WIRELESS NETWORKS PERFORMANCE STUDY

A THESIS SUBMITTED TO
THE GRADUATE SCHOOL OF NATURAL AND APPLIED SCIENCES
OF
MIDDLE EAST TECHNICAL UNIVERSITY

BY

YAKUP TAŞDELEN

IN PARTIAL FULFILLMENT OF THE REQUIREMENTS FOR THE DEGREE
OF
MASTER OF SCIENCE
IN
ELECTRICAL AND ELECTRONICS ENGINEERING

AUGUST 2004

Approval of the Graduate School of Natural and Applied Sciences

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ABSTRACT

WIRELESS NETWORKS PERFORMANCE STUDY

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August 2004, 63 pages

This thesis evaluates the performance of the 802.11 wireless network. The newly defined DCF+ protocol performance is studied and compared with the DCF protocol under different traffic conditions. A service differentiation mechanism is also applied to both protocols for basic access mechanism and the efficiency of the protocols is examined. As an additional work, the same study is performed for RTS/CTS access scheme and efficiency of the protocols is examined

As a result of this study, it is shown that the DCF+ protocol provides performance enhancement to the DCF protocol under different traffic conditions but fails to provide service differentiation because of its structure.

Keywords: Wireless networks, DCF+.

ÖZ

KABLOSUZ AĞLARDA PERFORMANSIN DEĞERLENDİRİLMESİ

Taşdelen, Yakup
Y. Lisans, Elektrik ve Elektronik Mühendisliği Bölümü
Tez Yöneticisi : Prof. Dr. Semih Bilgen

Ağustos 2004, 63 sayfa

Bu çalışma, 802.11 kablosuz ağların performansını incelemiştir. Yeni tanımlanmış DCF+ protokolünün performans değerleri incelenmiş ve farklı trafik koşullarında DCF protokolü ile karşılaştırılmıştır. Servis ayrım mekanizması her iki protokol için de uygulanarak, temel erişim metodunda, protokollerin servis ayrım etkinlikleri incelenmiştir. İlave çalışma olarak, RTS/CTS erişim metodunda aynı çalışma yapılarak her iki protokolün servis ayrım etkinlikleri birbiriyle karşılaştırılmıştır.

Bu çalışmanın sonucu olarak, DCF+ protokolünün, DCF protokolüne kıyasla performans iyileştirmesi sağladığı, ancak yapısından dolayı servis ayrımında başarısız olduğu gösterilmiştir.

Anahtar sözcükler: Kablosuz ağlar, DCF+.

To My Parents

ACKNOWLEDGEMENTS

I wish to express my deepest gratitude to my supervisor Prof. Dr. Semih Bilgen for his guidance, advices, criticism and encouragements and insight throughout the research.

My special thanks and appreciation goes to my family who has shared the entire burden of the study with me. Without their encouragement and support it would not be possible for me to complete this study.

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LIST OF ABBREVIATIONS AND ACRONYMS

ACK	Acknowledgement
ADB	Age Dependent Backoff
AP	Access Point
BSS	Basic Service Set
CFB	Contention-Free Burst
CSMA	Carrier Sense Multiple Access
CSMA/CA	Carrier Sense Multiple Access with Collision Avoidance
CW	Contention Window
DCF	Distributed Coordination Function
DFWMAC	Distributed Foundation Wireless MAC
DIFS	Distributed Interframe Space
DS	Distribution System
DSM	Distribution System Medium
DSS	Distribution System Service
DSSS	Direct Sequence Spread Spectrum
EDCF	Enhanced Distributed Coordination Function
ESS	Extended Service Set
FAMA	Floor Acquisition Multiple Access
FHSS	Frequency Hopping Spread Spectrum
HCF	Hybrid Coordination Function
IBSS	Independent Basic Service Set
IP	Internet Protocol
LAN	Local Area Network
MAC	Medium Access Control
MACA	Multiple Access with Collision Avoidance
PCF	Point Coordination Function
PDF	Probability Distribution Function
PHY	Physical Layer
QoS	Quality of Service

RTS/CTS	Request-to-Send/Clear-to-Send
SIFS	Short Interframe Space
STA	Station
UDP	Unified Data Protocol
TCP	Transmission Control Protocol
WLAN	Wireless Local Area Network
WM	Wireless Medium

CHAPTER I

INTRODUCTION

Wireless local area networking (WLAN) is a very dynamic field. Distributed contention-based medium access control (MAC) protocol research in wireless networks started with ALOHA and slotted ALOHA in the 1970s. Later, MACA, FAMA and DFWMAC were proposed by incorporating the carrier sense multiple access (CSMA) techniques as well as the RTS and CTS handshaking mechanism for collision avoidance [17]. Technological and regulatory developments have allowed the problems of high prices, low data rates and licensing requirements to be solved and the popularity of wireless LANs has grown significantly over the past few years [9]. Wireless networks face a trend of exponential traffic increase and growing importance to users. In some countries, such as Finland, the number of mobile subscriptions has exceeded the number of fixed lines [11].

To deal with the needs of wireless users, various wireless communication standards have been developed [1]. Currently, the IEEE 802.11 specifications are detailed and include both the MAC and the Physical Layer (PHY). The MAC incorporates two medium access methods, Distributed Coordination Function (DCF) and Point Coordination Function (PCF). DCF is an asynchronous data transmission function and it is appropriate for delay insensitive data. On the other hand, the optional PCF is designed for collision free and real-time applications [9]. There are two techniques used for packet transmittion in DCF. The default scheme is called basic access scheme and it is a two-way handshaking mechanism. In this scheme, a positive MAC acknowledgement (ACK) is transmitted by the destination station to confirm the successful packet transmission. The other optional scheme is a four-way handshaking mechanism, which uses request-to-send/clear-to-send (RTS/CTS) technique to reserve the channel before data transmission.

In the literature, several papers have studied the performance of the IEEE 802.11 protocol using analytical models or by means of simulation [2] [7] [8] [9] [10] [11] [12] [13]. Additionally, much research has been focused on improving the performance of the 802.11 MAC [15] [16] [17] [18] [19] [20]. To improve the performance of the IEEE 802.11 wireless network, some changes to the DCF protocol is proposed [3] [27] [28]. Another focus of research is service differentiation. In this subject, the way of providing different services to the different classes and the effectivity of the proposed solutions are discussed. The proposed solutions depend on making some changes to the PHY parameters such as SIFS duration and DIFS duration [22] [23] and [24]. For service differentiation, assigning a weight to each mobile station is also proposed [21]. Based on the demand of wireless users for receiving better services than the "best effort" services, IEEE has prepared and issued the enhanced DCF (EDCF) protocol which is a prioritization enhancement of the legacy 802.11 DCF. The performance and the possible improvements to the EDCF are discussed in [29] [30] [31].

In this thesis, the performance of DCF and new proposed protocol DCF+ are evaluated through simulation, based on the parameters in [2] and [3]. Based on these values, the performance improvement of DCF+ protocol is discussed. Also the DCF+ protocol is simulated under a disaster scenario [4]. Finally, the DCF+ protocol is implemented for a service differentiation application and the service differentiation efficiency of the DCF+ protocol is observed. The goal of the work is to evaluate DCF+ protocol in different aspects. For that reason, the performance of the DCF+ protocol under different traffic conditions is examined. The efficiency of the DCF+ in service differentiation applications is also discussed.

The outline of the thesis is as follows:

Chapter 2 gives detailed information about the DCF and DCF+ protocols and provides brief information about the studies on performance evaluation and performance improvement of the IEEE 802.11 wireless network. Chapter 2 also covers the studies on service differentiation and quality of service (QoS) enhancement.

Chapter 3 presents the simulation results and comparison of DCF and DCF+ protocols' throughput values. It also presents the simulation of DCF+ under disaster scenario and service differentiation application. The values for DCF+ obtained through the simulation and the values taken as a reference for DCF are also compared and the results are discussed.

Chapter 4 concludes the study and discusses some possible future work that can be added on this study.

CHAPTER II

LITERATURE SURVEY

2.1 802.11 Protocol for Wireless Local Area Networks

The study group 802.11 was formed under IEEE Project 802 to recommend an international standard for WLAN's and the first version of the standard was issued in 1999.

Wireless networks have fundamental characteristics that make them significantly different from traditional wired LANs. In wired LANs, an address is equivalent to a physical location and this is taken into account in the design of wired LANs. In IEEE 802.11, the addressable unit is a station (STA). The STA is a message destination but not (in general) a fixed location. [1]

2.1.1 The Effects of The Media on Design

The physical layer used in IEEE 802.11 is fundamentally different from wired media. The IEEE 802.11 PHYs;

a. Use a medium that has neither absolute nor readily observable boundaries.
b. Are unprotected from outside signals.
c. Communicate over a medium significantly less reliable than wired PHYs.
d. Have dynamic topologies.
e. Lack full connectivity. So the assumption normally made that every STA can hear every other STA is invalid.
f. Have time-varying and asymmetric propagation properties [1].

2.1.2 The Impact of Handling Mobile Stations

One of the requirements of IEEE 802.11 is to handle “mobile” as well as “portable” stations [1]. A portable station can be defined as a station that moves from location to location, but that is only used while at a fixed location. Mobile stations actually access the LAN while in motion.

It is not sufficient to handle only portable stations. Propagation effects blur the distinction between portable and mobile stations. Stationary stations often appear to be mobile due to propagation effects in wireless network.

2.1.3 Components of the IEEE 802.11 Architecture

There are several components in IEEE 802.11 architecture that interact to provide a wireless LAN that support station mobility transparently to the upper layers.

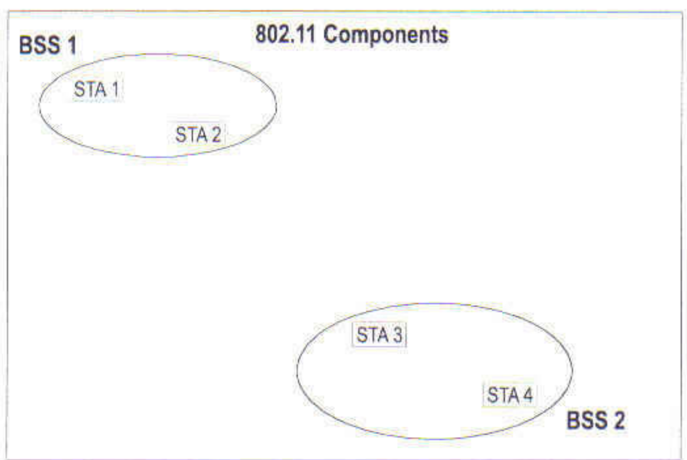

Figure 2.1 Basic Service Sets [1]

The independent basic service set (IBSS) is the most basic type of IEEE 802.11 LAN. A minimum IEEE 802.11 LAN may consist of only two stations. Figure 2.1

shows two IBSSs. This mode of operation is possible when IEEE 802.11 stations are able to communicate directly. The association between a STA and a BSS is dynamic. To become a member of an infrastructure BSS, a station shall become “associated”. These associations are dynamic and involve the use of the distribution system service (DSS) [1].

2.1.4 Distribution System Concepts

A BSS may form a component of an extended form of network that is built with multiple BSSs. The architectural component used to interconnect BSSs is the distribution system (DS). IEEE 802.11 logically separates the wireless medium (WM) from the distribution system medium (DSM). Each logical medium is used for different purposes, by a different component of the architecture and multiple media be either the same or different [1].

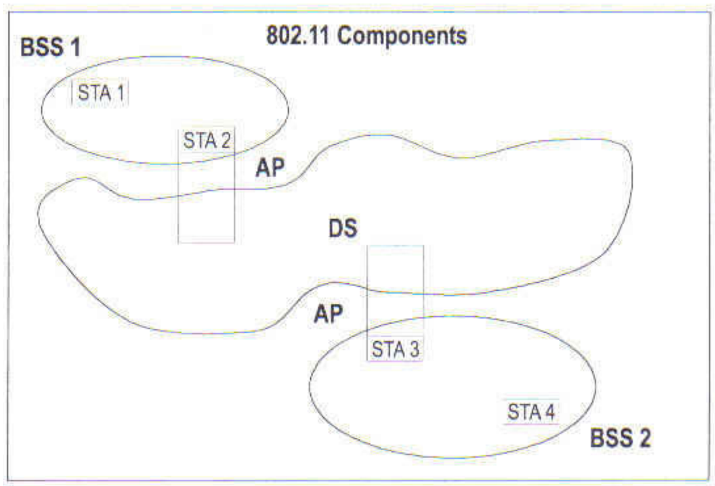

Figure 2.2 802.11 Distribution Systems and Access Points. [1]

The DS enables mobile device support by providing the logical services necessary to handle address to destination mapping and seamless integration of multiple BSSs. An access point (AP) is a STA that provides access to the DS by providing

DS services in addition to acting as a STA. Figure 2.2 adds the DS and AP components to the IEEE 802.11 architecture picture.

2.1.5 Extended Service Set (ESS): The Large Coverage Network

The DS and BSSs allow IEEE 802.11 to create a wireless network of arbitrary size and complexity. Stations within an ESS may communicate and mobile stations may move from one BSS to another (within the same ESS). To integrate the IEEE 802.11 architecture with a traditional wired LAN, a final logical architectural component is introduced-a portal. All data from non-IEEE 802.11 LANs enter the IEEE 802.11 architecture via a portal. The portal provides logical integration between the IEEE 802.11 architecture and existing wired LANs [1].

2.2 802.11 Distributed Coordination Function

2.2.1 General

In the 802.11 protocol, the fundamental mechanism to access the medium is called Distributed Coordination Function (DCF). DCF is a random access scheme, based on carrier sense multiple access with collision avoidance (CSMA/CA) protocol. Retransmission of the collided packets is managed according to the binary exponential backoff rules.

DCF describes two techniques to employ for packet transmission. The default scheme is a two-way handshaking technique, which is called basic access mechanism. This mechanism is characterized by the immediate transmission of a positive acknowledgement (ACK) by the destination station, upon successful reception of a packet transmitted by the sender station. Transmission of an ACK is required. Because, in the wireless medium, a transmitter cannot determine if a packet is successfully received or not, by listening to its own transmission [1] [2].

In addition to the basic access, an optional four way handshaking technique is introduced as RTS/CTS mechanism has been standardized. Before transmitting a packet, a station operating in RTS/CTS mode “reserves” the channel by sending a special Request-To-Send short frame. The destination station acknowledges the receipt of an RTS frame by sending back a Clear-To-Send frame, after which normal packet transmission and ACK response occurs. Since collision may occur only on the RTS frame, and it is detected by the lack of CTS response, the RTS/CTS mechanism allows increasing the system performance by reducing the duration of a collision when long messages are transmitted. As an important potential advantage, the RTS/CTS scheme designed in the 802.11 protocol is suited to combat the so-called problem of hidden terminals, which occurs when pairs of mobile stations result to be unable to hear each other [2] [26].

2.2.2 DCF Protocol-Working Principles and Details

A station, which has a new packet to transmit, monitors the channel activity. If the channel is idle for a period of time equal to a distributed interframe space (DIFS), the station transmits. Otherwise, if the channel is sensed busy (either immediately or during the DIFS), the station persists to monitor the channel until it is measured idle for a DIFS. After sensing the channel is idle for a DIFS, the station generates a random backoff interval before transmitting (collision avoidance feature of the protocol), to minimize the probability of collision with other packets that are being transmitted by other stations. Additionally, to avoid channel capture (to maintain the fairness between the stations), a station must wait a random backoff time between two consecutive new packet transmission, even if the medium is sensed idle in the DIFS time.

For efficiency reasons, DCF employs a discrete-time backoff scale. The time immediately following an idle DIFS is slotted, and a station is allowed to transmit only at the beginning of each slot time. The slot time size is set equal to the time needed at any station to detect the transmission of a packet from any other station. As given in Table 2.1, the slot time duration depends on the physical layer, and it accounts for the propagation delay, the time needed to switch from receiving to transmitting state (RX_TX_Turnaround_Time) and the time to signal to the MAC layer the state of the channel (busy detect time) [2] [22] [25]

2.2.3 Binary Exponential Backoff Scheme

An exponential backoff scheme is used in DCF protocol. At each packet transmission, the backoff time is uniformly chosen in the range (0, W-1). The value W is called contention window, and depends on the number of transmission failed for the packet.

Table 2.1 Slot Time, Minimum and Maximum Contention Window Values

PHY	Slot Time	CW_{min}	CW_{max}
FHSS	50 µs	16	1024
DSSS	20 µs	32	1024
IR	8 µs	64	1024

At the first transmission attempt, W is set equal to a value CW_{min} that is called minimum contention window. After each unsuccessful transmission, W is doubled, up to a maximum value $CW_{max}=2^m CW_{min}$, where m is the maximum backoff stage. The values CW_{min} and CW_{max} that are specified in the IEEE 802.11 Standard for Wireless LAN are PHY-specific and are summarized in Table 2.1 [2] [9] [11] [14] [15] [27] [30].

2.2.4 Basic Access Mechanism

The backoff time counter is decremented as long as the channel is sensed idle, “frozen” when a transmission is detected on the channel, and reactivated when the channel is sensed idle again for duration larger than DIFS. The station transmits when the backoff time reaches zero [2] [19] [22] [26].

Figure 2.3 illustrates this operation. Two stations A and B share the same wireless channel. At the end of the packet transmission, station B waits for a DIFS and then chooses a backoff time equal to 8 (uniformly chosen between 0 and CW_{min}), before transmitting the next packet. Assume that the first packet of station A arrives at the time indicated with an arrow in the figure. After a DIFS, the packet is transmitted. At the transmission time of the station A, the station B is in the middle of the Slot Time corresponding to a backoff value, equal to 5. As a consequence of the channel sensed busy, the backoff time is frozen to its value 5, and the backoff counter decrements again only when the channel is sensed idle for a DIFS [2].

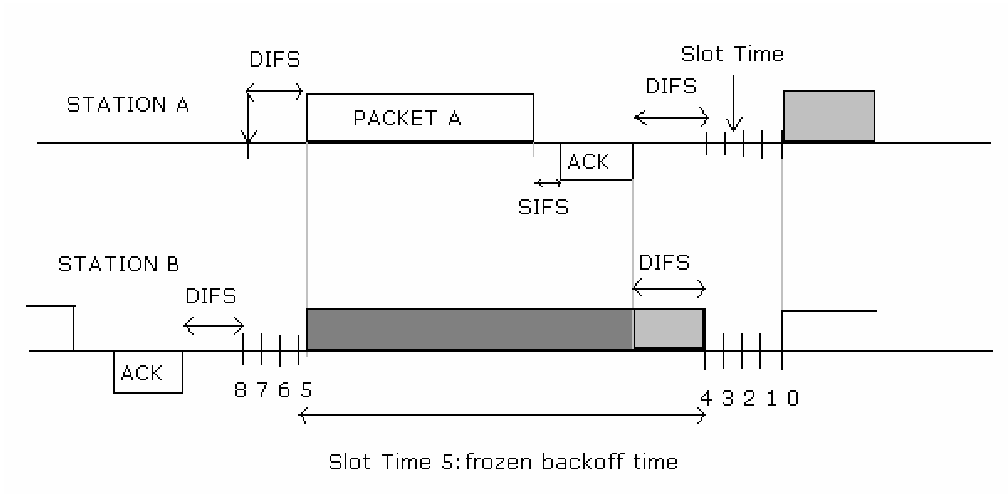

Figure 2.3 DCF Protocol with Basic Access Method [2]

The CSMA/CA does not rely on the capability of the stations to detect a collision by hearing their own transmission. For that reason, an ACK is transmitted by the destination station to signal the successful packet reception. The ACK is transmitted immediately at the end of the packet, after a period of time called short interframe space (SIFS). As the SIFS (and the propagation delay in total) is shorter than a DIFS, no other station is able to detect the channel idle for a DIFS until the end of the ACK. If the transmitting station does not receive the ACK within a specified ACK_Timeout or it detects the transmission of a different packet on the channel, it reschedules the packet transmission according to the given backoff rules [2] [19] [26] [27].

2.2.5 RTS/CTS Access Mechanism

DCF defines an additional four-way handshaking technique to be optionally used for a packet transmission. This mechanism, which is called as RTS/CTS (Request-to-Send/Clear-to-Send), is shown in Figure 2.4 [2]. A station that has a packet to transmit, waits until the channel is sensed idle for a DIFS, follows the backoff rules explained in 2.2.3. Then instead of sending the data packet preliminarily sends a special short frame called request-to-send (RTS). When the receiving station detects an RTS frame, it responds, after a SIFS, with a clear-to-send (CTS) frame. The transmitting station is allowed to transmit its packet only if the CTS frame is correctly received [2] [11] [14] [26].

The frames RTS and CTS carry the information of the length of the packet to be transmitted. This information can be read by any listening station and then that station can update a network allocation vector (NAV) containing the information of the period of time in which the channel will remain busy. Therefore, when a station is hidden from either the transmitting or the receiving station, by detecting just one frame among the RTS and CTS frames, it can suitably delay further transmission, and thus avoid collision [2] [14] [25] [26] [28].

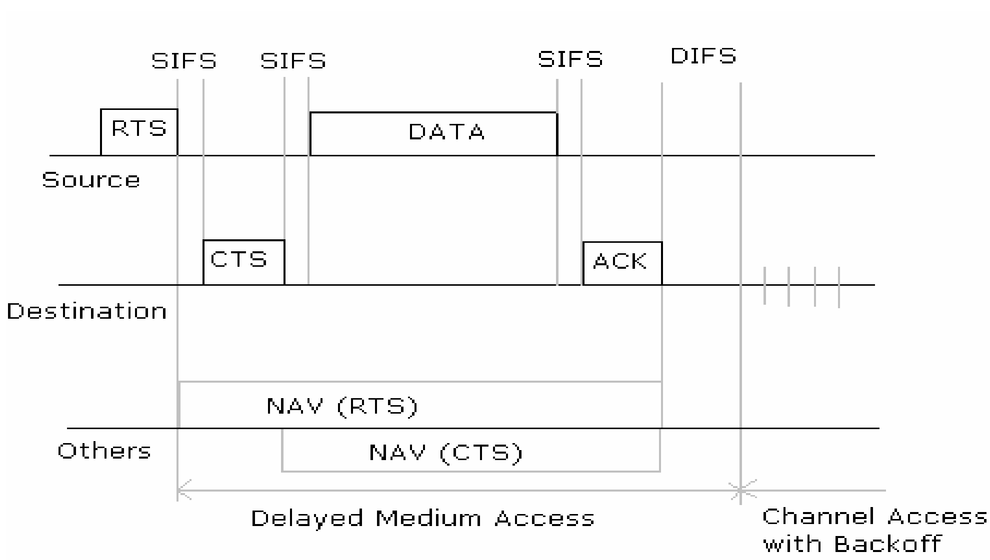

Figure 2.4 DCF Protocol with RTS/CTS Access Method [2]

The RTS/CTS mechanism is very effective in terms of system performance. Especially when large packets are considered, the RTS/CTS mechanism reduces the length of the frames involved in the contention process. In fact, in the assumption of perfect channel sensing by every station, collision may occur only when two or more packets are transmitted within the same slot time. If both transmitting stations employ the RTS/CTS mechanism, collision occurs only on the RTS frames and it is early detected by the transmitting stations by the lack of CTS responses [2] [7] [14] [15].

2.3 Enhancement of Reliable Transport Protocol-DCF+

2.3.1 Overview

A new scheme, DCF+ has been proposed [3] to improve the performance of reliable transport protocol over WLAN, which needs to receive the transport layer acknowledgement (ACK) on the backward direction. In the scenario of TCP over WLAN the forward TCP data and backward TCP ACK will compete for the channel. This may cause collisions and degrade the overall performance. The DCF+ scheme is able to improve the overall performance both in TCP and UDP. The performance improvement of DCF+ on different situations will be discussed in Chapter 3.

Note that, DCF+ is fully compatible with DCF. This means that if some stations support DCF+ while the others not, they can coexist and transfer data to each other. The access method in DCF+ can be considered as a data exchange on the backward direction after the original data exchange on the forward direction. DCF+ protocol is also applicable to both basic or RTS/CTS access methods [3].

2.3.2 How DCF+ Protocol Works

Suppose that the source station starts with the basic access method to compete for the channel and currently the destination station has a packet (DATA2 in Figure 2.5) to the source, which sends DATA1. [3].

In DCF+, the duration field in the MAC header is also used to set the NAV values as in DCF, so the destination needs to set the NAV of other stations by setting the duration field on the ACK field. After receiving the ACK packet, the source replies with the CTS. Then the destination could transfer the data packet (DATA2 in Figure 2.5) to the source station and the source replies with a normal ACK.

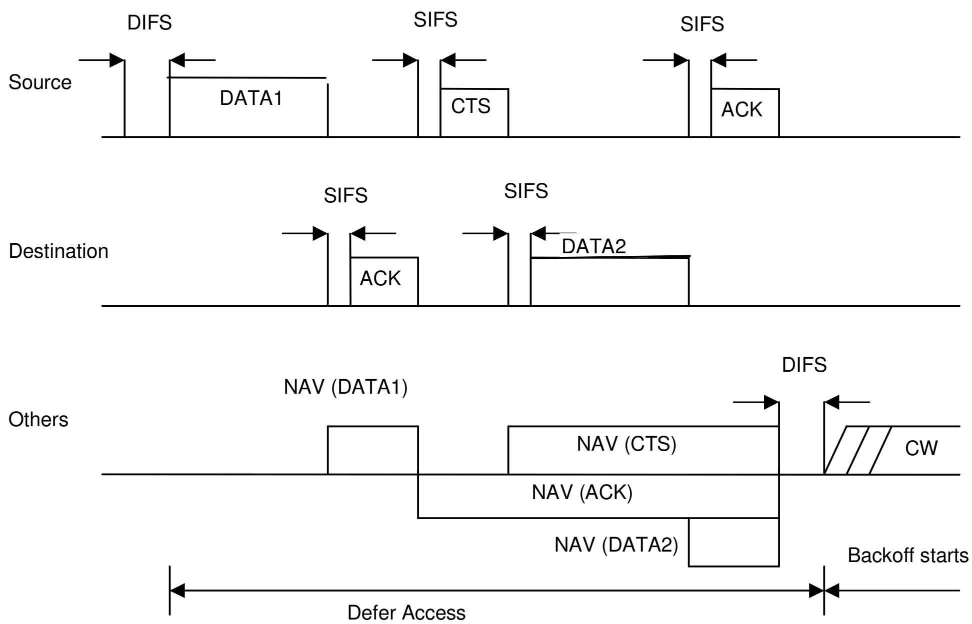

Figure 2.5 DCF+ Protocol with Basic Access Method [3]

Note that the first ACK in the procedure acts as an RTS sent by the destination station. Therefore, the second data transfer from destination to the source always deals with the hidden terminal issue as in RTS/CTS access method. The first ACK in the procedure is a normal ACK for the source station if the source station only supports 802.11 DCF, not DCF+. To keep the backward compatibility between 802.11 DCF and DCF+, the ACK frame is used after the successful transfer of the first data frame [3].

If the frame exchange starts with RTS/CTS access method, the procedure is similar. The frame exchange procedure in RTS/CTS access method is also shown in Fig. 2.6. Stations only supporting DCF and stations supporting DCF+ still can exchange frames by using DCF. Therefore, the backward compatibility is guaranteed. On the other hand there are still two issues that shall be discussed.

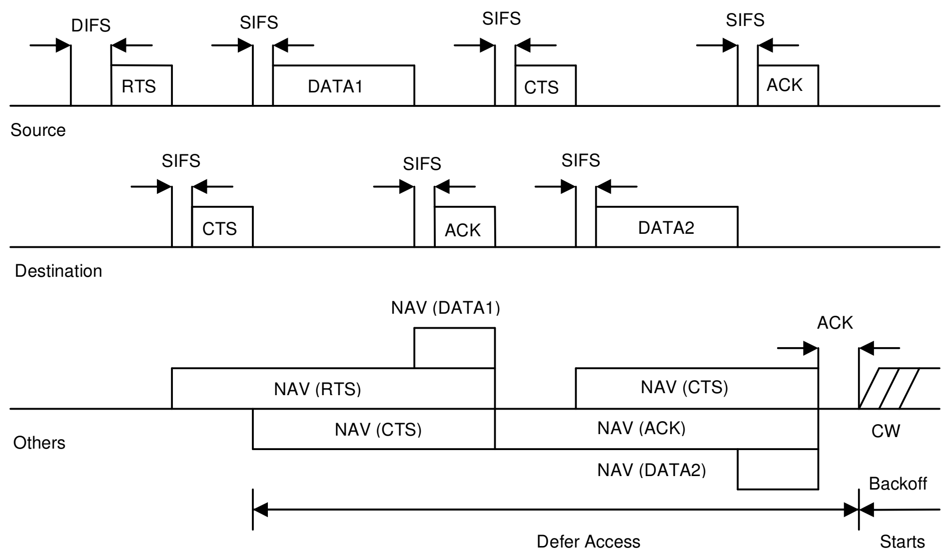

Figure 2.6 DCF+ Protocol with RTS/CTS Access Method [3]

These issues are:

1. In DCF+, it is assumed that the destination station has a data frame ready to be transmitted to the source station, but that is not always the situation. The destination station will always send an ACK after receiving the DATA1 frame correctly. Therefore, upon receiving an ACK, the source station using DCF+ must determine whether it needs to send the CTS to reserve channel for the second data frame. At this point, it is assumed that by examining the duration field of the ACK frame received, the source can determine whether the destination station has a data frame ready to send.

2. Assume that the destination station uses DCF+ but the source station only supports DCF. Supposing that whether or not the source station supports DCF+ is unknown at the destination station, the destination station may reserve the channel by the ACK and the bandwidth may be wasted. Two

alternatives are proposed to solve this issue: 1) A DCF station can make a record to determine whether another station is DCF+ capable. 2) Some reserved fields in the data frame can be used to indicate the source station is DCF+ capable or not. The reserved subtype value for a data frame can be used to fulfill this function [3].

2.4 Performance Evaluation of IEEE 802.11 Wireless Networks

2.4.1 Performance Evaluation for Different Aspects and Limits of IEEE 802.11 Wireless Networks

While some studies are based on existing rules that are defined by the IEEE 802.11 group, some are based on performance evaluation after making some changes on the defined working principles of the wireless networks. These changes mainly focus on the improvement of the network throughput.

In [7], the probability distribution of MAC layer service time is derived and based on this probability model, some performance metrics of the wireless LAN are analyzed under various traffic loads. The influence of a host with a lower bit rate on the throughput of other hosts that share the same radio channel is analyzed in [8]. In [9], the packet delays, the probability of a packet being discarded when it reaches the maximum retransmission limit and the average time to drop such a packet are investigated for the basic and RTS/CTS access mechanisms. The theoretical performance of CSMA/CA is investigated in [10]. The characteristics of both CSMA/CA methods are highlighted and the performance dependency on the number of stations and on traffic conditions is also evaluated in [10]. The behavior of the UDP transport protocol in terms of throughput and average delay over an IEEE 802.11 compliant wireless LAN is characterized in [11]. In [12], the system performance of a WLAN with CSMA/CA MAC protocol with different ARQ protocols is evaluated. The theoretical throughput upper limit and theoretical delay lower limits for the IEEE 802.11 protocols are evaluated. As a conclusion it is stated that, the existence of such limits indicates that by simply increasing the data rate without reducing overhead, the enhanced performance, in terms of throughput and delay, is bounded even when the data rate becomes infinitely high [13]. In [2] and [14], the saturated throughput analysis of 802.11 DCF is provided with the assumption of finite number of terminals and ideal channel conditions. In [2], an accurate analytical model is provided to compute the 802.11 DCF throughput. The throughput is calculated both for basic and RTS/CTS access methods. [2] covers the evaluation of different parameters that effect the network throughput. Basically different initial contention window sizes, maximum number of backoff stages, average number if idle slots per successful packet transmission and the average number of slot time units wasted on the channel because of packet collision per successful packet transmission are investigated both for basic and RTS/CTS access methods.

2.4.2 Performance Evaluation of IEEE 802.11 Wireless Networks With Modified Contention Window and Backoff Procedures

There are many studies about the improvement of the throughput by making some changes on the existing 802.11 backoff procedure. In [15], to improve the throughput, a new mechanism is proposed for resetting the contention window. After successful data transmission, instead of resetting the contention window to the initial value, the contention window is decreased to its half value or minimum contention value whichever is greater. In [16], the contention window is decreased according to a defined procedure, which is called Slow Contention Window Decrease Scheme. According to the defined procedure, after successful transmission of the data packet, the contention window is decreased according to the decrease factor value. Several parameters are investigated in this work such as the number of stations, the initial contention window, the decrease factor value, the maximum backoff stage and the coexistence of the RTS/CTS access mechanisms. Another study proposed for the throughput improvement is called Fast Collision Resolution MAC algorithm [17]. In this algorithm, the MAC backoff scheme is changed according to a complex procedure, which allows stations to change their contention window values according to the traffic conditions. In [18], an approach which is similar to the one used in [15] is applied. The difference is to halve the contention window after “c” (a predetermined value) consecutive successful transmissions. If the consecutive successful transmissions are less than “c” value, the contention window will remain unchanged or it will be doubled in case of collision. In [19], the probabilistic contention window control mechanism is introduced to improve the fairness of backoff procedure and the performance on real-time applications such a voice over IP and video conferencing is evaluated. Another study on the performance improvement of the network is presented in [20]. In this study, a contention window increase-decrease scheme is proposed. Two different algorithms, which are called Exponential Increase Exponential Decrease and Multiple Increase Exponential Decrease, are applied for performance comparison.

2.4.3 Quality of Service Differentiation in IEEE 802.11 Wireless Networks

The QoS differentiation is another popular research area in 802.11 wireless LANs. In [21], the QoS differentiation is performed by assigning different weights to stations. By changing backoff time and slot time according to the weight, QoS is controlled. Also fair throughput for each mobile station is obtained by assigning weight to each mobile station. Another study for QoS differentiation is focused on the change of MAC parameters. Each class is assigned different inter-frame space and contention window size. Due to the critical effect of these parameters in determining the system performance, the effective QoS differentiation between classes is obtained in [22]. The throughput performance of a p-persistent version of 802.11 MAC protocol with multiple QoS traffic classes is analyzed in [23]. In [24], a MAC architecture to support differentiated service in IEEE 802.11 wireless LAN is proposed. This architecture is based on employing the MAC-core as base and different adaptors as add-ons. The resulting MAC can provide prioritized services with different delays and throughputs. The QoS differentiation by setting different DIFS durations for different classes is evaluated in [25].

2.4.4 Performance Evaluation of IEEE 802.11 Wireless Networks by Changing DCF Protocol

The research in [26] is focused on the Transmission Burst issue. It means that, once a station has contended for the channel, it may transmit multiple data frames continuously to the same destination. The modified DCF can reduce channel contention time and increase the utilization ratio of the channel. Also, the modification does not introduce any additional control overhead. In [27], a modified

DCF function for real-time traffic in IEEE 802.11 wireless LAN is proposed. The proposed forward backoff scheme and call admission control employed by modified DCF can guarantee the throughput, delay bound and jitter of real-time traffic in the contention period. In [28], an extension of the IEEE 802.11 DCF protocol called PUMA (Priority Unavoidable Multiple Access) is proposed. It is shown that PUMA is fair, efficient, stable and allows for provision of time-bounded services.

2.4.5 A New Approach to Support Quality of Service in Wireless LANs-Enhanced DCF

IEEE 802.11e MAC is an emerging supplement to the IEEE 802.11 wireless LAN standard to support QoS. The 802.11e MAC is based on both centrally-controlled and contention-based channel access. In [29], enhanced distributed coordination function (EDCF) is compared with the 802.11 legacy MAC. Also, an optional feature of EDCF, called contention-free burst (CFB), which allows multiple MAC frame transmission during a single transmission opportunity is considered. In addition to the comparison of EDCF with the 802.11 legacy DCF, the Hybrid Coordination Function (HCF) mode of MAC operation is also compared with 802.11 legacy DCF and PCF in [30]. In [34], a retransmission scheme, known as Age Dependent Backoff (ADB), to decrease the delay and jitter of real time packets by adjusting the persistence factors dynamically based on the ages of the real-time packets in the transmission queues and the lifetime of the real-time packets.

2.5 Conclusion

As a result, we can say that extensive research has been performed about the performance evaluation of the 802.11 network. Some studies have focused on making changes in the main structure of the DCF MAC protocol for performance improvement and some have focused on service differentiation. To achieve service differentiation, the studies are focused on the tools that directly affect the performance parameters such as binary exponential backoff algorithm, SIFS duration, DIFS duration and initial contention window size. On the other hand, in some studies the performance of the wireless network under different scenarios is evaluated. These scenarios are also adapted from the problems faced in real life or made according to the predefined assumptions.

In the next chapter, performances of DCF and DCF+ protocols will be studied under various configurations to assess realistically the benefits brought by DCF+.

CHAPTER III

WIRELESS NETWORKS PERFORMANCE STUDY

3.1 Introduction

In this thesis, the performance of the 802.11 MAC protocol is evaluated in different aspects. First of all, based on the values in [2], the saturation throughput values are calculated for different numbers of stations. The verification of the simulation is obtained by getting very close throughput values to the results given in [2].

The performance of the 802.11 MAC protocol is evaluated widely by covering many different traffic conditions and also, taking into account all three different PHY layer specifications (different slot time, minimum contention window and maximum contention window).

The evaluation is focused on the saturation throughput, in the assumption of ideal channel conditions, i.e., no hidden terminals and capture. In the analysis, the saturation condition is assumed (fixed number of stations, each always have a packet for transmission). This is a fundamental performance figure defined as the limit reached by the system throughput as the offered load increases, and it represents the load that the system can carry in stable conditions.

In [3], a change on 802.11 MAC protocol is proposed to improve the performance of the wireless network, which is called DCF+. The main motivation to propose DCF+ protocol is getting a performance improvement without making any change to the existing network. Also the DCF+ protocol provides flexibility of the communication between the stations either uses DCF or DCF+ MAC protocol.

In our work, we have focused on the performance improvement of DCF+ MAC protocol under different scenarios. First of all, the 802.11 MAC protocol simulations are performed for both access mechanisms by using the same network conditions defined in [2]. Then, the simulation is repeated for the network conditions defined in

[3]. Additionally, the newly defined DCF+ protocol simulation is performed and the results are shown. After validating our simulation program through this process, we have investigated the performance improvement of DCF+ protocol under the conditions specified in [4] by comparing the DCF and DCF+ saturation throughput results.

Additional research made on QoS differentiation in 802.11 wireless networks includes [5] in which the effect of initial contention window size on the throughput is investigated. The QoS differentiation is presented in basic access mechanism by dividing the stations in the network into the two different classes and giving different initial contention window sizes. In our work, the same conditions are repeated for DCF+ protocol and the results are interpreted.

Moreover, the QoS differentiation mechanism is simulated for RTS/CTS access mechanism. The stations are divided into two classes and each class assigned different initial contention window sizes. The simulation is first performed for DCF protocol and then for DCF+ protocol. The results obtained for both protocols are compared.

3.2 Performance Evaluation of the IEEE 802.11 Distributed Coordination Function

In order to validate the simulation algorithm, the results obtained by using OMNet++ simulation program are compared with the results in [2]. The values of the parameters used to obtain numerical results are summarized in Table 3.1. Note that a fixed propagation delay implies equal distance between all node pairs.

Table 3.1 FHSS System Parameters and Additional Parameters Used for Simulation

Packet Payload	8184 bits
MAC Header	272 bits
PHY Header	128 bits
ACK	112 bits + PHY Header

RTS	160 bits + PHY Header
CTS	112 bits + PHY Header
Channel Bit Rate	1 Mbits/s
Propagation Delay	1 µs
Slot Time	50 µs
SIFS	28 µs
DIFS	128 µs
ACK_Timeout	300 µs
CTS_Timeout	300 µs

The values specified for the frequency hopping spread spectrum (FHSS) PHY layer [5] are used as system values. The channel bit rate has been assumed equal to 1 Mbits/s. The frame size is as specified in the 802.11 MAC specifications, and the PHY header has a value defined for the FHSS PHY. The values of the ACK_Timeout and CTS_Timeout are given in Table 3.1, and they have been set equal to 300 µs. When determining the numerical timeout value, the SIFS, ACK transmission and a round trip delay time have been considered. Packet destinations are randomly chosen and contention window size is uniformly distributed. In the simulation, constant packet payload size of 8184 bits is used.

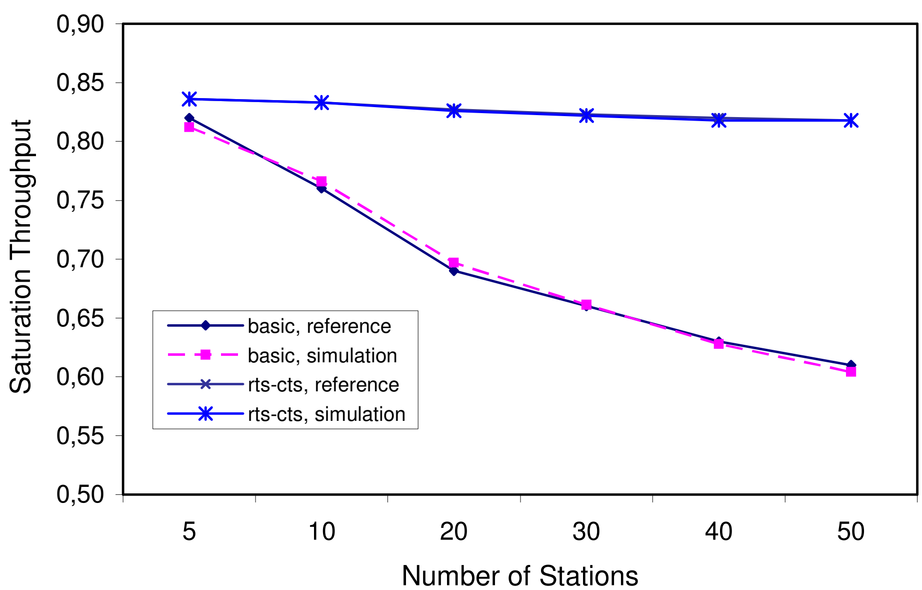

Figure 3.1 Saturation Throughput: Reference versus Simulation

Figure 3.1 shows the results obtained from the simulation and presented in [2]. All the simulation results have been obtained within a ±0.002 interval of 99 % confidence. As it is shown in the figure, the simulation results are very close to the results taken as a reference. Through this work, we can say that our simulation is working properly.

3.3 Performance Enhancement of 802.11 Distributed Coordination Function

3.3.1 General

In this section, to evaluate the performance enhancement of DCF+, the throughput results of DCF+ are compared with the results obtained in [2].

3.3.2 Performance Analysis for DCF

The same assumptions as in [2] are used. The contending stations are supposed to be a fixed number, n. Let b(t) be the stochastic process representing the back-off window size for a given station as slot time t. The slot time is referred to as the constant value σ and the variable time interval between two consecutive backoff time counter decrements. As in [2], the key approximation is that the probability p that a transmitted packet collides is independent from the state s(t) of the station.

All the parameters are based on the Direct Sequence Spread Spectrum (DSSS) PHY in 802.11. In DSSS, CW_{min} and CW_{max} equal to 31 and 1023 respectively. Therefore, we have

$$\begin{cases} W_i = 2^i W & i \le m' \\ W_i = 2^{m'} W & i > m' \end{cases} \qquad (1)$$

where $W = (CW_{min} + 1)$ and $2^{m'} W = (CW_{max} + 1)$, so for DSSS, we have m'=5. As specified in 802.11, the maximum backoff stage value could be larger than m', while the CW will be held after that, which is shown in equation (1). In fact, the m' also means the maximum retransmission count, which is different for data frame and RTS frame, i.e., 5 and 7 respectively. The key difference between [2] and [3] is that the Markov chain models are different, because the effect of frame retransmitting limit is considered in [3]. The details of the only non-null one-step transition probabilities in Markov chain and the calculation for collision duration and successful transmission duration are also given in [3]. The results of this calculation are provided as;

$$T_s^{bas} = DIFS + PHY_{hdr} + MAC_{hdr} + E[P] + \delta + SIFS + ACK + \delta$$

$$T_c^{bas} = DIFS + PHY_{hdr} + MAC_{hdr} + E[P^*] + SIFS + ACK$$

where the T_s and T_c are the average time the channel is sensed busy because of a successful transmission or a collision respectively and the E[P] is the average packet length. “bas” means basic access method, δ is the propagation delay and E[P*] is the average length of the longest packet payload involved in a collision. In this case, it is assumed that all the packets have the same length, therefore, E[P]= E[P*]=P.

For the RTS/CTS access method, assuming that all the station use the RTS/CTS for the data frame, then we have

$$T_s^{rts} = DIFS + RTS + SIFS + \delta + CTS + SIFS + \delta + PHY_{hdr} + MAC_{hdr} + E[P] \quad SIFS + \delta + ACK + \delta$$

$$T_c^{rts} = DIFS + RTS + SIFS + CTS$$

where rts means RTS/CTS access method. Note that the collision is supposed to be occurring between RTS frames and T_c^{rts} is different from that in [2] because CTS timeout effect is considered.

3.3.3 Markov Model Validation

It is assumed that each station has enough data to send to obtain saturated throughput performance of the new backoff scheme. The number of stations is varied to see the effect of throughput degradation due to increased collision probability. Packet destinations are randomly chosen and contention window size is uniformly distributed.

All the parameters used in simulation are according to the parameters in [5] for DSSS and are summarized in Table 3.2 It is assumed that the application data payload is 1000 bytes; IP header and UDP header are 20 and 8 bytes, so the packet payload at MAC layer is 1028 bytes.

Table 3.2 System Parameters for MAC and DSSS PHY Layer

Packet Payload	8224 bits
MAC Header	224 bits
PHY Header	192 bits
ACK	112 bits + PHY Header
RTS	160 bits + PHY Header
CTS	112 bits + PHY Header
Channel Bit Rate	1 Mbit/s
Propagation Delay	1 µs
Slot Time	20 µs
SIFS	10 µs
DIFS	50 µs

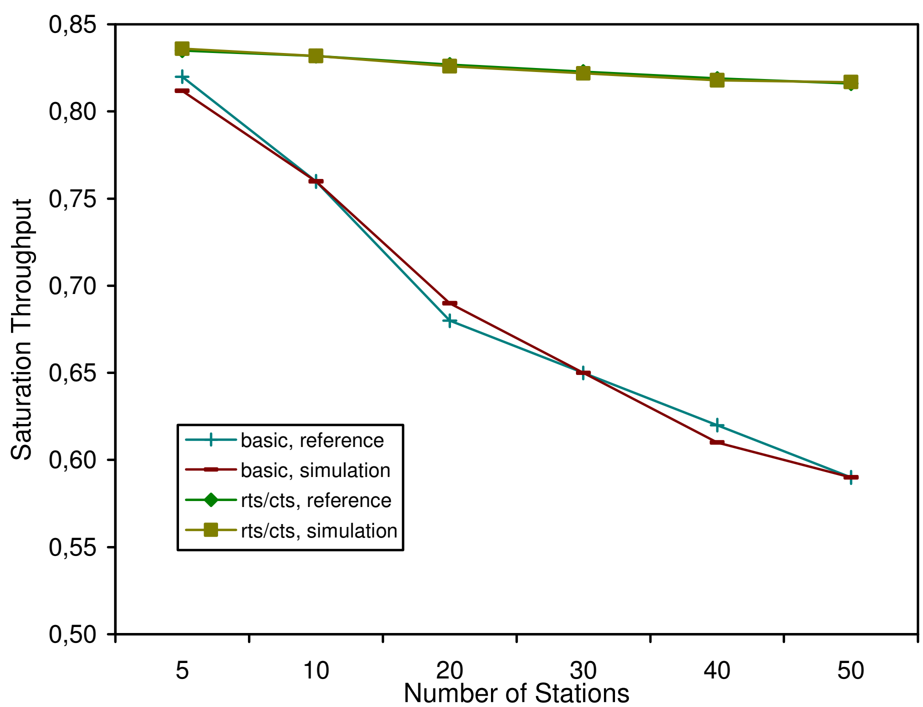

Figure 3.2 Saturation Throughput: Reference versus Simulation for DCF

The results for basic and RTS/CTS access methods are shown in Figure 3.2. The results shown as reference are the analysis results found in [3]. The results that are shown as simulation are the results of our simulation that is performed by using the parameters given in Table 3.2 As shown in Figure 3.2, the reference and simulation values are very close to each other. With this work, the accuracy of our simulation has been checked once more by using the parameters in [3].

3.3.4 DCF+ Analysis

The Markov chain model is used to analyze the performance of DCF+. Note that the DCF+ protocol depends on the packet transmission of destination station to source station, but destination station does not always have a packet for the source station. In this scenario, the access procedure is the same as that in DCF. For simplicity of the analysis, it is assumed that the destination always has such a packet to transfer. Therefore, the DCF+ throughput performance achieved here is the upper bound of DCF+ for two-way traffic.

TCP is used as the reliable transport protocol and suppose there is no ACK delay in the destination, that is, a TCP data packet always triggers a TCP ACK packet transfer on the backward direction. The application data packet is segmented at the TCP layer, each segment contains 1000 bytes, so a TCP data packet that arrives from the IP layer to the link layer is 1040 bytes, 40 bytes for IP and TCP header overheads totally. The TCP ACK packet is 40 bytes long, with no overhead introduced for options.

Suppose the length of the packet that arrives from the high layer to the MAC layer has probability distribution function (PDF) F(x). For simplicity it is assumed that the TCP sending window is large enough, thus the probability of data packet arriving at MAC layer and ACK packet arriving is the same, then for this case, we have

$$F(x) = \begin{cases} 0 & x < 40 \\ 1/2 & 40 \leq x < 1040 \\ 1 & x \geq 1040 \end{cases}$$

For simplicity, supposing the probability of three or more packets simultaneously colliding can be neglected, and then the PDF of the longest packet length for two packets in collision can be expressed as: $F^*(x) = F^2(x)$

The analysis procedure is repeated similarly as those for DCF in [3]. The analysis results for DCF and DCF+ taken from [3] and their simulation results obtained through this work is given in Figure 3.3.

As it is seen from the figure, there is a difference between the results obtained using the simulation and given in [3]. The results are obtained for different number of stations (5, 10, 20, 30, 40 and 50).

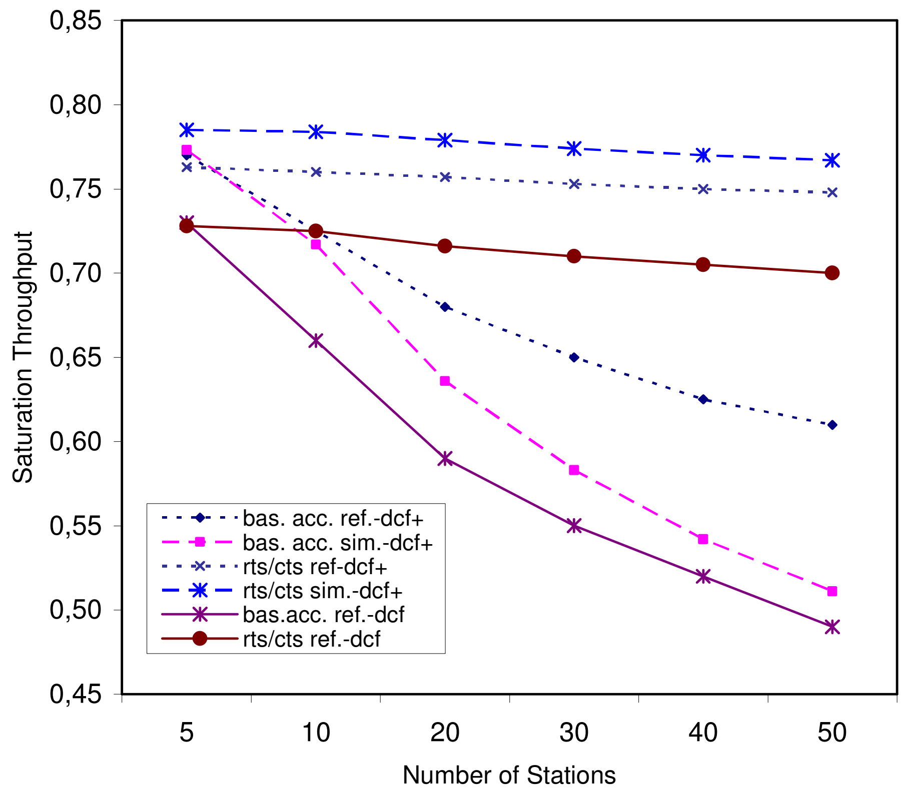

Figure 3.3 Saturation Throughput: Reference versus Simulation

In basic access, when the number of stations is less than or equal to 10, the results are close to each other. However, the results obtained for more than 10 stations are getting smaller values than the reference values. For 50 stations, the difference between the simulation and reference values are about 20 %. The reason behind this difference is insufficient information in [3] about the conditions of simulation to get those results. Additionally, as it is explained in [5], (no information has been given in [3] about this issue) in the proposed scheme for TCP, the TCP ACK is combined with the MAC ACK. This violates the layering principle that leads to the complication in MAC ACK message structure. In our simulation the TCP ACK is not combined with the MAC ACK because of the mentioned violation issue.

In our simulation, the main principle of DCF+ is applied, that is, in case of any packet arrival from source station, a packet is transmitted from destination station to source station. This situation depends on the assumption that the destination station always has a packet to source station to transmit as before. While applying this procedure, the explanation in [3] for the probability distribution function of packet length, which arrives from upper layer to the MAC layer, has been taken into account. For that reason, when the source station sends a data packet, the destination station sends back to the source station a data packet or TCP ACK packet based on a random selection. As seen from the Figure 3.3, the throughput values obtained with this approach are those lower than those given in [3]. But those values are still higher than DCF protocol.

The same approach is applied for the RTS/CTS access scheme in DCF+. As shown in the figure, the results obtained using simulations are about 3% higher than the values provided in [3].

As it is well known, in RTS/CTS access scheme, while the number of collisions increase with the number of stations, the system throughput decreases slightly. But in basic access scheme, the system throughput decreases dramatically with the increasing of number of stations. As seen from the figure, both basic and RTS/CTS access scheme results are slightly higher than the reference values for five stations. When the number of stations goes higher, the basic access scheme results go lower than the reference values while the RTS/CTS scheme results are higher than reference values. At this point, it can be said that, in our approach, the throughput of the network is more sensitive to the number of collisions than the proposed scheme in [3] and in parallel with the DCF protocol.

3.4 Performance Under Poisson Traffic

In this section, DCF and DCF+ performances are compared under Poisson traffic with average arrival rates below saturation values. The parameters in [2] are used for the simulation. The packet destination is randomly chosen between the stations and the contention window size is uniformly distributed. It is assumed that, the destination has a packet to transfer the source station if it has at least one packet ready to transfer at the time of data arrival. Basic and RTS/CTS schemes have been evaluated. The results are presented in Figure 3.4 and 3.5.

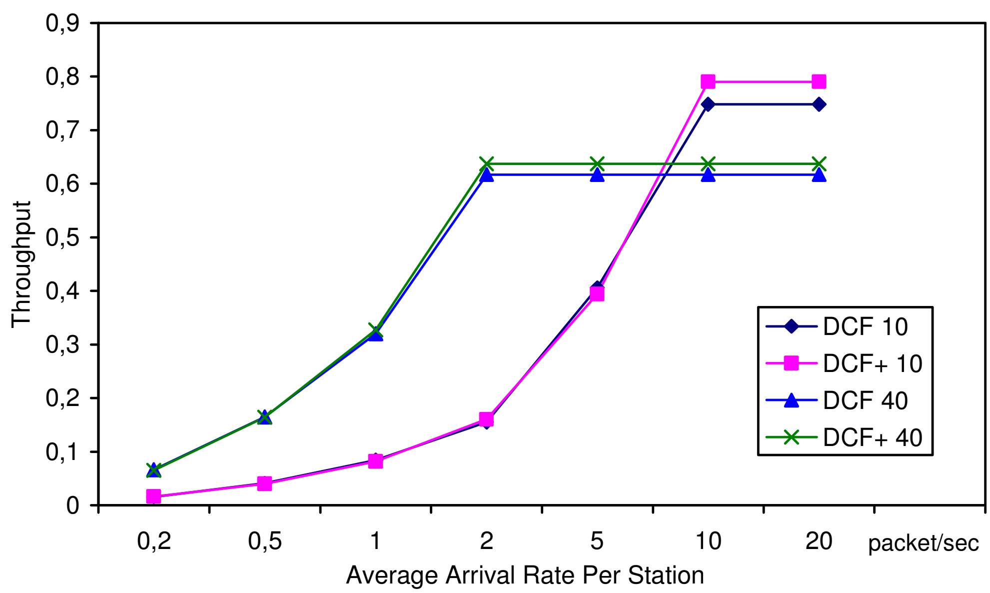

Figure 3.4 Performances under Poisson Traffic in Basic Access

It is seen that the DCF+ protocol provides higher throughput values than the DCF protocol under high packet arrival rates. The efficiency of DCF+ increases with the average packet arrival rate and maximizes at the saturation condition. At this point, the throughput reaches to highest value and remains constant, even though the average arrival rate goes higher. While the average packet arrival rate decreases, the efficiency of the DCF+ decreases and the DCF+ throughput values come closer to DCF throughput values. At the very low rates, DCF and DCF+ have almost the same throughput values. As a result we can say that, the DCF+ protocol has performance efficiency under saturation condition but in lower traffic conditions it has no performance improvement over DCF.

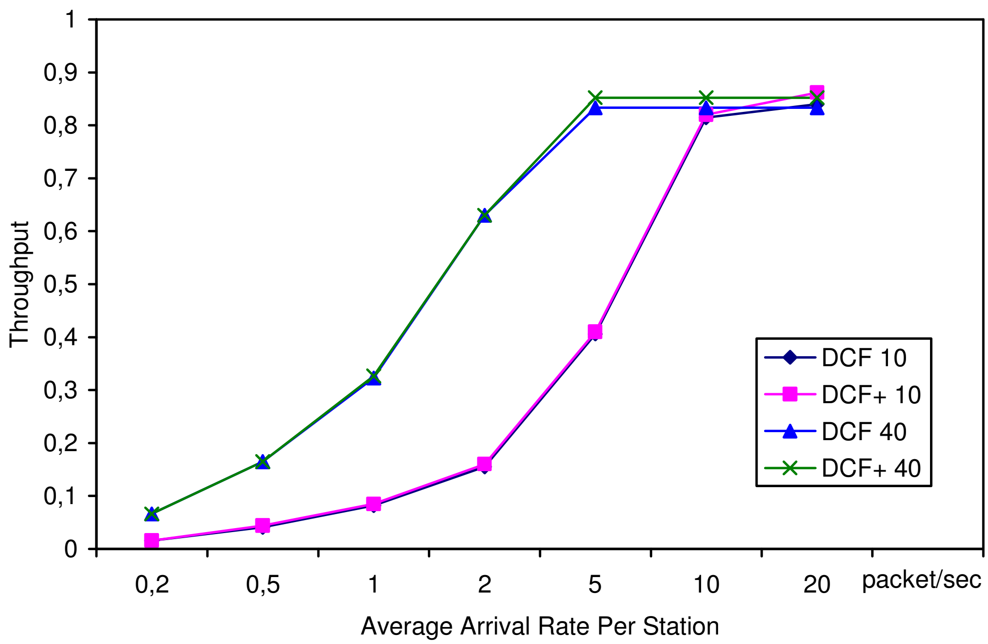

Figure 3.5 Performances under Poisson Traffic in RTS/CTS Access

In RTS/CTS access scheme, the performance of DCF and DCF+ is close to each other. Except the saturation condition, the throughput values of both protocols are almost same. We can say that the DCF+ has performance improvement only under saturation conditions. The Figure 3.4 and 3.5 shows that DCF+ has performance improvement under saturation condition in both basic and RTS/CTS access scheme. But under low traffic conditions, DCF+ has no performance improvement in RTS/CTS access scheme and it provides slight performance improvement in basic access scheme.

3.5 Disaster Scenario Analysis

3.5.1 General

Under the disaster scenario, there are *r* stations ready for new packet transmission at the same time. Each station has only one packet to transmit. The packet sizes are assumed to be fixed. The process for all *r* stations to successfully transmit their packets is defined as the *recovery process* of the disaster scenario. When the last station completes its packet transmission, the recovery process ends.

The technique used in [6] is adopted for evaluating the mean time of the recovery process for IEEE 802.11 MAC protocol. The technique requires two steps: 1) Computation of the attempt probability of a particular station, and 2) The mean time for all *r* stations to obtain a successful transmission each based on the attempt probability. The detailed information and calculation of attempt probability and mean time of the recovery process can be found in [4].

In this section, disaster duration and disaster throughput values for DCF+ protocol will be provided. Those values will be compared with the results that are obtained for DCF protocol in [4]. The performance improvement of DCF+ protocol will be evaluated also.

3.5.2 Numerical and Simulation Results

The parameters used for numerical calculations in [4] are used for our simulation also. Those values are all according to the specification of FHSS physical layer [1]. These parameters are also given in Table 3.3.

Let T_H be the transmission time of the MAC and PHY headers, $T_{payload}$ be the transmission time of the packet payload. Let T_{ACK}, T_{RTS} and T_{CTS} be the transmission time of the ACK, RTS and CTS frames respectively. For both the basic and four-way handshaking access methods, the time duration for a successful packet transmission (T_S), a collision (T_C) and an idle period (T_I) are given in Table 3.3.

Table 3.3 The T_S, T_C and T_I values

For basic access method:
$P_{tr}(i) = \tau_i(1-\tau_i)$ $T_S = T_H + T_{payload}$ + SIFS + τ + T_{ACK} + DIFS + τ = 8982 μsec
$T_I = T_{slot}$ =50 μsec
$T_C = T_H + T_{payload}$ + DIFS + τ = 8713 μsec
For RTS/CTS access method:
$T_S = T_{RTS}$ + SIFS + τ + T_{CTS} + SIFS + τ + T_H + $T_{payload}$ +SIFS + τ + T_{ACK} + DIFS + τ = 9568 μsec
$T_I = T_{slot}$ =50 μsec
$T_C = T_{RTS}$ + DIFS + τ =417 μsec

To validate the simulation algorithm, first the simulation for DCF protocol is performed and compared with the results in [4]. Then the simulation for DCF+ protocol is performed with the same parameters. The CW_{min} = 8 and CW_{max} = 256 values are used for both simulations. The simulation results are obtained by repeating the simulation until reaching the 95 % level of confidence.

The results presented in [4] and obtained through our simulation are given in Figure 3.6. Figure includes all disaster throughput values obtained for DCF and DCF+ for both access schemes. The simulation algorithm is validated by getting very close results to the reference values. After this process, the same algorithm is used for DCF+ protocol. When we compare the results obtained for DCF and DCF+ protocol, in DCF+ protocol we got higher disaster throughput values than in DCF protocol. To talk about the efficiency of DCF+ protocol in such disaster scenario, the throughput improvement of DCF+ protocol explained in the previous article shall be taken into account. If we compare the throughput improvement of DCF+ protocol, we can say that the level of improvement is almost same (as a percentage) in the normal network conditions and disaster scenario.

In DCF+ protocol, it is assumed that each destination station has a packet ready to be sent to source station. This is the main issue that is affecting the throughput efficiency of the DCF+ protocol. In order to adapt this assumption to the disaster scenario application, we assumed that;

- The source station has only one packet,
- The destination station shall have only one packet,
- The arrival address of the packet at the destination station is the source station,
- The packet at the destination station is ready to be sent at the time of received data from the source station.

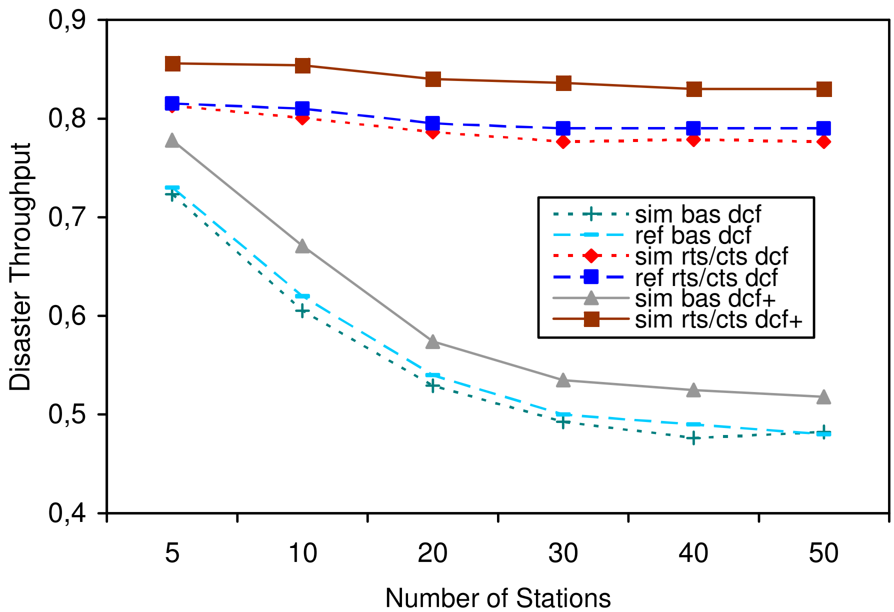

Figure 3.6 The Disaster Throughput of IEEE 802.11 Wireless Networks

Based on these assumptions, observing higher throughput performance of DCF+ protocol is an expected result. Because, in DCF protocol each station is contending to capture the environment for data transfer. After each successful data transmission, the total number of contending stations is decreasing by one. Other stations continue through their backoff procedure in accordance with the collision avoidance procedure of the DCF protocol. This process goes on until the last station sends its packet successfully.

In DCF+ protocol, after successful transmission of source station, the destination station sends its packet to the source station and the total number of contending station in the network decreases by two. If we compare this situation with DCF, we have two benefits in DCF+. These are;

1. After each successful data transmission, two stations will have completed their data transmission. This means that, only one back off procedure is applied for two stations' data transmission. The time that would be spent for the second station's backoff time is gained.
2. After each successful data transmission the total number of stations will be decreased by two instead of one. This will reduce the collision probability for the remaining stations and increase the total throughput.

The above issues are the general gains of DCF+ over DCF. If we compare the results of DCF+ in basic and RTS/CTS access methods, we can see that the performance improvement in RTS/CTS access method is slightly higher than basic access method. Because of the DCF+ protocol structure in basic access, the SIFS+CTS+SIFS time duration is spent before transmitting the second data frame from destination to source station. On the contrary in RTS/CTS access, the ACK frame for the first data frame is assumed to be the RTS frame and following a CTS frame the second data is sent from destination to the source station. This time difference between two access schemes causes a small difference in throughput improvement.

Another important data that will be examined for the disaster scenario is the *disaster duration.* The duration that the network shall sustain the disaster conditions is very important as well as disaster throughput values. The disaster duration data is also obtained in the same manner and provided in Figure 3.7.

The disaster duration values for DCF protocol are also taken from [4] as a reference. But when we compare the disaster throughput and disaster duration values in [4], we found that there is a conflict between those values for the same number of stations. As explained above, during the implementation of this scenario it is assumed that each station shall have only one packet having fixed size of 8184 bits. Based on this assumption, it is possible to calculate the disaster duration by using the disaster throughput value for the same number of stations. When we calculate those disaster duration values, we have found that the values given in [4] should be higher to match with the calculated values. For that reason, the calculated values are used in our figure to keep the consistency between the disaster throughput and disaster duration values.

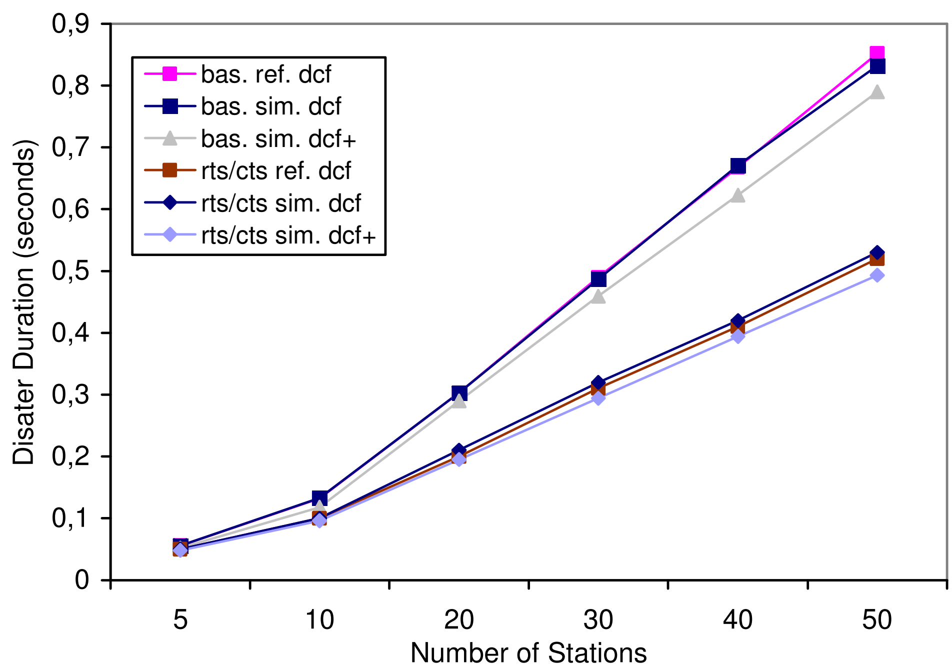

Figure 3.7 The Disaster Duration of IEEE 802.11 Wireless Networks

The disaster duration values depend on the disaster throughput values. So the results are in parallel with the disaster throughput results obtained by the simulation. The higher disaster throughput values cause the lower disaster duration values for the same number of stations. In DCF+ protocol, the disaster durations are shorter that the results obtained in DCF protocol and these results all comply with the saturation throughput values.

3.6 Service Differentiation Analysis in IEEE 802.11 Wireless LAN for DCF and DCF+ Protocols

3.6.1 General

The QoS issue in WLAN is more challenging than in wired networks because of the limited bandwidth and unreliable channel. It is very hard to predict the service level for different service classes in the presence of random access channels. In [5], the analytical model and simulation results which predict the saturation throughput of DCF with multiple service classes is introduced. In our work, the same simulation results are obtained for model validation. Afterwards, the same conditions are applied for DCF+ and the results are compared with those obtained for DCF.

3.6.2 Saturation Throughput

The detailed saturation throughput analysis is provided in [5] for the QoS differentiation issue, based on assigning different initial contention window sizes to each different class. As a result of the analysis, it can be shown that for large n_i and n_j, we have $p_i \approx p_j$ where p_i is the probability of a transmission from a station in the *i*th service class collides in any time slot and n_i is number of stations in the *i*th class. If service is differentiated using the initial contention window only, the transmission probability of the *i*th station is approximately inversely proportional to the initial contention window W_i. This means that the achievable bandwidth for users subscribing each service class is almost proportional to the inverse of their initial contention windows.

3.6.3 Model Validation and Simulation Results

The MAC system parameters used in the simulation are identical to those used in [2]. For model validation, the number of service classes is set to 2. In order to investigate the impact of the number of stations on service differentiation level, we set n1 = [2 10] and n2 = [2 10 20 40] where n1 and n2 are the number of stations for Class 1 and Class2 respectively. Only the initial contention window size will be used to differentiate between the two service classes. After each unsuccessful transmission attempt, the contention window will be doubled up to the 4 collisions. Three pairs of initial windows, {W_1 = 16, W_2 = 32}, {W_1 = 32, W_2 = 48}, {W_1 = 32, W_2 = 64}, are used. But the results for {W_1 = 16, W_2 = 32}, {W_1 = 32, W_2 = 48} will be investigated. In [5], the model validation is performed for only basic access mechanism. The same model will be applied for RTS/CTS access mechanism for DCF and DCF+. The results for RTS/CTS will also be discussed. First of all, the simulation presented in [5] is repeated to validate our model. Then the simulation is repeated for DCF+ with the same parameters. The results in [5] are taken as a reference. The reference values, our simulation results for DCF and DCF+ method covering two different initial contention window values are given in Figure 3.8 and 3.9 respectively. All of the simulations are performed for the basic access method.

The Figure 3.8 presents the bandwidth of each station from different classes. The values that are presented as reference are from [5]. The results represented, as "simulation dcf" are the results of simulation performed for the model validation. The results obtained by the simulation DCF+ protocol by using the same parameters are given as "simulation dcf+". All the results are provided for two classes, which are marked as S_1 and S_2.

As it can be seen from the figure, the simulation results of DCF protocol are very close to reference values for small number of stations. The difference between the simulation results and reference values are increasing with the increasing number of stations.

As stated before, the simulation results given in [5] are based on the same parameters given in [2]. So, the total throughput value calculated from the results in [5] should match the saturation throughput values given in [2] for the same number of stations. But when we compare the results, we saw that the values given in [5] are higher than the values in [2]. On the other hand, our simulation results are very close to the values given in [2].

Also, if we analyze simulation results for both classes, the bandwidth of first class (initial contention window=16) is almost twice of the second class's bandwidth. This result also shows that our simulation results are conforming to the analysis in [5]. The DCF+ simulation results are also given in the same figure for the same parameters. As shown from the figure, the DCF+ protocol does not give the desired result for the QoS differentiation. The second class's bandwidth is slightly higher than two third of the first class's bandwidth. For the first class, the expected result was having twice bandwidth of second class. The reason for this undesired result is the structure of the DCF+ protocol. If we remember the structure of the DCF+ protocol, it depends on the data frame transmission from destination to source station after receiving the data frame from the source station.

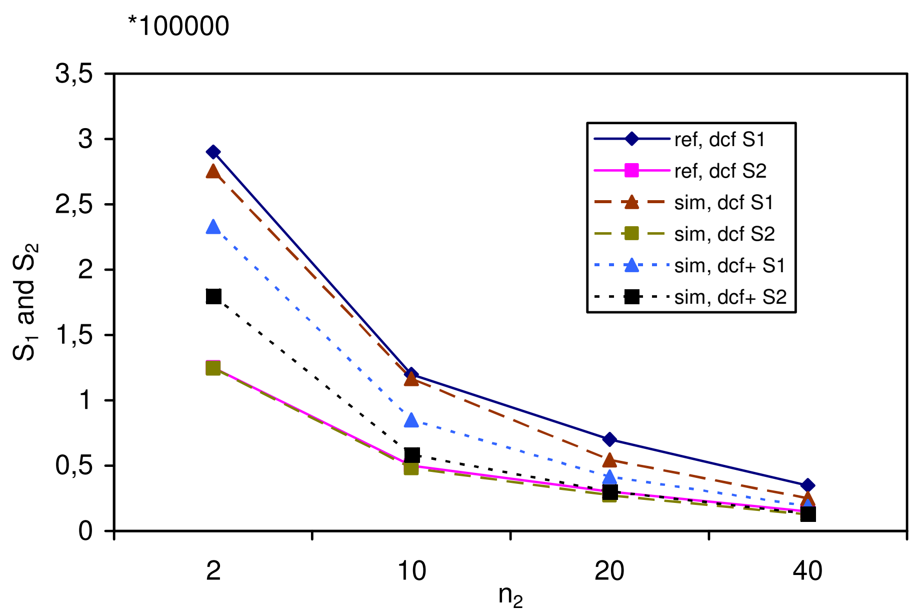

Figure 3.8 Comparison of reference values and simulation results for DCF and DCF+ in basic access method: In saturation condition, the bandwidth of S_1 and S_2 when n_1 = 2, W_1 = 16, W_2 = 32

This situation changes the QoS differentiation mechanism proposed in [5]. Because smaller initial contention window sizes provide priority to the stations specified as first class. But in DCF+, after any data transmission from the first class station, if the destination station is a second class station, this station automatically gets the priority to send data frame in the network. This causes the service classes to be mixed. The DCF+ curves in the figure also show this. There is still a difference between the bandwidth of first and second class station but this difference is not directly proportional to their initial contention window sizes.

The simulation results of DCF and DCF+ for different initial contention window sizes (W_1 = 32, W_2 = 48) are given in Figure 3.9.

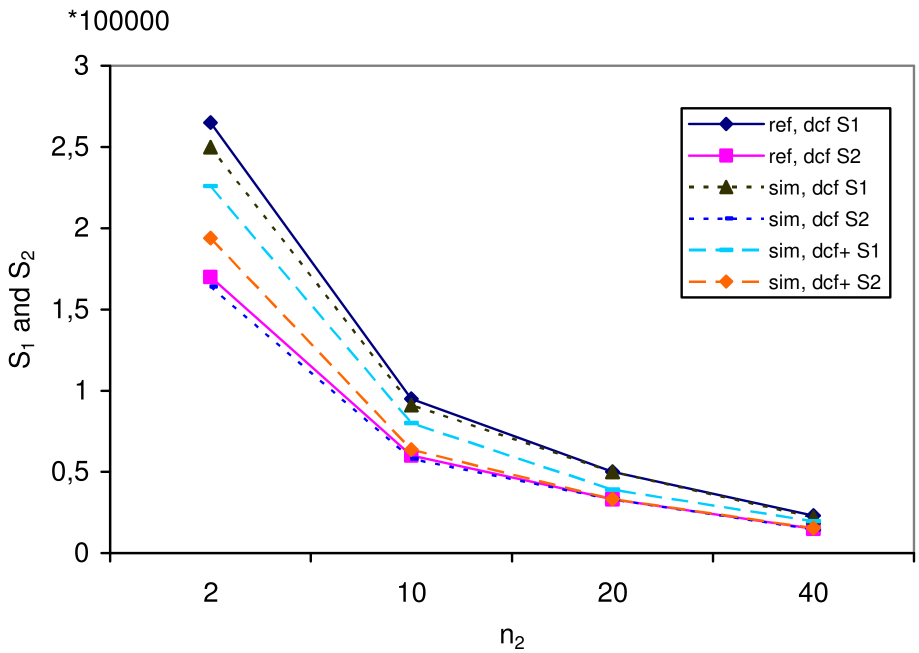

Figure 3.9 Comparison of reference values and simulation results for DCF and DCF+ in basic access method: In saturation condition, the bandwidth of S_1 and S_2 when n_1 = 2, W_1 = 32, W_2 = 48

The results shown in Figure 3.9 are consistent with the results given in Figure 3.8. Where the bandwidth ratio between two classes was 1/2 in Figure 3.8, this ratio is about 2/3 in Figure 3.9. This ratio is equal to the ratio between the initial contention window sizes of these two classes. So that, the result of the analysis in [5] is verified again with this simulation results. The DCF+ results in Figure 3.9 are also in parallel with the results in Figure 3.8. Due to the same reasons as explained above, the service differentiation mechanism using different initial contention window does not work in DCF+ protocol. The same simulations are repeated for the ten stations in the first class when changing the number of station in the second class from 2 to 40. There was no new outcome obtained from those simulations that can be added to the above results.

3.6.4 Additional Work on RTS/CTS Access Scheme and Simulation Results

The service differentiation in 802.11 WLAN is discussed above with the reference values taken from [5]. But in [5], this subject is evaluated only for basic access method.

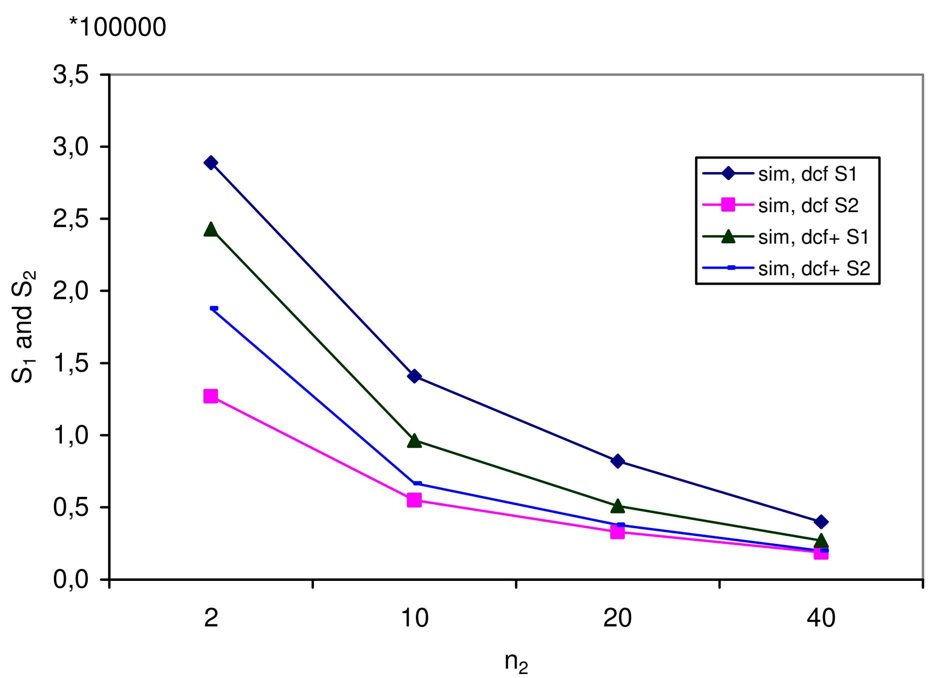

Figure 3.10 Comparison of simulation results for DCF and DCF+ in RTS/CTS access method: In saturation condition, the bandwidth of S_1 and S_2 when n_1 = 2, W_1 = 16, W_2 = 32

In this section, the same parameters will be used and the service differentiation issue will be discussed for RTS/CTS access scheme. The same parameters are used for obtaining the data and the results for W_1 = 16, W_2 = 32 and W_1 = 32, W_2 = 48 is given in Figure 3.10 and Figure 3.11 respectively.

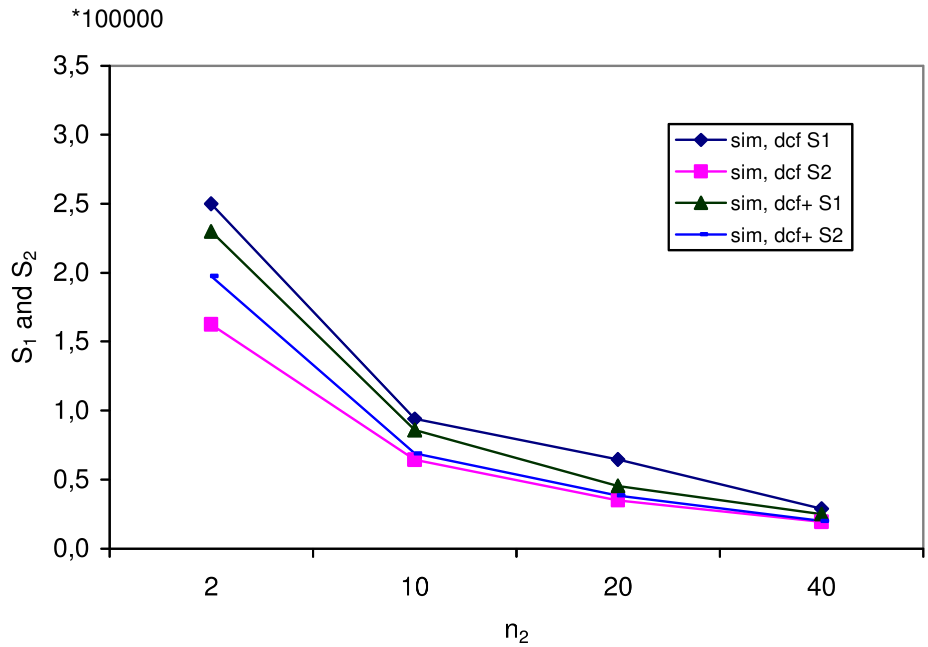

Figure 3.11 Comparison of simulation results for DCF and DCF+ in RTS/CTS access method: In saturation condition, the bandwidth of S_1 and S_2 when n_1 = 2, W_1 = 32, W_2 = 48.

The obtained data for RTS/CTS access method provides very close results to the basic access method. The total bandwidth is increased in RTS/CTS method, but the proportion of bandwidth shared by each class is still directly proportional to the initial contention window sizes in DCF protocol. Also in DCF+ protocol, the results are very similar to those obtained in basic access method. These results showed that the service differentiation by changing the initial contention window sizes does not work either in DCF or in DCF+ protocol.

CHAPTER IV

CONCLUSION

In this thesis, performance of the IEEE 802.11 wireless network is evaluated under different conditions. The throughput values obtained through DCF and DCF+ [3] are compared to evaluate the performance improvement of the DCF+ protocol. All of the results are obtained under saturation conditions i.e. all the stations have always a packet ready to transmit.

In order to evaluate the performance characteristics of the DCF+ protocol, the throughput values are obtained in different traffic conditions with predefined assumptions. For this study the scenario which is called as "disaster scenario" [4] is used.

The subject of service differentiation is also investigated in this study. The service differentiation mechanism is applied in DCF+ by dividing the stations in the network into two service classes and assigning different initial contention window sizes for these service classes. The obtained throughput values are compared with the results obtained in DCF and the successfulness/unsuccessfulness of the DCF+ protocol for service differentiation is discussed.

The throughput values are obtained for DCF+ not only for the conditions defined in [2] and [3] but also under Poisson traffic and special traffic conditions defined in [4]. Also service differentiation procedure defined in [5] is applied in DCF+. The main motivation behind this work is to analyze the throughput efficiency and characteristics of the DCF+ protocol under different conditions and determine the boundaries of the DCF+ protocol.

The thesis first introduces the throughput results for the DCF protocol obtained through the simulation program and the comparison of these results with the throughput values given in [2]. The simulation program is verified by getting the same results as in [2]. After having verified the simulation program, the simulation of the DCF+ protocol is performed according to the defined conditions in [3]. The reason of the difference between the results of DCF+ throughput values obtained through simulation and the values in [3] are discussed in Chapter 3. Additionally, the DCF protocol simulation is performed according to the conditions defined in [3]. The throughput results of DCF, DCF+ and the values taken as a reference from [3] is provided and the performance improvement of the DCF+ protocol is discussed.

In order to evaluate the performance of the DCF+ protocol in conditions other than the saturation condition, Poisson traffic and the scenario defined in [4] which is called as “disaster scenario” is applied in DCF+.

With Poisson traffic it is seen that, the DCF+ has no performance improvement in RTS/CTS access scheme other than saturation condition. In basic access scheme, while DCF+ has performance improvement under saturation condition, it has also slight improvement under lower traffic conditions.

The conditions defined in [4] which are also taken from [2] as a reference is used. The simulation of the disaster scenario is performed for both DCF and DCF+ protocols. The simulation is applied for both basic and RTS/CTS access schemes. Through simulation, the disaster throughput and disaster duration values are obtained for both access schemes. First of all, the disaster duration and disaster throughput values obtained by simulation are compared with the results in [4]. Then the disaster duration and disaster throughput values obtained for both DCF and DCF+ is compared and found that DCF+ has a performance improvement. The rate of improvement is in parallel with the improvement experienced in saturation conditions.

The performance characteristics of the DCF+ protocol for the service differentiation are also investigated. The analysis made in [5] is taken as a reference. For service differentiation, the initial contention window sizes are changed for different class of services. The same parameters are used for the simulation of the DCF and DCF+ protocols for basic access method. The simulation results obtained in DCF are compared with the values in [5]. Then the results obtained for DCF+ are compared with the DCF values provided in [5]. As an additional work, the simulation of the DCF and DCF+ protocols is performed for RTS/CTS access scheme and the same parameters stated in [5] are used for simulation. It is found that the DCF+ protocol cannot effectively provide service differentiation between the classes in both basic access and RTS/CTS scheme. In DCF, the bandwidth that is used for different classes are directly proportional to their initial contention window sizes. But in DCF+, the bandwidth of the first class is approximately 1.5 of second class while having 2 times smaller initial contention window size. Because of working principal of DCF+, any station from the second class may have priority to send a data frame whether it receives data from the station of first class or second class and this causes the undesired results.

In this thesis, the work in [3] has been moved one step forward. Some additional work is performed to assess the performance improvement of the DCF+ protocol under different traffic conditions.

In [3] it is shown that DCF+ has improved the throughput of the network under saturation conditions. In this thesis it is shown that DCF+ has performance improvement not only under saturation conditions but also under different traffic conditions and scenarios. The rate of performance improvement under different traffic conditions is also in parallel with each other. On the other hand, DCF+ simulations based on service differentiation application have shown that DCF+ fails to provide efficient differentiated services to the different classes.

As it is mentioned in Chapter 2, the DCF+ protocol is fully compatible with the DCF protocol. For that reason the implementation of the DCF+ protocol does not require any additional cost. If performance improvement is taken as an important measure, DCF+ should be used widely in the wireless network. On the other hand, the DCF protocol is more efficient in service differentiation applications and DCF+ should not be used for that kind of applications.

DCF+ was originally proposed for service to a reliable transport protocol. Our work shows that DCF+ has performance improvement at MAC layer. So, it can be used for both TCP and UDP applications and performance improvement can be obtained for unreliable transport protocol as well as reliable transport protocol.

The DCF+ protocol's efficiency depends on the presence of data to send on the backward direction after the original data exchange on the forward direction. On the other hand, the present TCP implementations may not need to receive TCP ACK for each successful packet transmission. Therefore, the conditions that provide main motivation for proposing the DCF+ have changed. So, the efficiency of DCF+ to improve the performance should be reevaluated and the necessity of the DCF+ protocol should be discussed with regard to the present TCP implementations.

As a future work, some changes can be applied to the DCF+ protocol and the performance of DCF+ can be compared with the DCF protocol. Further traffic types definitely merit performance investigation. Bursty traffic, in particular, needs to be studied. Also the EDCF protocol that is defined specifically for the service differentiation in wireless network applications can be adapted to DCF+.

The future trends in wireless telecommunications technology and services include improved coverage and universal roaming, increased integration of services, increased network based functionality, increased functionality of end user equipment, and improved spectrum efficiency. It appears that steady progress will be made toward the goal of being able to communicate anywhere, anytime, and in any mode such as voice, data, image, or video. These capabilities will eventually include multimedia and even broadband capabilities as well and that radio communications devices will increasingly be embedded in all types of equipment.

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