ANALYSIS AND ENHANCEMENT OF WIRELESS LANS IN NOISY CHANNELS

KHODER SHAMY

A THESIS
IN
THE DEPARTMENT
OF
ELECTRICAL AND COMPUTER ENGINEERING

PRESENTED IN PARTIAL FULFILLMENT OF THE REQUIREMENTS
FOR THE DEGREE OF MASTER OF APPLIED SCIENCE
CONCORDIA UNIVERSITY
MONTRÉAL, QUÉBEC, CANADA

JUNE 2008

Your file Votre référence
ISBN: 978-0-494-42497-1
Our file Notre référence
ISBN: 978-0-494-42497-1

Abstract

Analysis and Enhancement of Wireless LANs in Noisy Channels

Khoder Shamy

Without a doubt, Wireless Local Area Networks (WLANs) technology has been encountering an explosive growth lately. IEEE 802.11 is the standard associated with this promising technology, which enures shared access to the wireless medium through the distributed coordination function (DCF). Recently, the IEEE 802.11e task group has made extensions to WLANs medium access control (MAC) in order to support quality of service (QoS) traffic.

An inherited problem for WLANs, is the volatility of the propagation medium, which is a challenging issue that affects the system performance significantly. Consequently, enhancing the operation of the DCF in noisy environments is of great interest, and has attracted the attention of many researchers.

Our first major contribution in the presented thesis, is an analytical and simulation analysis for the binary exponential backoff (BEB) scheme of the DCF, in the presence of channel noise. We show that following the BEB procedure when a host encounters erroneous transmission is needed only if the channel was highly loaded. However, incrementing the contention window (CW) upon each packet failure, whether caused by instantaneous transmission (i.e. collision) or channel noise, will result in the waste of air time if the channel was lightly loaded. Accordingly, we present a hybrid access method that adapts the CW according to the channel load along with the frame error rate (FER).

Other means to overcome the channel noise is the adjustment of the transmission rate. Many rate adaptation (RA) algorithms were introduced in the past few years, including the Automatic Rate Fallback (ARF) which is currently implemented in the wireless cards. Yet, many drawbacks are associated with these RA algorithms; specifically, in regard to the techniques and events that should trigger the rate change. Moreover, the IEEE 802.11e QoS flows requirements were not considered with the latter schemes. Accordingly, our next major contribution in this work is the presentation of a novel rate adaptation scheme. The simplicity of the introduced rate adaptation scheme is that it relies on the MAC layer parameters rather than those of the PHY layer when adjusting the rate. Furthermore, our algorithm supports the IEEE 802.11e MAC extensions where QoS traffic requirements were integrated in the procedure of adjusting the bit rate. Hence, strict real-time flow parameters such as delay and maximum drop rate are respected. Finally, we enhance the dynamic assignment of transmission opportunities (TXOPs) in order to offer fair air-time for nodes facing high packet loss rate.

Acknowledgments

"Humankind's God-given creativity" — *New York Times*

Contents

List of Figures

List of Tables

List of Publications

- **Khoder Shamy**, Chadi Assi, Lei Guang, "A Study on the Binary Exponential Back-off in Noisy and Heterogeneous Environment", *in Proceedings of MSN'2007*, Beijing, China , December 2007.

- **Khoder Shamy**, Chadi Assi, "Efficient Rate Adaptation with QoS Support for Wireless Networks", submitted to *IEEE GLOBECOM'2008*, New Orleans, USA, December 2008.

Abbreviations

AC - Access Category

ACK - Acknowledgement Frame

AIFS - Arbitration Inter-frame Space

AIMD - Additive Increase Multiplicative Decrease

AOB - Asymptotically Optimal Backoff

AP - Access Point

AR - Auto Regressive

ARF - Automatic Rate Fallback

BEB - Binary Exponential Backoff

BER - Bit Error Rate

BS - Base Station

CAPs - Channel Access Parameters

CARA - Collision Aware Rate Adaptation

CBR - Constant Bit Rate

CCA - Clear Channel Assessment

CFB - Contention Free Bursting

CSMA/CA - Carrier Sense Multiple Access With Collision Avoidance

CTS - Clear-To-Send Frame

CW - Contention Window

DCF - Distributed Coordination Function

DIFS - DCF Inter-frame Space

DSSS - Direct Sequence Spread Spectrum

EDCA - Enhanced Distributed Channel Access

EDCF - Enhanced DCF

FER - Frame Error Rate

FHSS - Frequency Hopping Spread Spectrum

HCF - Hybrid Coordination Function

MAC - Medium Access Control
NAK - Negative Acknowledgment Frame
NAV - Network Allocation Vector
PCF - Point Coordinator Function
PCS - Physical Carrier Sensing
PHY - Physical (Layer)
QAP - QoS Access Point
QoS - Quality of Service
QSTA - QoS Station
RA - Rate Adaptation
RF - Radio Frequency
RTS - Request-To-Send Frame
SBA - Surplus Bandwidth Allowance
SIFS - Short Inter-frame Space
SLRC - Station Long Retry Count
SNR - Signal-to-Noise Ratio
SSRC - Station Short Retry Count
TXOP - Transmission Opportunity
VCS - Virtual Carrier Sensing
WLAN - Wireless Local Area Network

Chapter 1

Introduction

Currently, Wireless Local Area Networks (WLANs) are rapidly emerging as the technology of choice for the last hop access to the Internet. This is mainly due to the nature of wireless networks that gave users freedom and convenience of location while maintaining easy access to networking resources (e.g., the Internet). Other factors that led to the popularity of WLANs are ease of deployment and cost effectiveness. Unlike wired networks, which require physical cabling to individual workstations, a single access point can easily serve multiple mobile hosts equipped with wireless network cards. Consequently, most educational institutions now have wireless access points spread across classrooms and libraries (e.g., there are over 200 access points installed throughout both Concordia campuses [3]). Moreover, many Internet service providers are introducing wireless accessibility to clients across cities. Google Inc., for example, offers free wireless access to the residents of the city of Mountain View, the hometown of the corporation [2].

The IEEE 802.11 protocol represents the standard protocol that WLANs devices follow. The standard utilizes license-free radio frequency bands for its operations: 2.4 GHz for 802.11b and 802.11g, and 5 GHz in the case of 802.11a [1]. The medium access control

(MAC) represents one of the most important elements of 802.11. The MAC protocol is used to provide arbitrated access to a shared medium, in which several terminals compete for accessing the network resources using the distributed coordination function (DCF) as the primary access mechanism. It is based on the carrier sense multiple access with collision avoidance (CSMA/CA) protocol.

As today's real-time applications (e.g. voice and video conference programs) hold strict Quality of Service (QoS) requirements when admitted to transmission medium, further challenges have emerged for WLANs. In response, the IEEE 802.11e task group [4] was established, whose primary duty is to support QoS traffic extensions for the existing standard. Accordingly, the MAC layer access functions have been modified to provide differentiated services for audio, video and data flows.

1.1 Thesis Contribution and Motivation

Due to the shared access of hosts to the same wireless medium, one can find that the functionality of this emerging technology is a complicated matter. Moreover, unlike wired infrastructure, wireless medium is known to be very vulnerable to noise effects that can easily disrupt data transmission (e.g., fading, multi-path loss [35]). Consequently, the IEEE 802.11 task group has introduced advanced and unique mechanisms to handle the operations of the MAC layer. Binary exponential backoff (BEB), represents the DCF mechanism used to resolve failed transmissions due to packet collisions (i.e., instantaneous transmissions).

The major deficiency in BEB is that all failed transmissions are considered to be caused by collisions from simultaneous transmissions in the same time slot. However, this is not always true since failure can also result from other factors such as the channel noise or interference caused by neighboring transmissions (e.g., hidden nodes). The latter form of transmission failure occurs in multi-hop wireless networks and will not be considered in this thesis. Consequently, researchers have been working for several years to enhance the BEB functionality given the volatile nature of the wireless medium. More specifically, the BEB operation in the event of noisy environment has been the primary topic of several research work (e.g. [7], [8]). Our first contribution in the presented thesis tackles this issue. For this reason, we consider two binary exponential backoff (BEB) algorithms; a standard BEB where the Contention Window (CW) is doubled upon every packet loss, and another access method that triggers increasing CW only after a collision. It is therefore essential to distinguish among the types of frame losses and we present in this thesis such a method. We show, through theoretical analysis and simulation, that the second access procedure outperforms the standard BEB when the network is lightly loaded. However, as the number of nodes increases, the latter method yields poor system performance opposed to the standard BEB due to the high collision rate among contending nodes. Afterwards, we present a new hybrid access method that combines both methods advantages in order to achieve optimal throughput regardless of the network load.

A common property of the IEEE 802.11 physical schemes (802.11 a/b/g) is that each scheme offers several transmission rates that are utilized for data transmission over the channel. The different bit rates are characterized in turn by their modulation and coding which enable some to stand firm in highly error-prone wireless medium while others suffer severe degrading performance. Accordingly, link error detection and algorithms that decide when and how to switch to an upper or lower transmitting PHY rate have been of much interest lately. In our second contribution, we present a novel rate adaptation algorithm that can efficiently identify the threshold frame error rate (FER) at which link adjustment is required, based on a simple throughput analysis at the MAC layer. Additionally, when our rate adaptation algorithm operates on IEEE 802.11e supported MAC, it implements a differential treatment for QoS flows. When a real-time stream with QoS requirements is admitted, critical constraints such as delay bound and maximum packet drop count are integrated in the selection of the most convenient transmission rate that best respects the flow requirements.

1.2 Thesis Organization

The rest of this thesis is organized as follows. In the following chapter, besides reviewing the IEEE 802.11 and 802.11e medium access functions, we survey the literature studies that analyze and enhance the mechanisms of the access functions in noisy channels. Afterwards, we summarize various rate adaptation studies and evaluate their performance. In Chapter 3, we present our analysis for the BEB operations in noisy environment under different network loads. Accordingly, we draw our conclusions and analyze a novel access method that differentiates packet losses. Moreover, we study the BEB impact on nodes performance in heterogeneous environment (i.e., where nodes experience different frames error rate at the same time) in terms of fairness and overall system throughput. A new rate adaptation algorithm is introduced in Chapter 4. The algorithm switches between the different physical rates based on the error level estimated at the MAC layer along with a simple throughput analysis. Moreover, we integrate IEEE 802.11e QoS requirements, such as delay and maximum drop rate in the link adaption process. Dynamic bandwidth allocation of transmission opportunities (TXOPs) is introduced, as well, to better serve the differential needs of the multi-media applications. Finally, we conclude the thesis and present future research directions and suggestions in Chapter 5.

Chapter 2

IEEE 802.11/11e Access Functions and Rate Adaptation in Noisy Channels

2.1 Introduction

Throughout this chapter, we first examine both 802.11 and 802.11e standards. Then, we review the performance of the IEEE 802.11 standard in noisy channels. We primarily focus on the distributed coordination function (DCF) implemented at the MAC layer. Afterwards, we survey related literature studies that attempted to analyze and enhance the mechanisms of the access functions, specifically through modifying the operation of the binary exponential backoff (BEB) in the presence of transmission errors. Then, we review the methodologies used in implementing various rate adaptation techniques. Initially, we categorize RA mechanisms based on the approach used in the adjustment of the transmission rate, and then we evaluate their performances.

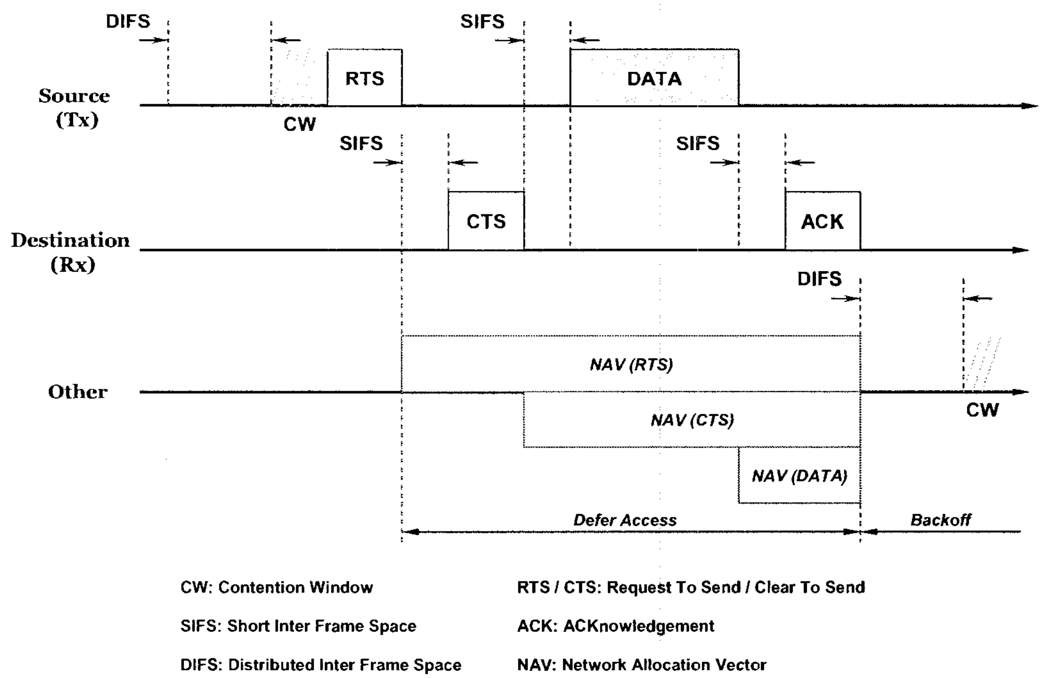

Figure 2.1: IEEE 802.11 DCF procedure

2.2 IEEE 802.11 standard

The IEEE 802.11 protocol [1] has become the predominant technology for Wireless Local Area Networks (WLANs). One of the most important elements of the 802.11 is its medium-access control (MAC); the MAC protocol is used to provide arbitrated access to a shared medium, in which several terminals access and compete for using the network resources. The IEEE 802.11 wireless networks employ the distributed coordination function (DCF) as the primary access mechanism; it is based on the carrier sense multiple access with collision avoidance (CSMA/CA) protocol and the binary exponential backoff. The performance of an IEEE 802.11 network largely depends on the operation of this backoff mechanism. Accordingly, there have been several research efforts for analyzing the saturation throughput achieved under DCF in single hop WLANs. The binary exponential backoff (BEB) of

DCF is used for resolving collisions among terminals attempting to simultaneously transmit on the channel. To ensure packet transmission reliability, MAC acknowledgement frames (ACK) are used to indicate the correct reception of the data packets. When an ACK frame is not received upon a transmission, the transmitting station assumes its packet has been lost due to collision and accordingly invokes the BEB mechanism for retransmission. However, packet losses may also occur from other transmission errors, which may arise from the combination of both path loss and receiver noise.

The IEEE 802.11 defines two basic access methods [1]:

1. A fully distributed mechanism called distributed coordination function (DCF), which allows contention access for wireless media.

2. A centralized mechanism called point coordinator function (PCF), which requires centralized access points.

DCF is the MAC layer basic access method for WLANs and ad hoc networks. It is also known as carrier sense multiple access with collision avoidance (CSMA/CA). CSMA/CA is designed to reduce collisions when multiple nodes access the shared medium. Carrier Sense is performed by both physical sense and virtual sense mechanisms. There are two communication options in DCF: (1) four-way handshaking, that is, RTS-CTS-DATA-ACK, which is suitable for long frame data transmission (as shown in Figure 1); (2) two-way handshaking, that is, DATA-ACK, which is suitable for short frame data transmission. A node with packets to transmit first senses the medium. If the medium is idle for at least a certain period, DCF inter-frame space DIFS, it will immediately request the channel by sending a short control frame request to send (RTS) to the receiver node. If the receiver correctly receives the RTS, it will reply with a short control frame clear to send (CTS). Once the sender receives the CTS, it will start to transfer DATA. After the successful reception of DATA, the receiver sends an ACK to the sender. The exchange of RTS/CTS prior to the actual data transmission reduces the high collision probability by distributing the medium reservation information and solves the hidden terminal problem. The RTS/CTS contains a duration field indicating the time (in microseconds) after the end of present frame transmission that the channel will be reserved to complete the data or management frame transmission. Any node within the transmission range of either the sending node or the receiving node hears the RTS/CTS exchange will learn about the medium reservation and adjust its network allocation vector (NAV), which indicates the amount of time that the node should defer. The collision will mostly happen when the current node completes its transmission and multiple nodes are waiting to contend for the channel. Thus, each node with data to transmit will generate a random backoff number from the range [0, CW] for an additional deferring time after the channel is idle for a DIFS time, where CW is the contention window size maintained by each node. The backoff counter is decremented as long as the channel is sensed idle, stopped when a transmission is detected on the channel, and restarted when the channel is sensed idle again for more than a DIFS. Once the backoff counter reaches zero, the sending node will reserve the channel by exchanging RTS/CTS as described above. If a node sends RTS but does not receive CTS within certain time, the node will defer by doubling its CW size and choosing a random value from the new range and retransmit RTS with limited times. If the RTS retry time is more than the station short retry count (SSRC) the sending node will drop the DATA packet and inform the network layer of a link breakage. Alternatively, if the ACK is not received within certain time, the sending node will retransmit the DATA packet for limited time (i.e., SSRC for a short frame DATA, or station long retry limit SLRC for a long frame DATA). The two-way handshake mechanism is similar to the 4-way handshake one, however there is no exchange of RTS/CTS messages preceding the DATA/ACK transmission.

2.2.1 Carrier Sensing Types:

It is worthwhile to define the two carrier sensing types, utilized by the DCF to determine the status of the wireless medium (i.e., busy or idle) upon initiating a frame transmission [1]:

1. The mandatory physical carrier sensing (PCS): Here, a node monitors the radio frequency (RF) energy level on the channel and initiates channel access attempt only if the power of the detected signal is below a certain carrier sense threshold (CS_{th}).

2. The optional virtual carrier sensing (VCS): Using this carrier sensing method, nodes observe the channel busy for a period indicated in the MAC frames defined in the protocol. More specifically, nodes hearing the RTS/CTS exchange messages will adjust their network allocation vector (NAV) according to the duration of the complete four-way handshake mechanism.

2.3 IEEE 802.11e standard

IEEE 802.11 generally treats all traffic flows with the same priority. A voice or video flow holding specific requirements will have to choose the same random waiting time as the other best effort flows. Hence the current DCF implemented in 802.11 is not suitable for

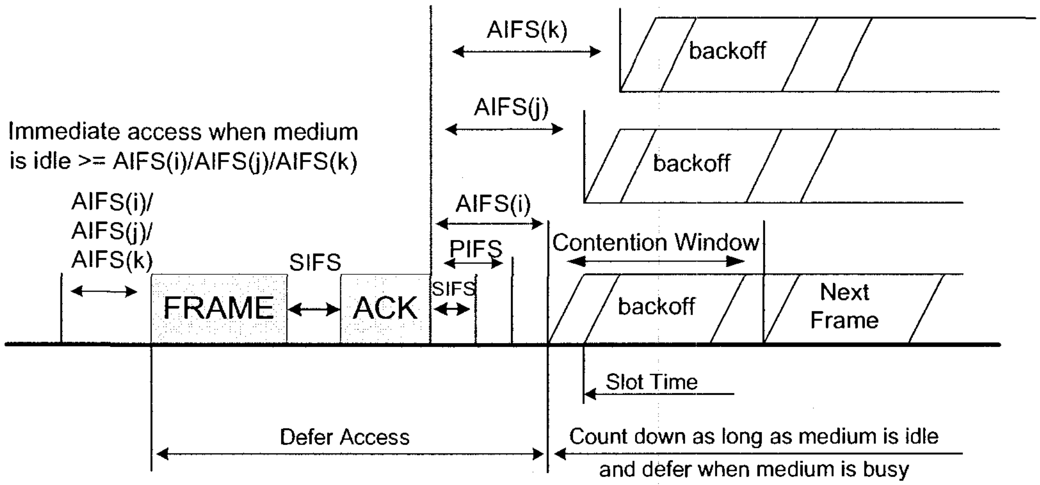

Figure 2.2: IEEE 802.11e EDCA

supporting multimedia applications with different quality of service (QoS) requirements.

Accordingly, the IEEE 802.11e task group was established to provide QoS support for applications at the MAC layer level. The new standard includes a novel access function called Hybrid Coordination Function (HCF). It has support for both, contention–based, Enhanced Distributed Channel Access (EDCA) and centrally–controlled (HCCA) channel access ([48], [49] and [47]).

Priority	**AC**	**Traffic Type**
1	0	Best Effort
2	0	Best Effort
0	0	Best Effort
3	1	Video Probe
4	2	Video
5	2	Video
6	3	Voice
7	3	Voice

Table 2.1: Traffic Types and Equivalent Access Categories for IEEE 802.11e

The operation of the EDCA requires four different access categories (ACs), in which each represents nothing but a virtual DCF. Here, the duration that a host uses to sense and wait for the expiry of the CW backoff counter will depend on the AC of the flow type. In fact, EDCA supports eight different priorities as shown in Table 2.1 and each of the four AC upholds flows according to the traffic type. In addition, QoS differentiation is met further through the channel access parameters (CAPs) which are assigned to the different ACs as well.

Hence, EDCA requires a packet to be labeled with the appropriate priority value when arriving from upper layers. Afterwards, it will be enqueued to the appropriate AC according to the traffic type.

In order to offer specific flows a higher chance of transmission in favor of low priority data, EDCA manipulates the following parameters (Figure 2.2):

1. Contention Window (CW): Each AC is assigned a different minimum CW, CW_{min}. Hence, high priority AC will have smaller CWs, whereas low priority flows will have to contend for accessing the medium while using larger CW.

2. Inter-Frame Space: The DCF uses the DIFS period as means to access the channel at the MAC level. In EDCA, however, Arbitration Inter-Frame space (AIFS) is used instead. Each AC will be given an AIFS value according to the equation:

$$AIFS = SIFS + AIFSN \times SlotTime \tag{2.1}$$

   where the *AIFSN* represents a specific AIFS number assigned for each AC and is based on physical settings. *SlotTime* is the duration of the time slot (e.g. $20\mu s$ for PHY 802.11b). Hence, the smaller the AIFS value, the higher the probability of accessing the channel.

2.4 DCF Access Methods in Noisy Channels

Due to the volatile nature of the wireless medium, enormous research efforts have focused on evaluating and enhancing Wireless LANs access methods in the presence of transmission errors. Many researchers have considered studying the performance and analysis of the access methods in noisy environments using analytical and simulation approaches. Among those many who suggested modifications to the operations of the DCF when high bit error rates (BER) influence transmission losses. For example, the authors of [7] evaluated, through simulations, the performance (throughput and fairness) of different access methods by varying the bit error rate (BER) of the channel. The authors noticed that the performance of the different access methods in terms of throughput and fairness radically changes when bit error rate increases. For small error rates, *Asymptotically Optimal Backoff*[1] (AOB) and *Idle Sense*[2] access methods provide good throughput performance, while AOB fails to attain good fairness. On the other hand, when error rates increase, only *Idle Sense* achieved both, good throughput and fairness. This is due to the fact that *Idle Sense* does not use the exponential backoff algorithm.

The authors of [8], [10] presented analytical studies (extensions to Bianchi's model [31]) for the saturation throughput of the IEEE 802.11 access method. Here, a known and fixed BER is assumed in [8] or a BER derived from the bit-energy-to noise ratio as perceived by the receiver is used to determine the packet error rate (PER) which is needed for the analytical model to derive expressions for the throughput. However, deriving the PER

[1]The AOB is an access method introduced by [11] in which the CW is adapted according to the network contention measured using the slot utilization and the average size of transmitted frames.

[2]Idle Sense was introduced in [16]. Exponential backoff is turned off and CW value is assigned using AIMD manner by comparing the number of idle slots to an optimal derived value.

from the BER does not seem to be a feasible and practical method since this information is not readily available at the transmitter. In other words, in order to enhance the access method of the IEEE 802.11 in noisy environment, the transmitter needs to separate losses due to collisions from those due to channel noise and other sources of error. Moreover, the performance evaluation studies focused mainly on the same analysis but tackling different degrading factors such as the PER influence on throughput, the packet delay and packet drop probability. The authors of [29] noted that when a packet is lost due to transmission error, doubling the contention window is inappropriate and could seriously degrade the performance. A negative acknowledgment frame (NAK) was introduced to the DCF operation in order to distinguish packet losses whether due to collision or transmission errors. The failure distinguishing strategy depends on the fact the packet collision will cause both, the header and the body of the packet to be corrupted. On the other hand, link-error failure usually affects only the body of the packet whereas the header remains intact since it has a relatively small size (i.e., small packet error rate (PER)). Hence, the modified DCF operation will act as follows [29]:

At the *Receiving Node*;

- If receiver receives a correct MAC header:
  - If MAC body is correct, an ACK frame is sent back;
  - If MAC body is wrong, a NAK frame is sent back.
- If wrong MAC header is received, the receiver sends nothing.

At the *Sending Node*;

- If an ACK frame is received, the transmission is successful;

- If a NAK frame is received, a link-error is assumed;
- If nothing is received, then a collision is assumed.

Accordingly, a suggestion to improve the standard backoff algorithm was proposed through doubling the backoff timer only when a collision loss is detected. However, no supporting analysis accompanied this proposition.

In [33], the authors presented an analytical study for the operation of the 802.11 in the presence of channel errors. Namely, the authors attempted to minimize the throughput penalty through optimizing the contention window value. However, the optimized CW value requires the knowledge of the number of active stations in the medium, which is not easily achievable. On the other hand, a very useful conclusion was noted in the latter work which states that the throughput penalty is attained because the BEB is not capable of differentiating between packet losses. This is because if a packet was lost due to link error, the CW will be increased causing significant packet delay.

The authors of [15] studied the performance of the standard BEB in noisy environments and showed that it results in poor performance since it does not have any means to differentiate between packet losses. They proposed an enhancement for BEB where a node treats any failed transmission due to channel corruption as successful and schedule the retransmission accordingly. Afterwards, they presented an error estimation method for differentiating between the causes of packet losses. However, their estimation method depends on measuring the number of active stations and their obtained results show a strong variation of the estimated channel error rate from the actual rate. Moreover, the presented study did not take into consideration the impact of the number of stations on the performance of the suggested enhancement. Indeed, in Chapter 3 of this thesis, we will prove that such an enhanced backoff operation upon a packet loss to link error is greatly dependant on the load on the channel.

2.5 Rate Adaptation in WLANs

Due to the changeable nature of the wireless channel, link error detection and algorithms that decide when and how to switch to an upper or lower transmitting data rate have been of much interest lately. In this section, we will introduce the different physical rates characteristics, then we review some of the most popular rate adaptation techniques.

2.5.1 IEEE 802.11 Physical Data Rates:

Each of the physical schemes presented by IEEE 802.11 (802.11 a/b/g), provides several data rates for the PHY layer to utilize for transmission. Each data rate is in turn characterized by its modulation and coding. Consequently, some bit rates can stand firm in highly error-prone wireless medium while others suffer severe degrading performance. Table 2.2 shows the physical specifications of IEEE 802.11b transmission rates: 11, 5.5, 2 and 1 Mbps. Bit rates using direct sequence spread spectrum (DSSS) will suffer higher transmission impairments as DSSS uses relatively wide frequencies (30 MHz) and hence signals reflected by obstacles will have high probability of being wrongly decoded. Whereas frequency hopping spread spectrum (FHSS) utilizes narrower transmission channels (1 Mhz) in addition to repeatedly changing the frequency. These factors give lower bit rates more robust transmissions, when nodes encounter multi path and high speed fading channels.

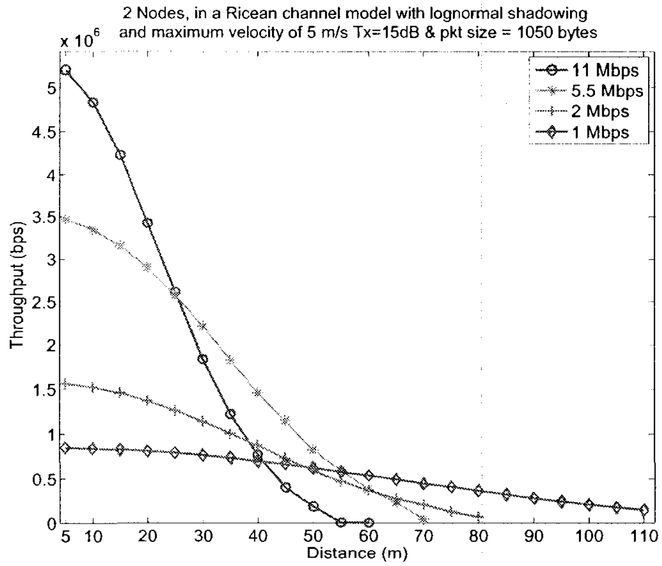

Figure 2.3: IEEE 802.11b Bit Rates performance in Ricean Fading Channel

Moreover, the higher the bit rate, the higher the bit error rate (BER) and hence, lower bit rates can endure deteriorating channel conditions. Figure 2.3 shows the performance of two saturated nodes with the different physical rates for IEEE 802.11b versus the distance separating them in a noisy channel with Ricean fading model[3] [36]. The figure shows that higher rates can be used efficiently only within short ranges, while lower rates (e.g. 1 Mbps) can maintain acceptable throughput for much longer distances which is the reason why control packets are usually transmitted using low data rates (1 or 2 Mbps).

Bit-Rate (Mbps)	Code Rate	Mandatory	Transmission Type
11	1/11	CCK	DSSS
5.5	1/11	CCK	DSSS
2	1/11	QPSK	DSSS/FHSS
1	1/11	BPSK	DSSS/FHSS

Table 2.2: IEEE 802.11b Data Rate Specifications.

[3]The Ricean fading model is a favored model for representing the indoor loss characteristics of the 2.4/5 GHz wireless channels of IEEE 802.11.

2.5.2 Survey of Popular Rate Adaptation Algorithms

The most common and basic rate adjustment algorithm is the *Automatic Rate Fallback* [40] (ARF) currently implemented in the IEEE 802.11 wireless cards. ARF alternates between the subsequent data rates based on a timing function and missed acknowledgment (ACK) frames [40] assuming that transmission failures have resulted from link error. The *Adaptive ARF* (AARF) was introduced in [24]. Contrary to ARF, AARF updates regularly the number of consecutive successful ACK frames (*SuccessThreshold*) needed to increase the rate. Upon a transmission failure, the rate is instantaneously decremented and the *SuccessThreshold* parameter is doubled. Similar work was performed in [37]. Here, two counters are kept updated; one for successful transmissions and one for failed transmissions with no differentiation between the failure types. Hence, the success counter is incremented if a frame is successfully transmitted, and the failure counter is reset to zero. Similarly, if an ACK timeout occurs, then the failure counter is incremented by one and the success counter is reset to zero. The rate is then adjusted when the counter reaches a specified threshold for failure or success. However, packets loss could have resulted from collisions with other transmissions in the channel. Decrementing the data rate in this case would worsen the collision probability because transmitting using lower rates will occupy lengthier air-time causing more contention and hence higher chance of collision. Hence, both approaches are inefficient as channel quality is not taken into consideration. The authors of [30] recognized this deficiency in ARF and accordingly proposed a rate adaptation method that differentiates between the causes of failed transmissions, whether from instantaneous transmissions (collision) or from poor channel conditions (noise). Hence, a transmitter would decrement its rate solely for subsequent failed ACK frames due to transmission errors. However, *Collision Aware Rate Adaptation* (CARA) presented in [30] uses RTS/CTS (Request to Send/Clear to Send) exchange mechanism to distinguish packet losses in addition to Clear Channel Assessment (CCA). Here, a failed RTS/CTS exchange may be considered as a collision and hence no link adaptation will be invoked; however, if multiple ACK frames failed, then data frames are considered as lost due to bad channel quality which requires using lower bit rate. However, in addition to that RTS/CTS handshake adds overhead to the transmitting time, RTS/CTS is switched off in commercial network cards. Other related studies in [45] and [46] require receiver to initiate the adjustment of the link rate based on channel condition observed. The receiver will rely either on RTS/CTS exchange or Signal-to-Noise Ratio (SNR) values to calculate the packet error rate (PER). Here, PER values along with thresholds are derived from SNR measurements. However, several real-scenario studies have recently proven that it is likely impractical to correlate loss probability with SNR and Bit Error Rate (BER) [18]. *Robust Rate Adaptation Algorithm* (RRAA) was presented in [25]; RRAA uses short-term loss estimation (over tens of frames) in order to assess the channel condition and accordingly trigger the transmission rate change. Moreover, an adaptive RTS (A-RTS) filter is introduced to subdue collisions. This A-RTS filter, however, leverages the need for the RTS/CTS exchange prior to every single packet transmission. Hence, the RTS/CTS mechanism is turned on only in case the throughput gain from the better rate adaptation outweigh the overhead of RTS. In [26], an *Opportunistic Auto Rate* (OAR) is proposed, where a station opportunistically transmits multiple back-to-back data packets whenever it experiences good channel conditions. Moreover, the latter method implicitly assumes that the PER is zero for a transmission rate in use [27].

The MAC layer protocol was modified in [38], where a negative acknowledgment (NAK) frame is sent from receiver upon link error detection and a counter is incremented to certain preset threshold upon which the physical rate would be decreased. Although the authors have shown that small overhead will be introduced by the NAK control frame, yet, it is not recommended to tamper such a widely implemented standard.

2.5.3 802.11e with QoS Rate Adaptation Work

Up to our knowledge, challenges facing link adaptation while providing QoS support for real-time multimedia flows were tackled so far by [39] and [44]. Authors of [44] suggested link adjustment based on service requirements of the flows; yet, no practical mechanism was provided but rather an extension to [46]. Threshold values for the PER critical to QoS constraints were considered in [39]. However, in addition to that the impractical correlation ([18]) between receiver sensitivity, Signal-to-Noise ratio (SNR) and BER values were used to derive the PER thresholds (QFER), no flow requirements were taken into consideration such as required delay bound or maximum drop limit.

2.5.4 Spatial Reuse and Rate Adjustment Schemes

In this section we review additional rate adaptation schemes that rely on specific physical tunable parameters for switching from one transmission rate to another. The concept of tuning carrier sensing[4] parameters along with the transmission rates was introduced in [43]. Here, the authors noticed that transmission space used by each transmission is directly dependent on the CS parameters, and accordingly, the proposed concept was called spatial

[4]In Section 2.2.1, we defined both carrier sensing types required by a node to contend for channel access.

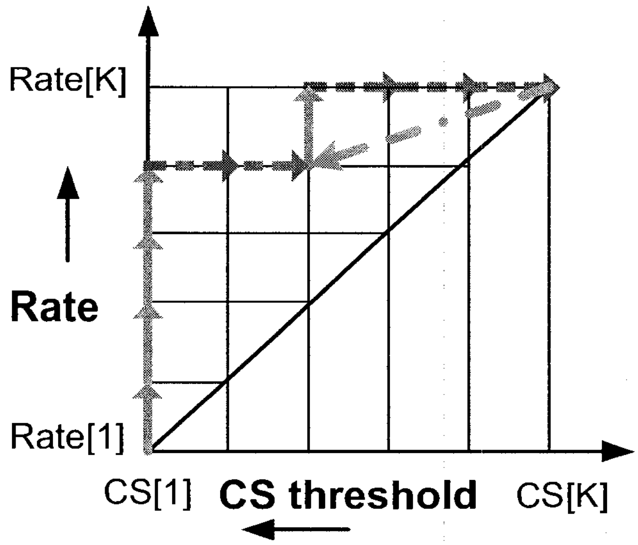

Figure 2.4: Spatial Backoff

backoff. Utilizing this concept, nodes will increase the carrier sensing threshold CS_{th}[5], in order to allow the initialization of more concurrent transmissions. On the other hand, in order to ensure that these simultaneous flows endure minimal transmission errors, the interference range[6] is decreased through decrementing the physical transmission rate. Hence, the proposed algorithm will improve spatial reuse through manipulating the CS_{th} along with the transmission rates as shown in Figure 2.4. The different transmitting rates are plotted in increasing order (Y axis) versus the carrier sensing thresholds which are plotted in decreasing order (X axis). Therefore, a node with coordinates (CS[i], Rate[j]) means that the node is currently using carrier sensing threshold CS[i] and a transmitting rate of value Rate[j]. The mechanism by which a node adjusts its transmitting rate and CS_{th} is performed as follows. Initially, a node starts transmitting on the lower physical rate. After a defined number of successful consecutive transmissions, the node increments its transmitting rate

[5]The carrier sensing threshold (CS_{th}) represents the signal strength above which a node determines the medium is busy [43].

[6]The "Interference Range" represents the radius of the area around the intended receiver, in which any transmission (other than the intended transmitter/sender) causes a frame loss at the receiver.

while maintaining the same CS_{th} value as presented by the upward solid arrows in Figure 2.4. On the other hand, the node will decrement the CS_{th} while maintaining the same transmission rate (demonstrated in the figure by the horizontal dotted arrows) in case the number of failed transmissions reaches a predefined value and CS_{th} does not reach the minimum value for the current transmission rate. Finally, if the node decrements its CS_{th} to the minimal value and yet faces transmission failure, then the transmission rate is decremented and the CS_{th} is assigned the value that was associated with that rate as shown through the dashed arrow in the figure.

Related work was performed in [42]. The authors suggest using the CS_{th} along with a fixed predefined interference range in order to select the optimal transmission rate based on the distance that separates the sending and receiving nodes.

2.6 Conclusion

In summary, many researchers have performed evaluation studies, analytically and through simulations, for WLANs in the presence of transmission errors in order to analyze their influence on the performance of mobile nodes. Accordingly, various enhancements were proposed to the the operation of DCF using the correlation between the binary exponential backoff (BEB) algorithm and the loss due to high PER of a transmitted packet. Specifically, there is a general belief that one way to overcome the loss caused by channel noise is to not apply the same methodology used for losses due to collision (i.e., increase the contention window), but rather reset or maintain the same value of CW. Yet, in addition to that, the related studies did not take channel capacity into consideration when drawing such conclusions, methods used for diagnosing transmission failures whether caused by noise in the medium or instantaneous transmissions (i.e. collisions) are inaccurate and add additional overhead to the current standard operation. In the next chapter, we will present a comprehensive analytical and simulation study that analyzes the correlation between the BEB of DCF, the error rate and load on the channel.

Chapter 3

Performance Analysis of the Binary Exponential Backoff in Noisy Channels

3.1 Introduction

In this chapter, we investigate the performance of the IEEE 802.11 access method in the presence of channel transmission impairments. We consider two binary exponential backoff (BEB) algorithms in our study: a standard BEB where a host increases its contention window (CW) upon every packet loss (collision or transmission error) and another access method with a capability to differentiate between the type of losses; here, a host experiencing a loss will increase its CW only after a collision and remain in the same backoff stage otherwise. Therefore, this method allows for quick recovery from channel impairments. Since it requires a host to differentiate between the causes of unsuccessful packet transmissions, we present and implement an accurate and robust online estimation method for that purpose. We show, through analysis and simulation, that the second access procedure outperforms the standard BEB when the network is lightly loaded. However, as the number of nodes increases, the quick recovery property results in intensifying the collisions among contending nodes and hence yields a poor system performance. We also compare both access methods in a heterogenous environment and study the per-host throughput. Our results indicate that the second access method always yields a better throughput for nodes experiencing noise as well as better system fairness. Finally, we propose a hybrid method that takes advantage of both access methods to achieve better throughput under various network loads.

3.2 Problem Statement

Data transmission over Wireless LANs usually encounters erroneous failures due to multiple factors, such as multi-path fading, mobility and thermal noise. In ideal channel conditions (i.e., noise-free medium), the standard operation of BEB implemented at the DCF will double the contention window upon a failed transmission due to two or more instantaneous transmissions (collision). This is to decrease the probability of collisions during the next round of the channel access competition. However, as the DCF lacks the capability of differentiating between packet losses, whether due to collisions or due to erroneous in the channel, the CW is doubled upon every failed transmission regardless of the cause of the failure. In the following sections, we prove that a simple access method (i.e., standard BEB) without any capability for differentiating between packet losses yields poor performance (overall system throughput) when the load on the network is light, as opposed to a

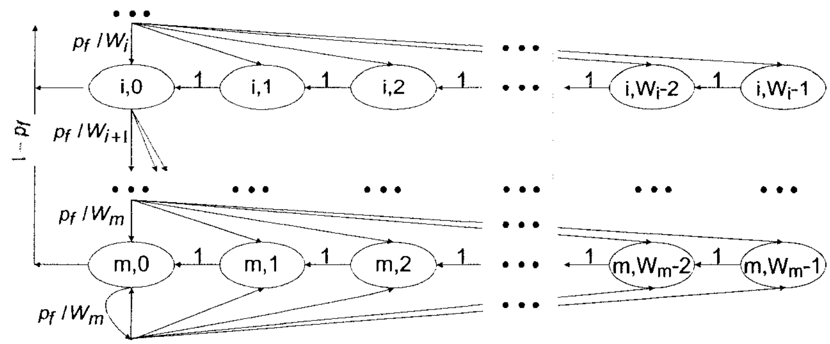

Figure 3.1: BEB_1 Markov Chain Model.

second access method where a node performs backoff only if the packet is lost due to collision and otherwise maintain the same backoff stage. That is, unnecessary backing off when there are link errors will leave much of the air time unused. However, when the load on the channel is high, the first access method achieves higher overall utilization; that is, trying to quickly recover from transmission errors by not backing off intensifies the contention on the channel and yields less air time to be efficiently used. Then we combine the advantages of both methods into a hybrid access scheme that probes the network load and applies the corresponding optimal backing-off technique.

3.3 Analysis of 802.11 in noisy environment

3.3.1 Transmission Probability and Throughput

The model we present for BEB is based on Bianchi's model [31]. Let $b(x)$ be the stochastic process representing the backoff time counter for a given station (N is the total number stations; a station is assumed to always have packets to send). A discrete and integer time scale is adopted, e.g., x and $x+1$ represent the beginning of two consecutive time slots. W is the contention window size, which is defined as $W_i = 2^i W_0$ where i is the backoff stage.

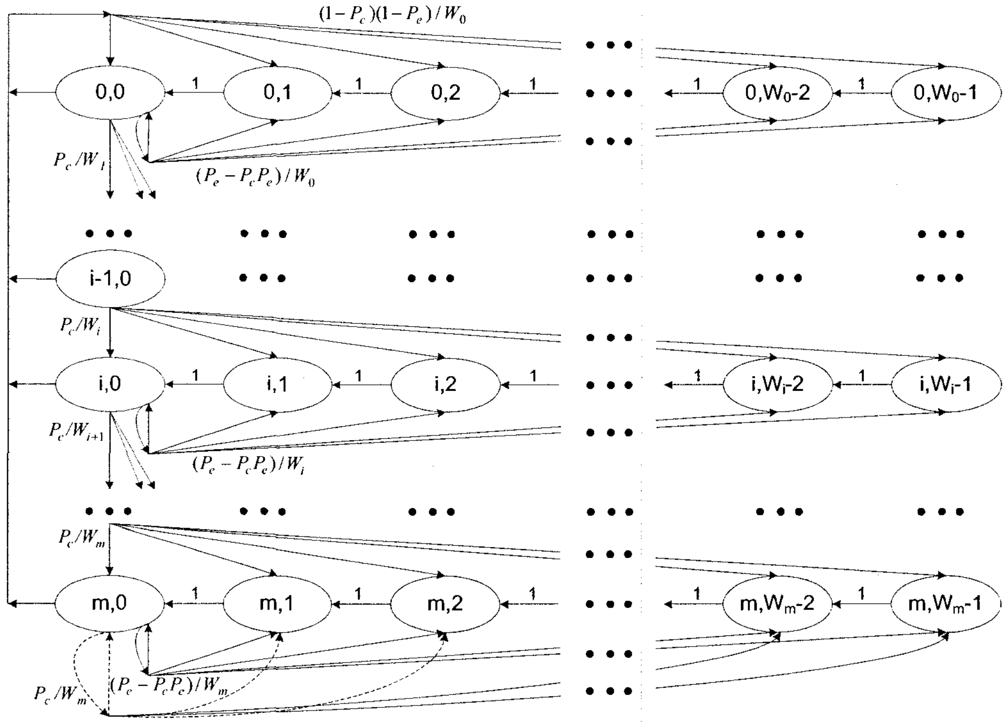

Figure 3.2: BEB_2 Markov Chain Model.

Let the initial contention window be the value of $W_0 = CW_{min}$ and maximum contention window be the value of $W_m = CW_{max}$. Let $s(x)$ be the stochastic process representing the backoff stage $(0, ..., m)$ of the station at time x. Denote p_c as the conditional collision probability which is constant and independent of the number of retransmissions incurred and let p_e represents the packet error rate (PER).

Upon detecting a packet loss, a station will decide whether the packet was lost due to collision or due to transmission error (the loss differentiation method is presented at a later stage of this chapter). If the station infers collision, then the standard BEB mechanism is followed and the contention window is doubled. If the packet is lost due to channel noise, then the station could either choose to double its backoff and schedule a retransmission or remain in the same backoff stage or treat the transmission as a successful one and accordingly resets the contention window and retries the transmission [15]. In the case where the station always doubles its contention window upon any kind of loss, the system can be modeled as a two dimensional stochastic process $\{s(x), b(x)\}$ ([15]), with the discrete time Markov chain depicted in Figure 3.1, where p_f is the transition probability and is approximated as:

$$p_f = 1 - (1 - p_c)(1 - p_e) \tag{3.1}$$

where,

$$p_c = 1 - (1 - \tau)^{N-1} \tag{3.2}$$

Denote by τ, the transmission probability of a station in a randomly chosen time slot:

$$\tau = \sum_{i=0}^{m} b_{i,0} = \frac{2(1 - 2p_f)}{(1 - 2p_f)(W_0 + 1) + p_f W_0 (1 - (2p_f)^m)} \tag{3.3}$$

Alternatively, if a station chooses to remain in the same backoff stage upon a packet loss due to noisy channel, the system can be modeled again as a two dimensional stochastic process with the discrete time Markov chain depicted in Figure 3.2 where p_s ($p_s = 1 - p_f$) is the success probability.

Our only non-null one-step transition probabilities are:

$$\begin{cases} P\{i,k|i,k+1\} = 1 & k \in (0, W_i - 2), i \in (0,m) \\ P\{0,k|i,0\} = \frac{(1-p_c)(1-p_e)}{W_0} & k \in (0, W_0 - 1), i \in (0,m) \\ P\{i,k|i-1,0\} = \frac{p_c}{W_i} & k \in (0, W_i - 1), i \in (1,m) \\ P\{m,k|m,0\} = \frac{p_c + p_e - p_c p_e}{W_m} & k \in (0, W_m - 1) \end{cases} \tag{3.4}$$

Denote by p_s the probability of a successful transmission then $p_s = (1-p_c)(1-p_e)$. From the transition states we have:

$$b_{i,0} = \alpha b_{i-1,0} \tag{3.5}$$

where,

$$\alpha = \frac{p_c}{p_s + p_c} \tag{3.6}$$

Then,

$$b_{i,0} = \alpha^i b_{0,0} \tag{3.7}$$

And for stage m we have,

$$b_{m,0} = b_{m-1,0}\frac{p_c}{p_s} = \alpha^{m-1}\frac{p_c}{p_s}b_{0,0} \tag{3.8}$$

The probability of being at stage *zero* is given by:

$$b_{0,k} = \frac{W_0 - k}{W_0}\left[p_s \sum_{j=0}^{m} b_{j,0} + p_e b_{0,0}\right] \tag{3.9}$$

The probability of being at stage i is given by:

$$b_{i,k} = \frac{W_i - k}{W_i}\left(p_c b_{i-1,0} + p_e b_{i,0}\right) \tag{3.10}$$

The probability of being at stage m is given by:

$$b_{m,k} = \frac{W_m - k}{W_m}\left(p_c b_{m-1,0} + p_c b_{m,0} + p_e b_{m,0}\right) \tag{3.11}$$

Moreover,

$$\sum_{j=0}^{m} b_{j,0} = \sum_{j=0}^{m-1} b_{j,0} + b_{m,0} = \frac{p_s + p_c}{p_s} b_{0,0} \tag{3.12}$$

Thus,

$$\begin{cases} b_{0,k} = \frac{W_0 - k}{W_0} \cdot b_{0,0} & i = 0 \\ b_{i,k} = \frac{W_i - k}{W_i} \cdot b_{i,0} & 0 < i < m \\ b_{m,k} = \frac{W_m - k}{W_m} \cdot b_{m,0} & i = m \end{cases} \tag{3.13}$$

Therefore,

$$b_{i,k} = \frac{W_i - k}{W_i} b_{i,0} \quad 0 \leq i \leq m, \quad 0 \leq k \leq W_i - 1 \tag{3.14}$$

Using normalization and the fact that: $\frac{p_c}{p_s} = \frac{\alpha}{1-\alpha}$, we proceed:

$$
\begin{aligned}
1 = \sum_{i=0}^{m}\sum_{k=0}^{W_i-1} b_{i,k} = \sum_{i=0}^{m} b_{i,0} \sum_{k=0}^{W_i-1} \frac{W_i-k}{W_i} = \sum_{k=0}^{m} b_{i,0}\frac{W_i+1}{2} \\
= \frac{b_{0,0}}{2}\left[W\left(\sum_{i=0}^{m-1}(2\alpha)^i + \frac{(2\alpha)^m}{1-\alpha}\right) + \frac{1}{1-\alpha}\right]
\end{aligned} \tag{3.15}
$$

The transmission probability of a station becomes:

$$
\tau = \sum_{i=0}^{m} b_{i,0} = \frac{2(1-2\alpha)}{(1-2\alpha)(W_0+1)+\alpha W_0(1-(2\alpha)^m)} \tag{3.16}
$$

Equations 3.1 and 3.3 (3.1, 3.16, and 3.6 respectively) represent a nonlinear system in two unknowns τ and p_f (τ and α respectively) which can be solved numerically. Let S be the normalized system throughput, defined as the fraction of time the channel is used to successfully transmit the payload bits. To compute S, one needs to analyze what happens in a randomly chosen slot time. Let P_{tr} be the probability that there is at least one transmission in the considered slot time:

$$
P_{tr} = 1-(1-\tau)^N \tag{3.17}
$$

The successful probability P_t is the probability that exactly one station transmits on the channel and is given by:

$$
P_t = \frac{N\tau(1-\tau)^{N-1}}{P_{tr}} \tag{3.18}
$$

We define the saturation throughput of the network as:

$$
\begin{aligned}
S &= \frac{E[\text{successfully transmitted payload bytes in a slot time}]}{E[\text{length of a slot time}]} \\
&= \frac{(1-p_e)P_s s_d}{P_i\sigma + (1-p_e)P_s T_s + P_c T_c + p_e P_s T_f}
\end{aligned}
\tag{3.19}
$$

where $P_s = P_{tr}P_t$ and $(1-p_e)P_s$ is the probability of a successful transmission, $P_i = 1 - P_{tr}$ is the probability of an idle slot, $P_c = 1 - P_i - P_s$ is the collision probability, s_d is the average packet payload size, and σ is the duration of an idle slot time. T_s, T_c and T_f are the average time the channel is sensed busy because of a successful transmission, a collision, or a corrupted transmission respectively. Let H be the packet header, $H = PHY_{hdr} + MAC_{hdr}$. In the case of accessing the channel using the *basic access mode*[1], we have:

$$
\begin{aligned}
T_s &= H + s_d + SIFS + \delta + ACK + DIFS + \delta \\
T_c &= H + s_d + DIFS + \delta \\
T_f &= H + s_d + DIFS + \delta
\end{aligned}
\tag{3.20}
$$

and in the RTS/CTS access mode, similar expressions can be obtained [31].

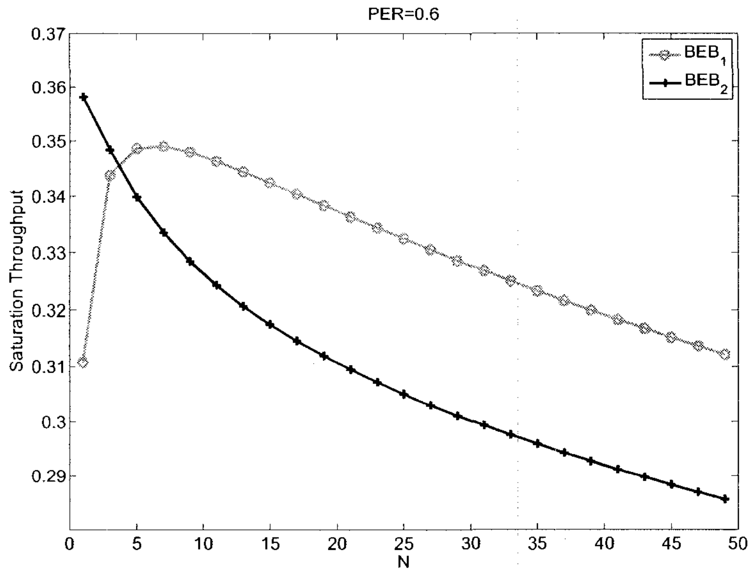

Figure 3.3: Saturation Throughput, 1Mbps channel rate

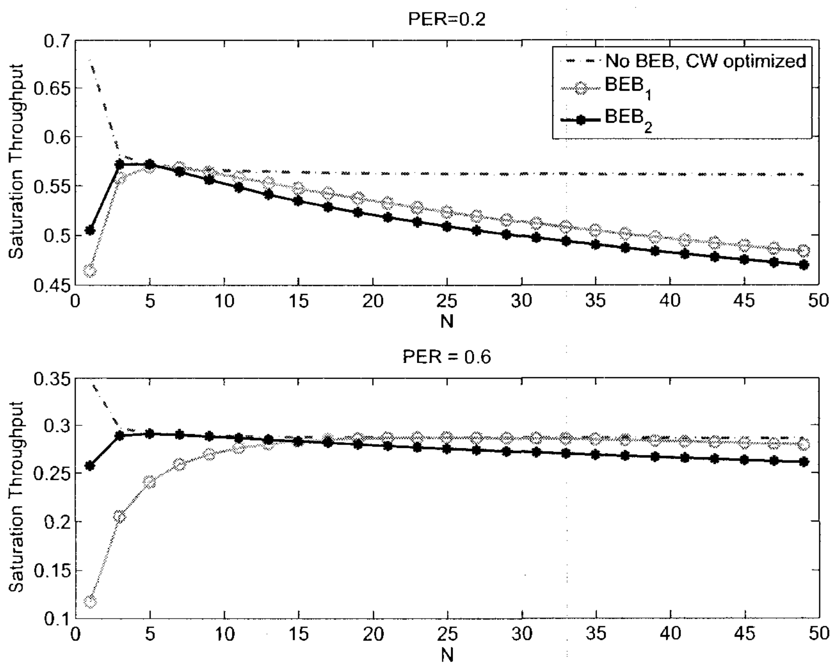

Figure 3.4: Saturation Throughput, 11Mbps channel rate

3.3.2 Analysis - Part I

In this section we present numerical results from the analytical models in order to better understand the behavior of the two access methods (BEB_1 that follows the standard operation of BEB and BEB_2 that doubles its contention window only upon collisions) presented earlier and expose the salient differences between them. We study the effect of packet error rates on the system performance; we consider the basic access mode and we assume here all the stations experience the same p_e. All the parameters used in our study follow those of DSSS [1]; we consider two channel rates, 1Mbps and 11Mbps, and data payload size of 1050 bytes. When the channel rate is low, e.g. 1Mbps, BEB_1 outperforms BEB_2 (see Figure 3.3), except when the number of stations is very low (e.g., below 3 stations when the PER is 0.6). These results suggest that it is advantageous to always double the contention window (i.e., follow the standard operation of BEB) upon a failed transmission regardless of the reason the transmission fails. This is mainly due to the fact that when the channel rate is low, a small number of stations saturates the channel; therefore, trying to quickly recover from transmission errors by not backing off (as in BEB_2) intensifies the contention on the channel and results in less air time that can be efficiently utilized. On the other hand, in the standard access method, stations double their contention window after every failed transmission and therefore stations operate at larger backoff to achieve better performance [16] (similar results are obtained for low p_e). Alternatively, when the channel rate is higher (e.g., 11Mbps), BEB_2 access method outperforms BEB_1 when the channel is lightly loaded (Figure 3.4). For example, when p_e is 0.6, it is advantageous to follow BEB_2 as long as

[1]The basic access mode is also known as the two-way handshaking mechanism (DATA-ACK). Using this access method, a node senses the channel for at least a DIFS idle period of time before initiating a transmission [1]. (For detailed explanation, you can refer to Section 2.2 of the thesis.)

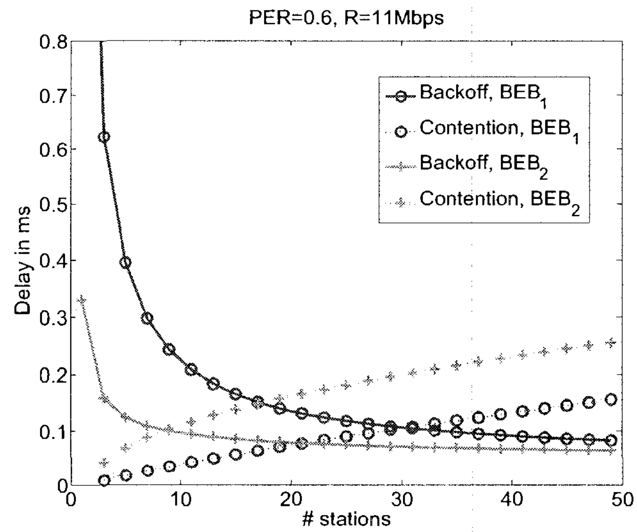

Figure 3.5: Backoff and collision time vs. number stations

the number of contending stations is less than 15. That is, when operating with unsaturated channel and in the presence of channel noise, it is advantageous for stations to quickly recover from packet losses, by retransmitting from the same backoff stage (maintain same CW), since unnecessary backing off when resources are available will only leave much of the air time unused and result in poor overall performance. However, as the number of stations increases, maintaining the same backoff stage upon a failed transmission (not collision) would exacerbate the contention and hence intensify collisions among contending stations. Therefore, at higher loads, the overall system performance would be improved if upon any failed transmission, a station schedules its retransmission after doubling the contention window.

In order to further understand these findings, we consider the average time needed to successfully transmit a packet; the authors of [32] have shown that the average time $E[T]$ for servicing a packet in an N user WLAN can be written as:

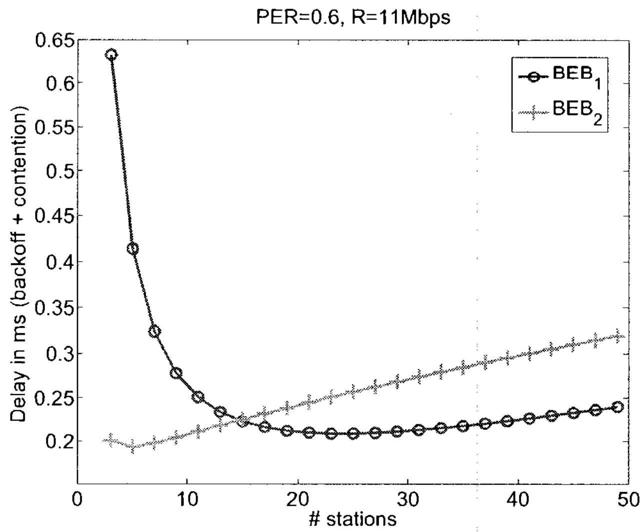

Figure 3.6: Sum of backoff and collision times vs. number stations

$$E[T] = K + \frac{E[B]}{N} + \frac{E[C]}{N_c} \tag{3.21}$$

where K represents the sum of the time incurred in sending physical layer headers, the payload and the MAC layer headers, and the time spent when the system incurs a failed transmission due to channel noise. $E[B]$ is the average time a station spends in backoff, $E[C]$ is the average time a station spends in collision and N_c is the average number of stations involved in a collision and are determined as follows:

$$E[B] = \frac{\sigma}{\tau(1-p_c)} \tag{3.22}$$

$$E[C] = \frac{p_c}{1-p_c} T_c \tag{3.23}$$

$$N_c = \frac{N\tau - N\tau(1-\tau)^{N-1}}{1-(1-\tau)^N - N\tau(1-\tau)^{N-1}} \tag{3.24}$$

Clearly, the term $\frac{E[B]}{N} + \frac{E[C]}{N_c}$ in the expression of $E[T]$ is the only term affected by the choice of the access method. Therefore, the lower is this term, the lower the time it takes to service successfully a packet and hence the higher is the overall achieved throughput. In Figure 3.5, we show the backoff time normalized by N and the collision time normalized by N_c for both access methods in the case where the channel error rate is 0.6. We see that stations under BEB_2 constantly have lower backoff delays; this is because stations suffering packet losses always attempt to recover from these losses by retransmitting from their current backoff stage. Alternatively, stations under BEB_1 experience much larger backoff delays, especially when the number of stations is small. This is mainly attributed to the fact that packets that are lost due to channel error will force stations to continuously backoff, although contention on the channel is not the problem. On the other hand, the average time spent in collisions under BEB_2 is larger; this is due to the higher frequency of access on the channel by stations following BEB_2, as opposed to BEB_1, which results in intensifying the collision. Indeed, it is the reduction in the time spent in backoff (when the number of stations is small) that is allowing BEB_2 to achieve higher throughput. Here, although the time spent in collision is higher, the quick retransmission attempt by stations to recover from packet loss will yield an improvement in the system performance, especially when the channel is lightly loaded. This is shown further in Figure 3.6 where we look at the sum of both times; when the load is light (e.g., $N \leq 15$), the total time is smaller than that of BEB_1 which explains the higher throughput achieved in Figure 3.4 (note that the point at which the throughput of BEB_1

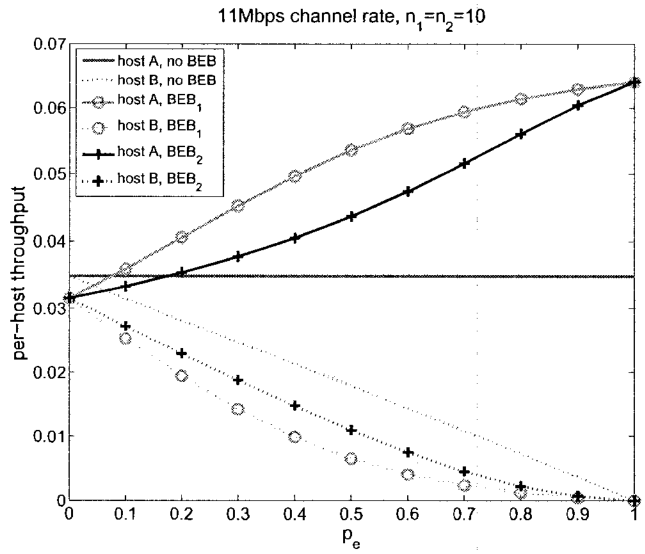

Figure 3.7: Per-Host Normalized Throughput

starts outperforming BEB_2 corresponds to $N = 15$). Alternatively, when the load is higher, the figure shows it is not advantageous to resort to quick recovery from packet loss (i.e., using BEB_2 access method) as this will only exacerbate collisions on the channel and result in lower overall system performance.

Now, in order to maximize the system throughput, the authors of [33] minimized the term $\frac{E[B]}{N} + \frac{E[C]}{N_c}$ with respect to τ; the optimal transmission probability is obtained by solving:

$$(1-\tau^*)^N = \frac{T_c^*+1}{T_c^*}(1-N\tau^*) \tag{3.25}$$

where $T_c^* = \frac{T_c}{\sigma}$. The authors accordingly suggested a method to optimize the performance of 802.11 by turning off the BEB access method and derived the optimal initial contention window CW_{min}^* (from τ^*) that all stations should utilize. It is clear however that τ^* is strongly dependent on the number of stations, N, which is not trivial to estimate. We

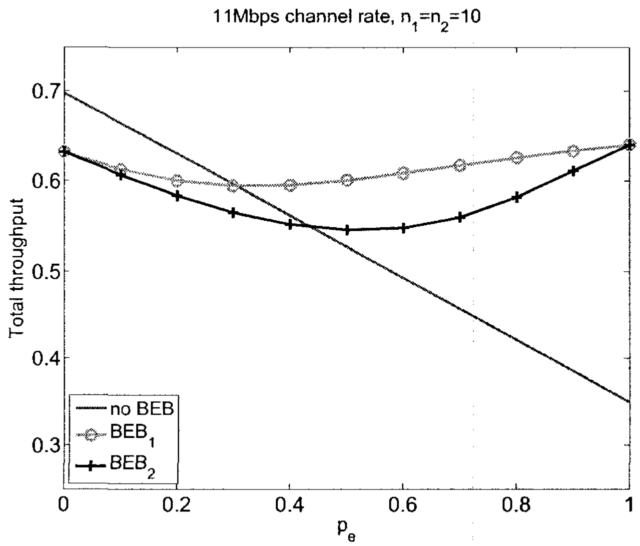

Figure 3.8: Total Normalized Throughput

show in Figure 3.4 the throughput obtained by this access method; clearly, when the error rate is small, the performance is significantly improved in comparison with the other two methods. Note that in this method, all stations equally attempt to access the channel so that they achieve the maximum possible throughput; therefore, a station does not need to differentiate between the causes of packet losses. As the error rate increases, we see that the optimized access method performs similar to BEB_2 when the load is light and similar to BEB_1 when the number of stations is higher. Similar to [33], the authors of [16] suggested to turn off BEB and derived the optimal number of idle slots that yields optimal performance. In their new access method, termed *Idle Sense*, each host measures the average number of consecutive idle slots between transmission attempts and make sure that this number is close to an optimal number (the optimal number is derived from the analysis that maximizes the throughput) by either increasing or reducing the contention window in an additive increase, multiplicative decrease (AIMD) manner. This method is more practical than that of [33] as it does not require any knowledge about the number of stations in the network.

3.3.3 Analysis - Part II

We extend our analysis to consider a network with heterogenous conditions wherein hosts (labeled $l = 1, ..., N$) experience different transmission errors. For simplicity, we consider two classes of nodes, where all stations (n_1) within one class (C_1) do not suffer any channel impairments and the remaining stations (C_2, $n_2 = N - n_1$) experience all the same transmission error, p_e. Here, under BEB_1, the transmission probability ($\tau_{l,i}^{(1)}$, $i = 1, 2$) for a station l in class i is given by (3.3) and its transition probability ($p_{f,l}^{(1)}$) is:

$$p_{f,l}^{(1)} = 1 - (1 - \tau_1^{(1)})^{n_1 - 1}(1 - \tau_2^{(1)})^{n_2} \tag{3.26}$$

if node $l \in C_1$; otherwise:

$$p_{f,l}^{(1)} = 1 - (1 - p_e)(1 - \tau_1^{(1)})^{n_1}(1 - \tau_2^{(1)})^{n_2 - 1} \tag{3.27}$$

With N stations, equations (3.3), (3.26), and (3.27) provide a set of non linear equations that can be solved numerically for $\tau_1^{(1)}$, $\tau_1^{(2)}$, and $p_{f,l}^{(1)}$ ($l \in C_1$ or C_2). Clearly, hosts belonging to different classes will access the channel with different transmission probability, as determined by the access method. The probability that a host l successfully transmits is:

$$p_s^l = \begin{cases} \tau_1^{(1)}(1-\tau_1^{(1)})^{n_1-1}(1-\tau_2^{(1)})^{n_2} & \text{if } l \in C_1 \\ \tau_2^{(1)}(1-\tau_1^{(1)})^{n_1}(1-\tau_2^{(1)})^{n_2-1} & \text{if } l \in C_2 \end{cases} \tag{3.28}$$

The sum of the probabilities of successful transmissions for all stations is $P_s = \sum_{l=1}^{N} p_s^l$ and the probability of at least one station attempting transmission is:

$$P_{tr} = 1 - P_i = 1 - (1-\tau_1^{(1)})^{n_1}(1-\tau_2^{(1)})^{n_2} \tag{3.29}$$

where P_i is the probability that the medium is idle. Hence, the collision probability is $P_c = 1 - P_i - P_s$. Finally, one can determine the per-host (l) throughput:

$$S_l = \begin{cases} \dfrac{p_s^l s_d}{P_i\sigma + P_s T_s + P_c T_c} & \text{if } l \in C_1 \\ \\ \dfrac{(1-p_e)p_s^l s_d}{P_i\sigma+(1-p_e)P_s T_s+P_c T_c+p_e P_s T_f} & \text{if } l \in C_2 \end{cases} \tag{3.30}$$

and the normalized throughput for the system is $S = \sum_{l=1}^{N} S_l$. In a similar way, one can determine the individual throughput for a host using the second access method. We will compare the per-host throughput under these two access methods to that achieved in a network where BEB is absent [33]. For that last method, all nodes are forced to use the same access probability $\tau_l = \tau^*, l = 1...N$ derived using (3.25) and the per-host throughput is computed using (3.30).

Figure 3.7 shows the per-station throughput for two nodes (A, a station with ideal channel conditions, and B, a station with bad channel quality) when varying the error transmission rate in a network with 20 nodes, 10 of which experience transmission errors (used parameters are shown in Table 3.1) . Under BEB_1 , the throughput for host B decreases quickly as p_e increases while the throughput for host A increases; this is because in this scenario, 50% of the hosts are experiencing transmission errors and hence are forced to continuously backoff upon every packet loss thereby leaving the channel idle for other hosts (50%) to collectively access with less contentions. On the other hand, under BEB_2, host B (A) obtains a higher (lower) throughput than it would under BEB_1. Here, a host experiencing transmission impairments enjoys higher access to the channel in order to overcome the channel error and therefore achieve higher throughput as opposed to a host experiencing the same error under BEB_1. These results demonstrate that although BEB_2 may not achieve higher overall system throughput than BEB_1, especially when the number of nodes is high (as shown in Figure 3.8), BEB_2 enables stations experiencing transmission errors to obtain a better access to the channel and hence improve their throughput. Finally, it is to be noted that BEB_2 improves slightly the fairness among stations as opposed to BEB_1 where some hosts obtain large throughput at the expense of starving other hosts. Figure 3.7 shows also the per-host throughput achieved by an optimal access method where BEB is absent [33]; the figure shows that when there is no error (or small error), this system achieves better individual and overall system throughput. However, as the channel condition deteriorates, host A consistently receives the same access to the channel and hence attains the same throughput while host B, who is equally accessing the channel as host A, receives a decreasing throughput as p_e increases. Note, however, that the throughput achieved by host B under this access method is always higher than that obtained under the other access methods due to the fact that the access probability is derived to achieve maximum throughput. This result also shows that this access method yields the best fairness among different hosts.

With respect to the overall system throughput, Figure 3.8 shows that the optimized access method achieves the highest throughput when the error rate for stations in C_2 is small. However, when the channel conditions deteriorate, the optimal access method severely degrades the network performance since hosts are not able to *dynamically* adjust their access to efficiently utilize the channel. However, the other two access methods allow stations not experiencing any channel error to increase their access and hence achieve higher overall throughput. Note that BEB_1 is a more greedy access method than BEB_2, since stations experiencing transmission impairments are forced to unnecessarily backoff, thereby allowing other stations better access the channel, and accordingly yielding a higher overall throughput. Similar observations are obtained when we vary the percentage of nodes experiencing errors and for different number of nodes in the system.

3.4 Channel Quality Measurement at the MAC Layer

Many studies have shown that the wireless channel exhibits time varying characteristics; namely, the quality of received signals changes dramatically even over short time intervals due to multiple causes (noise, attenuation, interference, multipath propagation, and host mobility) [14], [23]. Poor transmission conditions may result in incorrectly received frames which in turn may force some hosts to lower their transmission bit rate (e.g., [24]) in order to obtain better throughput [16]. It is therefore of great interest to develop techniques for direct measurement of channel conditions.

Previous work have focused on the physical (PHY) layer approach based on signal to noise ratio (SNR) and RSSI measurements [28]; however, real life measurements have shown that the correlation between SNR and the PER perceived at the MAC layer is rather low [14]. The authors of [29] have proposed a feedback message (NAK frame) that enables the sender to distinguish between losses due to collisions and losses due to channel noise. This, however, requires a reverse signaling channel between the receiver and the sender in addition to modifications to the existing standard which is not immediate in the current standard. Distinguishing between packet losses can easily be achieved when using RTS/CTS (Request To Send/Clear To Send) handshake mechanism for each transmitted packet; here a failed RTS/CTS exchange may be considered as a collision and no ACK for a data frame is considered as a loss due to bad channel quality [16], [30]. We note that RTS/CTS reservation mechanism is usually turned off in wireless cards because of the overhead. For the basic access method, the authors of [15] presented a method for estimating the PER that requires the knowledge of the number of active stations and is based on some observations from a derived Markov model in noisy environment. The number of active stations (N) is estimated by either probing the channel continuously to monitor the ongoing transmissions or is provided by the Access Point (AP) through the management frames. Using N, the transmission probability (τ) is computed (similar to Equation 3.16). Afterwards, the error probability can be easily calculated. This method, however, does not give accurate estimates of the PER (10%-20% difference with the actual data). A novel loss differentiation method for dense networks was proposed in [13]. The authors defined three packet loss types; 1) Collision: here, both, the interference signal and the desired signal start at the same time slot. 2) Type-1 Interference: the interference signal arrives prior to the desired signal. 3) Type-2 Interference: the interference signal arrives after the arrival of the desired signal. Afterwards, the PER was estimated by the differentiating the loss types through exploiting the timings of arrival of the interference signal relative to desired signal. Here, it is assumed that energy detection based carrier sensing is implemented and observed at each node pair prior to a transmission. Recently, the authors of [12] presented a method for estimating the channel quality at the MAC layer. The method is based on few observations about the operation of CSMA/CA method. The CSMA/CA creates well-defined times at which packet transmissions by a station are admissible and at all other times transmissions are disallowed. To specify the admissible transmission times, the following four events were defined; 1) a station has seen the medium as idle and, if its backoff is in progress, has decremented its backoff counter. These are called *idle slots*. 2) A station station has detected the medium as busy due to one or more transmissions, and has suspended its backoff counter based on the network allocation vector (NAV); DIFS/EIFS indicate the backoff can resume. These slots are called *other transmissions*. 3) A station has transmitted, received an ACK and is about to resume its backoff; these slots are called *successful transmissions*. 4) Finally, a station has transmitted, timed-out while waiting for an ACK and is about to resume its backoff; these slots are called *unsuccessful transmissions*. The probability of a collision by one node is then precisely the probability that at a slot boundary the channel is busy due to a transmission by one or more other stations. The estimator works as follows; suppose over some period of time, a station transmits T times and of these A are successful because an ACK is received. Suppose there are also R slots in which the station does not transmit and that I of these are idle. Denote by p_c, p_e the collision and error probabilities respectively. Hence, if a station does not transmit, then it sees the medium busy with a probability p_c.

Accordingly, the authors of [12] obtained the following expressions for both p_c and p_e:

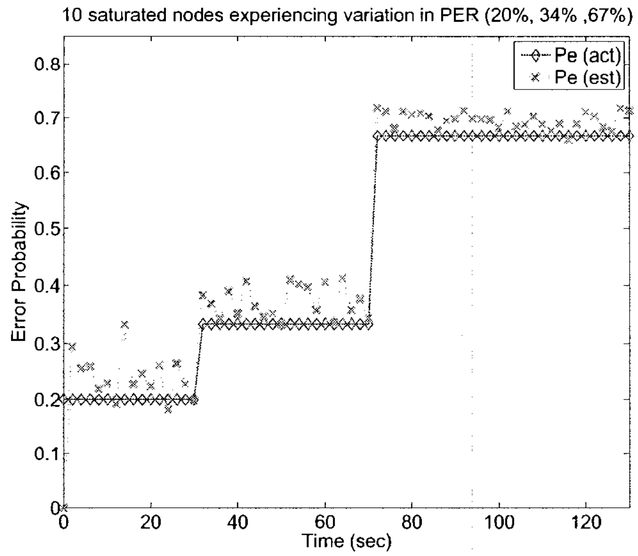

Figure 3.9: Estimating p_e while varying actual packet corruption with time.

$$p_c = \frac{R-I}{R} = \frac{\#\text{other transmits}}{\#\text{idle} + \#\text{other transmits}} \tag{3.31}$$

$$p_e = 1 - \frac{1 - \frac{T-A}{T}}{1-p_c} \tag{3.32}$$

providing that $0 \leq p_e \leq 1$.The collision probability is estimated as the proportion of busy slots due to other transmissions by other stations. To determine p_e, one needs to determine the station successful probability:

$$p_s = \frac{\#\text{successful transmits}}{\#\text{attempted transmits}} = (1-p_c)(1-p_e) \tag{3.33}$$

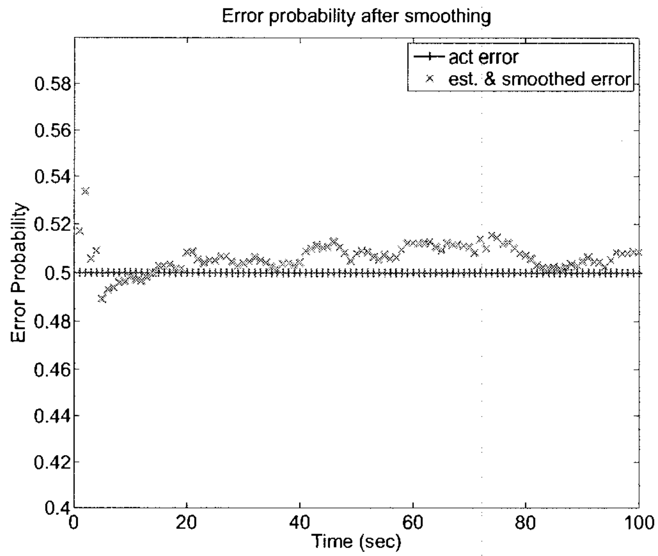

Figure 3.10: Smoothing measured error probability using moving average method.

3.5 Online Error Estimator

In this section we discuss the implementation of the online estimator used in our work and validate its accuracy. We use the online estimator presented in equations (3.31), (3.32) and (3.33); p_s and p_c can be measured and p_e is determined by solving (3.33). We note here that the authors of [12] have provided an off-line validation of the estimator. Hence, in order to retrieve the slots information and compute the corresponding noise and collision probabilities, one will have to wait for the end of the experiment to analyze the trace files. This way, the estimated probabilities can not be used for adjusting the DCF operation in the real-time. The key issue in the estimator is to keep track of both busy slots (*other transmissions*) as well as the idle slots during a predetermined time interval of our choice from which p_c and p_e are estimated. The widely used network simulator (*ns*-2) only implements virtual carrier sensing wherein a host relies on the NAV in received RTS/CTS messages to determine whether a slot is idle or not. However, in order to implement the online estimator, a station must accurately determine whether a slot is busy or not and this is readily available in *QualNet* (a commercial network simulator [17]) since it implements the physical carrier sensing (PCS) scheme of the IEEE 802.11. Here, the status of a given time slot will depend on the air-link energy level. To validate the error estimator, we tested it under a wide range of scenarios with different channel noise, number of nodes and channel physical data rates. Figure 3.9 shows the temporal behavior of the estimated error for a network of 10 saturated nodes experiencing varying channel error rates (20%, 34% and 67% respectively). Clearly, the figure shows that the estimator performs well; the estimated p_e closely follows the actual error rate as it increases over time with a maximum difference less than 10% and an average difference less than 5%. Additionally, the figure shows a very good response time of the estimator to channel variations. Now, in order to reduce the variations of the estimated error, we use the following Auto Regressive (AR) filter:

$$\hat{p}_e(n) = \delta \times \tilde{p}_e(n) + [1-\delta] \times p_e(n-1) \tag{3.34}$$

where $\hat{p}_e$ is the AR smoothing of the error probability and $\tilde{p}_e$ is the measured error probability for an individual node taken at the n^{th} time interval and δ is the correlation coefficient ($\delta = 0.125$ was used throughout the experiments). $\tilde{p}_e$ is measured according to the estimation method explained in the previous section. Figure 3.10 shows the estimated $\hat{p}_e$ versus time when the channel rate is 11Mbps and the actual packet error rate is fixed at 50%. The estimated $\hat{p}_e$ is sampled every 1-second; we can clearly see now the remarkable performance of the online estimator where a variation of about 1% between the actual and the estimated error rate is observed. Finally, Figure 3.11 plots the estimated collision and error

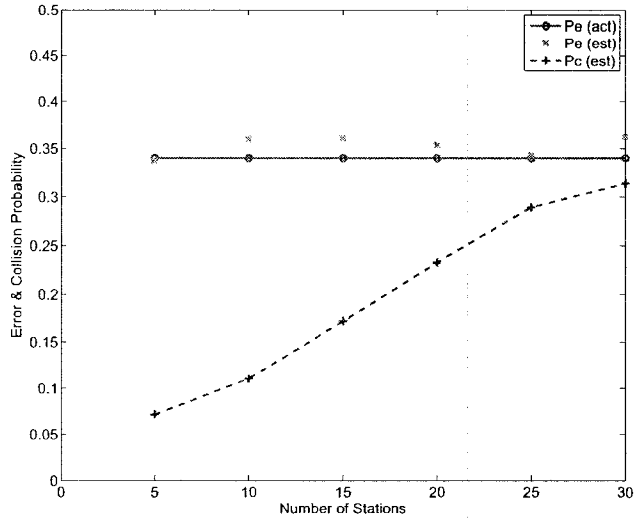

Figure 3.11: Estimating error and collision as load increases in the channel.

probabilities as the number of active stations in the WLAN is varied. The figure shows that as the number of stations increases, the collision probability increases, as expected, whereas the error probability remains unchanged.

3.5.1 Loss Differentiation

It is worthwhile to discuss how a station effectively differentiates between collisions and errors upon transmitting or retransmitting a packet. A station periodically estimates, using the online estimator, both the collision and the error probabilities (p_c and p_e) perceived by the station. These probabilities will be used by the station in order to determine whether a packet loss in the following cycle is due to collision or due to channel error. Therefore, upon an unsuccessful packet transmission, the station will double its backoff window (i.e., loss due to collision) with a probability $\varphi = p_c/(p_c + p_e)$ and remain in the same stage (i.e., loss due to error) with a probability $1 - \varphi$.

3.6 Performance Analysis

We validate the analytical model presented earlier with results obtained from simulations using *QualNet* simulator [17]. The IEEE 802.11b was used throughout the experiments. We assume that all stations experience the same packet corruption rate and that a station has always a packet to transmit (of 1050 bytes payload size) using the basic access mode. The channel rate is fixed at 1Mbps and the number of nodes is varied from 2 to 35. We obtain results (normalized system throughput) under different packet error rates and the results are presented in Figure 3.12. The figure shows that the analytical results are comparable to those of the simulation (both results are for BEB_2) as the load increases for various channel conditions (i.e., different p_e).

Parameters	**Value**
Payload	1050 Bytes
MAC Header	30 Bytes
PHY Header	28 Bytes
RTS	20 Bytes
CTS	14 Bytes
ACK	14 Bytes
Propagation Delay	1 μs
SIFS	10 μs
DIFS	50 μs
Slot Time	20 μs
CW_{min}	32
CW_{max}	1024

Table 3.1: Parameters for Analytical and Simulation Results

Next, we compare the performance of the two access methods BEB_1 and BEB_2. The MAC and PHY system parameters used throughout the experiments are shown in Table 3.1 and we consider only basic access mode. We use the throughput enhancement [15] as a metric for our comparison: $\nabla S = \frac{S'-S}{S} \times 100$ where S' and S are the system throughput

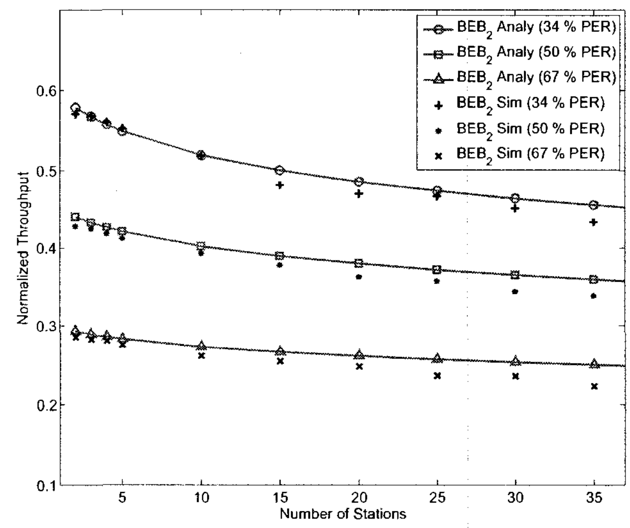

Figure 3.12: Comparing simulation and analytical throughput for various packet error rates while increasing the number of active stations. (1Mbps data rate)

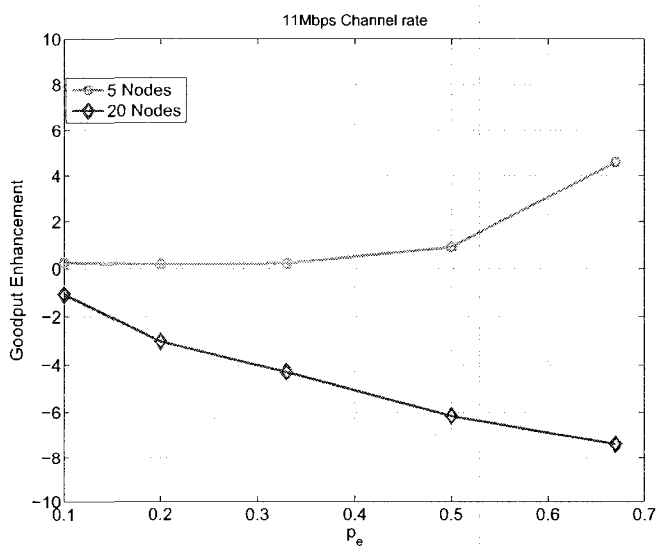

Figure 3.13: Goodput enhancement for BEB_1 and BEB_2, 11 Mbps channel rate

achieved under BEB_2 and BEB_1 respectively. Figure 3.13 shows the throughput enhancement obtained from simulation results for two networks of 5 and 20 nodes while varying channel error rates. Note that at any given time, all nodes in the network experience the same transmission impairments. Clearly, the figure shows that when the number of nodes is low ($N = 5$), BEB_2 outperforms BEB_1 under various channel conditions. However, when the number of hosts increases ($N = 20$), the throughput enhancement decreases (below zero) and BEB_1 outperforms the second access method. This has also been shown from the analytical results in Figure 3.4. The conclusion from these results is that it is not always good to recover from channel noise by performing quick retransmission because the channel may be congested and fast retransmission would only exacerbate the collision among contending stations on the channel. Our result counters the claim of [15] where the authors designed a smart access method and concluded that it is always advantageous to reset the CW upon an unsuccessful transmission due to transmission errors. The authors of [33] measured the throughput penalty obtained when default BEB (N=10, 11Mbps channel rate) is operating under the presence of transmission errors. They concluded that this throughput penalty is attained because BEB is not capable of differentiating between packet losses. Accordingly, our results showed the same (i.e., for N=10); we have also shown that even with the presence of loss differentiation, the throughput penalty will be higher when the number of nodes accessing the medium is large.

Next, we compare the performance of the two access methods in a heterogenous network. We first consider two hosts (A and B), where A experiences varying channel error rates (0% when $0 \leq t < 70s$, 67% when $70 \leq t < 140s$ and 34%, when $140 \leq t < 250s$ where t is the simulation time) and the PER for B is 0 all the time. Figure 3.14 shows

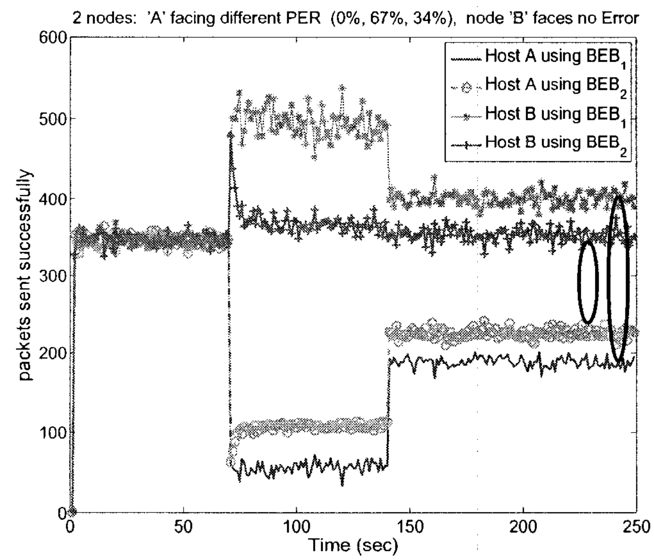

Figure 3.14: Throughput of a channel having two transmitting nodes (A & B), where only host 'A' faces (varying) PER.

the number of successfully transmitted packets per second versus time. Clearly, when the PER is 0, both stations equally share the access to the channel. Alternatively, when host A experiences transmission errors, host A is forced, under BEB_1, to double its CW upon each packet loss and hence the channel appears idle more often for host B. Host B takes therefore advantage and transmits more packets which will eventually lead to a long term unfairness. On the other hand, when BEB_2 is used as an access method, host A will have more chance to utilize the channel due to its packet loss differentiation mechanism (CW is doubled only upon collision) and accordingly, the number of successfully transmitted packets is fairly higher than that under BEB_1. Additionally, host B will receive a fair share of the channel without penalizing the access of host A.

Now, we consider a network with 10 competing stations where only one node suffers variation in transmission errors (Fig. 3.15) and another scenario of 10 competing nodes where 5 of them face transmission errors (Fig. 3.16). Figure 3.15 shows the throughput acquired by host A (experiencing errors) and host B (no error); the figure shows that host A receives a lower throughput under BEB_1 access method, especially at higher channel error rates. However, the host throughput improves under the second access method (BEB_2) due to the faster retransmissions upon packet loss due to channel error. It is interesting to see that host B will receive the same throughput under the two access methods. This is due to the fact that although BEB_1 penalizes host A, the gain in channel access is shared among all competing hosts (9 in this case) and accordingly, the gain per host is relatively minor.

Alternatively, when the number of stations experiencing transmission errors is higher (e.g., 5 in Fig. 3.16), host A (any host experiencing transmission errors) obtains much better throughput under BEB_2 than that obtained under BEB_1. When the PER increases, the throughput of host A is further impacted and host B will gain more access to the channel, since 5 nodes (out of the 10) are suffering transmission errors which leads to a significant increase in channel idle time and hence, host B (with the other 4 nodes competing under error-free channel condition) will have better chance for successful transmissions especially as the PER increases. This would actually seem like the channel is operating with less number of nodes. The gain in throughput is attributed to the reduction in access from hosts experiencing transmission errors (both under BEB_2 and even more under BEB_1). The figure shows that host B obtains slightly higher throughput under the first access method than the second one. Overall, the results reveal that BEB_2 yields better per host throughput and hence better long term fairness as opposed to BEB_1 where hosts with transmission errors will be severely penalized although the overall system performance of BEB_1 may be slightly better than that under BEB_2. These results have been corroborated from our analysis of the analytical model.

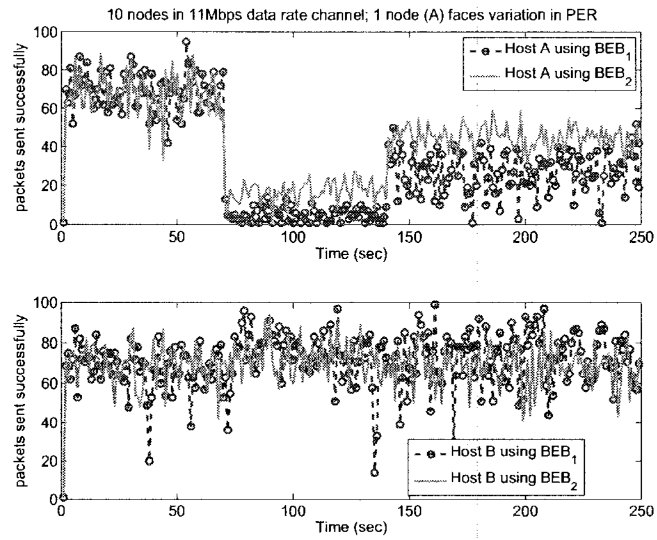

Figure 3.15: Only 1 node suffers noise (host A), the other 9 nodes have error-free channel conditions (host B one of them).

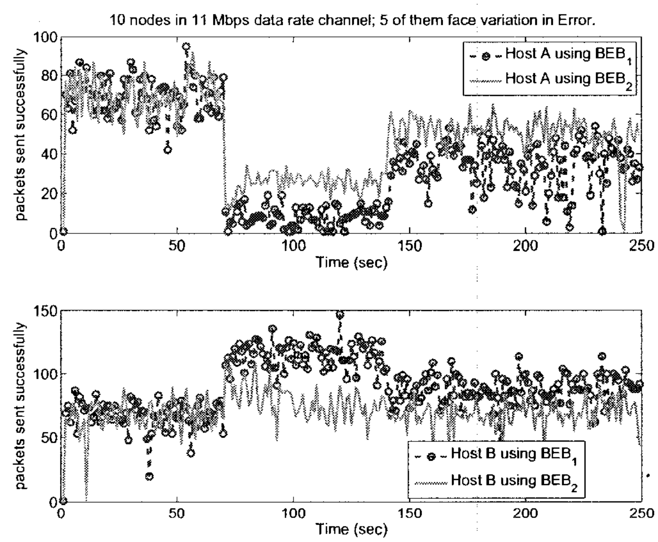

Figure 3.16: 5 nodes suffer noise (host A one of them), the other 5 compete in error-free channel (host B one of them).

3.7 An Adaptive Access Method

Clearly, it has been shown that both BEB_1 and BEB_2 offer some advantages depending on the load of the network. We will attempt to derive an enhanced access method that combines the advantages of BEB_1 and BEB_2.

3.7.1 Probing the Medium

First, we note that the authors of [16] derived an expression for the optimal number of idle slots ($\overline{n}_{i\infty}^{opt}$) between two transmission attempts after obtaining the optimal transmission probability that maximizes the throughput. The derived $\overline{n}_{i\infty}^{opt}$ enabled them to determine when the channel is used in an optimized manner based on the transmission attempt probability, denoted by τ. To obtain the optimal value for τ (τ^{opt}), the following cost function derived from the throughput function needs to be minimized:

$$Cost(\tau) = \frac{\frac{T_c}{T_{SLOT}} P_c + P_i}{P_t} \tag{3.35}$$

where $P_c = 1 - P_t - P_i$ represents the collision probability, T_c is the average collision time, T_{SLOT} is the slot time, P_t represents the successful transmission probability and $P_i = (1-\tau)^N$ is the probability of an idle slot.

When the first order derivative of the above cost function is set to zero, we get:

$$1 - N\tau^{opt} = \eta(1-\tau^{opt})^N \tag{3.36}$$

where $\eta = 1 - \frac{T_{SLOT}}{T_c}$ and can be computed based on the 802.11 parameters.

As N $\longrightarrow \infty$, the authors concluded that $\bar{n}_{i\infty}^{opt}$ attains value:

$$\bar{n}_{i\infty}^{opt} = \frac{e^{-\zeta}}{1 - e^{-\zeta}} \tag{3.37}$$

where $\zeta = N\tau^{opt}$ (the complete derivation of $\bar{n}_{i\infty}^{opt}$ can be found in [16]). Note that $\bar{n}_{i\infty}^{opt}$ is a computable constant; (e.g., it is equal to 5.68 for the 802.11b parameters).

The authors of [16] developed an access method, termed *Idle Sense*, based on the above observations. Hosts implementing the Idle Sense access method are forced to compete with the same contention window (*CW*). The value of *CW* depends only on the network load and is derived after measuring the mean number of idle slots ($\bar{n}_i$) between two transmission attempts and comparing it with the optimal one ($\bar{n}_{i\infty}^{opt}$), which is derived from the previous relations and using the fact that $\bar{n}_i = \frac{P_i}{1-P_i}$.

In our work, the hybrid access method we propose utilizes features used in probing the load on the channel similar to what we explained above. Hence, we keep track of the mean number of idle slots between two transmission attempts. When the channel is highly loaded, we turn on BEB_1 mechanism upon a packet loss since the relatively large backoff offered by BEB_1 is essential to avoid contention in this case. On the other hand, BEB_2 is used if the channel is lightly loaded. This is because when the idle slots observed in the medium are beyond the optimal derived value, it means that it is beneficial not to increase the backoff interval upon an erroneous transmission as the channel can accommodate more transmissions. The algorithm can be summarized as follows:

- If measured $\bar{n}_i < \bar{n}_{i\infty}^{opt} \longleftarrow$ *Access channel using* BEB_1.
- If measured $\bar{n}_i > \bar{n}_{i\infty}^{opt} \longleftarrow$ *Access channel using* BEB_2.

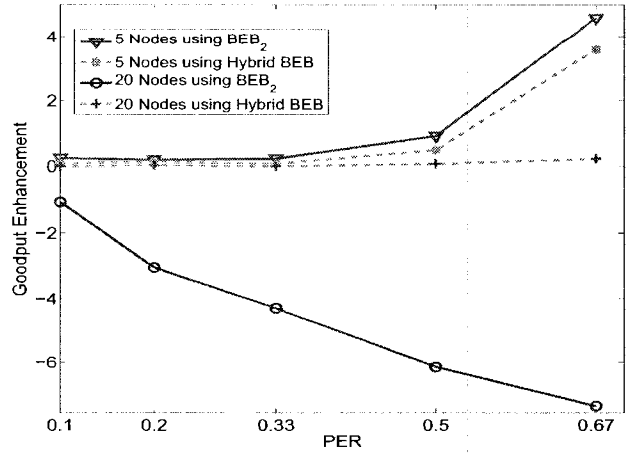

Figure 3.17: Throughput enhancement attained by 5-nodes and 20-nodes while varying PER (0%, 67%, & 34% respectively) under both BEB_2 and Hybrid BEB (11Mbps data rate).

To validate the proposed hybrid access method, we run simulation scenarios in which nodes keep track of the mean number of ideal slots along with the optimal one. This is to ensure nodes will select the relevant access method according to the channel load using the proposed approach. Figure 3.17 plots the throughput enhancement presented in the previous section for 5-nodes and 20-nodes network when implementing BEB_2 and the hybrid BEB while varying the transmission corruption rate. Both methods achieve almost the same enhancement when the number of nodes is low. However, it is clear that the performance of BEB_2 deteriorates when the number of nodes is 20 (negative enhancement), while the hybrid BEB performs the same as BEB_1 (zero enhancement).

3.7.2 Adjusting the Contention Window (CW)

Now, we consider extending our proposed hybrid access method to modify the way the contention window (CW) is reset when operating in a noisy congested channel. First, we recall that using the standard BEB, nodes will attempt to increment the CW upon each transmission failure; the CW is instantly reset after a successful transmission and CW_{min} will be utilized for the first transmission attempt for a new packet. However, when operating in a congested environment (e.g., large number of nodes), backing off using the CW_{min} will lead to excessive transmission failures.

The authors of [16] suggest to turn off BEB and force all stations to use the same CW, whose value is adapted in an additive increase multiplicative decrease (AIMD) manner according to the network state. On the other hand, *Asymptotically Optimal Backoff*[2] (AOB) defined in [11] is not a practical access method, since it requires the estimation of slot utilization probabilities (e.g., the transmission probability (τ)). Related work was presented in [21] and [22], where the CW incrementing mechanism is kept intact upon a transmission failure. However, a multiplicative decrease factor was used to reset the CW slowly in the case of [21]; this will tamper the CW known stages for the BEB. Furthermore; the CW is decremented instantly to a prior stage in [22], without a direct correlation with the usefulness of the new stage.

Now, we extend our proposed hybrid access method to intelligently reset the CW upon a successful transmission while operating in a noisy congested channel. The proposed scheme is fairly simple. Each node keeps track of the transmission successes ($Success_{cw[i]}$) and failures ($Failure_{cw[i]}$) along with the CW stage i used for each transmission. Accordingly, a reputation factor $Rep_{cw[i]}$ is calculated for each CW stage i as follows:

[2]The AOB is an access method introduced by [11] in which the CW is adapted according to the network contention measured using the slot utilization and the average size of transmitted frames.

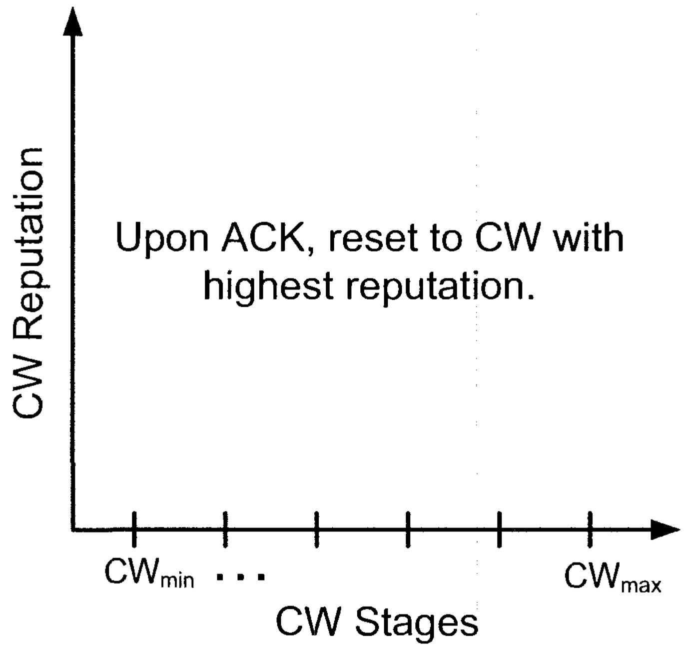

Figure 3.18: Reputation-based CW resetting.

$$Rep_{cw[i]} = \frac{Success_{cw[i]}}{Success_{cw[i]} + Failure_{cw[i]}} \tag{3.38}$$

Now, when a successful transmission occurs, rather than directly resetting the CW value to CW_{min}, the contention window will be reset to a lower stage with the highest reputation factor (Figure 3.18). Indeed, the term $Rep_{cw[i]}$ computed in Equation 3.38, will help the node elect the most suitable CW at a given network state. We note that the reputation factor $Rep_{cw[i]}$ is reset after a selected duration along with the success and failure counters. Here, the node ensures that the next transmission will utilize the optimal CW stage, which is contrary to the previous mentioned studies that require random and undefined CW stages for the backing off mechanism.

Now, we validate the hybrid access method in a congested environment while enabling the reputation-based CW resetting scheme. Figure 3.19 plots the throughput enhancement

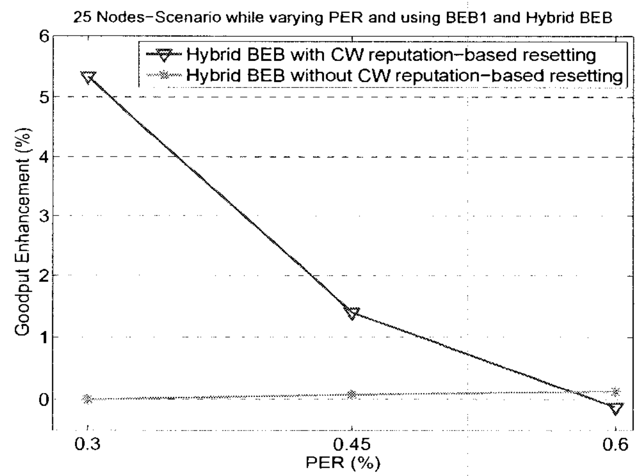

Figure 3.19: Throughput enhancement attained when operating 25 nodes using Hybrid BEB with and without reputation-based CW resetting.

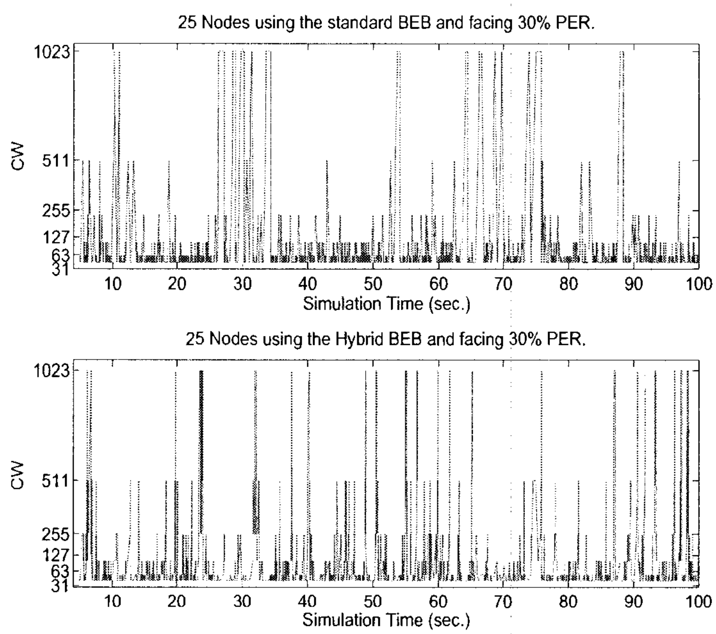

Figure 3.20: Observing CW variation for a node using both the standard BEB and the hybrid BEB with reputation-based CW resetting.

of the proposed method over the standard BEB while varying the PER. First, it is interesting to see the performance of the hybrid BEB for the scenario while not using the proposed CW resetting mechanism. There, the hybrid BEB performance is almost the same as BEB_1 and zero enhancement is observed. On the other hand, when integrating the reputation CW resetting in the hybrid BEB, a significant enhancement is noticed as shown in Figure 3.19, where the goodput enhancement reaches about 5.4 % at PER of 30 %. This due to the fact that contention is enormous in a 25-node scenario. Hence, when the CW_{min} is used for backing off after a successful transmission, high collision rates will occur. However, the reputation-based CW resetting mechanism will attempt to reset the CW to the value with highest known success rate and hence less collision will occur. This can be clearly seen in Figure 3.20 where CW values are plotted for a node while using the standard BEB and then when using the hybrid BEB. We see that the CW values are concentrated around the lower stages in the case of the default BEB. In contrary, when the hybrid method is used, we observe that CW values attain higher values due to the reputation-based CW resetting.

CW Stage	**CW value**
0	31 (CW_{min})
1	63
2	127
3	255
4	511
5	1023 (CW_{max})

Table 3.2: Contention Window default values for IEEE 802.11

3.8 Conclusion

We investigated the performance of two Wireless LAN access methods in the presence of channel transmission impairments; a standard binary exponential backoff (BEB) that increases the contention window (CW) after every packet loss and another access method with a capability of differentiating between transmission errors and collision. Hosts implementing the second access method will increase their CW only after collision, reset if the transmission is successful and maintain the same backoff stage otherwise. This enables hosts experiencing channel impairments to quickly recover from transmission errors. We have also presented an accurate online estimation method for loss differentiation to enable the operation of the second access method. Our results revealed that the second access method outperforms the standard BEB when the network load is light; however, as the load increases, the quick recovery property of the second access method intensifies collisions among contending hosts especially at higher packet error rates. We validated these results using both analytical and simulation methods. We also analyzed the per-host achieved throughput in a heterogeneous network; the second access method always achieved better individual performance for hosts experiencing transmission error and better overall system fairness. Finally, we presented an adaptive approach that integrates the operations of both access methods by deploying some network probing scheme to measure the load on the network and accordingly adapt the access of the nodes. Moreover, we integrated a novel CW resetting scheme to the hybrid method, where the CW stage with the highest success reputation will be elected after a successful packet transmission.

Chapter 4

Rate Adaptation for Noisy Wireless LANs

4.1 Introduction

As we mentioned in the previous chapter, Wireless LANs (WLANs) suffer degrading performance when operating within domestic areas due to multiple reasons such as: multi-path, fading, path loss and user mobility. To overcome this, transmission rate is usually adjusted to a more error-resistant rate. In this chapter, we first present a novel rate adaptation algorithm for IEEE 802.11 that can efficiently identify the threshold packet error rate (PER) at which link adjustment is required, based on a simple throughput analysis at the MAC layer. Then, we extend our rate adaptation algorithm to support IEEE 802.11e *quality of service* (QoS) requirements. When a real-time stream with QoS requirements is admitted, critical constraints such as delay bound and maximum packet drop count are integrated in the selection of the most convenient transmission rate that best respects the flow requirements. Moreover, we use dynamic bandwidth allocation rather than the default *transmission opportunities* (TXOPs) in a way that best offers a flow the required time for retransmissions due to packet failure based on the variant loss rate present in the channel. We validate our proposed rate adaptation algorithms via simulations where the efficiency and effectiveness of the algorithm are noticed for both best effort and QoS flows.

4.2 Problem Statement

The 802.11 standard provides several rates for data transmission, where the maximum is 11 Mbps for 802.11b and can reach up to 54 Mbps for 802.11a. These data rates utilize efficient but different physical modulation schemes. In ideal channel conditions where nodes are within transmission range, the higher the data rate, the better the throughput attained at the receiver. However, when operating in error-prone channels, lower data rates can overcome the channel conditions to achieve more reliable transmissions than the higher data rates; this is due to the relatively narrow channels used at lower rates, and the frequent change in transmitting frequencies which yield more robust transmission links. Recent studies have proven that link errors are very common in domestic environments [35], where multi-path and fading signals are very common reasons for the degraded wireless performance. Hence, in order to have an efficient rate adaptation algorithm, nodes need to recognize the right time at which it is beneficial to decrement (or increment) the transmission rate for attaining the optimal performance based on the channel conditions. Additionally, QoS flows requirements are treated differentially with the proposed rate adaptation technique, where critical constraints such as delay bound and maximum packet drop count are used to select the most convenient transmission rate at specific channel conditions. It is essential as well to dynamically allocate bandwidth for QoS stations (QSTAs) through the assigned transmission opportunities (TXOPs), in a way that reflects the additional time needed to retransmit packets due to ACK timeouts caused by either collision or channel noise.

Priority	AC	Traffic Type
1	0	Best Effort
2	0	Best Effort
0	0	Best Effort
3	1	Video Probe
4	2	Video
5	2	Video
6	3	Voice
7	3	Voice

Table 4.1: Traffic Types and Access Categories And Priorities for 802.11e

4.3 Proposed Rate Adaptation Algorithm

4.3.1 Preliminaries

Implementing an efficient link rate adaptation algorithm requires a methodology for wireless nodes in order to decide the most suitable transmission rate. For example, a node may decide to switch to a lower rate if the throughput (X_{low}) is at least equal or more than that (X_{high}) obtained at higher rate where X_{low} and X_{high} represent the throughput values attained at lower and higher data rates respectively. The throughput, however, depends on the duration spent by a station to transmit its packets successfully after applying the DCF mechanism. This time can be expressed as follows:

$$T_s = T_B + DIFS + (T_P + T_H) + \frac{L}{R} + T_{ACK} \tag{4.1}$$

where T_B represents the time spent in backoff, $DIFS$ is the DCF Interframe space, T_P, T_H and T_{ACK} represent the transmission time of preamble, header and ACK frame respectively, L is the payload size and R is the bit rate used. From the previous equation, we see that decreasing the data rate would result in a longer duration for transmitting a packet which in turn leads to a lower throughput and more collisions. Hence, stations normally transmit at the maximum possible transmission rate; 11 Mbps in case of PHY 802.11b and 54 Mbps for PHY 802.11(a/g). However, this is true only when nodes communicate over a medium with ideal characteristics. Nevertheless, many experimental studies (e.g. [35]) have shown that the wireless channel exhibits time varying characteristics due to multiple causes (noise, attenuation, multi-path, and host mobility) that results in poor transmission conditions which leads to erroneous frames at the receiving side. Hence, in order to maintain an acceptable performance, adjusting transmission rates is crucial when nodes face highly impaired channel conditions. The packet error rate (PER) is considered a reliable measurement of the channel quality at the node level; when link rate adjustment is performed for new transmission, one expects that the PER suffered at the lower rate (e_l) to be much less than that experienced at the higher rate (e_h) due to the more robust modulations. Accordingly, a useful expression for the throughput in an error-prone medium can be used [16]:

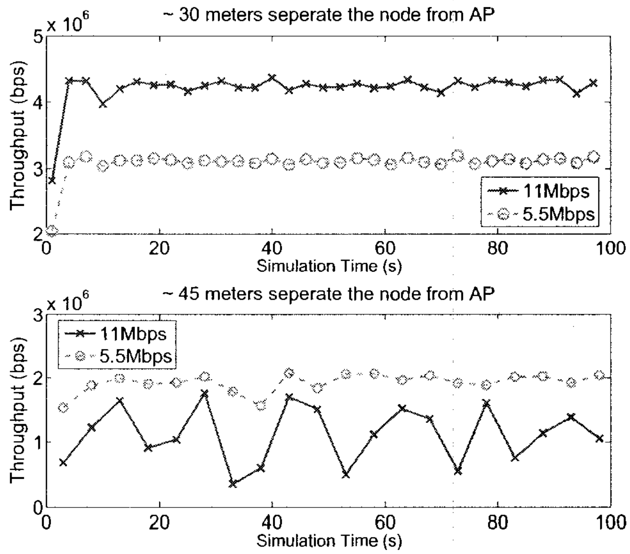

Figure 4.1: System Performance for different data rates as PER increases. Upper figure PER for 11 (5.5) Mbps 14 % (8%), downfigure 58 % (28%) .

$$\begin{cases} X_{low} = r_l \sigma_l (1 - e_l) \\ X_{high} = r_h \sigma_h (1 - e_h) \end{cases} \tag{4.2}$$

where r_l (r_h) is the low (high) bit rate, σ_l and σ_h are the portions of the useful throughput that are independent from the transmission rate used. Assuming $\sigma_l = \sigma_h$, the PER threshold value (e_{max}) at which a station should decrement its bit rate will be:

$$e_{max} = e_h = 1 - \frac{r_l}{r_h}(1 - e_l) \tag{4.3}$$

This implies that as long as the PER exhibited observed at a specific bit rate is lower than e_{max}, it is more beneficial to maintain the same transmission rate. However, if the PER reaches a value that is greater than or equal to e_{max}, then decrementing the bit rate would be essential to achieve better performance. As an example, assume that a transmission rate of 11 Mbps bit rate is being used for a certain data transmission; in order to switch to the next lower bit rate (5.5 Mbps), the node should wait until the error rate value reaches 50% (initially e_l is set 0). Figure 4.1 plots the throughput of two communicating nodes in two different scenarios for both 11 Mbps and 5.5 Mbps bit rates while assuming a Ricean fading channel [36], using *QualNet* network simulator. First, when the distance that separates the nodes and the access point (AP) is around 30 meters, we notice that when the nodes use the 11 Mbps data rate, 14% PER is suffered, and the throughput is much higher than that obtained when 5.5 Mbps bit rate was used (8% PER at 5.5 Mbps). However, as the distance from the AP increases to 45 meters in the second scenario, attenuation and multi-path losses increase the PER to 58% at 11 Mbps, while the lower rate faces a lesser PER value of 28% resulting in a better and more stable throughput over time. Indeed, the derived threshold, e_{max}, can be an accurate indicator about the event of decrementing the PHY transmission rates. Alternatively, a wireless node should also decide when to increase or restore its previous transmission rate in response to an improved channel quality. For instance, when a node moves closer to a Base Station (BS) or an Access Point (AP) , it may need to increase its transmission rate to obtain a higher throughput. The performance enhancement can be seen after, for example, a defined number of consecutive successful transmissions (ACK count). Here, the station would try to transmit using the higher bit rate while measuring the PER. If the corruption rate is beyond the threshold, then the station would switch to the lower rate. Otherwise, if the experienced PER at the higher rate is less than the threshold, then the station should continue using this rate. We note here that PER can be easily estimated using the online estimator introduced in the previous chapter.

4.3.2 Algorithm Overview

Based on the events defined for the thresholds at which transmission rate should be adjusted, we have implemented a link adaptation algorithm (*Algorithm 1*) that operates at the MAC layer as follows; initially active nodes would be listening to the channel regardless of whether a node is in transmitting, receiving or idle mode. Hence, continuous estimation of the channel condition (i.e., PER) using the online error estimator is possible, since the status of all slots can be determined at all times. A node wishing to transmit will apply the DCF contention mechanism first and when the medium is idle, it will start transmitting using the highest available physical data rate. The algorithm assumes that the PER faced at the next lower rate (e_l) is initially zero, as the lower rate has not been used yet, and hence the node could not run the estimator to determine the PER value. If the transmission were successful, then the node proceeds with using that rate. Otherwise, if the ACK timeout duration expired, and the current measured PER ($e_{current}$) is greater than the calculated PER threshold (e_{max}), then this indicates that the transmitted frames are subject to excessive corruption that is beyond the threshold, and hence a retransmission is scheduled with the next lower rate. Here, another extension we added is that the node would not double the contention window CW but maintain it after decrementing the rate due to a transmission failure, since there is no need to increase the CW on the lower rate. This is because the transmission failure is due to the noise in the channel. Therefore, the error faced at the lower rate will be reduced greatly and consequently, there is no need to use larger CW for the next transmission. Similarly, the process is repeated for lower bit rates and the estimated values of the PER are saved in an array for each bit rate as they are required later in

Algorithm 1 Basic Rate Adaptation Algorithm

$r_{max} \leftarrow Max\_PHY\_Rate$
$r_{min} \leftarrow Min\_PHY\_Rate$
$r_{current} \leftarrow r_{max}$
$e_l \leftarrow 0$
for *each transmission* **do**
    $e_{current} \leftarrow$ *retrieve from estimator && update array*
    $r_{lower} \leftarrow r_{current} - 1$
    //calculate max tolerable PER for current rate
    $e_{max} \leftarrow 1 - \frac{r_{lower}}{r_{current}}(1 - e_l)$
    if $ACK\_num \geq 10$ **then**
        //increase rate
        $r_{current} \leftarrow min(^{++}r_{current}, r_{max})$
        $ACK\_num \leftarrow 0$
    end if
    if $ACK\_Timeout$ && $e_{current} \geq e_{max}$ **then**
        //decrement transmission rate
        $r_{current} \leftarrow max(^{--}r_{current}, r_{min})$
        Maintain same CW //i.e. do not double the CW right after decrementing the rate
        Backoff & Transmit packet
    end if
    //trace ACK count for rate increase
    if *Current_Tx && Previous_Tx were successful* **then**
        ACK_num^{++}
    else
        $ACK\_num \leftarrow 0$
    end if
end for

the calculation of the PER threshold (e_l) which was set initially to zero. For increasing the transmission rate, each node keeps track of the successful ACK frames; then, a sequence of consecutive acknowledgments (e.g., 10 consecutive ACKs) would indicate that the channel condition has improved and the next higher rate could be used.

4.4 QoS–Based Rate Adaptation Strategy

4.4.1 Preliminaries

The Wireless LAN presents unique challenges to multimedia applications of strict QoS requirements; this is due to the limited capacity and shared medium of WLANs that is subject to unpredictable erroneous transmissions. Today's real-time applications require service differentiation when admitted to the medium, in contrast to the best effort flows. Hence, it is essential to integrate QoS requirements in any rate adaptation algorithm to ensure real-time flows are treated differentially.

QoS streams with real−time traffic differs from other data flows, mainly, in the delay requirements they are obliged to meet; here, the receiver demanding real−time traffic consider late packets as lost and will eventually discard them. Accordingly, minimizing the delay time of QoS streams is a key issue for today's real-time applications whose boundaries differ from one application to another, and are usually included in the table of specifications (TSPEC) [4]. The delay associated with packet transmission in WLANs can be divided mainly into queueing delay and medium access delay. The queueing delay resembles the time the packet spent since it entered first the transmission queue till time it got on the head of queue and start contending for channel access. Medium access delay, however, is composed of the total time the packet waited while contenting for idle medium time till it gets transmitted. Hence, the total delay is the summation of queueing delay and medium access delay and is usually referred to as end−to−end delay as the propagation time of packet is neglected.

4.4.2 Deriving the Essential QoS Requirements

Now, we attempt to find the relation between the retransmission limit and the end-to-end delay for the purpose of the rate adaptation technique which will follow. First, it is important to note that the medium access delay, mentioned in the previous paragraph, for a packet of priority class i is directly influenced by the retransmission limit (R_i), which is bounded by [5]:

$$0 \leq R_i \leq R_i^{max} \tag{4.4}$$

Where R_i^{max} represents the maximum retransmission limit For instance, $R_i = 0$ corresponds to the case where the queuing delay is very high and the system cannot guarantee the delay requirement of the arriving traffic stream. Here, a packet is not allowed any retransmission and it is dropped upon the first collision. Alternatively, if the queueing delay is negligible (e.g., when the load is light), then R_i converges towards its limit, R_i^{max} , for which an exact expression may be derived. Using the fact that a packet is dropped when it reaches the last backoff stage and experiences another transmission failure; the average time to drop a packet is estimated [20] as:

$$E[T_{drop}] = \frac{CW_{min,i}(2^{R_i^{max}+1} - 1) + (R_i^{max} + 1)}{2} E[slot] \tag{4.5}$$

$CW_{min,i}$ represents the lower bound of the contention window for priority class i and $E[slot]$ is the average slot time (whether an idle slot, a busy slot with a successful transmission or a busy slot with a failed transmission), whose value can be measured by a station. Subsequently, if an arriving flow is of *constant bit rate* (CBR) type and delay bound D_i, the station will use the measured $E[slot]$ in order to estimate the retry limit for the incoming flow. By setting $E[T_{drop}] = D_i$, one can solve for the maximum retry limit [5]:

$$R_i^{max} = \left\lfloor log_2 \left\{ \frac{2D_i}{CW_{min,i}E[slot]} + 1 \right\} - 1 \right\rfloor \tag{4.6}$$

for priority class i with delay bound D_i and slot time $E[slot]$.

In addition to the strict delay requirements of real–time streams, a packet dropping threshold may be specified to ensure the stability of QoS application. A packet is usually dropped at the MAC layer when the retransmission limit is reached. Hence, after several consecutive transmission failures, due to either collision or transmission corruption, the packet is discarded and dropped. Depending on the type of the real-time traffic, drop rate can be tolerated up to a fixed threshold. Certain value of dropped packets may be tolerable for most multimedia traffic flows, however, if the dropped packets portion surpass a preset limit, degrading performance of the real-time application will be noticed on the receiving side. Although the drop rate is not included in TSPEC table of IEEE 802.11e, applications can specify it in their traffic requests. The probability that a given packet of traffic flow of priority i is dropped after R_i attempted transmissions is given by:

$$p_i^{drop} = (1-(1-p_i^{error})(1-p_i^{col}))^{R_i+1} \tag{4.7}$$

where p_i^{error} and p_i^{col} represent the packet error and packet collision probabilities respectively.

Now, using the relation in (4.6), one can easily derive the retransmission limit (R_i^{max}) from the delay requirements. Additionally, given that p_i^{col} is always less than 1, and with a specified maximum allowable packet drop $p_i^{max\_drop}$ provided by the QoS application, one can obtain the value for the maximum tolerable noise level from (4.7):

$$e_{max} = \frac{\sqrt[(R_i+1)]{p_i^{max\_drop}} - p_i^{col}}{1-p_i^{col}} \tag{4.8}$$

Subsequently, as the channel gets impaired and its conditions deteriorate, more packets will be dropped. This will lead to low performance at the receiver side since the QoS flow requirements were violated. Hence, it is crucial to diagnose such degrading factors. Here, equation (4.8) identifies the maximum corruption rate at which delay and maximum dropping rate are critical. This threshold (e_{max} in Equation 4.8) will indicate the level of the frame loss that a certain real-time application can maintain. Indeed, this threshold e_{max} will be an important factor in our QoS rate adaptation algorithm, *QoS−Rate Adapation algorithm, (QoS−RA)* which we present in *Algorithm 2*.

4.4.3 Dynamic Bandwidth Allocation

Transmission opportunity (TXOP) is a new concept that was introduced in the recent IEEE 802.11e draft [4]. The TXOP represents the allowable time period at which nodes associated with a QoS access point (QAP) can initiate transmission during a service interval (SI). Two modes of TXOP are defined; the initiation of the TXOP period at which the medium permits access and then the multiple frames transmission exchange and is often called contention free bursting (CFB) as there is no immediate contention between packets transmission. By default, TXOP values are assigned fixed values by QAPs according to the traffic type requirements which are shown in Table 4.2 [6]. We note that a TXOP limit of 0 value for background and best effort traffic types indicates that only a single frame may be transmitted during a TXOP period. As frames transmissions over the wireless medium are usually associated with missed ACKs that are lost due to either collision or noisy channel, the Surplus Bandwidth Allowance (SBA) parameter was introduced in the

TXOP assignment to ensure the requesting QoS station (QSTA) is allocated a minimum amount of excess time by the QAP such that the retransmissions can take place within the same service interval. The SBA factor is calculated by each QSTA based on the loss probability and then sent through the management frame to the QAP, which in turn will consider when assigning the TXOP limit for the specific QSTA. The IEEE 802.11e draft suggests the SBA_i for traffic of AC i be calculated as follows:

$$SBA_i = \frac{N_{allocated}}{N_{payload}} \tag{4.9}$$

where $N_{allocated}$ represents the number of allocated packets that can be sent during a service interval and $N_{payload}$ represents the actual packets needed to be transmitted during the service interval. $N_{allocated}$ can be represented as "$N_{payload} + Packets\ in\ excess$".

Some related studies suggest the QAP administrator assigns the SBA an approximate fixed value (e.g., 1.1 in [9]). However, this is not accurate as the wireless channel conditions are not constant due to various reasons (e.g., QSTAs mobility). In addition to that, the standard suggests the SBA be set by the QSTAs rather than the QAP. Therefore, the SBA should reflect the variant transmission failures at each QSTA. Accordingly, the authors of [5], have suggested calculating the SBA according to the following equation:

$$P_i^c = 1 - \frac{1}{SBA_i} \tag{4.10}$$

where P_i^c is the collision probability for the flow of AC i. Here, the authors assume ideal channel conditions and hence they neglected the packet loss due channel noise. In practice, however, erroneous transmissions constitute a major portion of the failed transmissions. Hence, it is of great interest to extend the SBA calculation to include not only the collision probability of the QoS flows, but also the error probability P_i^e, which is easily evaluated using the online estimator introduced in the previous chapter. Thus, SBA can then be expressed as:

$$SBA_i = \frac{1}{(1-P_i^c)(1-P_i^e)} \tag{4.11}$$

AC	TXOP for PHY 802.11b	TXOP for PHY 802.11a/g
AC_BK	0	0
AC_BE	0	0
AC_VI	6.016ms	3.008ms
AC_VO	3.264ms	1.504ms

Table 4.2: TXOP limit default parameters.

Consequently, QSTAs would be calculating their SBA_i during each SI. This dynamic SBA will reflect their need for the extra retries essential to overcome the failed transmissions due to either collision or channel noise. Afterwards, the QAP can assign dynamically the corresponding $TXOP_i$ for each QSTA as follows:

$$TXOP_i = TXOP_i^{default} \times SBA_i \tag{4.12}$$

Note that $TXOP_i^{default}$ corresponds to the default TXOP values (Table 4.2). Now, using the dynamic TXOP assignment approach, nodes will have a relatively fair channel access duration. This is because nodes that suffer high frame loss rate, and require extra SBA will receive larger TXOP so that packets will have higher retransmission chances. On the other hand, if the dynamic allocation of TXOP was not used, then stations facing poor channel conditions would retain the same retransmission chances as any other station with ideal medium conditions.

4.4.4 QoS−Rate Adaptation Algorithm:

Now, we elaborate on our approach of rate adjustment (*Algorithm 2*). *Algorithm 2* differs from the basic rate adaption mechanism presented in *Algorithm 1*, in that it can provide differentiated treatment for traffic flows based on their priorities requirements. Initially, the average slot time $E[Slot]$ is calculated at every node by periodically keeping track of the slots (whether an idle slot, a busy slot with a successful transmission or a busy slot with a failed transmission, i.e., collision or corruption) and counts their number per period of measurement (e.g., N_i idle slots, N_s successful slots, N_c collided slots, and N_e for corrupted slots) and measures the total idle period (e.g., $\Gamma_i = N_i \times \sigma$), the total successful period (Γ_s), collision period (Γ_c) and corruption period (Γ_e). $E[slot]$ is then estimated as $(\Gamma_s + \Gamma_i + \Gamma_c + \Gamma_e)/(N_s + N_i + N_c + N_e)$ ([5]). Afterwards, admitted data streams are classified according to their type and priority. Hence, if the admitted flow specifies the delay and the maximum tolerable packet drop parameters, then the maximum retry limit will be derived according to Equation 4.6. In addition, if the QAP starts a new service interval, then the QSTA will calculate the SBA value according to equation 4.11, where loss caused by both, collision and channel noise will be considered. This SBA value shall be sent through management frame to the QAP which in turn assigns the dynamic TXOP limits for the associated QSTA. Afterwards, having the maximum retry limit R_i^{max}, it is feasible to obtain the error threshold value at which the rate should be adjusted using the derived threshold in Equation (4.8).

We note here that the collision and error probabilities are both determined from the online estimator. Hence, upon a failed transmission (ACK timeout), a station will select a lower bit rate for its future transmission in case the experienced PER was greater or equal to the calculated threshold that complies with its differential requirements. On the other hand, if the admitted flow carries best effort data, or no delay or maximum packet drop limit were specified, then *Algorithm 1* is applied in estimating the channel quality as well as the PER threshold values at which adjusting data rate would be triggered. For incrementing the transmission rate, the algorithm uses the same approach of the basic algorithm in keeping track of the count of the successful ACK frames, where a station will switch to the next higher rate if a certain desired number of consecutive ACK frames is received, which is an indicator of good channel conditions. We have included the state flow diagram of the rate adaptation algorithm in Figure 4.2 for further clarification purposes where both QoS and non-QoS flows were taken into consideration.

4.5 Performance Analysis

4.5.1 Basic Algorithm

Extensive experiments through computer simulations were generated in order to validate the correctness and effectiveness of the proposed rate adjustment (RA) methodology. Initially, we simulate scenarios with saturated best effort flows of 1050 bytes payload size (no QoS) using *QualNet* simulator through two topologies depicted in figures 4.3 and 4.4. We consider 5 nodes, 30 meters initially far from the AP in a Ricean fading channel with K

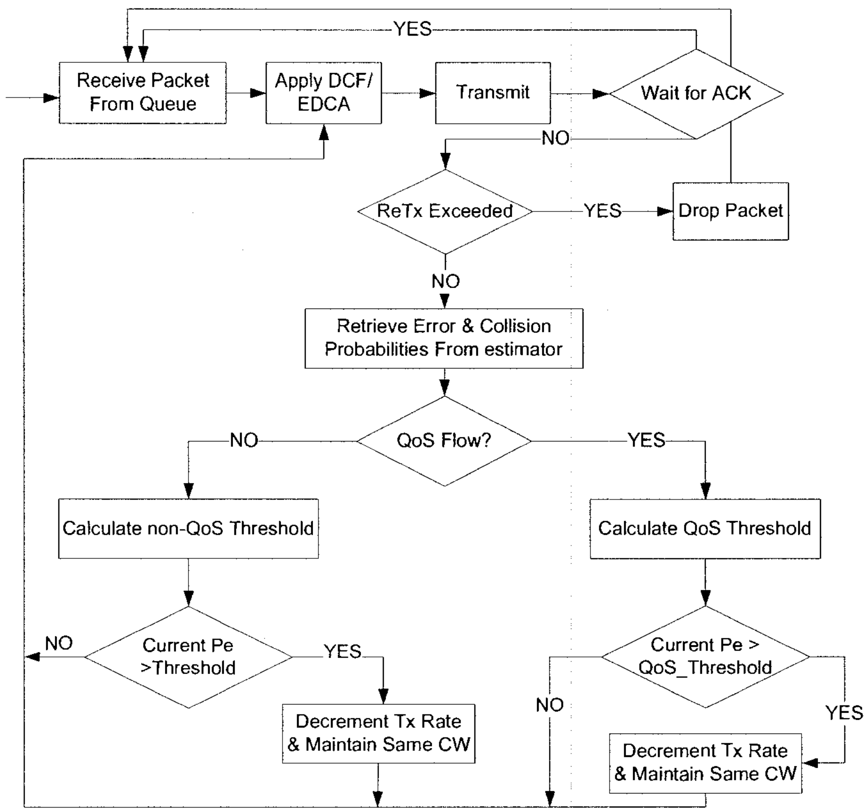

Figure 4.2: Rate Adaptation Algorithm State Flow Diagram

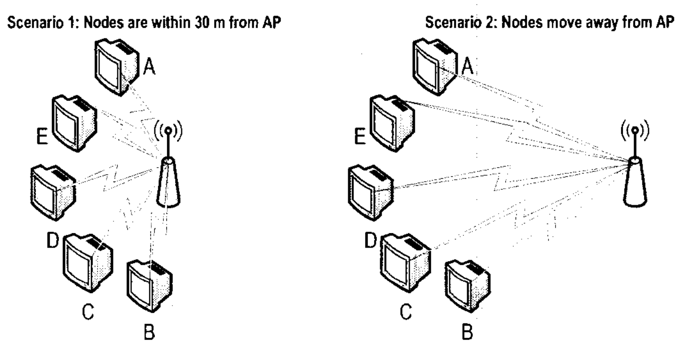

Figure 4.3: Initial Simulation topology.

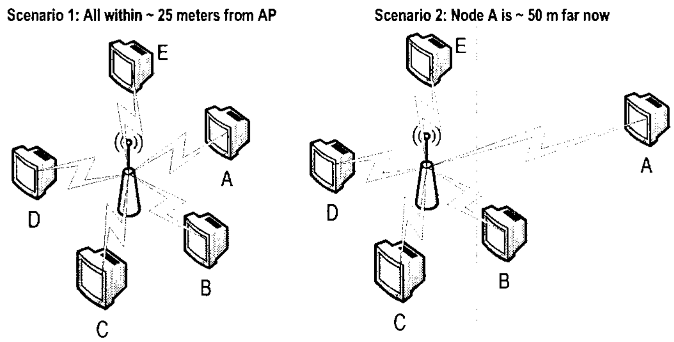

Figure 4.4: 802.11 anomaly scenario.

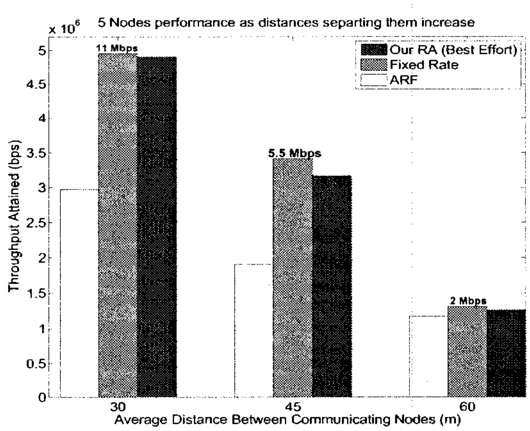

Figure 4.5: Performance as function of distance.

factor [36] 10^{-4}, and Doppler frequency of 20 Hz; we compare the performance of Auto Rate Fallback (ARF) [40], our rate adaptation algorithm, in addition to the optimal fixed PHY rate[1]. The performance of the system throughput is then recorded as nodes move 45 m and then 60 m away from the AP. The performance results are shown in Figure 4.5, where we notice a remarkable out-performance of the proposed link adaptation algorithm over the standard ARF method (close to that of the fixed rate). This is due to the fact that ARF relies on counting the number of consecutive missed acknowledgments which have resulted from transmission failures due not only to channel conditions but also to simultaneous transmissions (collisions); our rate adaptation algorithm, however, is capable of measuring the noise threshold at which it is crucial to adjust the transmission rate without confusion with collisions and hence, a better performance is clearly noticed when the latter method is used. Additionally, when a node running our RA chooses to decrement its transmission rate upon reaching the computed corruption threshold, it refrains from doubling its contention window as the cause of transmission failure (i.e., channel noise) is diminished; this indeed minimizes unnecessary further backoff delays. Moreover, one can notice that as the distance increases between the transmitting nodes and the access point, the difference between throughput of both mechanisms decreases; this is due to the lower rates used that will increase the contention and hence, limit the available bandwidth.

Now, we consider another scenario where the node mobility is more practical, as shown in Figure 4.4. Initially, all nodes (A through E) are within 25 meters from the AP; then, node A chooses to move to a new location that is 50 meters far. Figures 4.6 and 4.7 show

[1]The optimal fixed rate represents the PHY transmission rate used when all link adaptation schemes are turned off. Hence, all fixed rates (e.g., 11/5.5/2/1Mbps) will be used for simulations separately. Afterwards, the highest throughput will be plotted and used for comparing purposes.

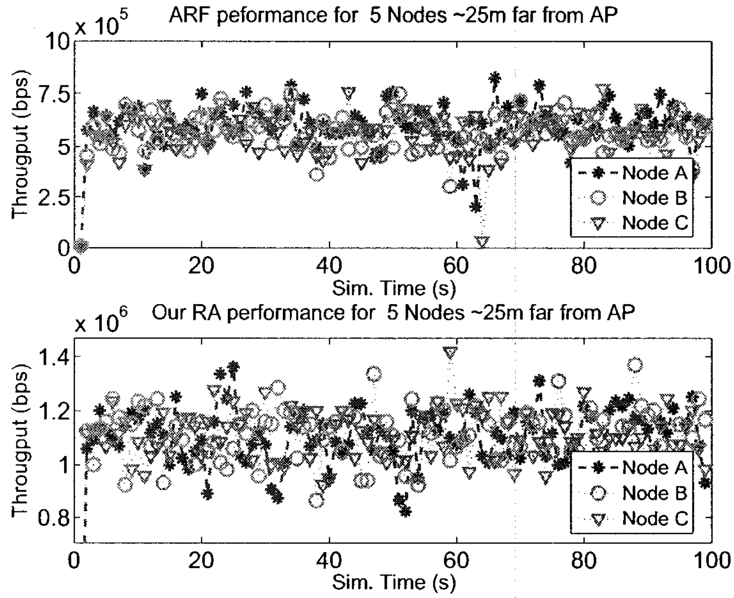

Figure 4.6: Performance as nodes are close to AP

the individual throughput of 3 nodes in the scenario versus time before and after node A moves. It is clear that the nodes achieve higher throughput while using our RA algorithm rather than ARF in the first topology; the reason behind this is that while all nodes are relatively close to the AP, packet corruption rate would be negligible since channel signals maintain their strength over small distances and hence the proposed RA will not attempt to decrease the bit rate, as opposed to ARF, which would decrement the link rate due to collisions. Now, we can see that as node A moves away, our RA achieves about 250 kbps individual throughput enhancement over ARF; it is interesting here to observe that although nodes B and C maintained their positions (i.e., did not move) while node A moved away from the AP, all nodes achieved almost the same individual throughput due to the increased duration of transmission required by node A; indeed, this time is reused from other stations time. This phenomenon is known as anomaly in IEEE 802.11 based networks [50].

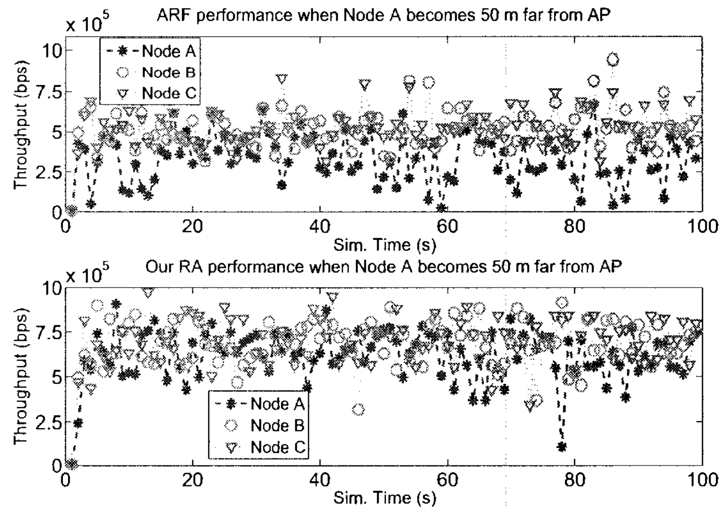

Figure 4.7: Performance as a node moves away from the AP

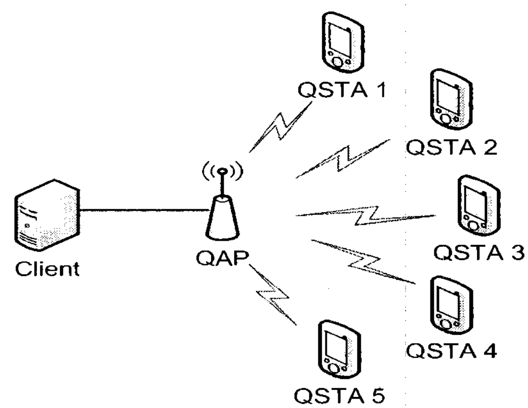

Figure 4.8: QoS simulation topology.

4.5.2 QoS-RA Verification

In this section, we present the simulation results for the proposed rate adaptation scheme in the presence of flows with *quality of service* requirements. As *QualNet* does not support the bandwidth allocation extension of the IEEE 802.11e, we used a modified version of ns-2 implemented by [6], in order to perform the required verifications. We set the simulation scenario as shown in Figure 4.8. We consider 5 QoS stations (QSTA), each carrying two almost saturated video flows (i.e., AC=2) of payload size 900 bytes. The QSTAs need

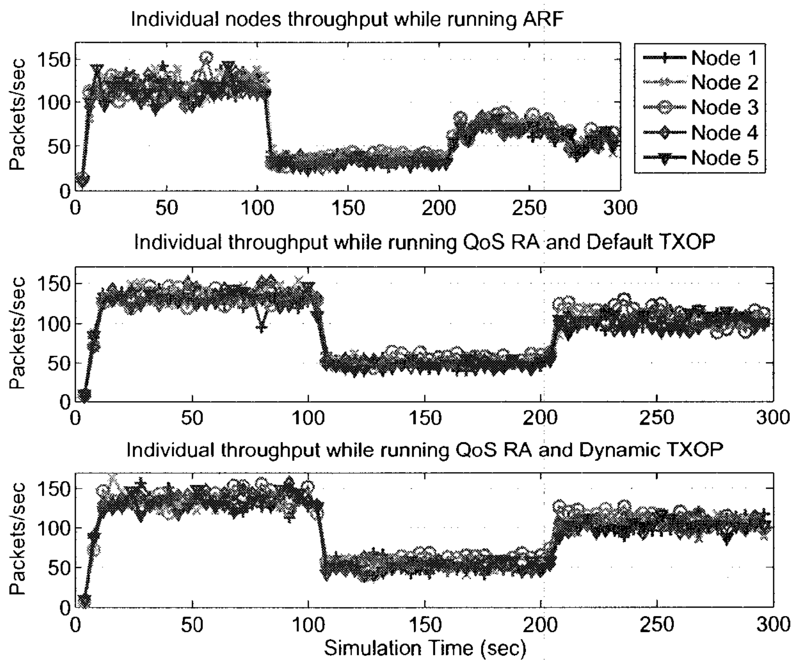

Figure 4.9: QoS flows under variable channel characteristics

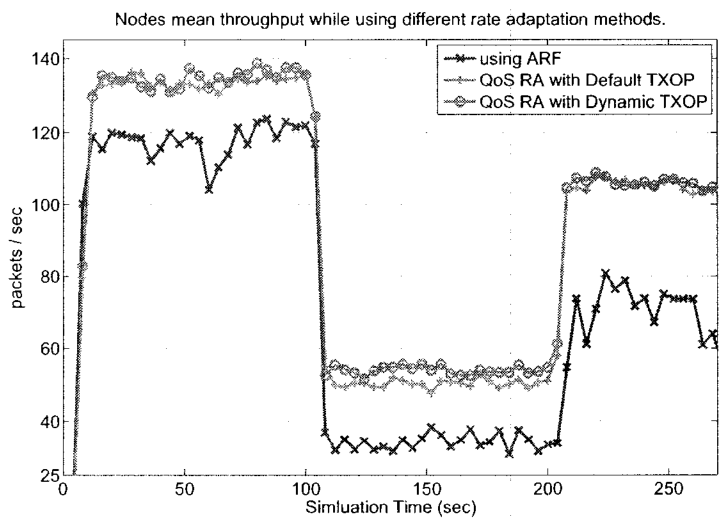

Figure 4.10: Comparing mean QSTAs throughput

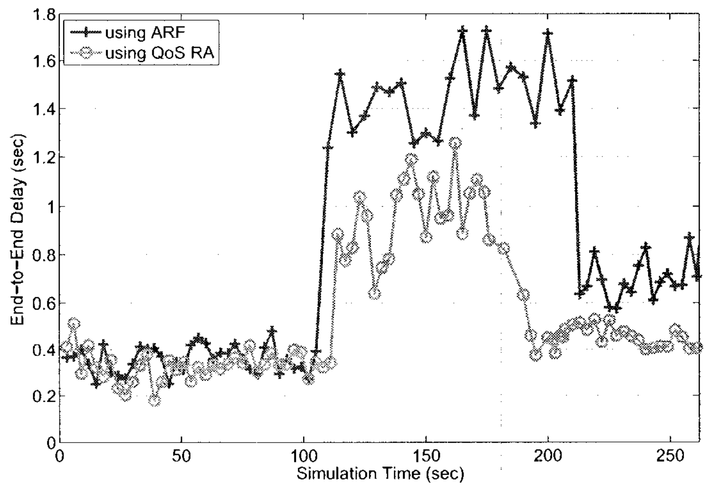

Figure 4.11: Average end-to-end delay

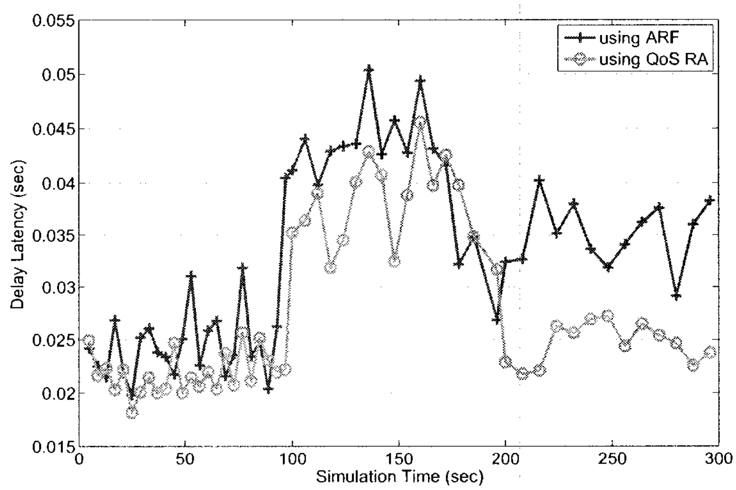

Figure 4.12: Jitter

to deliver their QoS packets to the wired client after coordinating with the QAP. We run the simulations for a duration of 300 seconds, which in turn was split into three intervals according to different Ricean channel conditions: during the first interval (0-99s), the nodes are close to the QAP ($\sim$ 20m), i.e., erroneous transmissions are negligible. Afterwards, the distance is increased to 50 meters in the next interval (100-199s), where the corruption rate is high. Finally, the distance between the QAP and QSTAs is decreased to around 35 meters at the last interval (200-300s) in order to enhance the medium conditions.

Figure 4.9 shows the individual node throughput versus simulation time for three link adjustment methods: the standard ARF, the proposed QoS rate adaptation method (QoS-RA) while using the default set TXOPs according to Table 4.2, in addition to the QoS-RA with the proposed dynamic bandwidth allocation technique. As we expected, we notice that individual nodes throughput is optimal during the first simulation time interval in the three cases when the QSTAs are close to the QAP. The throughput then drops down to below 50 packets per second as the channel conditions deteriorate due to QSTAs mobility away from the QAP. Finally, the throughput improves as the channel noise is reduced. We then compare the mean nodes throughput for the three link adaptation methods as show in Figure 4.10. It is clear that the QoS-RA proposed method outperforms the ARF throughout all the experiment. This is expected and is similar to the previous scenarios (with no QoS flows) as ARF is incapable of genuinely differentiating the causes of the transmissions failures. On the other hand, it is worthwhile to notice the out-performance of QoS-RA when dynamic bandwidth is enabled over that when the default preset TXOP limits are used. This is mainly apparent in the second interval. When the PER faced by QSTAs is negligible (first time interval), both the dynamic and constant bandwidth allocation QoS-RA achieve almost the same performance. However, as the corruption rate increases (second and last time interval), it is apparent that the rate adaptation algorithm with dynamic bandwidth allocation outperforms the one with the default TXOP values. This is because when the loss rate is low, the dynamically set SBA factor has insignificant effect. However as the corruption rate increases, the need for extra bandwidth becomes essential in order to achieve the desired performance while overcoming the failed retransmissions. Here, the dynamic SBA allocation takes this extra retransmission into consideration as we already presented and longer transmission opportunities will be assigned for that purpose, as opposed to the case where the constant default SBA values are used all the time.

Furthermore, the average end-to-end delay and jitter are affected as well. It can be easily noticed in Figure 4.11 that the proposed rate adjustment technique along with the dynamic TXOP assignment achieve a lower delay than the standard method. This is mainly due to two reasons; the QoS-RA decrements the transmission rate only when the threshold PER is reached rather than relying on missed ACKs whether from channel noise or collisions, which in turn create higher contention and hence, higher packet loss and retransmissions (i.e., large delays) as in the case of ARF. Moreover, after QSTAs choose to decrement the transmission rate, CW value is maintained rather than doubled because the transmission failure cause is reduced. Therefore, less pending time will be wasted in backing off using larger contention window, which results in lower delays.

Finally, we study the behavior of the QoS-RA method in heterogeneous environment (i.e., nodes experience different PER values at the same time). Figure 4.13 plots the individual throughput results of five QSTAs, where only one node (QSTA 2) moves away and towards the AP according to the three time intervals described in the previous scenario; all

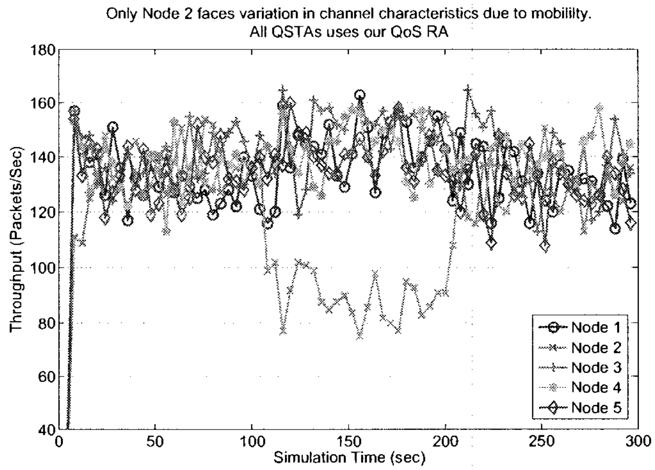

Figure 4.13: Testing anomaly problem for QoS flows

other nodes maintain the same distance close to the QAP (i.e., they face negligible PER all the time). It is apparent that QSTA 2 mobility did not degrade the performance of the other nodes through the whole simulation, as opposed to the situation when non-QoS were admitted to the channel as we have already seen in Figure 4.7.

The explanation for this behavior is that TXOPs restrict the transmission limit of a given QSTA by a time bound which should not be surpassed. Accordingly, QSTA 2 cannot initiate any further transmission when its TXOP limit is exceeded. Thus, flows associated with other nodes will not be affected by the degrading performance of QSTA 2. On the other hand, non-QoS flow having TXOP of 0 value (Table 4.2), which means that packet transmission is allowed any time the medium is sensed idle, during a service interval. Thus, the anomaly problem exists only for such flows.

4.6 Conclusion

In this chapter, we presented a novel and practical approach for link adaptation based on channel quality measurement. A basic version of the rate adaptation algorithm was introduced for adjusting the transmission rates based on threshold values of throughput enhancement along with maximum tolerable corruption rate in case of having best effort flows. Moreover, we integrated QoS requirements of delay and maximum drop rate along with dynamic bandwidth allocation in the link adaption process in a way that best serves the needs of real−time applications. Finally, we validated the algorithms through extensive simulations.

Algorithm 2 QoS−RA Algorithm

$r_{max} \leftarrow Max\_PHY\_Rate$
$r_{min} \leftarrow Min\_PHY\_Rate$
$r_{current} \leftarrow r_{max}$
$e_l \leftarrow 0$
for *each transmission* **do**
  $E[Slot] \leftarrow$ *average slot time*
  retrieve $e_{current}$ & p_i^{col} *from estimator && update array*
  $r_{lower} \leftarrow r_{current} - 1$
  if *QAP requests SBA value* **then**
    // QAP will use this SBA to assign dynamic TXOP limit for QSTAs
    calculate and send the SBA value relevant to current channel conditions
  end if
  if $ACK\_num \geq 10$ **then**
    //increase rate
    $r_{current} \leftarrow min(^{++}r_{current}, r_{max})$
    $ACK\_num \leftarrow 0$
  end if
  if *flow specifies delay_bound && max drop* **then**
    Derive retry limit R_i *from delay & max drop*
    //set threshold for QoS flow
    $e_{max} \leftarrow \left(\frac{\sqrt[(R_i+1)]{p_i^{max\_drop}} - p_i^{col}}{1 - p_i^{col}} \right)$
  else
    //follow best effort approach
    $e_{max} \leftarrow 1 - \frac{r_{lower}}{r_{current}}(1 - e_l)$
  end if
  if $ACK\_Timeout$ && $e_{current} \geq e_{max}$ **then**
    //decrement transmission rate
    $r_{current} \leftarrow max(^{--}r_{current}, r_{min})$
    Maintain same CW //i.e. do not double the CW right after decrementing the rate
    Backoff & Transmit packet
  end if
  //Trace ACK count for rate increase
  if *Current_Tx && Previous_Tx were successful* **then**
    ACK_num^{++}
  else
    $ACK\_num \leftarrow 0$
  end if
end for

Chapter 5

Conclusion and Future Work

5.1 Dissertation Summary

Wireless Local Area Network (WLAN) has been a rapidly emerging technology, particularly during the past few years. Among the numerous challenges encountered by WLANs are the operations of its medium access control schemes in noisy channels. The binary exponential backoff function, responsible for resolving network access conflicts, treats all transmission failures as caused by collisions. However, due to the volatile nature of the wireless medium, failed packets exchange due to erroneous transmissions are very common. As a result, it is essential that the MAC layer identifies the causes of packet losses in order to improve the performance of the DCF. Additionally, due to the different modulations and codings of the various 802.11 PHY rates, rate adaptation algorithms are of great interest. The key principle in any efficient RA technique is to identify the event at which performance on a lower rate is favorable.

In the first part of this thesis, we presented an analytical and performance analysis for the binary exponential backoff. It is apparent now, that incrementing the contention window upon each transmission failure is essential only in the case the channel is highly loaded. On the other hand, when the number of active stations communicating in the medium is low, following the DCF procedure upon each failed transmission due both collided and corrupted packets, will result in the waste of air time. Hence, in the latter case, contention window size should be maintained if the channel noise was behind the unacknowledged packets. Accordingly, a hybrid access method was presented that achieve optimal performance in different channel loads.

New rate adaptation scheme was introduced in the second part of the this work. The efficiency of the link adjustment algorithm resides in the simplicity of its operation, which is based on simple throughput analysis long with PER estimation at the MAC layer level. Moreover, our algorithm supports the IEEE 802.11e MAC extensions. In fact, we integrated the QoS requirements in the procedure of adjusting the transmission rate so that strict parameters such as delay and maximum drop rate are respected. Moreover, we implemented the dynamic assignment of transmission opportunities (TXOPs) in way that provide extra reserved duration for nodes facing high packet loss rate.

5.2 Future Work

This dissertation work has presented thorough investigation and enhancement for the IEEE 802.11 DCF operations at the MAC layer. The work was validated through analytical and computer simulation models. Furthermore, a rate adaptation technique was introduced that integrated the IEEE 802.11e QoS flow types and requirements along with dynamic bandwidth allocation. Yet, the wireless networks architecture can take complex configurations such as multi-hop or ad hoc networks where transmitted packets are relayed through intermediate nodes to reach their final destination. Hence, future work is needed to validate and extend the presented work in such networking topologies. Other areas that need further research in the future can be summarized as follows:

- Testbed Usage: The presented modifications of the binary exponential backoff function can be studied using testbed scenarios in domestic environment where channel noise can be very realistic. Moreover, QoS rate adaptation validation in such mediums would be of much interest as well.

- Multi-hop Networks: As mentioned earlier, the study of the BEB operations in multihop networks is essential, especially that the need for such topologies is emerging rapidly. In particular, many factors need to considered in such environments, such as the aggregate effect of interfered signals.

- Misbehavior in IEEE 802.11e: Misbehavior in IEEE 802.11 shared medium has attracted lots of researchers lately. Here, selfish hosts may achieve higher and an unfair throughput share by not respecting the MAC protocol operations. The use of surplus bandwidth allocation (SBA) by IEEE 802.11e has created a new vulnerability issue, since selfish nodes can cheat simply by requesting a large and unjustified surplus bandwidth. Hence, resolving such deficiency is crucial.

- IEEE 802.11e Anomaly: As we have seen in Chapter 4, IEEE 802.11e TXOPs protect the QoS flows from the anomaly phenomena through the specified transmission time limit which can not be surpassed. However, non-QoS flows have TXOP of 0 value, i.e. packet transmission is allowed any time the medium is sensed idle. Thus anomaly effect still exists for such flows and requires further investigations and resolution.

Bibliography

[1] IEEE 802.11 Specification, *"Wireless LAN Media Access Control (MAC) and Physical Layer (PHY) Specifications"*, 1999.

[2] Google Inc., *"Google WiFi"*, http://wifi.google.com/.

[3] Concordia University, *"Instructional and Information Technology Services (IITS)"*, http://iits.concordia.ca/.

[4] IEEE 802.11e WG, *"Wireless LAN Medium Access Control (MAC) Enhancements for Quality of service (QoS)"*,IEEE 802.11e Draft 8.0.

[5] Y. Liu, C. Assi, and A. Agarwal *"Enhanced Per-Flow Admission Control and QoS Provisioning in IEEE 802.11e Wireless LANs"*, in IEEE Transactions on Vehicular Technology, to appear, 2007.

[6] S. Wiethlter, C. Hoene, A. Wolisz, *"Perceptual Quality of Internet Telephony over IEEE 802.11e Supporting Enhanced DCF and Contention Free Bursting"*, Telecommunication Networks Group, Technische Universitt Berlin, 2004.

[7] Elena Lopez-Aguilera, Martin Heusse, Franck Rousseau, Andrzej Duda, Jordi Casademont, *"Evaluating Wireless LAN Access Methods in Presence of Transmission Errors"*, IEEE INFOCOM'2006, Barcelona, Spain, Poster, 2006.

[8] P. Chatzimisios, A. C. Boucavalas, V. Vistas, *"Performance Analysis of IEEE 802.11 DCF in Presence of Transmission Errors"*, IEEE ICC'2004, June, 2004.

[9] K. Kumar Gavini, V. Apte and S. Iyer,*"PLUS-DAC: A distributed admission control scheme for 802.11e WLANs"*, in the Proc. of ICON, Kuala Lumpur, Malaysia, Nov 2005.

[10] Q. Ni, T. Li, T. Turletti, Y. Xiao, *"Saturation Throughput Analysis of Error-Prone 802.11 Wireless Networks"*, Journal of Wireless Communications and Mobile Computing, Vol.5, Issue 8, pp. 945-956, December, 2005, .

[11] L. Bononi, M. Conti and E. Gregori, *"Design and performance evaluation of an asymptotically optimal backoff algorithm for IEEE 802.11 Wireless LANs"*, in the Proc. of the 33rd Annual Hawaii International Conference on System Sciences, 2000.

[12] D. Malone, P. Clifford, D. J. Leith, *"MAC Layer Channel Quality Measurement in 802.11"*, IEEE Communications Letters, Vol. 11, No. 2, February, 2007.

[13] H. Ma, J. Zhu, and S. Roy, *"On Loss Differentiation for CSMA-Based Dense Wireless Network"*, IEEE Communications Letters, Vol. 11, No. 11, November, 2007.

[14] D. Aguayo, J. Bicket, S. Biswa, G. Judd, R. Morris, *"Link Level Measurements from an 802.11b Mesh Network"*, in Proc. of ACM SIGCOMM, Boston, USA, 2004.

[15] Tamer Nadeem, Ashok Agrawala, *"IEEE 802.11 DCF Enhancements for Noisy Environments"*, in Proc. of IEEE PIMRC'2004, Barcelona, Spain, September 2004.

[16] Martin Heusse and Franck Rousseau and Romaric Guillier and Andrzej Duda, *"Idle Sense: An Optimal Access Method for High Throughput and Fairness in Rate Diverse Wireless LANs"*, in Proc. of ACM SIGCOM, Philadelphia, USA, August, 2005.

[17] Scalable Network Technologies, *"QualNet Simlator"*, http://www.scalable-networks.com/.

[18] D. Aguayo, J. Bicket,S. Biswas, G. Judd, and R. Morris, *"Link-Level Measurements from an 802.11b Mesh Network"*, in Proc. of ACM SIGCOMM, Boston, USA, August, 2004.

[19] Engelstad, P.E., sterb, O.N., *"Analysis of the Total Delay of IEEE 802.11e EDCA and 802.11 DCF"*, in Proc. of IEEE International Conference on Communication (ICC06), Istanbul, June, 2006.

[20] P. Chatzimisios, A.C. Boucouvalas, V. Vitsas, *"IEEE 802.11 Packet Delay - A Finite Retry Limit Analysis"*, in GLOBECOM03, Vol. 2, pp. 950-54, December, 2003.

[21] I. Aad, Q. Ni, C. Barakat and T. Turletti, *"Enhancing IEEE 802.11 MAC in Congested Environments"*, in Proc. of IEEE ASWN, Boston, USA, 2004.

[22] P. Chatzimisios, A. C. Boucouvalas, V. Vitsas, A. Vafiadis, A. Oikonomidis and P. Huang, *"A Simple and Effective Backoff Scheme for the IEEE 802.11 MAC Protocol"*, in Proc. of the International Conference on Cybernetics and Information Technologies, Systems and Applications (CITSA 2005), Orlando, USA, July 2005.

[23] A. Miu, G . Tan, H. Balakrishnan, and J. Apostolopoulos, *"Divert: Fine-Grained Path Selection for Wireless LANs"*, in Proc. of MOBISYS'2004, pp. 203-216, Boston, USA, June, 2004.

[24] Mathieu Lacage, Mohammad Hossein Manshaei, and Thierry Turletti, *"IEEE 802.11 Rate Adaptation: A Practical Approach"*, in Proc. of MSWiM'2004, pp. 126-134, October, 2004.

[25] Starsky H.Y. Wong, Hao Yang, Songwu Lu and Vaduvur Bharghavan, *"Robust Rate Adaptation for 802.11 Wireless Networks"*, in Proc. of MOBICOM, 2006.

[26] B. Sadeghi, V. Kanodia, A. Sabharwal and E. Knightly, *"Opportunistic media access for multirate ad hoc networks"*, in Proc. of MOBICOM, Atlanta, Georgia, USA, September, 2002.

[27] Tarun Joshi, Anindo Mukherjee, Younghwan Yoo, and Dharma P. Agrawal, *"Airtime Fairness for IEEE 802.11 Multirate Networks"*, in IEEE Transactions On Mobile Computing, Vol. 7, No. 4, April, 2008.

[28] D. Qiao and Sunghyun Choi, *"Goodput Enhancement of IEEE 802.11a Wireless LAN via Link Adaptation"*, in Proc. IEEE ICC, Finland, 2001.

[29] Qixiang Pang, Soung C. Liew, and Victor C. M. Leung, *"Design of an Effective Loss-Distinguishable MAC Protocol for 802.11 WLAN"*, IEEE Communications Letters, Vol. 9, No. 9, September, 2005.

[30] Jongseok Kim, Seongkwan Kim, Sunghyun Choi, Daji Qiao, *"CARA: Collision–Aware Rate Adaptation for IEEE 802.11 WLANs"*, in Proc. of IEEE INFOCOM'2006, Barcelona, Spain, April, 2006.

[31] G. Bianchi, *"Performance Analysis of the IEEE 802.11 Distributed Coordination Function"*, IEEE Journal on Selected Areas in Communications, Vol. 18, No. 3, March, 2000, pp. 535-547.

[32] Kamesh Medepalli and Fouad A. Tobagi, *"Throughput Analysis of IEEE 802.11 Wireless LANs using an Average Cycle Time Approach"*, in Proc. of IEEE GLOBECOM'2005, St. Louis, USA, 2005.

[33] Kamesh Medepalli and Fouad A. Tobagi, *"On Optimization of CSMA/CA based Wireless LANs: Part I – Impact of Exponential Backoff"*, in Proc. of ICC'2006, June, 2006.

[34] D. Chizhik, J. Ling, P. W. Wolniansky, R. A. Valenzuela, N. Costa, and K. Huber, *"Multiple–Input–Multiple–Output Measurements and Modeling in Manhattan"*, IEEE Journal on Selected Areas in Communications, Vol.21, No. 3, April, 2003.

[35] Daniel B. Faria, *"Modeling Signal Attenuation in IEEE 802.11 Wireless LANs – Vol. 1"*, Computer Science Department, Stanford University.

[36] Charalambos D. Charalambous and Nickie Menemenlis, *"Stochastic Models for Short-Term Multipath Fading Channels: Chi-square and Ornstein-Uhlenbeck"*, in Proc of the Conference on Decision and Control, Phoenix, Arizona, USA, December 1999

[37] P. Chevillat, J. Jelitto, A. Noll Barreto, and H.L. Truong, *"A Dynamic Link Adaptation Algorithm for IEEE 802.11a Wireless LANs"*, in Proc of ICC '03, Seatle, USA, May, 2003.

[38] Qixiang Pang, Victor C.M. Leung and Soung C. Liew *"A Rate Adaptation Algorithm for IEEE 802.11 WLANs Based on MAC–Layer Loss Differentiation"*, in Proc of Broadband Networks'05, Boston, USA, October, 2005.

[39] Tsung–Han Lee, Alan Marshall and Bosheng Zhou *"A QoS–based Rate Adaptation Strategy for IEEE a/b/g PHY Schemes using IEEE 802.11e in Ad–hoc Networks"*, in Proc of the International conference on Networking and Services, July, 2006.

[40] Ad Kamerman and Leo Monteban *"WaveLAN–II: A High–Performance Wireless LAN for the Unlicensed Band"*, in Bell Labs Technical Journal, Summer 1997.

[41] D. Qiao, S. Choi, A. Jain, and K. G. Shin, *"MiSer: an optimal low-energy transmission strategy for IEEE 802.11a/h"*, in Proceedings of the ACM International Conference on Mobile Computing and Networking -Mobicom, San Diego, USA, September 2003, pp. 161175.

[42] Ma, H. and Roy, S., *"Simple and Effective Carrier Sensing Adaptation for Multi Rate Ad-Hoc MESH Networks"*, in Proceedings of the IEEE International Conference on Mobile Ad-hoc and Sensor Systems -MASS, Vancouver, Canada, 2006, pp. 795800.

[43] Yang, X. and Vaidya, N., *"A Spatial Backoff Algorithm Using the Joint Control of Carrier Sense Threshold and Transmission Rate"*, in Proceedings of the IEEE International Society Conference on Sensor, Mesh and Ad Hoc Communications and Networks-SECON, San Diego,USA, June 2007, pp. 501511.

[44] Jong-Ok Kim, Hideki Tode, and Koso Murakami, *"Service–based Rate Adaptation Architecture for IEEE 802.11e QoS Networks"*, in Proc. of IEEE GLOBECOM'2005, St. Louis, USA, 2005.

[45] Michael R. Souryal and Nader Moayeri *"Joint rate adaptation and channel–adaptive relaying in 802.11 ad hoc networks"*, in Proc. MILCOM, October. 2006.

[46] G. Holland, N. Vaidya, and P. Bahl *"A rate-adaptive MAC protocol for multi-hop wireless networks"*, in Proc. MobiCom, 2001, pp. 236 251.

[47] Yang Xiao *"QoS guarantee and provisioning at the contention-based wireless MAC layer in the IEEE 802.11e wireless LANs"*, IEEE, Wireless Communications, Vol.13, pp. 14-21, Feb. 2006.

[48] Sunghyun Choi, del Prado Javier, Sai Shankar N and Mangold Stefan, *"IEEE 802.11e Contention-Based Channel Access (EDCF) Performance Evaluation"*, Communications, 2003. Proc. IEEE ICC, Vol.2, pp. 1151-1156, May 2003.

[49] Yang Xiao *"Performance Analysis of Priority Schemes for IEEE 802.11 and IEEE 802.11e Wireless LANs"*, IEEE Transactions on Wireless Communications, Vol.4, pp. 1506-1515, July 2005.

[50] M. Heusse, F. Rousseau, G. Berger-Sabbatel, A. Duda *"Performance Anomaly of 802.11b"*, in Proc. IEEE INFOCOM, 2003.