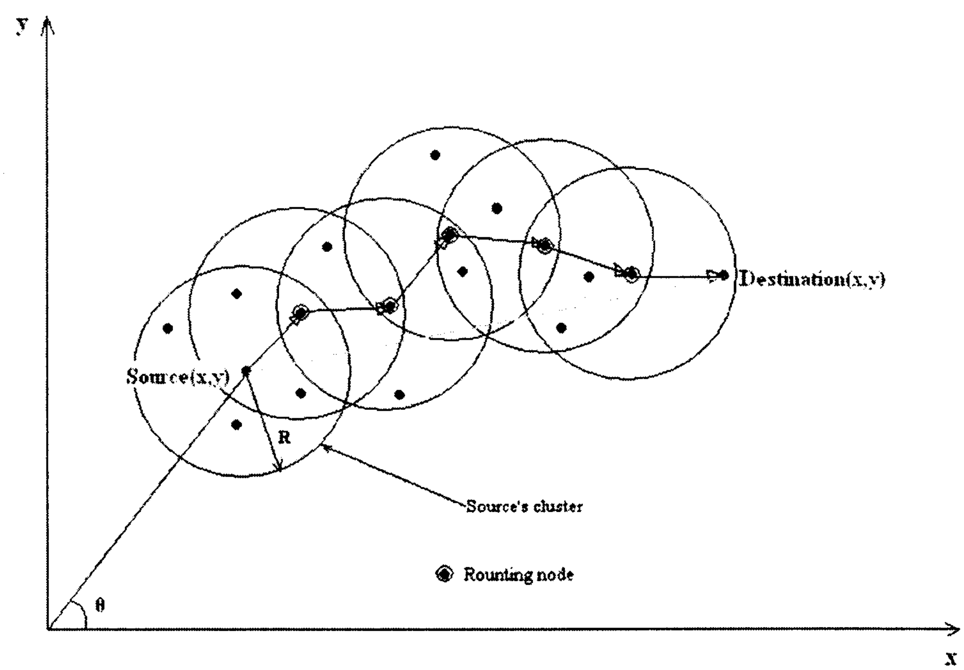

Figure 3.3 An Example of Routing Pattern

In the thesis, the comparison of simulation results confirms the improvement of performance to IEEE 802.11 backoff scheme. The binary exponential backoff scheme can cause too many collisions and idle periods during transmission cycle. Increasing CW reduce collision failures only when a station fails to transfer a packet because of collision. However, there are also many reasons of failures including collision, noise interference, and channel fading. These reasons of failures besides collision also cause CW's increase, so it leads to wasted idle time. This is one of the reasons to introduce CSI to reflect a station's ambient congestion level.

The following input parameters are used in the simulation to obtain the performance results presented in this thesis except those specified. Various input parameters combinations are also studied to achieve optimal performance results. Some parameters are set with different values to analyze the performance under various scenarios:

1) Number of stations N: 200

2) Transmission range radius R: 150 meters

3) Testing area RANGE: $RANGE^2=1000X1000m^2$

4) Station's coordinates x, y: randomly chosen in the range of (0, 1000) meters respectively.

5) Station' s moving velocity V: randomly chosen from 0 to 30 meters a second, i.e. MAXSPEED=108Km/h.

6) Station moving direction angle θ : randomly chosen in the range of 0-360°.

7) Channel rate: 1Mb/s

8) Average data packet size: 1KBytes.

9) Station buffer size BUFFERSIZE: 4, 8, 16, 32, 64, 128, 256 packets

10) Channel packet success probability Pc: 0, 0.1, 0.2, ..., 1.

11) Packet generating probability Pg: 0, 0.1, 0.2, …, 1.

12) Slot time T_UNIT: 10μs.

13) SIFS time: 10μs.

14) DIFS time: 50μs.

15) Main iteration time TP: 800SIFS=8000μs

16) Buffer algorithm: FIFO, a node sends its packet from its buffer according to FIFO.

17) Minimum contention window size CW_{min}: 8 in IEEE 802.11, CW1 and CW2

18) Maximum contention window size CW_{max}: 256 in IEEE 802.11, $CW1_{max}$=64

19) Maximum number of retries for a packet to contend for channel access (MAXRETRYNUM): 3

20) Total simulation time TOT_RUNTIME: 800000 Slot time

3.3.2 Performance Criteria

1) Average end-to-end traffic delay statistics:

Average end-to-end traffic delay is defined as a period of time between the time a packet is generated and the time the packet reaches its destination and successfully acknowledged. The delay consists of propagation delay, transmission delay, and queuing delay, assuming propagation delay is small enough to be ignored. Average end-to-end traffic delay can be calculated as the result of the sum of successful transmission time of all successful total end-to-end transmitted data packets divided by the number of successful transmitted data packets'. The end-to-end transmission time starts from the moment a data packet is generated and ends when the sending station receives ACK successfully. The end-to-end transmission time counts transmission delay and total queuing delay of source, intermediate routing nodes, and destination without considering propagation delay of all packets NP generated by all users.

$$E[DT]=\frac{\sum_{i=1}^{NP} DT_i}{NP}$$

DT_i is the end-to-end traffic delay of data packet number i, NP is the total number of successful transmitted data packets for all users.

2) End-to-end traffic delay variance:

Traffic delay variance is defined as the sum of the squares of difference between one packet i's end-to-end traffic delay DT_i and average traffic delay E [DT] divided by NP. Traffic delay variance is calculated using following equation:

$$\delta_{DT}^2=\frac{\sum_{i=1}^{NP}(DT_i-E[DT])^2}{NP},$$

DT_i is the end-to-end transmission delay of data packet number i, NP is the total number of successful transmitted data packets for all users.

3) Average end-to-end queuing delay:

Average queuing delay is defined as the average total waiting time in buffers for data packets to be transmitted from the source station, intermediate routing stations, and the target station. It can be calculated by using the following equation:

$$E[DQ]=\frac{\sum_{i=1}^{N} DQ_i}{N}, \text{ where } DQ_i=\frac{\sum_{t=1}^{T} DQ_{i,t}}{T},$$

$DQ_{i,t}$ is station i's queuing delay at time t . It can be represented by counting the number of data packets in the buffer of station i at time t, DQ_i is the average queuing delay at station i during the whole simulation period. It can be represented by the average number of data packets in the buffer of station i at any time during the whole simulation period, T is whole simulation period, N is total number of stations in simulation area.

4) Queuing delay variance

Queuing delay variance is defined as the sum of the squares of difference between station i's queuing delay DQ_i and station's average queuing delay E [DQ] divided by N. Queuing delay variance can be entitled as the sample variance i.e.

$$\delta_{DQ}^2 = \frac{\sum_{i=1}^{N}(DQ_i - E[DQ])^2}{N}$$

Where i is the station number and N is total number of stations in simulation area.

5) Average buffer overflow

Average buffer overflow is defined as the average number of times the buffer overflows for all stations in testing area during the whole simulation period.

$$E[BO] = \frac{\sum_{i=1}^{N} BO_i}{N}$$

BO_i is the total number of time station i's buffer content exceeds its limits during the simulation time. N is total number of stations in simulation area.

6) Buffer overflow variance

Buffer overflow variance is defined as the sum of the squares of difference between station i's buffer overflow BO_i and station's average buffer overflow E[BO] divided by N. Buffer overflow variance can be entitled as the sample variance i.e.

$$\delta_{BO}^2 = \frac{\sum_{i=1}^{N}(BO_i - E[BO])^2}{N},$$

Where N is the number of stations.

7) Average number of dropped packets

Average number of dropped packets is defined as the average number of dropped data packets including those packets loss due to buffer overflow, those dropped because the number of contention retrying for access exceeds retrying limits, and those lost due to channel failures for all stations during the whole running time. When Pc=1, no packet loss occurs due to channel failure. DP_i is defined as the total numbers of dropped packets of station i during the whole running time.

$$E[DP] = \frac{\sum_{i=1}^{N} DP_i}{N}$$

Where N is the number of stations.

8) Dropped packets variance

Dropped packets variance is defined as the sum of the squares of difference between station i's average dropped packets number DP_i and station's average dropped packets number E[DP] divided by N. Dropped packets variance can be obtained by

$$\delta_{DP}^2 = \frac{\sum_{i=1}^{N}(DP_i - E[DP])^2}{N},$$

Where N is the number of stations.

9) Average Traffic Throughput

Average throughput $E[\eta]$ is defined as station's average percentage of the packets that are successfully transmitted to destination during the simulation period over the packets that are generated, and where η_i is station i's average throughput,

η_i=(The number of data packets of source station i that reach the destination successfully)/(The number of total data packets that station i generates).

$$E[\eta] = \frac{\sum_{i=1}^{N}\eta_i}{N}$$

Where N is the number of stations.

10) Throughput variance

Throughput variance is defined as the sum of the squares of difference between station i's average throughput η_i and all stations' average throughput $E[\eta]$ divided by N. Throughput variance can be obtained by

$$\delta_{\eta}^2 = \frac{\sum_{i=1}^{N}(\eta_i - E[\eta])^2}{N}$$

Where N is total number of stations.

3.3.3 TWMA Simulation Model and Control Rules

Figure 3.4 depicts a general structure of TWMA simulation diagram. At the first iteration, a station is generated with ID, initiated position coordinates (x, y), moving speed and direction all obtained by repeatedly generating uniformly distributed random variables in the range of (0,1). The variables are multiplied by N, RANGE, MAXSPEED, and 360° to get user ID, position coordinates (x, y), velocity v, and moving direction angle θ respectively. Every 10 main program iterations, stations' position coordinates, moving direction and cluster information are updated, and cluster information indicates the current number and ID of nodes in the cluster. Every main iteration, each station is checked if it generates a data packet. The probability of packet generation Pg is introduced to control the generation of packet for a station; when the station has a random outcome larger than Pg, it generates a packet. Pg is an input parameter defined as packet generation probability that can be set in the range from 0.1 to1. The next step is to generate a destination station with information (x, y, v, θ) for the generated packet. Then, the active station starts to contend for channel access if the destination is found in the reachable range. Routing for next node is needed if the destination is out of the range. After gaining the channel, the station starts to transmit control and data packets. If the receiver is not the destination station for the successful transmitted data packet, the next routing table is consulted and so on. Figure 3.4 shows that the transmitted data packet will be transmitted to the intermediate routing nodes that are found using the shortest distance routing algorithm (explained later) until it reaches the destination. If the data packet transmission is successful and the receiver is the destination for the data packet, the whole transmission cycle is finished. If the station fails to access the channel or to transmit, it will retry to contend for channel access. Meanwhile, the retry limit is checked, if it is exceeded, the packet will be dropped. When the total time is exceeded, the simulation ends. If not, the simulation continues to run.

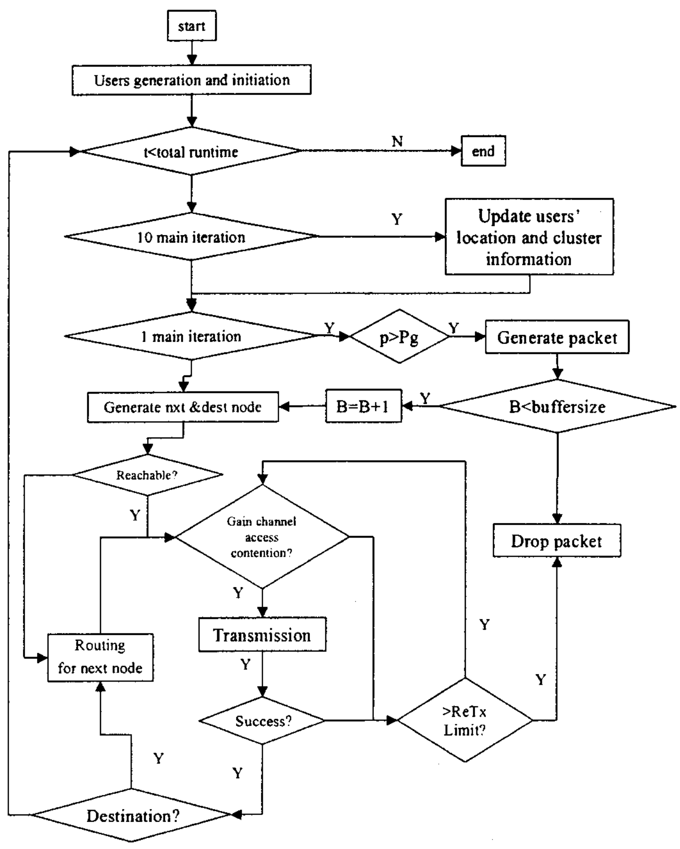

Figure 3.4 TWMA Simulation Flow Diagram

In the simulation, twelve states are defined as IDLE, NAV, Waiting For Contention (WFCONTND_S), BACKOFF, Transmitting RTS (TXRTS_S), Receiving RTS (RXRTS_S), Transmitting CTS (TXCTS_S), Receiving CTS (RXCTS_S), Transmitting DATA (TXDATA_S), Receiving DATA (RXDATA_S), Transmitting ACK (TXACK_S), and Receiving ACK (RXACK_S). In the state IDLE_S, The station doesn't have any packet to send. In WFCONTND_S, the station has a data packet to send and is trying to obtain the channel. TX means that the station is transmitting a packet that may be a data or control packet. RX means that the station is receiving a data or control packet. BACKOFF_S state means that the station is in backoff procedure. In NAV_S, the station is restricted from contending for channel until NAV period is finished. The sender's state is set as Transmitting RTS, Receiving CTS, Transmitting DATA, and Receiving ACK (TXRTS_S, RXCTS_S, TXDATA_S, and RXACK_S) respectively according to different sections in the transmission cycle whereas the next node's state is set as Receiving RTS, Transmitting CTS, Receiving DATA, Transmitting ACK (RXRTS_S, TXCTS_S, RXDATA_S, and TXACK_S) correspondingly.

Figure 3.5 to Figure 3.14 show the details of TWMA states changes during a transmission cycle starting with sensing the channel, contending for the channel, backoff contention, RTS/CTS handshaking, transmitting, receiving data packet to the time of successfully receiving the acknowledgement of transmitted packet.

Figure 3.5 shows details of packet generation and distance checking. The new generated packet is put into the buffer and its status is set as Waiting For Contention (WFCONTND_S) and excess packets are dropped if the station's buffer overflows. If the destination is out of radio coverage, next intermediate routing node needs to be found and added to routing table of the packet.

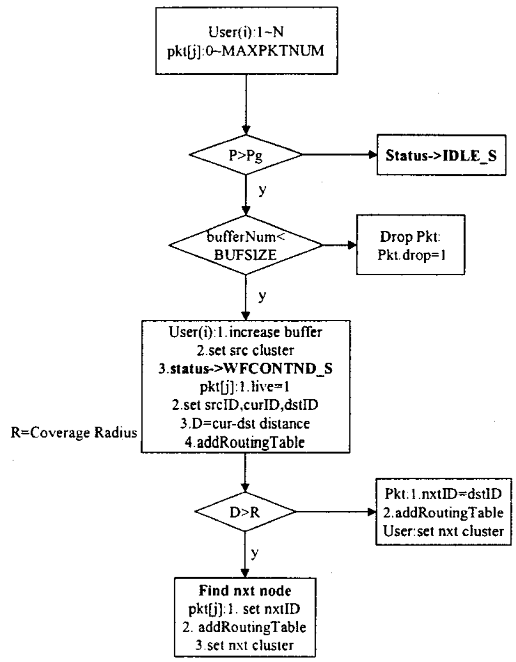

Figure 3.5 MAC control rules 1-Packet Generation and Distance Check

Figure 3.6 shows the detailed finding of the next node procedure. The packet has an information structure including source ID, current node ID, next node ID, and destination ID etc. The intention is to check whether the destination is in the range of radio coverage (D≤R). Next intermediate routing node is needed if the destination is not reachable. A shortest distance routing algorithm for simulation is used to find the next node. Shortest distance routing algorithm means finding a station that has shortest distance to the destination as next node in the sender's cluster. The cluster of a station is a circle area where the station is at the center and radio coverage distance is the radius.

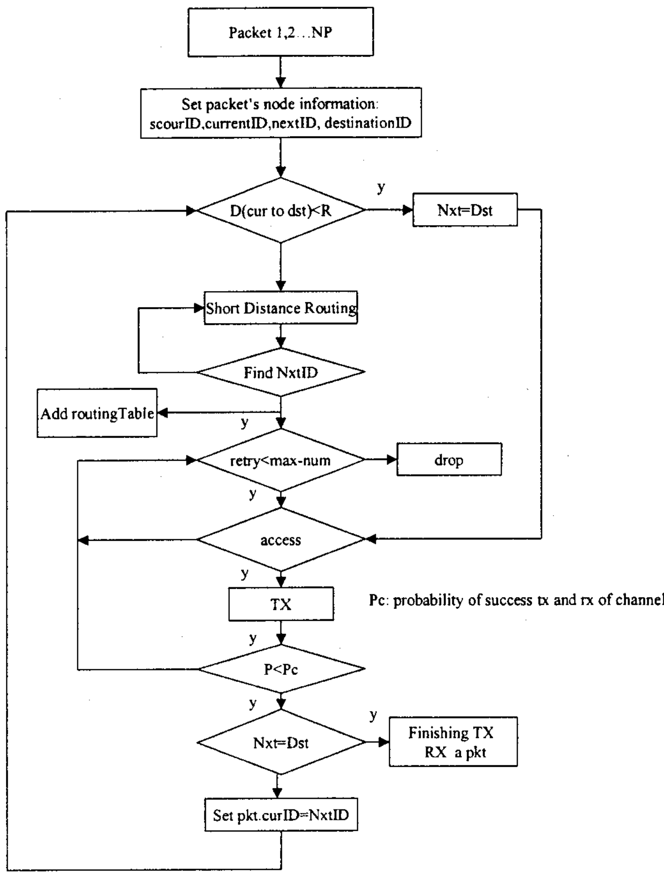

Figure 3.6 MAC control rules 2-Routing Procedure

When the sender finds its next node, it begins to contend to access the shared channel until it gains the channel within its retry limit. Its status is set as Backoff (BACKOFF_S) as long as it senses the channel idle for DIFS. If the sender wins the backoff contention during the contention cycle then it gains the channel. If the next receiver is the destination, the packet transmission cycle is finished after a successful transmission. If not, the packet has to update its information structure. The packet's current ID is changed to its next node's ID i.e. set the next node as its new sender and find a new intermediate routing node as its next one. Finding routing procedure and transmission procedure are repeated until the packet reaches the destination.

Figure 3.7 shows that the transitions take place if a station in idle state reaches the end of NAV, its state is changed to WFCONTND_S if in addition it has a packet in buffer to send. The program checks whether it is the receiver of other stations, if it is, its state may be changed to RXRTS_S, RXCTS_S, RXDATA_S, and RXACK_S accordingly depend on the type of packet it will receive.

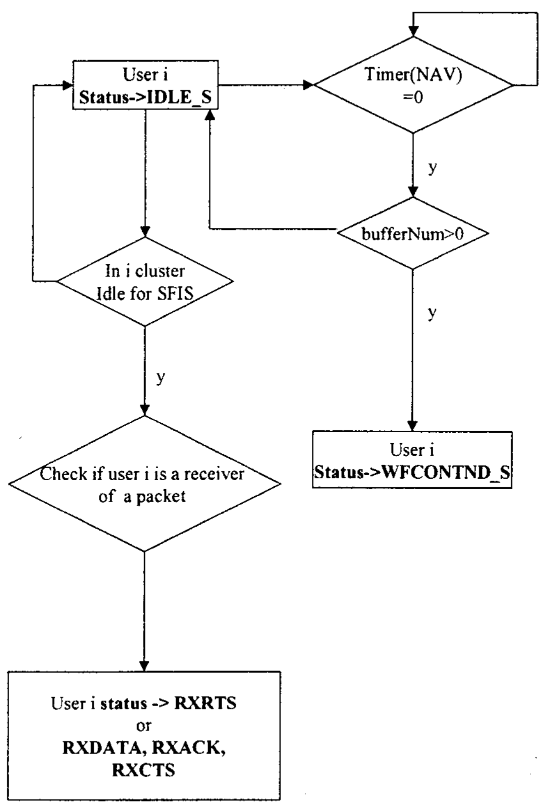

Figure 3.7 MAC control rules 3-States Transitions

Figure 3.8 shows the procedure that a station changed its state from WFCONTND_S to BCKOFF_S. When a station is in state WFCONTND_S, CW and backoff counter value are set first. If it is the first time to contend to access, CW is changed to CW_{min}. Otherwise, CW is doubled if the CW does not exceed the CW_{max} and $CW=CW_{max}$ if $CW \geq CW_{max}$. After getting its CW and backoff count value, the station starts to checks if the shared channel in its cluster has been idle for DIFS slots. The station is ready to go for backoff contention procedure when the channel is idle for DIFS.

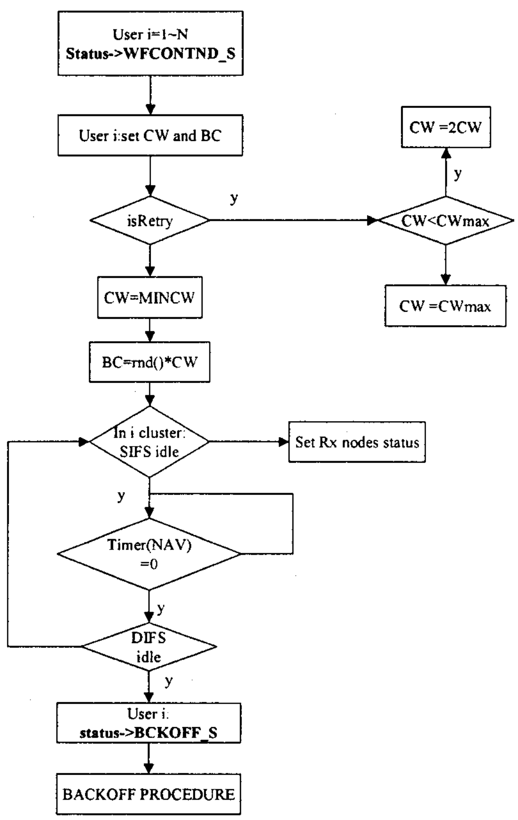

Figure 3.8 MAC Control Rules 4- Wait for Contention to Backoff Procedure

Figure 3.9 shows the backoff contention procedure. The station in state of BCKOFF_S checks if its backoff counter reaches zero at every time slot and decrement it by one after it senses the channel is idle for each SIFS slot. If the channel is not idle, the station's backoff counter will be frozen by the other users that are currently trying to transmit the packet in the c luster. If b ackoff c ounter o f o ne u ser r eaches z ero w hereas t he c hannel i s i dle, i t changes its state from BCKOFF_S to TXRTS_S or TXDATA_S, which freezes other active stations backoff counter. Otherwise, its state is changed to WFCONTND_S.

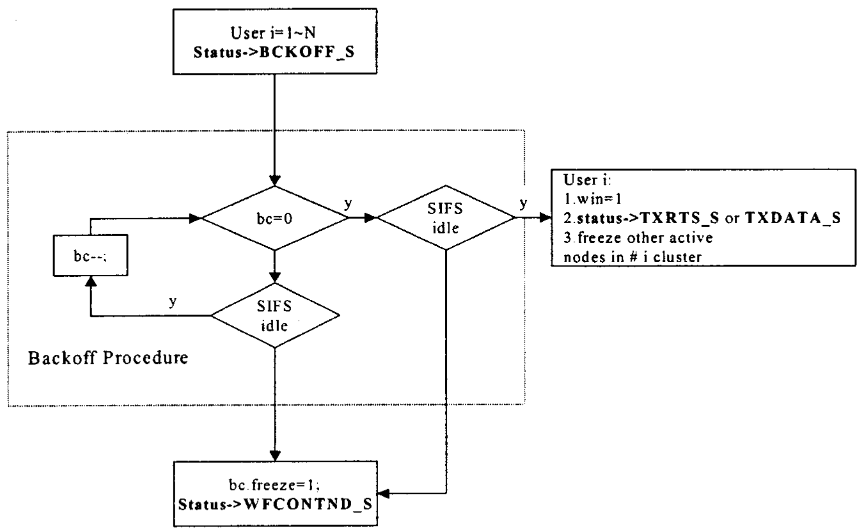

Figure 3.9 MAC control rules 5-Backoff Contention Procedure

Figure 3.10 shows the RTS and CTS control packet pair i.e. transmission and receiving. For example, a station i in TXRTS_S state transmits a data packet to the next node j, meanwhile sets the next receiving node j in RXRTS_S state and other stations in station i's cluster NAV_S states. After finishing transmission and receiving of RTS, transmission and receiving of CTS begins. In jth cluster, every station that receives the CTS except station i sets its NAV state. The transmission and receiving of DATA and ACK packet pair is very similar to the RTS and CTS control packet pair transmission and receiving, so it is not repeated here.

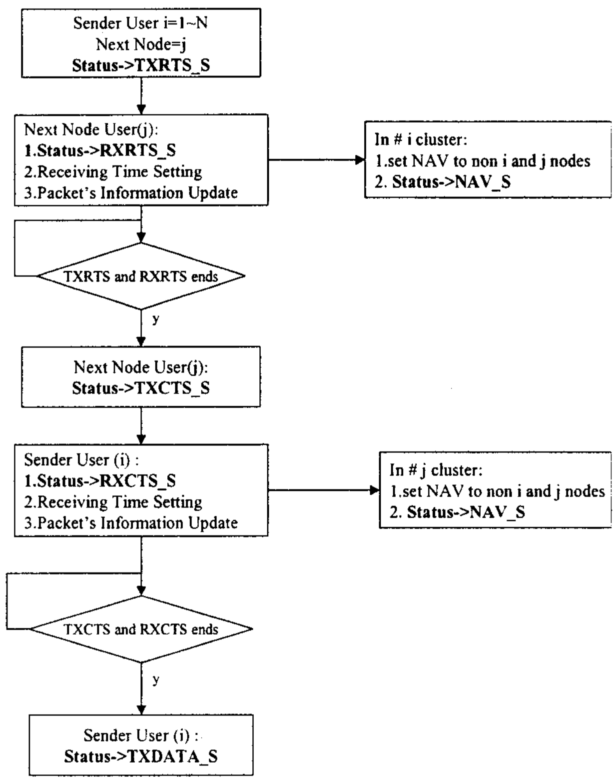

Figure 3.10 MAC control rules 6-Packet Transmission and Receiving

Figure 3.11 shows the states transitions between RXDATA_S and TXACK_S. A station in RXDATA_S sets its state to TXACK_S after it finishes receiving the data packet. In the simulation, a timer is used to count the transmission period of the DATA. The states transition between RXRTS_S and TXCTS_S is the same. So is the transition between RXCTS_S and TXDATA_S.

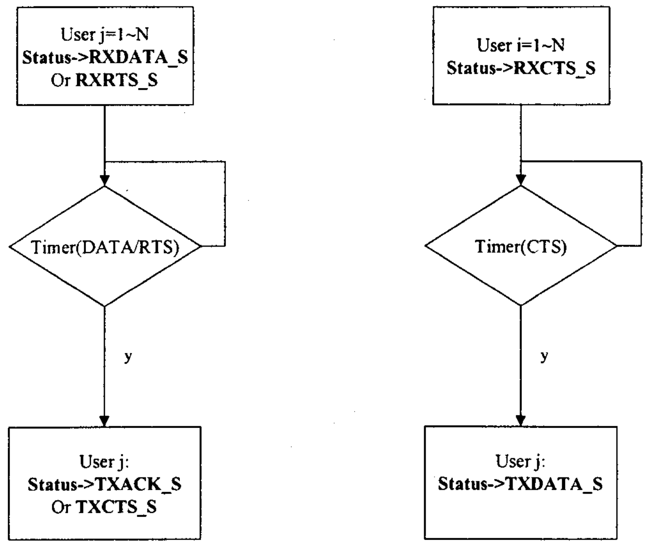

Figure 3. 11 MAC control rules 7-States Transitions

Figure 3.12 shows the states transitions between TXACK_S, RXACK_S, WFCONTND_S, and IDLE_S. Station i in state TXACK_S sends ACK to station j and sets station j to state RXACK_S. If station i is not a destination node of the transmitted packet, the station i's state is set to WFCONTND_S. If station i is destination for the transmitted packet, when station i receives ACK successfully, it will check whether it has more packet waiting to be sent in its buffer. If it has, its state will be changed to WFCONTND_S, if not, its state will be changed to IDLE_S.

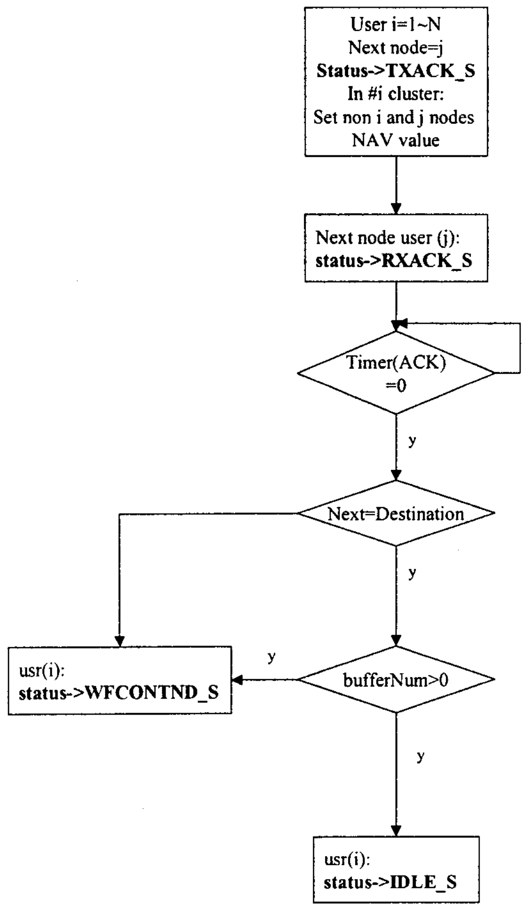

Figure 3.12 MAC control rules 8- States Transitions

Figure 3.13 shows the states transition between NAV_S, WFCONTND_S, and IDLE_S. A station in NAV_S sets its state to WFCONTND_S after its NAV ends if there are one or more data packets waiting to be sent in its buffer, otherwise, it is set to state IDLE_S.

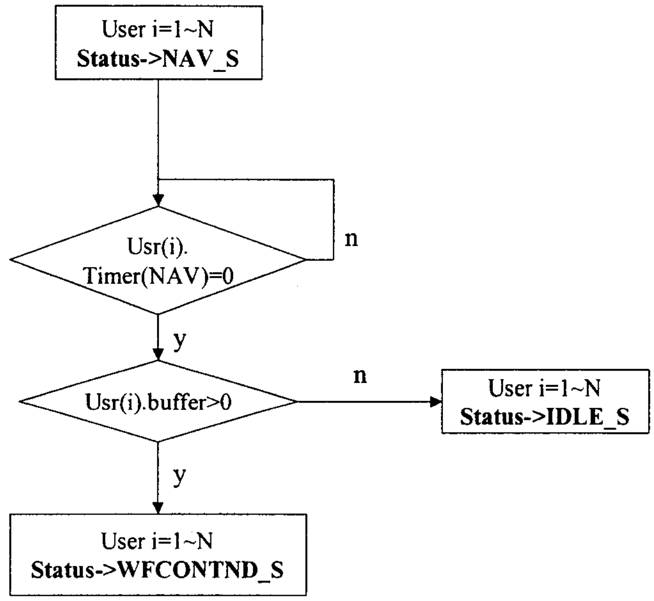

Figure 3.13 MAC control rules 10-States Transitions

Figure 3.14 shows the detailed final ACK checking procedure of the simulation. If station i in RXACK_S successfully receives ACK, in other words, successfully transmits its packet to the next node without buffer overflow, it will set its CW size and get its backoff counter randomly according to the used algorithm. Accordingly it needs to update the packet's information including success or failure counting, success or fail flag setting, increasing and decreasing relative station's buffer depending on successful or failed transmission and receiving of the ACK. If station i is the destination, it will update its buffer data by increasing its buffer by one and decreasing the original packet sender and current sender's buffers by one respectively. If station i in RXACK_S fails to receive ACK, it will increase

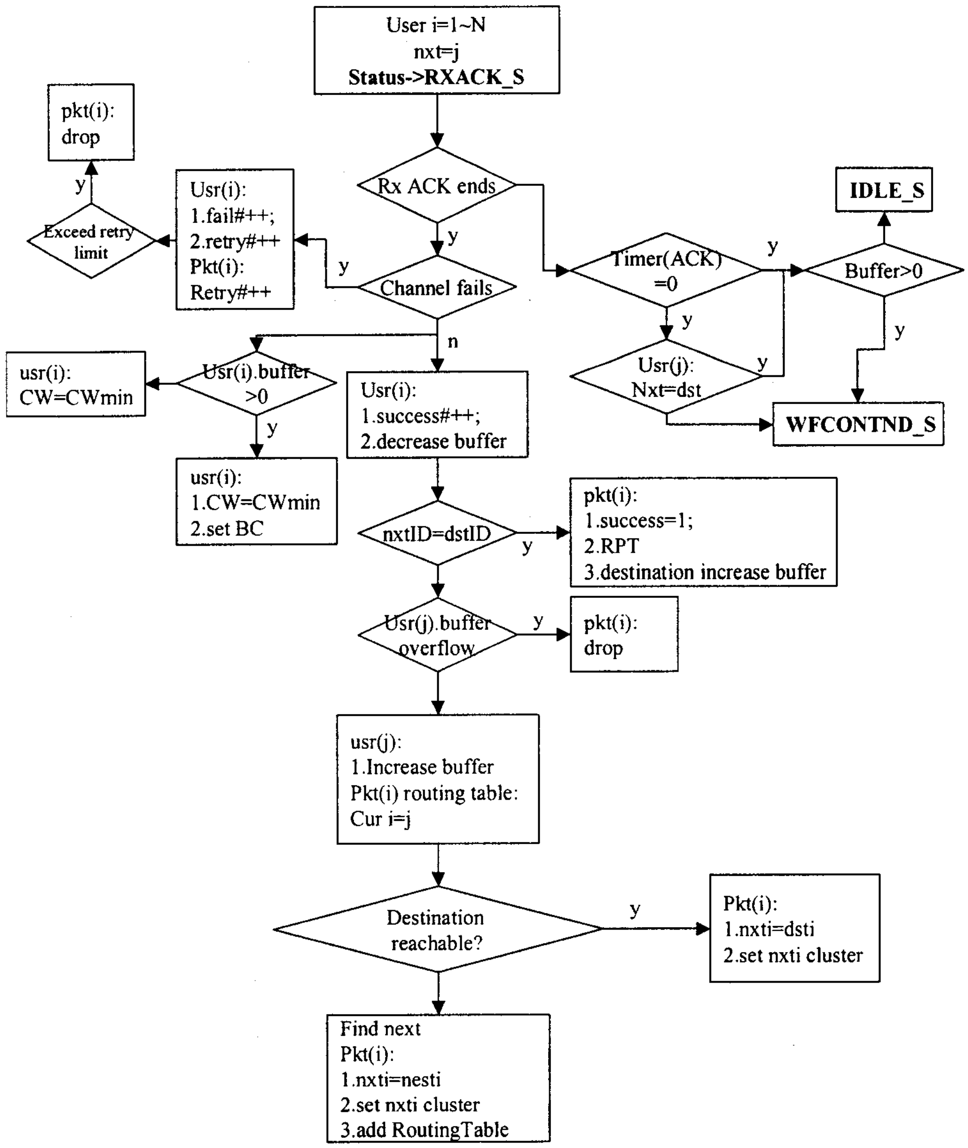

Figure 3.14 MAC control rules 9-ACK Check Procedure

its retry number and increase its CW size according to the used algorithm. If the receiver is buffer overflowed, the transmitted data packet will be dropped. The following procedures are distance checked for the destination and finding next intermediate node if it is needed.

If the destination is in its cluster, set the destination as next node and update next node's cluster information. If not, it has to find its next intermediate routing node using the same shortest distance routing algorithm. It needs to update routing table of the packet, and update station i's cluster information as well. These routing and transmission procedures are repeated until the packet reaches the destination. In the cases of buffer overflow on the next node, exceeding the retry limit for contending access, life timeout, and channel failure (Pc is an input parameter defined as probability of successful channel), the transmitted packet will be dropped. In the simulation, Pc is usually set as 1 so a perfect channel condition is assumed.

3.4 Performance Results

Both TWMA and IEEE 802.11 MAC simulations have been modeled in order to make comparison between two MAC schemes with respect to throughput, latency, queuing delay, buffer overflow, and etc. Among the metrics stated above the most important performance ones to be used to measure the effectiveness of media access control algorithm are throughput and latency. The following performance data is obtained under some conditions and circumstances specified earlier such as channel success probability Pc=1 which means there are no interference, no fading in the channel to cause transmission failure. Simulation results show that input parameters such as probability of packet generation Pg, buffer size, Pc, and etc have expected relationship with end-to-end traffic delay, queuing delay, and throughput etc. Simulation results also confirm that it is possible for TWMA to achieve less delay and higher throughput performance than IEEE 802.11 MAC by searching for optimum combination of the new algorithm variables such as W, delta $\triangle$, and CSI values. In the simulation, the variables such as W is chosen from 4, 6, 8, 10, delta $\triangle$ from 4, 6, 8, 10, and CSI from (1,2,3) and (1,2,4). Optimum combinations of those variables W=6, $\triangle$=6, and CSI=(1,2,4) are used to generate best performance.

3.4.1 Average Traffic Delay

Average end-to-end traffic delay consists of propagation delay, transmission delay, and queuing delay. Increasing the probability of packet generation intensifies the traffic load in the shared channel. As a result, transmission delay and queuing delay will certainly grow. As stated earlier, the increases or decreases of buffer size directly increase or decrease the queuing delay in each station whether it is a sender, or a intermediate router, or a final receiver and more obviously in higher traffic load. When Pg=0.1, the end-to-end traffic delay is at its lowest point and then increases as Pg increases until it reaches the highest point though there are some small ups and downs along the way. For a certain Pg value, it can be seen that the traffic delay increases as long as buffer size increases (mainly because the involved queuing delay increases). The traffic delay reaches its highest point as expected when buffer size is 128 or 256 for a certain Pg value. Traffic delay variance has very similar graphical pattern as average traffic delay in Figure 3.16. Traffic delay variance increases as Pg increases. For a fix Pg value, traffic delay variance increases as buffer size increases from 4 to 256.

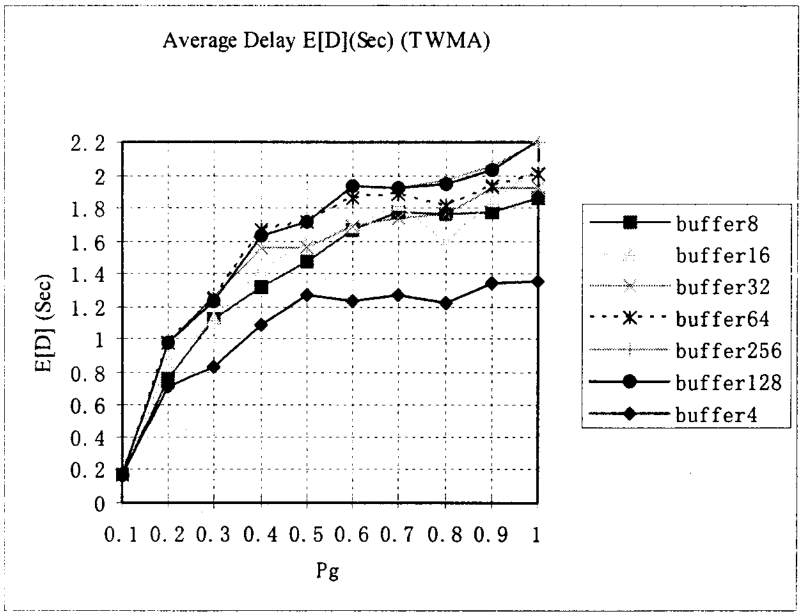

Figure 3.15 Average Traffic Delay E[D] Versus Probability of Packet Generation Pg

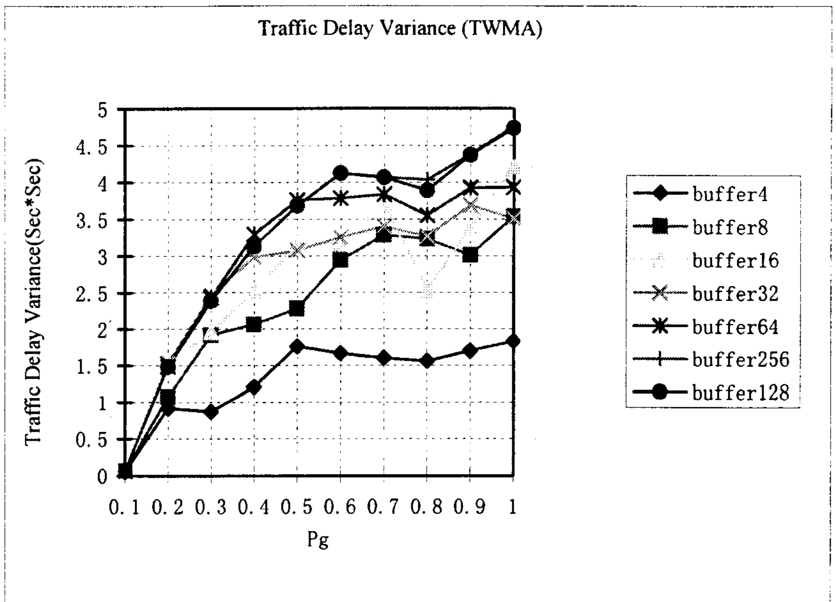

Figure 3.16 Traffic Delay Variance δ^2_{DT} Versus Pg Probability of Packet Generation

3. 4.1.1 Delay Comparison between TWMA and IEEE 802.11 MAC

3.4.1.1.1 End-to-end Delay Comparison for Different Buffer Sizes and Packet Generation Probabilities Pg

Figures 3.17 to 3.26 show the comparison of the delay performance between TWMA and 802.11 MAC for 10 different values of Pg in the range 0.1 to 1. From these Figures, we can see that TWMA with optimum combination of variables has much better performance than 802.11 standard MAC in term of average end-to-end traffic delay. The delays for both TWMA and 802.11 MAC rise slightly in general with buffer size from 4 to 256 for a certain Pg value. The reason for this is that queuing delay increases as buffer size increases where the network density remains the same i.e. Pg is a constant value chosen from 0.1 to 1. When Pg=0.1 in TWMA, average end-to-end traffic delay does not change much for a buffer size larger than 16. The reason is that for light traffic, the queuing delay is relatively small and there is no buffer overflow, so it does not change much when the buffer size is larger than 16. When Pg=1 in TWMA, because of the heavy traffic, average traffic delay increases as buffer size increases.

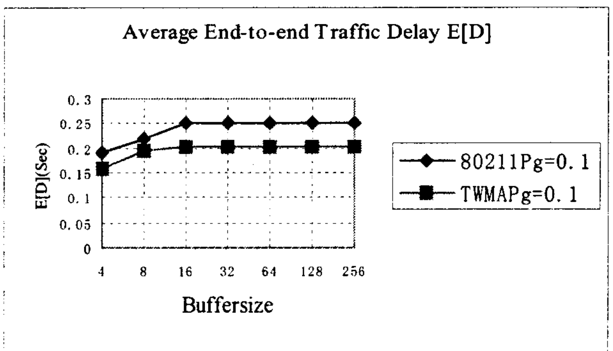

Figure 3.17 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Packet Generation Probability Pg = 0.1.

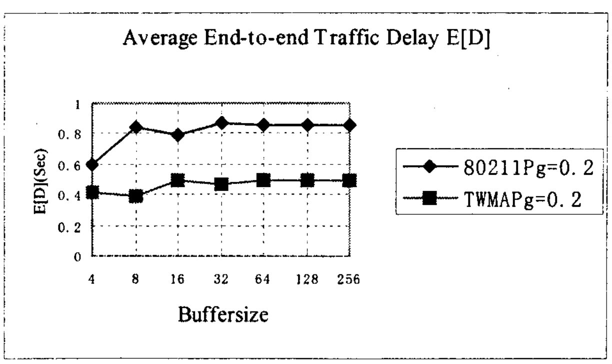

Figure 3.18 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Packet Generation Probability Pg = 0.2.

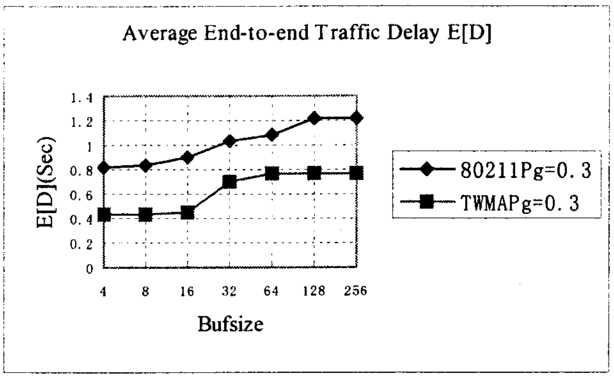

Figure 3.19 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Packet Generation Probability Pg = 0.3.

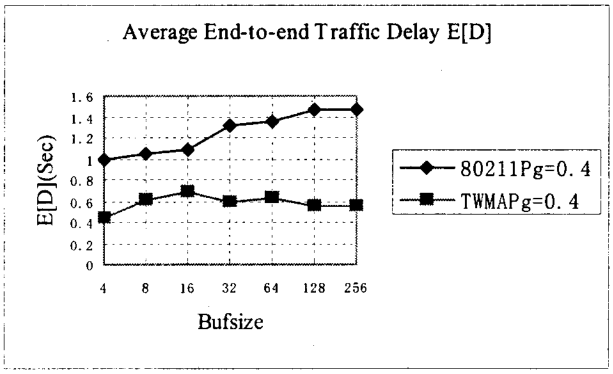

Figure 3.20 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Packet Generation Probability Pg = 0.4.

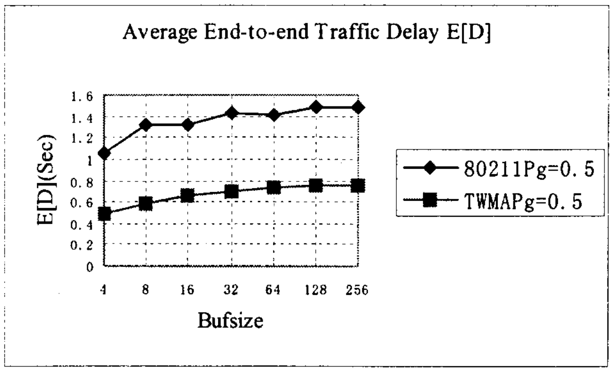

Figure 3.21 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Packet Generation Probability Pg = 0.5.

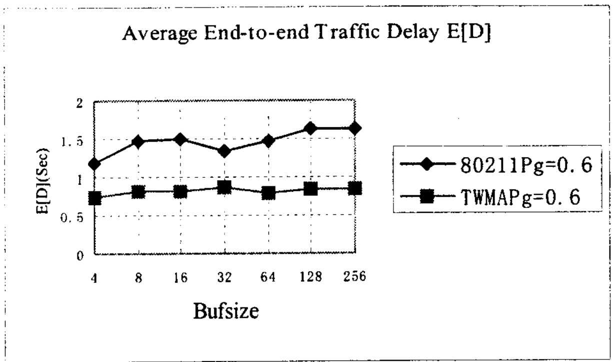

Figure 3.22 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Packet Generation Probability Pg = 0.6.

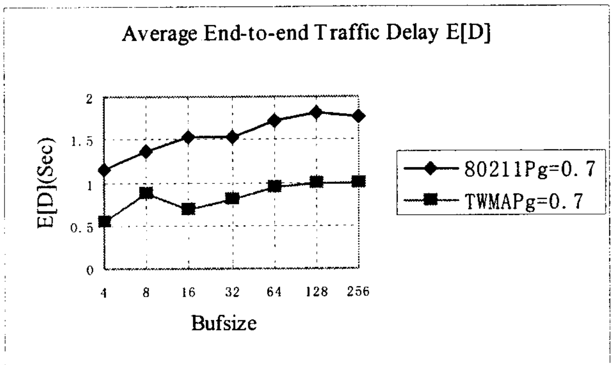

Figure 3.23 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Packet Generation Probability Pg = 0.7.

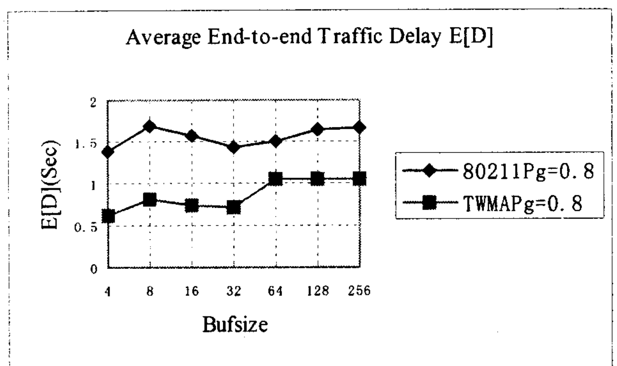

Figure 3.24 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Packet Generation Probability Pg = 0.8.

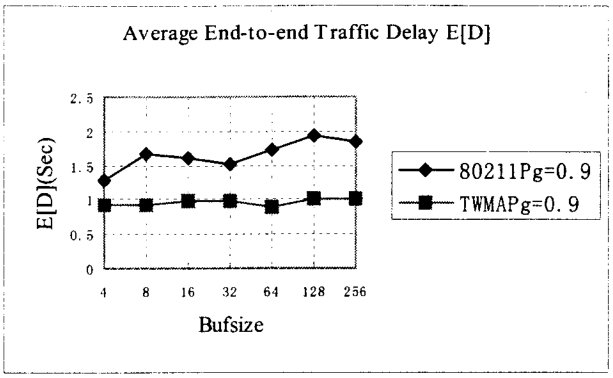

Figure 3.25 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Packet Generation Probability Pg = 0.9.

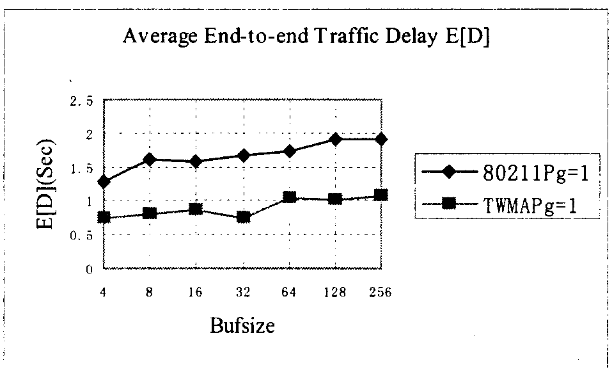

Figure 3. 26 Average End-to-end Traffic Delay Comparison Between TWMA and 802.11 MAC for Different Buffer Size 4 to 256 at Packet Generation Probability Pg = 1.

In the following Figures 3.27 to 3.33, these figures come from the same performance results but they show the comparisons from different perspectives. The average delays of both IEEE 802.11 and TWMA are very similar when traffic is light (Pg=0.1). It is because that initial contention window sizes in both IEEE 802.11 and TWMA access algorithms are the same (CW initial value CW_{min}=8) and they have similar number of few collisions in both cases. With the increases of Pg, the difference of average delays between IEEE 802.11 and TWMA becomes larger gradually for all buffer sizes in the range of 4 to 256.

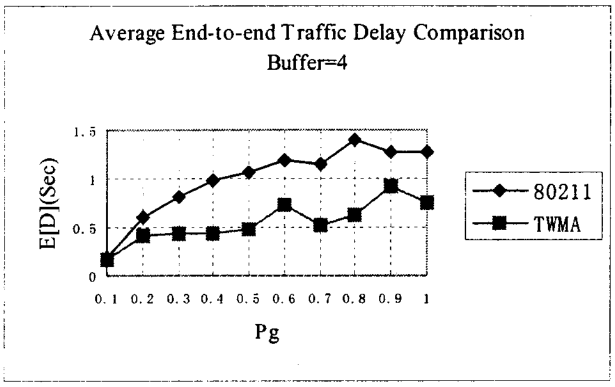

Figure 3. 27 Average End-to-end Traffic Delay Comparison at Buffer Size =4

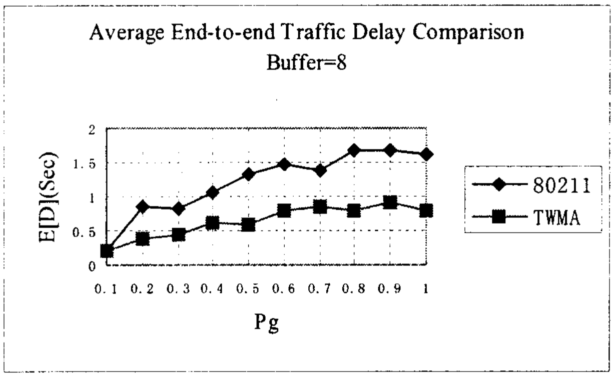

Figure 3. 28 Average End-to-end Traffic Delay Comparison at Buffer Size =8

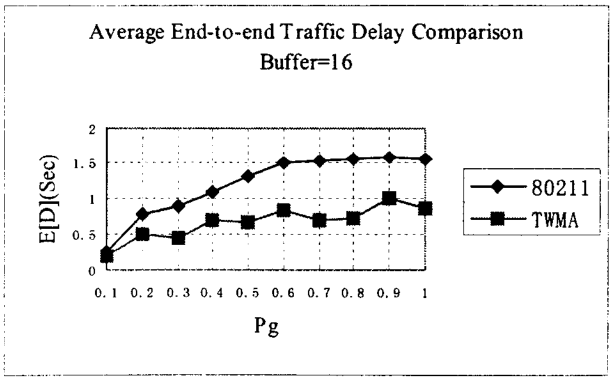

Figure 3. 29 Average End-to-end Traffic Delay Comparison at Buffer Size =16

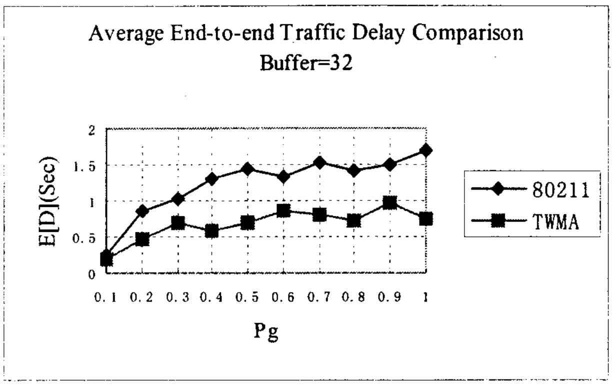

Figure 3. 30 Average End-to-end Traffic Delay Comparison at Buffer Size =32

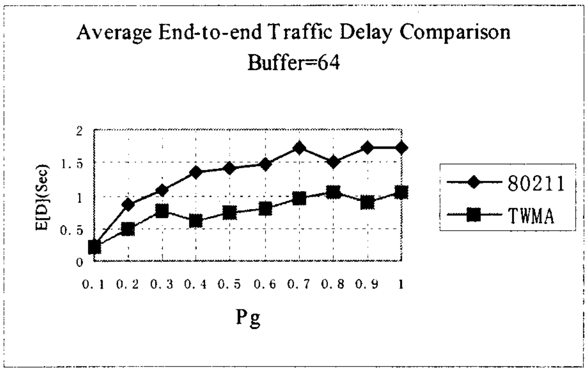

Figure 3. 31 Average End-to-end Traffic Delay Comparison at Buffer Size =64

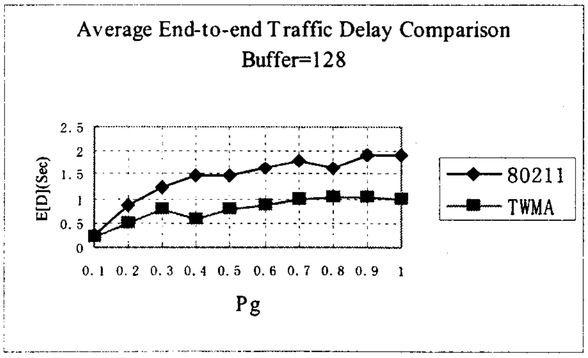

Figure 3. 32 Average End-to-end Traffic Delay Comparison at Buffer Size =128

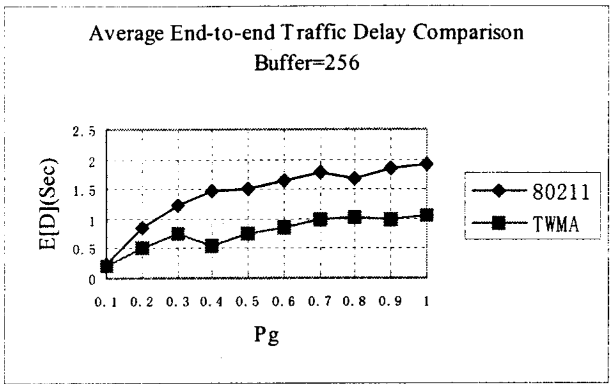

Figure 3. 33 Average End-to-end Traffic Delay Comparison at Buffer Size =256

3.4.1.1.2 Delay Distribution Comparison

Figures 3.34 to 3.36 show the average end-to-end traffic delay distribution of the 802.11 MAC and TWMA for different packet generation probabilities Pg from 0.1 to 1 i.e. the distribution of percent (%) of packets arriving their destinations over total number of all successful packets in the range of certain delay period, less than 5TP to more than 50TP (TP= length of a packet). If the retransmission times of a packet exceed maximum retransmission limit 3, if the transmission time of a packet exceeds the TTL (Time To Live), which is 4s in the simulation, and if the transmission of a packet fails because of collision, buffer overflow, and channel failure, the packet is dropped. In most cases, the simulation results confirm the performance as expected - that the proportion of packets over total number of all successful packets arriving to their destinations in less than a period of time, for example, 5TP or 10TP in TWMA is higher than in 802.11 MAC. In other words, the percentage of packets that arrives earlier over total success packets in TWMA is higher than that of 802.11 MAC.

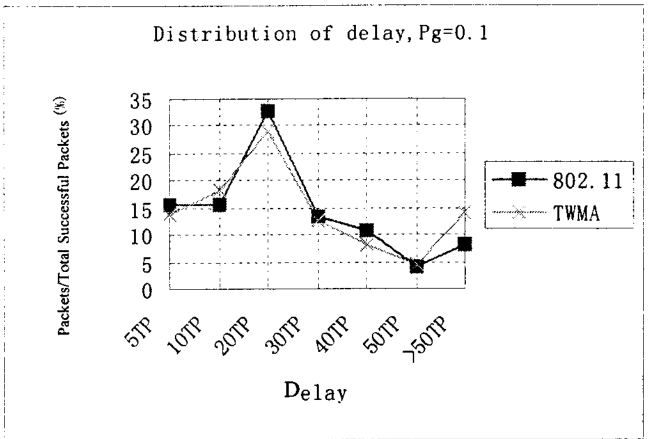

Figure 3.34 Distribution of Delay at Pg=0.1

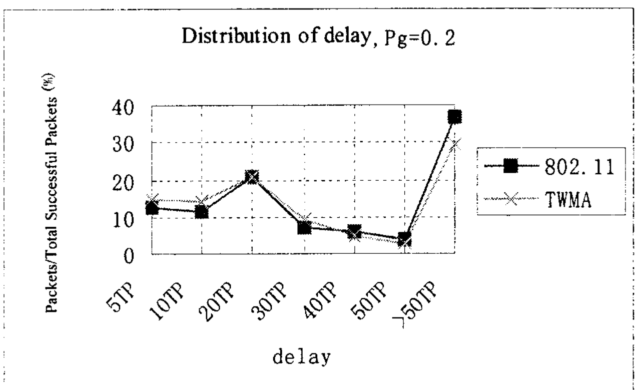

Figure 3.35 Distribution of Delay at Pg=0.2

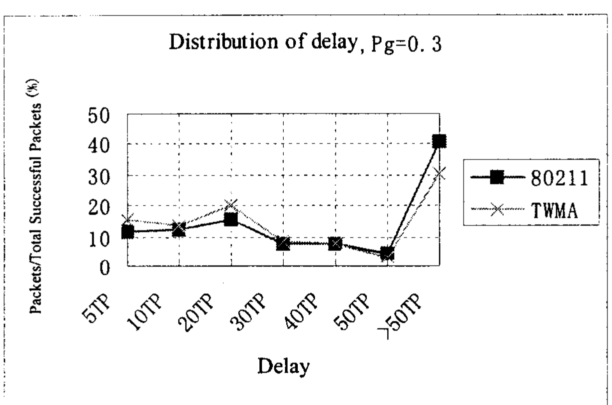

Figure 3.36 Distribution of Delay at Pg=0.3

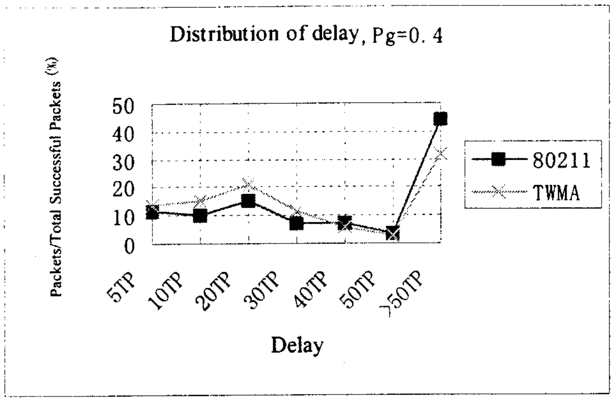

Figure 3.37 Distribution of Delay at Pg=0.4

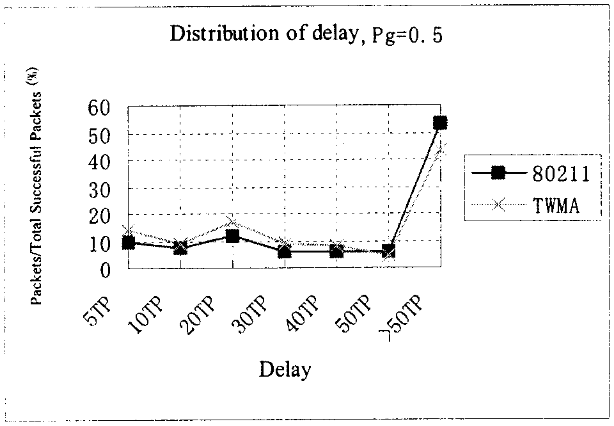

Figure 3.38 Distribution of Delay at Pg=0.5

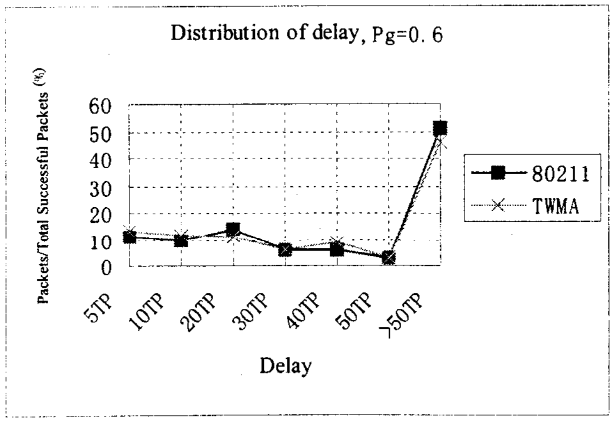

Figure 3.39 Distribution of Delay at Pg=0.6

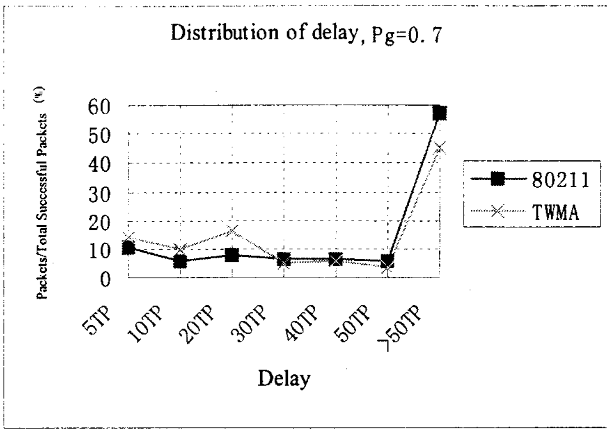

Figure 3.40 Distribution of Delay at Pg=0.7

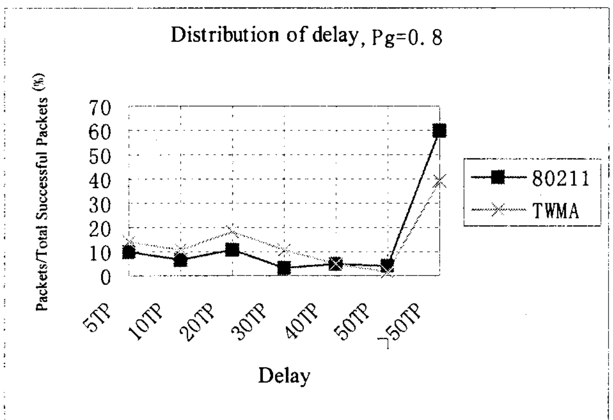

Figure 3.41 Distribution of Delay at Pg=0.8

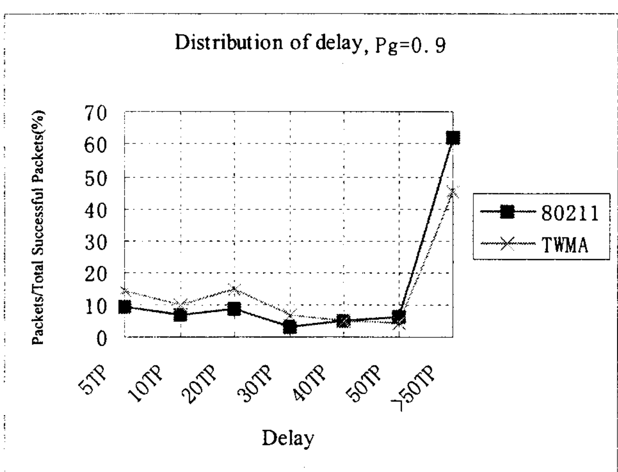

Figure 3.42 Distribution of Delay at Pg=0.9

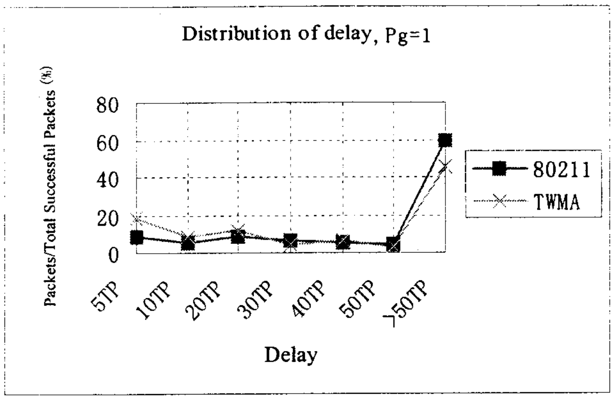

Figure 3.43 Distribution of Delay at Pg=1

3.4.2 Average End-to-end Queuing Delay

Figures 3.44 and 3.45 show the simulation results reflecting the relationship between queuing delays' increase along with the increase of buffer size and probability of packet generation Pg. In Figure 3.44, it is obvious that the queuing delay increases in proportion to the probability of packet generation and growth of stations' buffer size. As the probability of packet generation increases, for a certain buffer size, the number of queued packets in the buffer to be transmitted increases, therefore queuing delay increases. As the buffer size gets larger and stations' buffer have larger capacity to hold more packets, more packets in each buffer will increase the queuing delay as well. We can see that queuing delay is very small at the beginning when Pg is 0.1 for a fixed buffer size. When Pg value increases, the queuing delay increases as well until it reaches its top as Pg approaches 1.

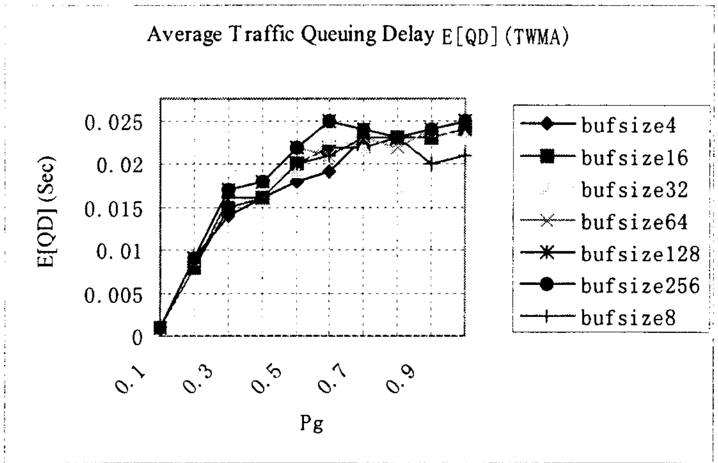

Figure 3.44 Average End-to-end Queuing Delay E [QD] Versus Probability of Packet Generation Pg

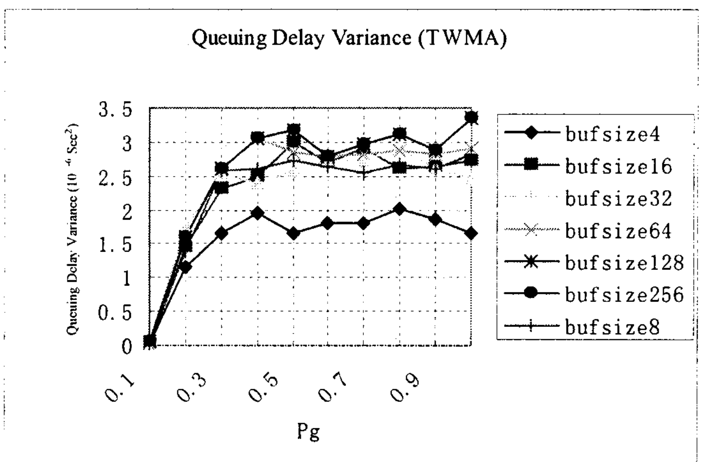

Figure 3.45 Queuing Delay Variance δ_{DQ}^2 Versus Probability of Packet Generation Pg

3.4.3 Average Buffer Overflow

Figures 3.46 and 3.47 depict the relationship between the average buffer overflow and traffic load changes represented by Pg. As the probability of packet generation Pg increases, i.e. more packets are generated, average buffer overflow increases. The curves in Figure 3.46 confirm that the average overflow increases almost linearly with the density of network traffic. On the other hand, it can be seen that when the buffer size of stations increases from 4 to 256, the average buffer overflow will be reduced dramatically. Average buffer overflow reaches its highest point when the probability of packet generation Pg is 1 for every buffer size from 4 to 64, a station with Pg=1 means that station generate a packet for every main iteration (a packet length). When buffer size is 128 or 256, no buffer overflow takes place. When the buffer size is 4 and Pg is 1, the number of buffer overflow reaches it's highest level around $9x10^{-5}$, i.e. every 10^5 time slots there would be average of 9 buffer overflows for each station. As we can see from Figure 3.47, the average buffer overflow variance has similar graphical tendancy as that of the average buffer overflow.

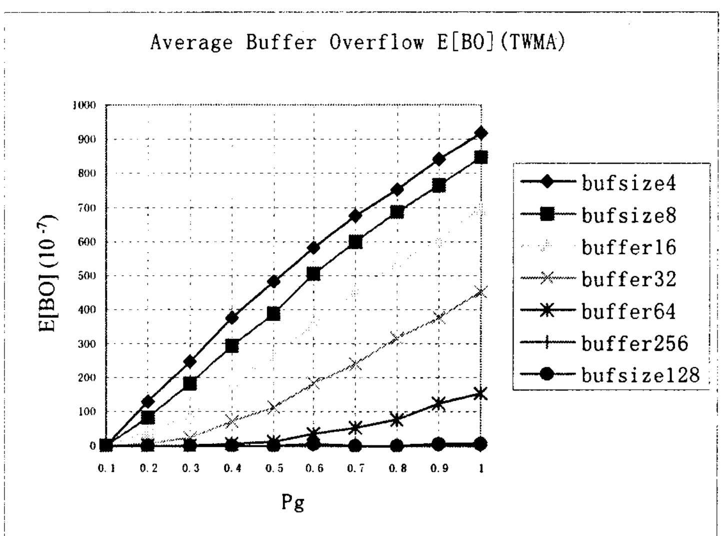

Figure 3.46 Average Buffer Overflow E [BO] VS Probability of Packet Generation Pg

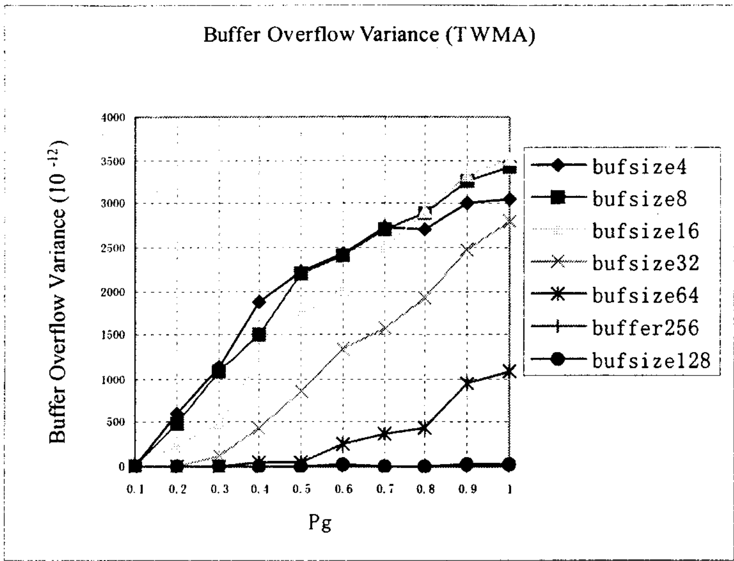

Figure 3.47 Buffer Overflow Variance δ^2_{BO} Versus Probability of Packet Generation Pg

3.4.4 Average Number of Dropped Packets

Figure 3.48 shows that the proportion of dropped packets out of the total generated packets during the simulation increases as the probability of packet generation Pg increases from 0.1 to 1 and the buffer size decreases from 256 to 4. The number of dropped packet is at its lowest level when Pg=0.1 and gradually increases as Pg increases. When Pg=1, the dropped packet number reaches its top. The reason is that as the probability of packet generation Pg grows, more packets participate in contention for accessing the shared resource, directly resulting in more lost packets because of collisions. When the station buffer size decreases, the buffer overflow of stations increases, which furthermore increases the number of dropped packets. The heavy traffic load increases the contention time for gaining the shared channel. As a consequence, more packets are dropped because more packets exceed the life time limit during transmission resulting from heavy traffic load. Figure 3.48 clearly explains the relationship between the cause and the effect as expected. Dropped packet variance has similar graphical pattern as the average number of dropped packet, as in Figure 3.49.

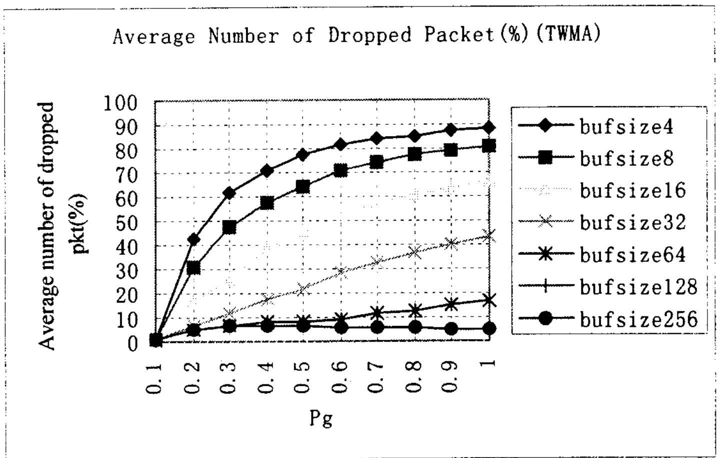

Figure 3.48 Number of Dropped Packet Over Total Packet Number Versus Probability of Packet Generation Pg

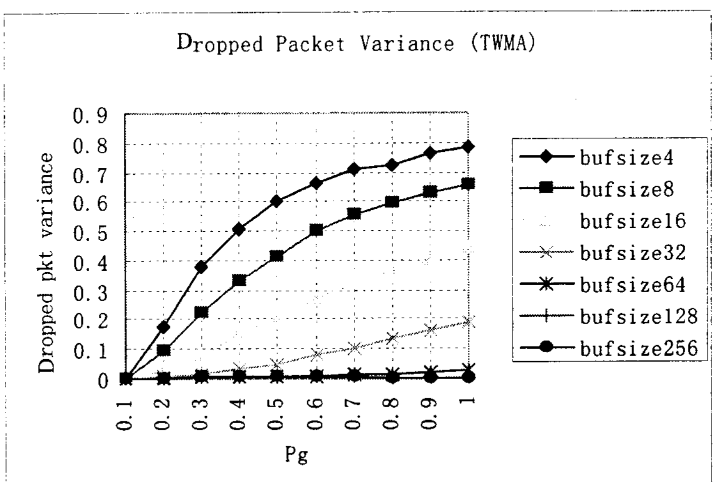

Figure 3.49 Dropped packets variance δ_{DP}^2 Versus Probability of Packet Generation Pg

3.4.5 Average Traffic Throughput

3.4.5.1 Throughput Comparison 1

Figure 3.50 depicts the tendency that the average traffic throughput decreases as probability of packet generation Pg for each station increases. This is because more packets are generated during the simulation as the probability of packet generation increases. Heavier traffic load certainly causes more collisions in contention for accessing the shared channel, more buffers will overflow, and longer queuing delays are experienced on the shared channel. On the other hand, a larger buffer size is likely to increase the queuing delay due to the fact that there are more queuing packets waiting to be sent in large buffers than in small buffers. It is easy to see that the performance measures are deeply related to the traffic channel load represented as Pg. Throughput has its highest point for each station with different buffer size from 4 to 256 when probability of packet generation Pg is 0.1. After that, the throughput decreases dramatically as the probability of packet generation Pg increases from 0.1 to 0.4, and then gradually decreases as Pg approaches 1. When probability of packet generation Pg reaches 1, the throughput reaches its lowest point for all different buffer size circumstances. The reason for this is that when Pg is 0.1, there are only a smallest number of stations (compared to other Pg values) participating in contending to access the shared channel, so there is a fewest number of collisions, buffer overflows, and queuing delays that cause packets being dropped. Therefore, the throughput gets its maximum value. When Pg is 1, there is the largest number of stations taking part in contending to access the shared channel, it causes serious traffic jams of more collisions, buffer overflows, and longer delays that result in large amount of dropped packets and furthermore reduce the throughput to its minimum. For different Pg values, the throughput in the situation where each station has largest buffer sizes 256 is higher than the case where a station has a buffer size in the range 4 to 128. The larger the buffer size a station has, the fewer packets are dropped because of buffer overflow, i.e. the more packets reach the destinations successfully, the larger the throughput becomes. Figure 3.50 shows that the throughput (i.e. number of successful packets over total number of generated packets) increase as the buffer size of each station increases. The reason for this is that the stations that have large buffer sizes reduce the buffer overflows resulting from limited buffer size and furthermore reduce the number of packets that are dropped at the sender or receiver or intermediate nodes.

Figures 3.52 to 3.58 show that: when buffer size is 4, TWMA has higher throughputs than IEEE 802.11 MAC for every Pg value in the range from 0.1 to 1. When the buffer size is 8, TWMA almost has higher throughputs than IEEE 802.11 MAC for every Pg value in the range from 0.1 to 1 except Pg=0.3. The result at Pg=0.3 maybe overcame by increasing the number of running simulation to get an average result. When the buffer size is 16, TWMA almost has higher throughputs than IEEE 802.11 MAC for every Pg value in the range from 0.1 to 0.3 and a slightly higher from 0.4 to 1. When the buffer size is 64, TWMA has higher throughputs than IEEE 802.11 MAC for Pg value is 0.1, 0.2, and 0.3 and a slightly higher in the range from 0.4 to 1. When the buffer size is 128, TWMA has a little bit higher throughputs than IEEE 802.11 MAC for every Pg value in the range from 0.1 to 1. When the buffer size is 256, TWMA has very similar throughputs to IEEE 802.11 MAC for each Pg value in the range from 0.1 to 1. In cases of different Pg values and buffer sizes, most of the throughputs of TWMA are higher than that of IEEE 802.11, a few have not significant differences or are lower. The key thing to notice in the graphs from Figures 3.52 to 3.58 is that the throughput degrades as Pg gets larger. This makes sense as explained earlier-as Pg increases, traffic becomes heavy and congested, and so there will be more collisions in contention for channel access, more dropped packets resulting from longer queuing delay, buffer overflows, and limit number of contentions for channel access after failure transmissions. From the results of throughput shown in the simulation, we can find that TWMA has better performance in most circumstances than IEEE 802.11 MAC in term of the average traffic throughput. The simulation confirms that with some optimum parameters combinations TWMA can achieve better and stable performance in term of both average traffic delay and average traffic throughput.

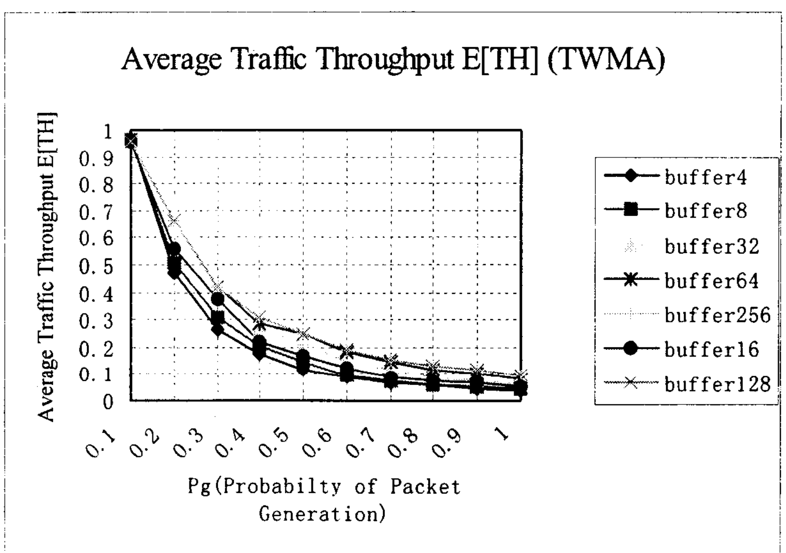

Figure 3.50 Average Traffic Throughput E [TH] Versus Probability Of Packet Generation

Pg

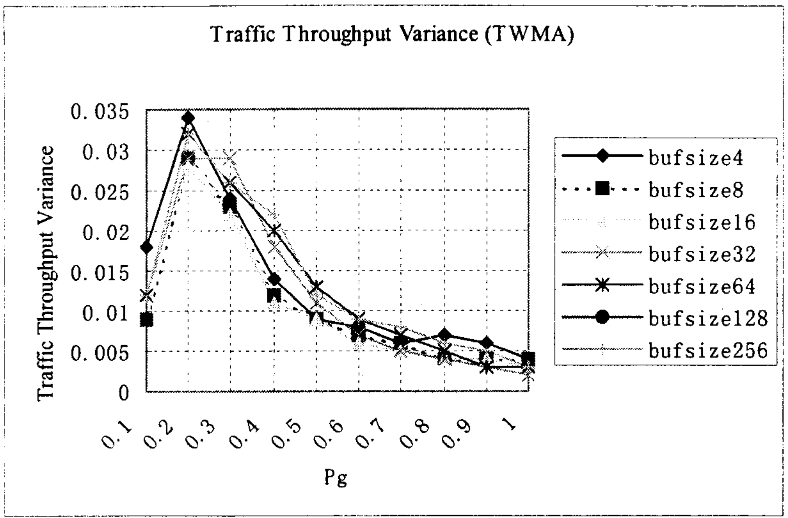

Figure 3.51 Average Traffic Throughput Variance δ_η^2 Versus Probability of Packet Generation Pg

The simulation environment for Figures 3.52 to 3.58: Total simulation time: 800000 time units. The channel success probability: Pc=1. Other input parameters are set as same as described earlier in chapter 2 except those specified in Figures. Average traffic throughput is plotted on the y-axis and probability of packet generation Pg on x-axis. Same x-y-axis settings are for Figures 3.52 to 3.58.

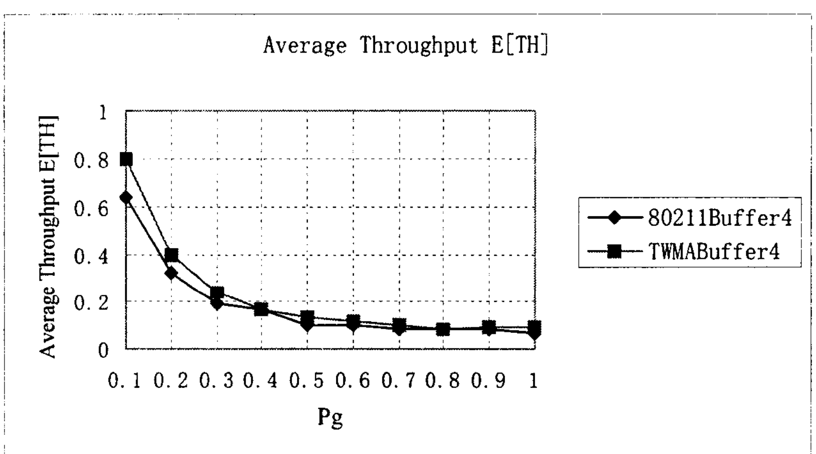

Figure 3.52 Throughput Comparison between TWMA and IEEE 802.11 MAC with

Different Pg in the range 0.1 to 1 at Buffer Size is 4

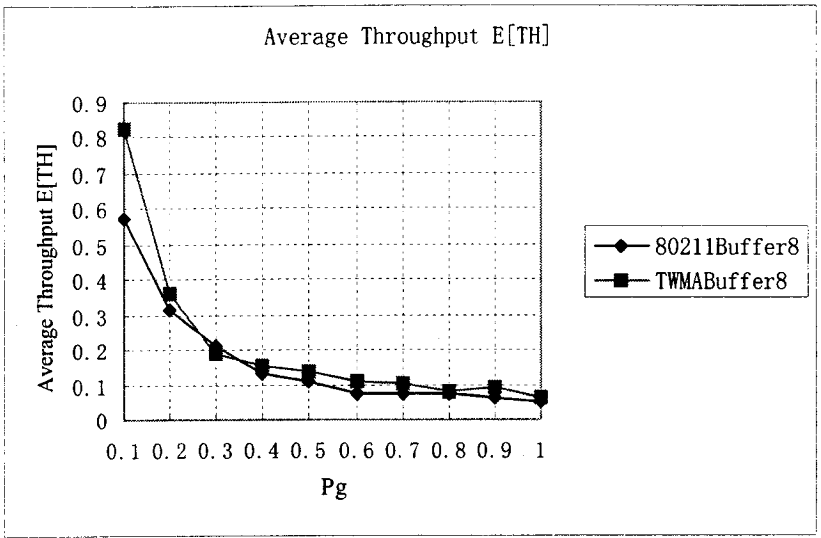

Figure 3.53 Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Pg in the range 0.1 to 1 at Buffer Size is 8

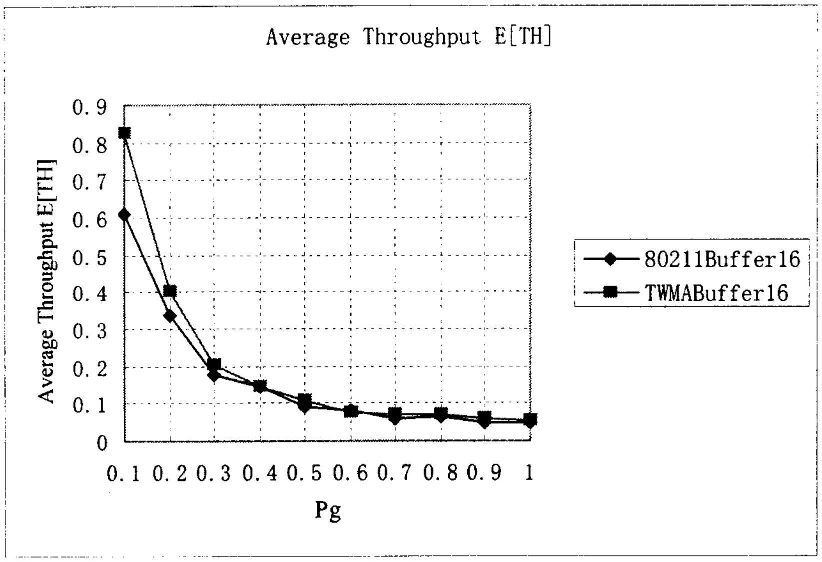

Figure 3.54 Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Pg in the range 0.1 to 1 at Buffer Size is16

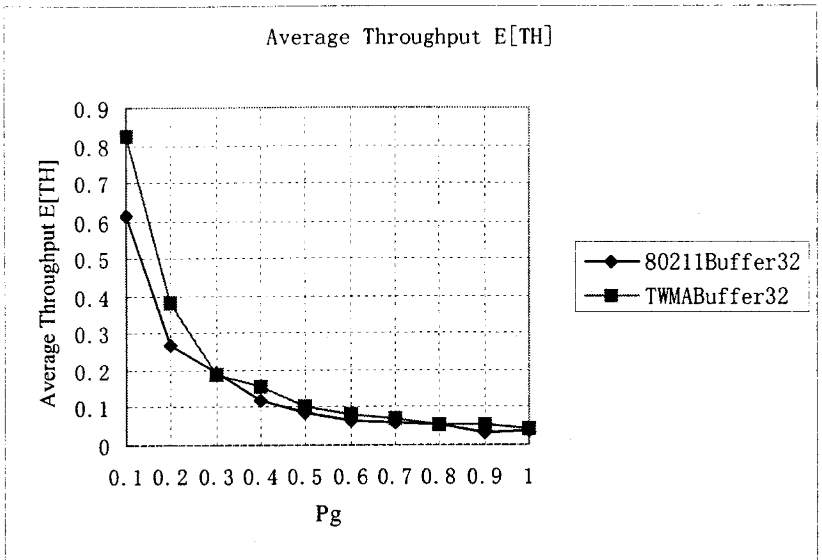

Figure 3.55 Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Pg in the range 0.1 to 1 at Buffer Size is 32

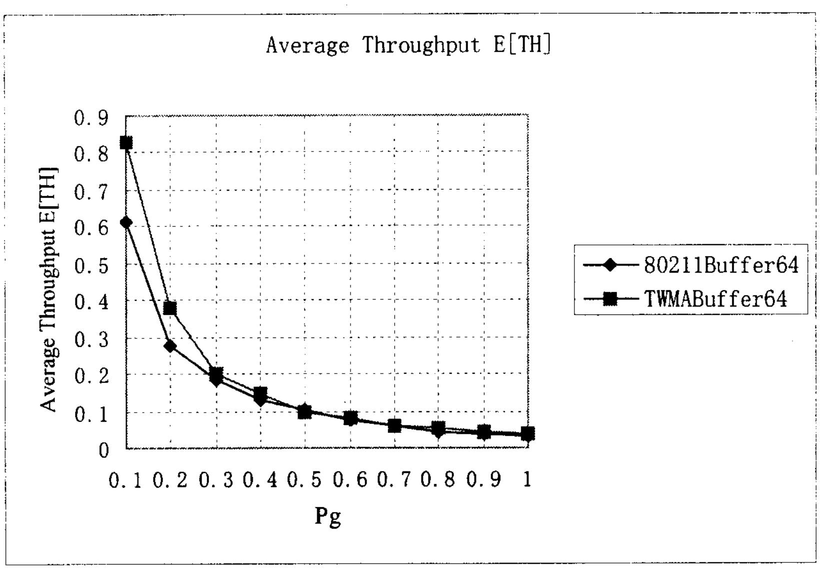

Figure 3.56 Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Pg in the range 0.1 to 1 at Buffer Size is 64

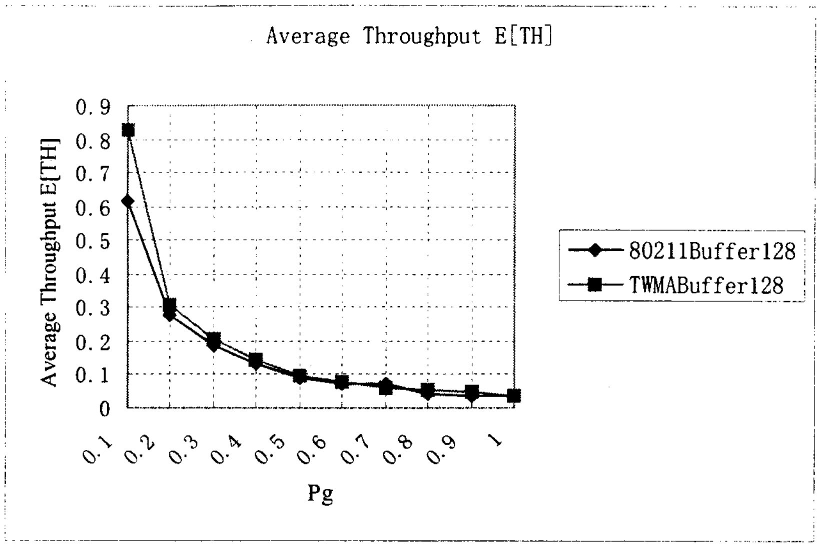

Figure 3.57 Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Pg in the range 0.1 to 1 at Buffer Size is 128

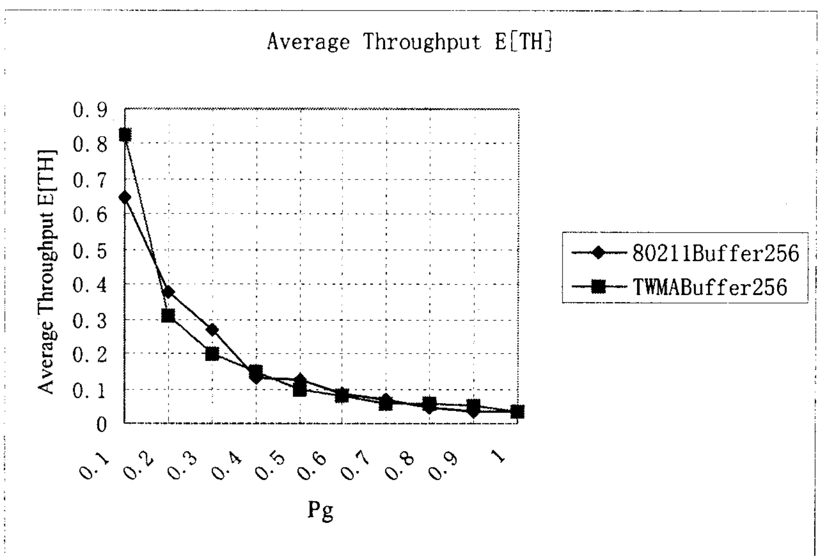

Figure 3.58 Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Pg in the range 0.1 to 1 at Buffer Size is 256

3.4.5.2 Throughput Comparison 2

In Figures 3.59 to 3.68, average throughput is plotted on the y-axis and buffer size on x-axis. The simulation setting is the same as stated earlier in chapter 2 except those specified. These Figures show that when probability of packet generation Pg gets different values in the range 0.1 to 1, the average throughput of TWMA has better performance than that of IEEE 802.11 MAC for most buffer-sizes in the range 4 to 256. For example, in Figure 3.59, when probability of packet generation Pg=1, TWMA average traffic throughput has better performance than that of IEEE 802.11 MAC for each different buffer size in the range 4 to 256. In Figure 3.64, when Pg=0.6, TWMA throughput has better or same performance than that of IEEE 802.11 MAC for buffer size 4, 8, 32, 64, 128 except buffer size 16, 256.

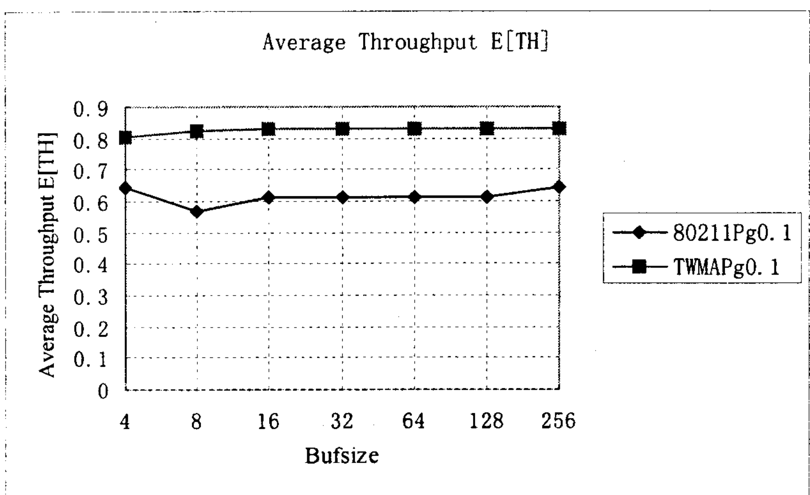

Figure 3.59 Average Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Buffer Size in the range 4 to 256 at Pg=0.1.

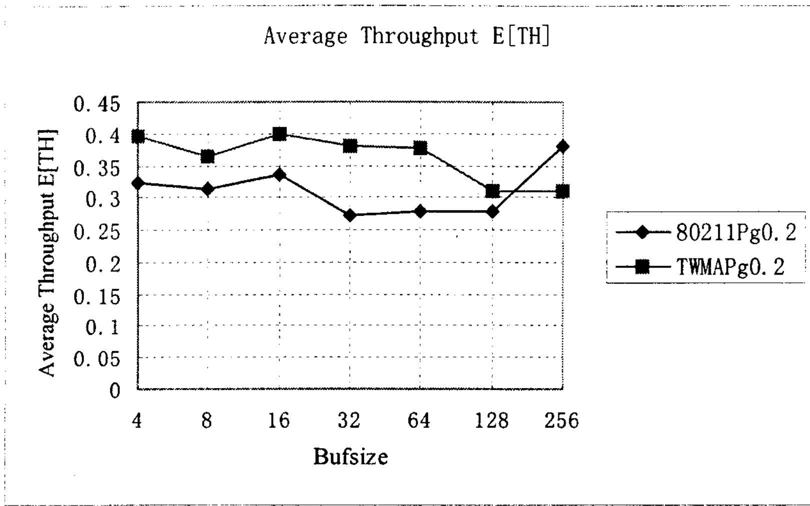

Figure 3.60 Average Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Buffer Size in the range 4 to 256 at Pg=0.2.

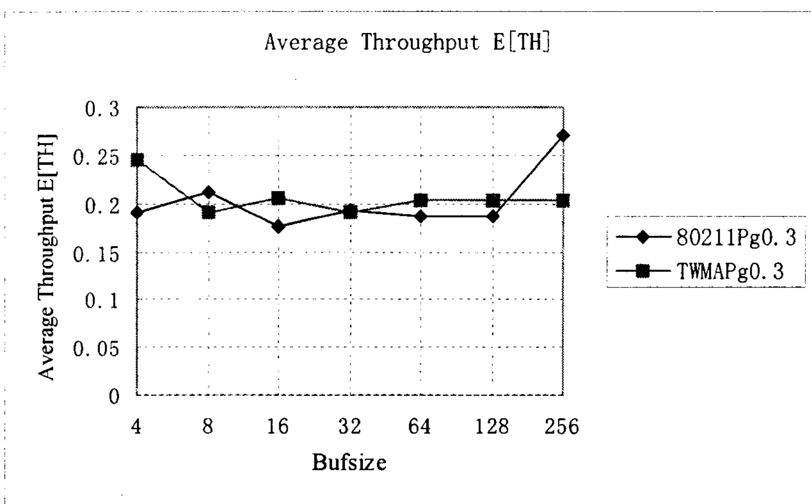

Figure 3.61 Average Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Buffer Size in the range 4 to 256 at Pg=0.3.

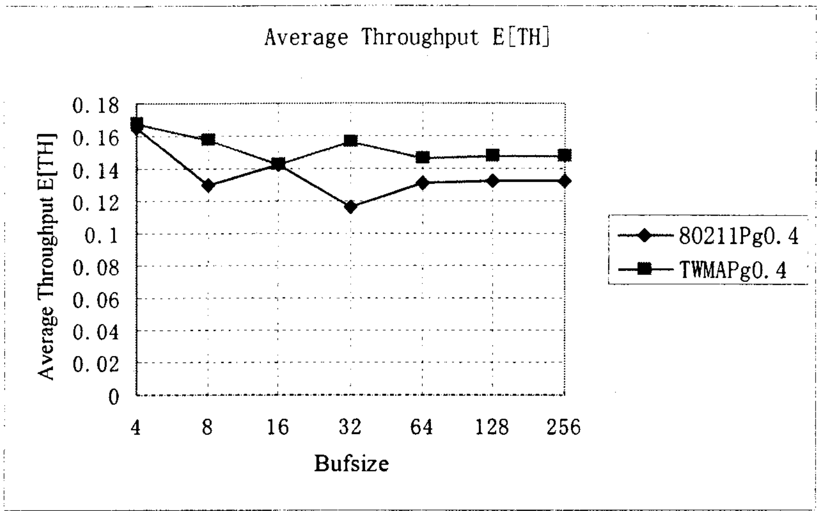

Figure 3.62 Average Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Buffer Size in the range 4 to 256 at Pg=0.4.

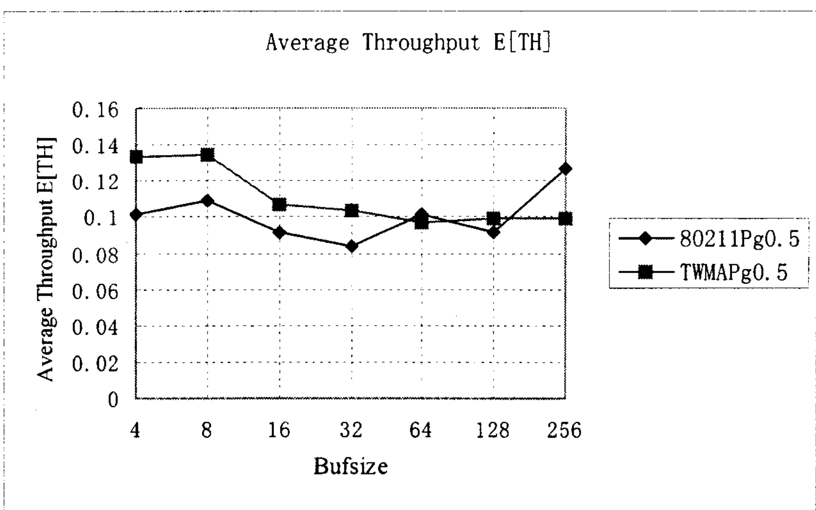

Figure 3.63 Average Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Buffer Size in the range 4 to 256 at Pg=0.5.

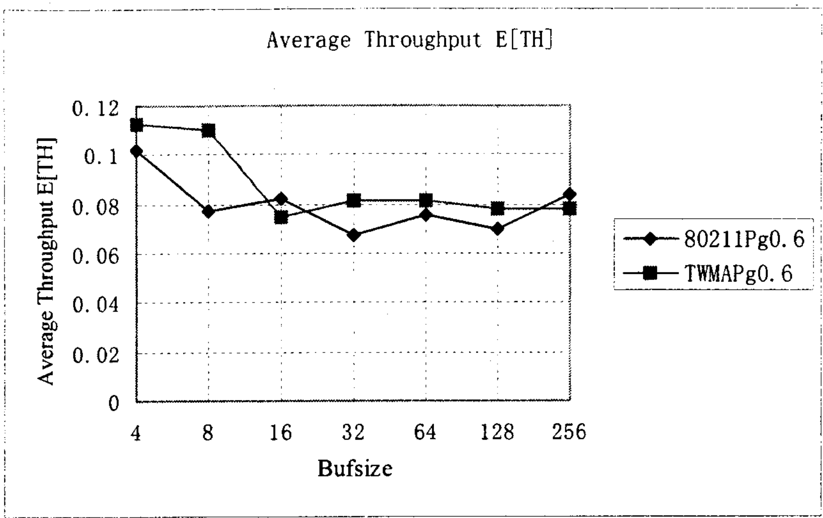

Figure 3.64 Average Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Buffer Size in the range 4 to 256 at Pg=0.6.

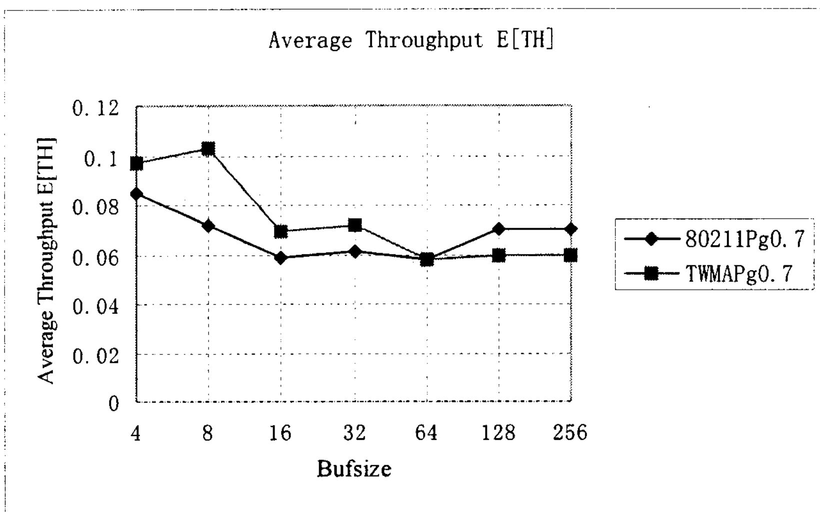

Figure 3.65 Average Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Buffer Size in the range 4 to 256 at Pg=0.7.

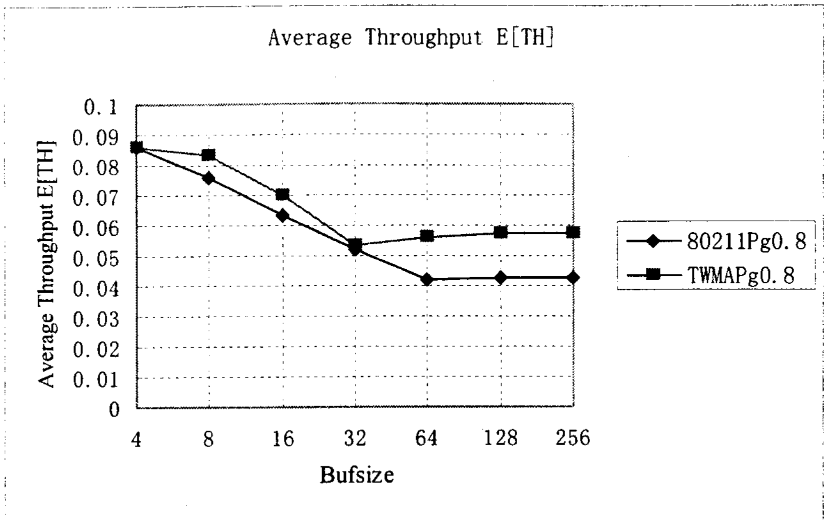

Figure 3.66 Average Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Buffer Size in the range 4 to 256 at Pg=0.8.

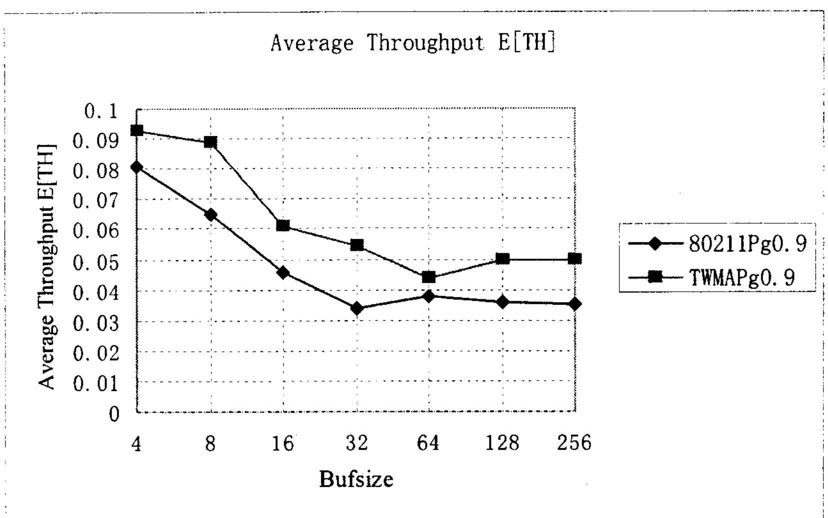

Figure 3.67 Average Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Buffer Size in the range 4 to 256 at Pg=0.9.

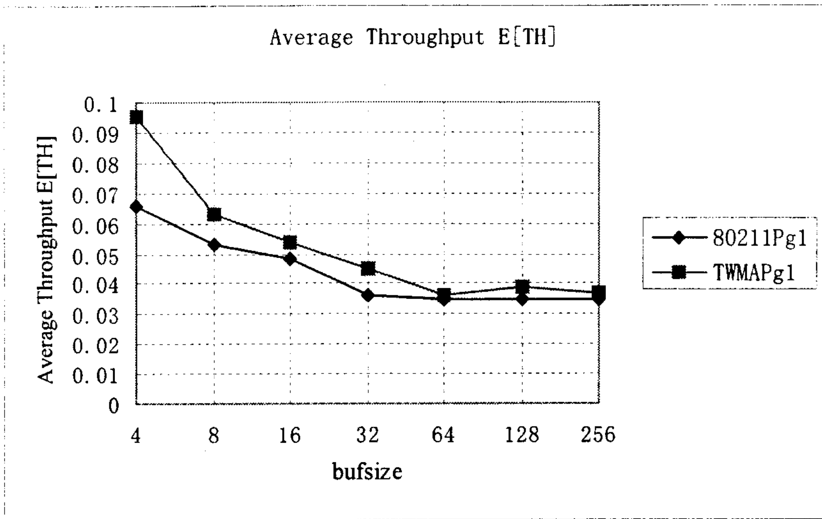

Figure 3.68 Average Throughput Comparison between TWMA and IEEE 802.11 MAC with Different Buffer Size in the range 4 to 256 at Pg=1.

Chapter 4 Conclusions and Future Consideration

4.1 Conclusion

The objective of the research reported in this thesis is to investigate the proposed method of improving the fairness in medium access and the efficient utilization of bandwidth of the shared medium under control of DCF in wireless ad hoc networks. In the thesis, both TWMA and IEEE 802.11 MAC models have been simulated and studied in order to make a comparison between them in term of latency and throughput. In order to realize the objective, two windows contention algorithm (TWMA) is proposed to deal with the fairness of medium access and efficient utilization of the shared medium among users. In TWMA, a parameter channel status indicator (CSI) is introduced to represent adjacent channel status of an active user in order to adjust contention window size to minimize the idle waste and reduce collisions in backoff contention procedure. A typical station updates its CSI every time before starting contention for channel access. Two windows contention is deployed to solve the fairness problem in contention for channel access, in other words, to balance the sharing of the medium among users who enter as new comers and old users who had experienced successful transmission of data or failures. Two window sizes, increase of the first contention window size, and W_{th} are related to the CSI value so the window size is determined by the adjacent channel status. When the traffic load increases, the window size increases to reduce the collisions according to the adjacent channel status, and when the traffic becomes light, the window size decreases to reduce the waste of idle slots. It is shown in the simulation the proposed algorithm finally increases the traffic throughput and decreases the end-to-end latency as the results of reduction of the idle slots and collisions wasted in the backoff contention procedure.

The simulation results present end-to-end traffic delay, queuing delay, buffer overflow, dropped packets, and throughput performance of TWMA and comparisons with IEEE 802.11 MAC scheme. The performance of TWMA with various input parameters are evaluated and optimal combination of input parameters has been found. The results show that the backoff algorithm in MAC is of major important to the performance of the MAC protocol. The main difference between TWMA and IEEE 802.11 MAC is the modification of the backoff algorithm. The performance with optimal parameters results confirm that the proposed TWMA is effective in achieving higher throughput and lower latency performance than that of IEEE 802.11 standards MAC algorithm. From the delay observation, the end-to-end traffic delays in TWMA are less than those in IEEE 802.11 in all cases where the buffer size is in the range 4 to 256 and Pg in the range 0.1 to1. The percentage of packets arriving the destination in less than 5TP in TWMA is more than that in IEEE 802.11 in 90% of all cases. Though, whether or not it is better to use TWMA for time-bounded packet transfer than IEEE 802.11 MAC is left for further investigation because the advantage is not so noticeable as we see from those results. For the throughput comparison between TWMA and IEEE 802.11, the throughputs are better than those in IEEE 802.11 in 81% of all cases where buffer size is in the range 4 to 256 and probability of packet generation Pg is in the range 0.1 to1.

The proposed TWMA is compatible with IEEE 802.11 standards MAC protocols without adding much complexity and major modification to IEEE 802.11 standards.

4.2 Future Consideration

Because the thesis mainly focuses on the MAC algorithm, there are several aspects being simplified, like deploying uniform packet size that is larger than RTS threshold. In order to be closer to practical wireless situation, variable data packets should be taken into consideration and investigated. To be noticed is that only unicast is applied in this simulation, multicast transmissions shall be studied too if the simulation is to be applied to more general practical cases and circumstances. PCF shall be studied if the simulation program is to be extended to the wired infrastructure of wireless LANs through access points.

The same simulation model can be used to get the performance results of different number of users with different Pg values without any modification to the simulation model except a few input parameter settings.

Wireless LAN will continue to play increasing and essential important roles in realizing its functionalities of mobility, relocation, and ad hoc networking requirements and a method to cover areas where are difficult for wired networks. With great efforts to improve data rates and efficiency of media utilization, the future of wireless LAN is unlimited.

References

[1] Neeli Prasad, Anand Prasad, "WLAN Systems and Wireless IP for Next Generation Communications". 2002

[2] B.P.Crow, I.Widjaja, J.G.Kim, P.T.Sakai, "IEEE 802.11 Wireless Local Area Networks". 1997

[3] B. Bensaou, Y.Wang, C.C.Ko. "Fair medium access in 80211 based wireless ad-hoc networks" 2000

[4] V. Bharghavan, A. Demers, S.Shenker, L.Zhang "MACAW: a medium access protocol for wireless LANs". 1994

[5] P. Karn. "MACA: A new channel access method for packet radio".

[6] T. ozugur, M. Naghshineh,P. Kermani, J.A. Copeland "Fair media access for wireless LANs." 1999

[7] T. Nandagopal, T.E. Kim, X. Gao, V. Bharghavan. "Achieving MAC layer fairness in the wireless packet networks."

[8] T. Ozugur, M. Naghshineh, P. Kermani, C. Olsen, B. Rezvani, J. Coperland, "Balanced media access methods for wireless networks."

[9] N.H.Vaidya, P. Bahl, "Fair scheduling in broadcast environments" 1999

[10] Y. Kwon, Y. Fang, H. Latchmen, "Fast collision resolution MAC algorithm for wireless LAN". 2002

[11] ANS/IEEE 802.11,1999 Edition, "Wireless LAN medium access control (MAC) and physical layer (PHY) specifications."

[12] I. Aad, C. Castelluccia, "Differentiation mechanisms for IEEE 802.11" 2001

[13] H. Wu, Y. Peng, K. Long, S. Cheng, J. Ma, "Performance of reliable transport protocol over IEEE 802.11 wireless LAN: analysis and enhancement". 2002

[14] Mario. JN, IT Lu, "Spread spectrum medium access protocol with collision avoidance in mobile ad-hoc wireless network" 1999

[15] F. Calli, M. Conti, E. Gregori, "IEEE 802.11 wireless LAN: capacity analysis and protocol enhancement". 1998

[16] P. Brenner, "IEEE 802.11 protocol", 1997

[17] X. yang, N H. Vaidya, "DSCR: A more stable MAC protocol for wireless networks"

[18] William Stallings "IEEE 802.11: Moving closer to practical wireless LANs ", 2001

[19] Juha A., Jouni M., Jyri R. "Wireless LAN access network architecture for mobile operators" 2001

[20] Anurag D., Nitin V., Paramvir B. "Priority and fair scheduling in a wireless LAN" 2001

[21] Geng sheng K., Po-Chang K. "A collision-free medium access control protocol for flow oriented ad hoc wireless LAN"

[22] X.Yang, Nitin H.V. "Pipelined packet scheduling in wireless LANs", 2002

[23] Timucin Ozugur, "Optimal MAC-layer Fairness in 802.11 Networks" 2002

[24] D. Qiao, S.Choi, "Goodput enhancement of IEEE 802.11a wireless LAN via link adaptation", 2001

[25] H. Woesner, J. Weinmiller, J. Ebert, A. Wolisz, "Modified backoff algorithm for DFWMAC's distributed coordination function"

[26] S. Sheu, Y. Lee, M. Chen, "Providing multiple data rates in infrastructure wireless

networks", 2001

[27] V. Vitsas, A.C. Boucouvalas, "Performance analysis of the collision avoidance procedure of the advanced infrared (AIr) CSMA/CA protocol for wireless LANs", 2002

[28] A. Spyropoulos, C.Raghavendra, "A token-based greedy chain scheduling algorithm for situation aware wireless LANs", 2002

[29] G M Lundy, M. Almquist, T. Oruk, "Specification, verification and simulation of a wireless LAN protocol: MACAW"

[30] P. Liu, "IEEE 802.11 wireless LAN: can we use it for multimedia", 1998

[31] A.G. Savvides, "Designing and prototyping a wireless point-to-point link with multimedia application support", 2000

[32] K.Kim, A.Ahmad, Kiseon Kim, "A wireless multimedia LAN architecture using DCF with shortened contention window for QoS provisioning", 2002

[33] Y.Wang, B. Bensaou, "Achieving fairness in IEEE 802.11 DFWMAC with variable packet lengths", 2001

[34] V. Hayes, "Tutorial on 802.11 to 802", 1996

[37] W. Ajib, P. Godlewski, "A proposal of an access persistence protocol over data wireless networks", 2000

[38] C. Lin, J. Liu, "Bandwidth routing in ad hoc wireless networks", 1998

[39]J. K-Jush, Peter S., "Structure and performance of the HIPERLAN/2 physical layer", 1999

[40] R.V. Nee, G. Awater, M. Morikura, H.Takanashi, M. Webster, K. Halord, "New high-rate wireless LAN standards", 1999

[41] R. V. Nee, "A new OFDM standard for high rate wireless LAN in the 5 GHz band"

[42] A. Ganz, Kitti W., "IEEE 802.11 wireless LAN association procedure for multimedia applications", 1999

[43] E. M. Royer, C.Toh, "A review of current routing protocols for ad hoc mobile wireless networks", 1999

[44] A.L-Garcia, "Communication networks", 2000