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Design and analysis of MAC protocols for wireless networks

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Design and analysis of MAC protocols for wireless networks

by

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DEDICATION

To my parents.

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ABSTRACT

During the last few years, wireless networking has attracted much of the research and industry interest. In addition, almost all current wireless devices are based on the IEEE 802.11 and IEEE 802.16 standards for the local and metropolitan area networks (LAN/MAN) respectively. Both of these standards define the medium access control layer (MAC) and physical layer (PHY) parts of a wireless user. In a wireless network, the MAC protocol plays a significant role in determining the performance of the whole network and individual users. Accordingly, many challenges are addressed by research to improve the performance of MAC operations in IEEE 802.11 and IEEE 802.16 standards. Such performance is measured using different metrics like the throughput, fairness, delay, utilization, and drop rate.

We propose new protocols and solutions to enhance the performance of an IEEE 802.11 WLAN (wireless LAN) network, and to enhance the utilization of an IEEE 802.16e WMAN (wireless MAN). First, we propose a new protocol called HDCF (High-performance Distributed Coordination Function), to address the problem of wasted time, or idle slots and collided frames, in contention resolution of the IEEE 802.11 DCF. Second, we propose a simple protocol that enhances the performance of DCF in the existence of the hidden terminal problem. Opposite to other approaches, the proposed protocol attempts to benefit from the hidden terminal problem. Third, we propose two variants of a simple though effective distributed scheme, called NZ-ACK (Non Zero-Acknowledgement), to address the effects of coexisting IEEE 802.11e EDCA and IEEE 802.11 DCF devices. Finally, we investigate encouraging ertPS (enhanced real time Polling Service) connections, in an IEEE 802.16e, network to benefit from contention, and we aim at improving the network performance without violating any delay requirements of voice applications.

CHAPTER 1. Introduction

During the last few years, wireless networking has attracted much of the research and industry interest. In addition, almost all current wireless devices are based on the IEEE 802.11 and IEEE 802.16 standards for the local and metropolitan area networks (LAN/MAN) respectively. Both of these standards define the medium access control layer (MAC) and physical layer (PHY) parts of a wireless user.

In a wireless network, the MAC protocol plays a significant role in determining the performance of the whole network and individual users. Accordingly, many challenges are addressed by research to improve the performance of MAC operations in IEEE 802.11 and IEEE 802.16 standards. Such performance is measured using different metrics like the throughput, fairness, delay, utilization, and drop rate.

1.1 IEEE 802.11

The 802.11 standard defines two modes of operation: DCF (Distributed Coordination Function), and PCF (Point Coordination Function). Alternatively, the new Hybrid Coordination Function (HCF) is introduced in the 802.11e standard to provide different mechanisms to meet the growing demand of users for real-time application. HCF includes two modes of operation: Enhanced Distributed Coordination Access (EDCA), and HCF Controlled Access (HCCA).

PCF and HCCA are centralized controlled access methods that exist at a coordinator node, the access point (AP). The AP uses polling to assign the right to access the channel following a predetermined schedule. Both operations have the drawbacks of requiring a coordinator node, and adding the overhead of polling messages that are usually transmitted using lower physical rates. On the other hand, DCF and EDCA are distributed contention-based access functions in which the right to access the wireless channel is determined by different local contention parameters used by every user. Extending DCF, EDCA introduces different QoS (quality of service) mechanisms like priority levels and transmission time bounds. Consequently, much attention is given to the distributed operations of IEEE 802.11 especially DCF which is the basic operation of the MAC protocol defined in all IEEE 802.11 standards including the IEEE 802.11e.

Using the IEEE 802.11 DCF, stations compete for the channel using a random backoff access scheme. Therefore, there is an overhead of idle slots and collisions which degrade the performance of DCF. Such degradation increases with higher loads and network sizes, and with the existence of hidden terminal problem. In addition, wireless networks are expected to have a mix of IEEE 802.11 and IEEE 802.11e standards. Hence, there has been an interest in the performance of such networks due to the deference between EDCA and legacy DCF.

We propose new protocols and solutions to enhance the performance of an IEEE 802.11 WLAN (wireless LAN) network. First, we propose a new protocol called HDCF (High-performance DCF), to address the problem of wasted time, or idle slots and collided frames, in contention resolution of DCF. Second, we propose a simple protocol that enhances the performance of DCF in the existence of the hidden terminal problem. Opposite to other approaches, the proposed protocol attempts to benefit from the hidden terminal problem. Finally, we propose two variants of a simple though effective distributed scheme, called NZ-ACK (Non Zero-Acknowledgement), to address the effects of coexisting IEEE 802.11e EDCA and IEEE 802.11 DCF devices.

We implemented all of these proposed protocols using Opnet Modeler by modifying the existing 802.11/802.11e models which represent the MAC and PHY layers. In addition, the wireless medium is presented via a number of pipeline stages. These stages allow for determining propagation delays, transmission ranges, out of range stations, and different properties of all transmitted and received signals (like SNR and power). We modified some of these stages to add the capture effect feature and the hidden terminal problem.

High-Performance Distributed Coordination Function (HDCF)

The performance of 802.11 DCF degrades especially under larger network sizes, and higher loads due to higher contention level resulting in more idle slots and higher collision rates. We propose HDCF to address the problem of wasted time in contention resolution of DCF via classifying stations into active and inactive ones. The objectives are to coordinate transmissions from different active stations with no collisions or idle slots, and limit the contention to newly transmitting stations. HDCF utilizes an interrupt scheme with active transmissions to enhance the fairness and eliminate, or reduce much of, the costs of contention in DCF (idle slots and collisions) without adding any assumptions or constraints to DCF.

We provide a simple analytical description of HDCF compared to DCF. We use a simple but a well-known and an accurate model of the IEEE DCF which is presented in (2), and we start with assumptions like that used in (2). We explain how new arrivals affect the probability of collision, and how the collision level is reduced. We also show that like DCF, HDCF operation consists of cycles such that each cycle includes on average a transmission by each user in the network. While DCF achieves this fairness property with the cost of idle slots and collisions, HDCF reduces much of such overheads, and thus is expected to enhance the throughput and fairness of the network.

In general, HDCF has the following advantages: 1) No idle slots wasted when there are no new stations trying to transmit, or no need to stop active transmissions. 2) Fairness to new stations as they can contend for the channel directly (like in DCF) without long delays as the contention cost is much smaller. 3) Stations transmit in random order without the need for a slotted channel, reserved periods, time synchronization, central control, or knowledge of number of active users.

Finally, we use Opnet to provide a simulation study for networks of two different PHYs (the IEEE 802.11b and 802.11g). In addition, the experiments consider different loads, network sizes (number of users in the network), noise levels, packet sizes, and traffic types. Results illustrate that HDCF outperforms DCF with gains up to 391.2% of throughput and 26.8% of fairness level.

Taking Advantage of the Existence of Hidden Nodes

When wireless users are out of range, they would not be able to hear frames transmitted by each other. This is referred to as the hidden terminal problem, and significantly degrades the performance of the IEEE 802.11 DCF because it results in higher collision rates.

Although the problem is addressed by different works, it is not totally eliminated. Hence, we propose a simple protocol that enhances the performance of DCF in the existence of hidden terminal problem. Opposite to other approaches, we propose to take advantage of the hidden terminal problem whenever possible. We investigate if non-hidden stations could help each other retransmit faster to enhance the performance of 802.11 WLANs. Such cooperative retransmissions are expected to be faster since with DCF a non-collided station mostly transmits earlier than collided stations that double their backoff values. The proposed scheme modifies 802.11 DCF, is backward compatible, and works over the 802.11 PHY. We also present an analysis model to calculate the saturation throughput of the new scheme and compare it to that of DCF.

Capture effect is an advancement in wireless networks that allows a station to correctly receive one of the collided frames under some conditions like a threshold of the received signal's SNR (signal-to-noise ratio). Thus, captures would enhance the throughout of the network while decreasing the fairness level. Consequently, we consider capture effect as it may reduce the gains of the proposed scheme, and would make it possible for the new scheme to be used even in a fully-connected WLAN where there are no hidden nodes.

Using Opnet simulation, we evaluate the new scheme with and without the capture effect for different topologies. Results show gains of the number of retransmissions per packet, throughput, fairness, delay, and packet drops. However, there is a small trade-off regarding fairness in some scenarios. Finally, simulation is used to validate the analytical model.

Non-Zero ACK (NZ-ACK)

The 802.11e standard is designed to be backward compatible with the 802.11. Thus wireless networks are expected to have mix of EDCA (802.11e) and legacy DCF (802.11, 802.11b, 802.11g, and 802.11a) users. As a result, EDCA users' performance may be degraded because of the existence of legacy users, and therefore would get a lower priority of service. The main reason for such influence is due to the fact that EDCA users are controlled through the use of different contention parameters, which are distributed by a central controller. Nevertheless, there is no control over legacy users because their contention parameters are PHY dependent, i.e. they have constant values.

We provide an insight on the effects of coexisting legacy DCF and EDCA devices, and present general desirable features for any proposed mitigating solution. Based on these features, we then propose a simple distributed scheme, called NZ-ACK (Non Zero-Acknowledgement), to alleviate the influence of legacy DCF on EDCA performance in networks consisting of both types of users.

NZ-ACK introduces a new ACK policy, and has the following features: 1) Simple and distributed. 2) Fully transparent to legacy DCF users, and thus backward compatibility is maintained. 3) No changes to the 802.11e standard frames formats. 4) Minimal overhead to EDCA users as all processing is at the QAP. 5) Adaptively provide control over legacy stations, and reserve more resources for the EDCA users as necessary.

Two variants of NZ-ACK are proposed. First, we use a simple intuition based on number of users of both types and expected traffic at EDCA users. This variant requires the AP to maintain virtual buffers for EDCA flows, and update these buffers depending on admission information. Second, we provide a model for solving the main challenges of NZ-ACK such that the priority of EDCA users is maintained. The model includes contention parameters, the number of users, and transmission activities of both types of users without the need for any virtual buffers or admission information.

Opnet simulation is used to evaluate both variants of NZ-ACK. Simulation results show that NZ-ACK maintains the priority of service and delay bounds of EDCA users while providing acceptable throughput for legacy users.

1.2 IEEE 802.16

The IEEE 802.16 provides a promising broadband wireless access technology. Using advanced communication technologies such as OFDM/OFDMA and MIMO, the IEEE 802.16 is capable of supporting higher transmission rates, provides strong QoS mechanisms, and extends the service ranges. Moreover, the IEEE 802.16 is evolving toward supporting mobility, and using relay devices. As a result, it it expected to replace or extend the already existing broadband communication, or DSL and cable.

IEEE 802.16 defines both the MAC (medium access control) and PHY (physical) layers of a broadband wireless network. The IEEE 802.16s MAC is a connection-oriented reservation scheme in which the subscriber stations (SSs) have to reserve any required bandwidth for transmissions. The BS (base station) coordinates reservations for all transmissions and receptions. A connection is used to uniquely identify a flow from, or to, a SS. Hence, the standard also specifies bandwidth request/allocation mechanisms for different traffic service types. Therefore, efficient bandwidth requests, bandwidth allocations, scheduling at both BS and SSs sides, QoS architectures, admission control, and traffics classifications are all essential for 802.16 networks.

The IEEE 802.16 introduced different QoS classes which characterize different QoS requirements including UGS (Unsolicited Grant Services), rtPS (real time Polling Services), nrtPS (non real time Polling Services), and BE (Best Effort). The IEEE 802.16e added the ertPS (enhanced real time Polling Service) class as an enhancement for UGS and rtPS. Hence, it is expected that different real-time applications will be using ertPS class. On the other hand, many applications are using BE and nrtPS connections. For ertPS, the BS allocates bandwidth based on the negotiated characteristics. However, when used for VBR (variable bit rate) applications, such allocation may not be fully used due to the variability of traffic at a SS side. Hence, the total efficiency or utilization of the network may be degraded. Therefor, we consider the performance of an IEEE 802.16 network with ertPS connections because it is critical for VoIP applications. Thus, our work focuses on ertPS for voice applications using the well-known ON-OFF model. Such model has proven to be practical and accurate. Our main objective is to improve the network performance without violating the delay requirements of voice applications.

Since the IEEE 802.16 allows ertPS to use both contention and unicast polling, we investigate encouraging ertPS connections to benefit from contention. Instead of always allocating bandwidth to ertPS connections, we propose an algorithm that adaptively uses a mix of contention and polling. The new algorithm adapts to different parameters like the number of SSs and delay requirements. However, as there is no differentiation between different classes in contention in the current standard, a problem occurs when ertPS connections compete with many low priority connections within a contention region. This would cause more collisions, idle slots, and delays to get the required bandwidth. To overcome this problem, we propose to implement a mechanism at the SS's UL scheduler of bandwidth requests to maintain the priority of the delay-sensitive ertPS connections in contention. While UGS connections are granted bandwidth without any request, rtPS connections are polled periodically to request bandwidth, and nrtPS connections are polled but less frequently than rtPS. On the other hand, BE connections will be using contention most of the time as they are provided with no guarantees. Hence, we consider the performance of ertPS and BE connections in an IEEE 802.16e network. Finally, we use Qualnet Modeler for the performance evaluation. Results show that the proposed scheme improves the jitter (with gains around 60%) measures and the throughput performance (about 2% to 155% of gain) without violating any latency requirements.

1.3 Organization

In the following chapters, we provide description of each of the proposed schemes. This includes the problem statement, background information, related work, and performance analysis. HDCF is presented in Chapter 2. Then, NZ-ACK is illustrated in Chapter 3, and its modification is presented in Chapter 5. In Chapter 4, we present the proposed protocol for taking advantage of hidden terminals. Then Chapter 6 includes the new scheme proposed for enhancing the bandwidth utilization in IEEE 802.16e. Finally, conclusion remarks and future directions are in Chapter 7.

CHAPTER 2. The Design and Analysis of a High-Performance Distributed Coordination Function for IEEE 802.11 Wireless Networks

Submitted to the IEEE/ACM Transactions on Networking (ToN)

Haithem Al-Mefleh [1,3], J. Morris Chang [2,3]

2.1 Abstract

IEEE 802.11 wireless local area networks (WLANs) are becoming more popular. The performance of 802.11 DCF (Distributed Coordination Function), that is the basic MAC scheme used in wireless devices, degrades especially under larger network sizes, and higher loads due to higher contention and so more idle slots and higher collision rates. In this chapter, we propose a new high-performance DCF (HDCF) scheme that achieves a higher and more stable performance while providing fair access among all users. In HDCF, the transmitting stations randomly select who is the next transmitter and so active stations do not have to contend for the channel, and an interrupt scheme is used by newly transmitting stations without contending with the existing active stations. As a result, HDCF achieves collision avoidance and fairness without idle slots added by the backoff algorithm used in DCF. For evaluation, we provide an analytical model to discuss different issues of HDCF. Also, we utilize Opnet Modeler to provide simulation that considers both saturated and non-saturated stations. The results show that HDCF outperforms DCF in terms of throughput, and long-term and short-term fairness. The simulations show gains up to 391.2% of normalized throughput and 26.8% of fairness index.

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[2] Associate Professor.
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2.2 Introduction

The IEEE 802.11 (3; 4; 13) standard is becoming the most popular medium access control (MAC) protocol used for wireless local area networks (WLANs). The standard defines two modes of operation: DCF (Distributed Coordination Function), and PCF (Point Coordination Function). While PCF is optional and cannot be used for Ad-Hoc networks, DCF is mandatory and is the only option for 802.11-based ad-hoc networks. Infrastructure WLAN benefits from PCF where no contention, and so no collision, is needed as the AP assigns the right to access the channel following a predetermined schedule. PCF provides a higher efficiency than that of DCF but it is not attractive since it is a centralized operation. Nevertheless, DCF is the basic operation for all the 802.11 standards including the 802.11e-2005 (13).

Other than being simple and distributed, DCF is most popular because it assures long-term fairness where each station has the same opportunity to access the channel. However, DCF performance is degraded by collisions and idle backoff slots. Moreover, collisions and idle slots increase as the number of contending stations increases. Using the same analysis found in (2), Fig. 2.1 shows the probability of collision as a function of the number of contending stations.

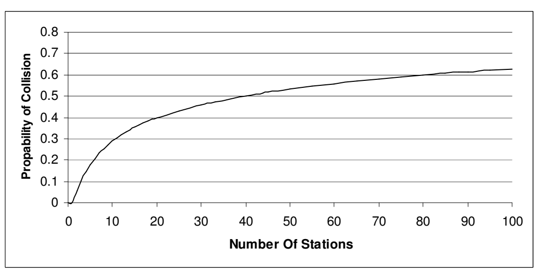


Figure 2.1 Probability of collision in DCF

In DCF, Binary-Exponential-Backoff (BEB) procedure is used to resolve collisions, and a uniform distribution is used to provide fairness property for all users. A station with a packet to transmit will do so if the medium is sensed idle for a period of DIFS. Otherwise, the station sets its backoff counter by randomly choosing a number following a uniform distribution: $NumberOfBackoffSlots \sim U(0, CW)$ where CW is called the contention window and is initially set to CW_{min}. The station decrements its backoff counter by one for every time slot the medium is sensed idle, and transmits when this counter reaches zero. The destination responds by sending an acknowledgment (ACK) back. The packets transmitted carry the time needed to complete the transmission of a packet and its acknowledgement. This time is used by all other stations to defer their access to the medium and is called Network Allocation Vector (NAV). Collisions occur when two or more stations are transmitting at the same time. With every collision, the station doubles its CW unless a maximum limit CW_{max} is reached, and selects a new backoff counter from the new range. The process is repeated until the packet is successfully transmitted or is dropped because a retry limit is reached. Unfortunately, such behavior degrades the performance of the network especially under higher loads due to collisions and idle slots. Even if the network has only one station transmitting, that station still has to backoff for a number of slots. In addition, collisions occur more frequently when the number of contending users increases. This results in an unstable behavior of DCF under very high loads.

In this chapter, we propose a new contention management scheme named High-performance DCF (HDCF), which addresses the problem of wasted time in contention resolution via classifying stations into active and inactive ones. Our objectives are to coordinate transmissions from different active stations with no collisions or idle slots, and limit the contention to newly transmitting stations. HDCF is a distributed random access scheme that achieves a higher throughput while providing long-term and short-term fairness among all users. In general, each station maintains a list of active users. The transmitting station chooses randomly the next station to transmit from its own list of active users following a uniform distribution: $NextStationToTransmit \sim U(first, last)$ where first and last are the first and last entries of the active list. The selected station transmits after a PIFS period following the last transmission, and other active stations will defer their attempts to transmit the same way NAV is used in DCF. Thus, there are no collisions or redundant idle slots due to active transmissions. On the other hand, a newly transmitting station uses an interrupt scheme. Thereafter, active stations stop their active transmissions and only new stations would contend for the channel using DCF. As a result, HDCF reduces the number of contending stations, and so collision rates, and backoff slots. Results show that HDCF outperforms DCF in terms of throughput, and fairness index with gains up to 391.2% and 26.8% respectively.

With HDCF, stations transmit in a uniform random order using a single channel with no central control, no time synchronization, no slotted channel, and no periods' reservations. In addition, HDCF utilizes an interrupt scheme so that active stations (one or more) keep transmitting unless there are new stations welling to transmit, and that those new stations (one or more) can contend directly to assure fairness preventing unbounded delays for new stations. Finally, HDCF works using the 802.11 PHY and MAC attributes (like NAV, retry limits, fragmentation, and others), introduces no additional packets, and works with or without RTS/CTS mode (e.g. used for hidden-terminal problem).

HDCF is presented in (59). In this chapter, we provide an analytical description of HDCF compared to DCF. We use a simple but a well-known and an accurate model of the IEEE DCF which is presented in (2). The analysis illustrates different issues of HDCF like fairness and how collisions are reduced. In addition, the simulation experiments consider more results of the IEEE 802.11b, VBR (variable rate traffic), and a mix of VBR and CBR (constant bit rate) traffics.

The rest of this chapter is organized as following. Related work is summarized in section 2.3. In section 2.4, HDCF protocol's details and rules are defined. In section 2.5, a simple analytical analysis is provided to discuss performance and design issues of HDCF. In addition, a simulation study is presented in section 2.6 to evaluate HDCF and compare it to DCF. Finally, concluding remarks are given in section 2.7.

2.3 Related Work

To enhance DCF, many researchers proposed schemes that mainly attempt to reduce collision rates, adapt CW to congestion levels, or find optimal values of CW. However, collisions and wasted times still exist because some approaches solve one problem and leave another (e.g., (5; 6; 7)), and optimal values are approximate and oscillate with the network conditions that are variable (e.g., (5; 6; 7; 8; 9)). In addition, some schemes require the existence of an access point (AP) or complex computations (e.g., (8; 9)). Instead of providing a history of all such proposals, we will give examples that fall into these categories.

SD (5) divides CW by a factor after a successful transmission to improve fairness. FCR (6) achieves a high throughput by having each station reset its CW to a minimal value after a successful transmission, and double the CW exponentially after a collision or losing contention. Thus, FCR requires the use of another mechanism to provide fairness. CONTI (7) attempts to fix the total number of backoff slots to a constant value. Hence, there are always idle slots and collisions may occur. In (8), the authors argued that the backoff value must be set equal to the number of stations to maximize the throughput. This algorithm requires an AP to broadcast the number of stations. Hybrid protocols (e.g. (9; 10)) divide the channel into consecutive reserved contention and contention-free periods. Such protocols require a central controller, reservation, multi-channels, the use of RTS/CTS, slotted channels, and/or time synchronization. Also, new stations first wait for the contention-free periods to end resulting in unbounded delays and unfairness especially when a new station waits more than one contention-free period. Therefore, most of these schemes limit the number of active users and lengths of different periods.

2.4 HDCF Details

HDCF utilizes an interrupt scheme and active transmissions to enhance fairness and eliminate, or reduce much of, the costs of contention of DCF (idle slots and collisions) without adding any assumptions or constraints to DCF. The following subsections describe how HDCF works.

2.4.1 Definitions

1. *Active Stations and Active-List:* active stations are those added to Active-List. Active-List contains a list of stations that have more packets to transmit, hence the name Active stations. Each station will maintain its own Active-List, and each entry of an Active-List

has the format $< ID >$ where ID is the MAC address of an Active station. Active lists may not be the same in all stations; active lists could be partial.

2. *Next-Station:* the station that is supposed to be the next transmitter and that is selected by the currently transmitting station.

3. *Active Transmissions:* an active transmission is started by Next-Station after a PIFS ($PIFS = SLOT + SIFS$) following the last transmission.

4. *Idle Stations:* stations that have no data to transmit.

5. *New Stations:* stations that were idle because they did not have data to transmit, and at current time are having data to transmit. This includes mobile stations that move into the network and have data to transmit, and stations that were turned off or in a sleep mode and are turning on. New stations are also referred to as new arrivals.

2.4.2 Next-Station Selection

The current transmitting station, the source, will randomly select an entry from its Active-List, and announce that ID as Next-Station. To provide fairness, a uniform distribution is used:

$$\text{Next-Station} = \text{Uniform}(A[0], A[Size - 1]) \tag{2.1}$$

where $A[0]$ is the first entry and $A[Size - 1]$ is the last entry of the station's Active-List. The announcing station does not have to be active. A transmitting station will make an announcement even if it will not become active. This eliminates the need for active stations to contend to get back into active mode.

Using the uniform distribution, an active station may choose itself as the next transmitter. This assures the property provided by DCF which states that each station has the same opportunity to access the channel. In addition, it prevents a station from wasting any idle slots, no need to go through the backoff stages, if there are no other active stations.

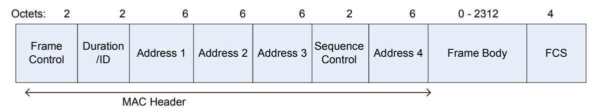


Figure 2.2 MAC frame format

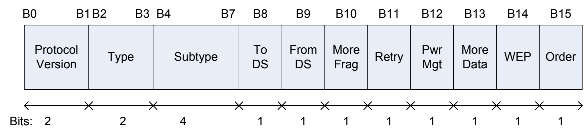


Figure 2.3 Frame Control field

2.4.3 Announcement

A station announces its future status by informing its neighbors, using broadcast nature of wireless medium, that it does have or does not have more packets to transmit. In addition, a station announces Next-Station; the next station that has the right to access the channel.

An announcement is performed by a station while it is transmitting. The advantage of this behavior is that there is no need for any special frames or messages to be exchanged. Whenever a station wins the right to access the channel, it will transmit a packet. The same packet can be used for announcement.

Using 802.11 packet formats, the "More Data" bit of the Frame Control field, Fig. 2.3, can be used for announcing that a station is active. The "More Data" bit can be used since it is used in PCF but not in DCF. Another bit called "More Fragments" is used when more fragments are to be transmitted with DCF. In addition, the header's "Address4" field of the data frame, Fig. 2.2, can be used to announce Next-Station. This means an overhead of 6 bytes, the size of the MAC address which is small compared to the average packet size.

When a station receives, or overhears, a packet with the "More Data" bit set to "1", it adds an entry to its Active-List unless that entry already exists. The entry will be $< ID >$, where ID is the MAC address of the transmitting node. On the other hand, if the "More Data" bit is set to "0" then the entry, if exists, that has the MAC address of the transmitting node will be removed from all overhearing stations' Active-Lists. Note that for a station to be removed from all Active-Lists, it needs to announce it only once; during the transmission of its last packet.

2.4.4 HDCF Rules

When a station wins the right to transmit, it will also announce Next-Station that is selected randomly from its own Active-List. As shown in Fig. 2.4, active stations use PIFS as an inter frame spacing (IFS): Next-Station starts transmitting PIFS after the end of the last active station's transmission. For the IFS between packets of the same transmission, SIFS is used as in DCF. In addition, DCF NAV is still used, so stations will defer to the end of the ongoing transmission.

A new station initially assumes DCF; it transmits if the channel is idle for a period of DIFS followed by backoff slots determined by Binary Exponential Backoff as shown in Fig. 2.5. If there are active stations, then a new station will detect at least one active transmission since PIFS is used as the IFS between any two consecutive active transmissions and PIFS is shorter than DIFS. Therefore, following DCF rules would block a new station. There are two options to allow a new station's transmission:

1. Force active stations to switch back to DCF, or a silent mode, every while: every specific limit of active transmissions or every time limit. Active stations need to wait only long enough to check if there is any new station trying to transmit using DCF. A problem with this approach is the overhead of time wasted when no new stations are arriving. In addition, if there are more than two new stations, some of them may have to wait long time before being able to start transmitting. This results in unfairness and unbounded delays for new stations.

2. Allow a new station to interrupt active stations before the end of PIFS when they detect

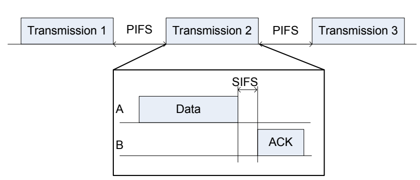


Figure 2.4 Active transmissions and their components

active transmissions. This is similar to the behavior of DCF since it allows new stations to contend for the channel as soon as the ongoing transmission ends. Therefore, we propose to use an interrupt scheme, Fig. 2.6, by which a new station uses a jam signal (the jam signal is a special signal used by many wireless MAC protocols, for instance, different jam periods are used by Black Bursts (like (11; 63)) to provide levels of priority) to stop active transmissions. If there is more than one new station interrupting, they will collide resulting in longer time spent contending for the channel. Hence, a new station starts transmitting after the jam only if the medium is idle for a period of one slot followed by backoff slots. The backoff procedure will follow the Binary Exponential Backoff procedure.

When active stations including the Next-Station detect a busy medium before the end of PIFS, as described in Fig. 2.6, then there is at least one new station trying to transmit. Therefore, all active stations switch back to DCF to give new stations the chance to transmit. To prevent long delays and for practical issues, active stations follow DCF after the jam signal but with EIFS ($EIFS = DIFS + SIFS + T_{ACK}$, with ACK sent using lowest PHY rate) instead of DIFS. EIFS is used only one time after the jam signal. This also provides much higher priority for new stations that use one slot after the jam. Active transmissions are reactivated by the interrupting station since it knows about at least one active station; the last announced Next-Station.

In the following we consider different optimizations of HDCF. This includes dropping stations from active lists, and scenarios of mobility and hidden nodes.

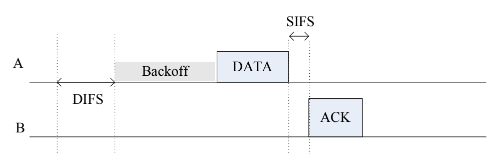


Figure 2.5 Basic operation of DCF

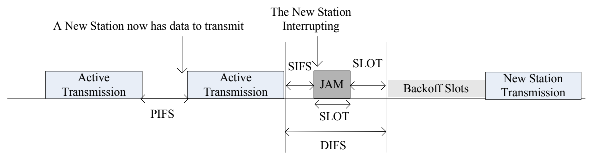


Figure 2.6 The interrupt scheme

2.4.4.1 Mobility

When considering mobility, HDCF may be optimized by allowing the drop of a station from the local active list if it performs handover (or operations like disassociation), or if it does not start transmitting for a number of times. A conservative value would be 1. However, a station may not start an active transmission due to other reasons like those mentioned in subsection 2.4.6. Hence, we suggest the use of a higher value which also should not be very large - like 3.

2.4.4.2 Hidden Nodes

Like DCF, RTS-CTS operation should be used for the hidden terminal problem. Then when a collision occurs due to the hidden terminal problem, all stations would switch back to DCF and the collided transmitters would start backoff procedure. However, to enhance the performance of HDCF when hidden nodes exist, we propose that the receiver rebroadcast the next station address in the ACK frame. Accordingly, all stations within the ranges of the receiver and the transmitter are aware of the address of the next station. Thereafter, a station would defer accessing the channel if no activity is sensed for a period of $RTS{+}SIFS$ when the next station address is not within the active list. This would protect the transmission of a hidden active station preventing a collision when the receiver is not hidden, i.e. waiting long enough to hear the CTS frame.

Another improvement is to use ACK frames to rebroadcast the future status announcement, i.e. having more data or not, of the transmitter. Thus, a node that is hidden (to the transmitter) may add the transmitter to the active list. The address of the transmitter is already included in the ACK frame. On the other hand, one control bit, in the ACK frame, can be used to announce the future status, and is simply copied from the status announced in the data frame. In addition, the ACK can be modified to re-announce the next station address. Thus when selected by a hidden transmitter, a station can be selected to be the transmitter.

Finally, HDCF stations may adapt their transmissions according to network and channel characteristics using different techniques used for the 802.11 DCF like the use of RTS/CTS operation for hidden nodes, RTS threshold, and fragmentation.

2.4.5 An Example

Fig. 2.7 is a simple example that illustrates HDCF operation.

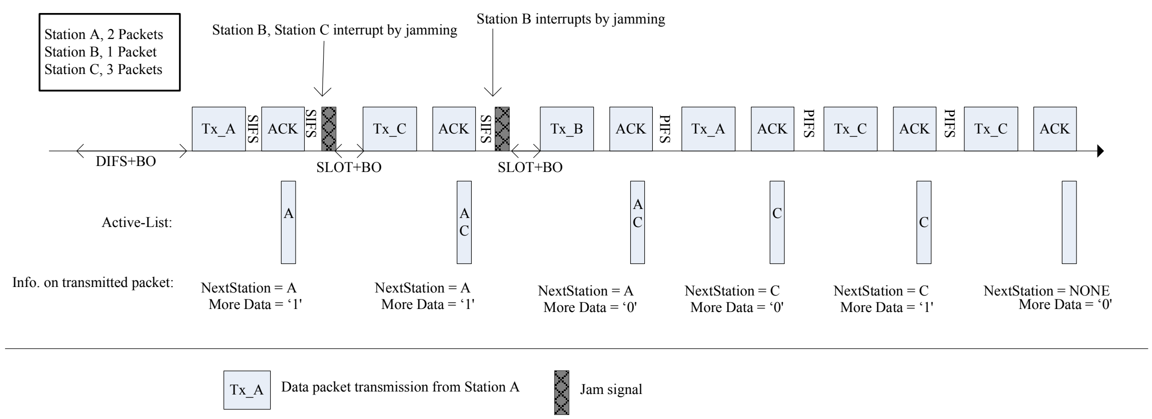


Figure 2.7 An example with three stations joining network at different times

In the example, there are three stations that have data to transmit: A with 2 packets, B with 1 packet, and C with 3 packets. Initially, all three stations will contend for the channel using DCF since they do not overhear any active transmission. Assuming that A wins the contention, A transmits one packet, and adds itself to its Active-List since it has another packet to transmit. The packet transmitted by A will inform all neighbors that A has more packets to transmit. In addition, A announces that the next transmitter is A. The reason that A chooses itself is the uniform distribution; there is one entry in the list and so that entry will be selected with a probability of 1. Stations B and C overhears that announcement, and hence each one adds A to its own Active-List.

Stations B and C jam for one slot SIFS after the end of the transmission of A. After jamming, both stations attempt to transmit after waiting for one slot followed by a random number of backoff slots. Assume that C wins the contention. C adds itself to its Active-List, and transmits while announcing A as the next transmitter, assuming that A was selected. The active list is updated at each of the three stations to include both: A, and C. Station B jams the channel for one slot SIFS following the transmission of C, then it transmits after a period of one slot and a number of backoff slots. Note that station B announces that it has no more data to transmit. Now the active list at each station includes stations A and C. Moreover, B announces A as the next transmitter, and so A transmits PIFS after transmission of B as no more stations are interrupting.

Node A transmits while announcing that it has no more data to transmit and that C is the next transmitter. As a result, Active-Lists at all stations are updated to include only C. Station C transmits PIFS after the transmission of A while announcing itself as the next transmitter, and that it has more data to transmit. Station C transmits its last packet without any interrupt announcing no more data to transmit. All three stations update there Active-lists that become empty.

2.4.6 Recovery Mechanism

Because of hidden terminal problem, channel errors, mobility, and the sudden shut down (turning power off) of any station, it is possible that the next selected station would not be able to start its transmission. In such case, all other active stations would notice the absence of Next-Stations's transmission just after a PIFS period by SIFS. Therefore, active stations are required to temporarily contend for the channel using DCF as a recovery mechanism, see figure 2.8 for the timings of switching to DCF (note there is no overhead to original timing in DCF since $DIFS = PIFS + SIFS$). Once active transmissions are recovered, active stations will switch back to the active state.

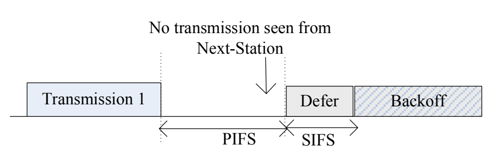


Figure 2.8 Recovery from a lost announcement

Moreover, different scenarios may arise because of wireless channel conditions. One scenario may occur when an active station, other than the next selected one, can overhear but cannot decode a packet that carries an announcement. This active station should temporaliy switch back to DCF operations. As a result, the announced next station would start transmitting with no problems, if it does hear the announcement. Another situation is when an active transmitter does not receive an ACK. This would be seen as a collision by this transmitter. Moreover, a station may handover and later is selected as the next transmitter. Hence, that station may not be able to start an active transmission. In summary, recovery is achieved by having active stations switch back to DCF operations and active stations will re-follow HDCF rules as soon as Next-Station is announced.

2.4.7 Summary, and Advantages

An HDCF station operates in one of two modes: active mode, and contending mode. In active mode, there are no backoff, no collisions, and no idle slots. On the other hand, contending mode uses legend DCF but with much lower collision rate because almost only new stations contend for the channel. The way Next-Station is selected, and the interrupt scheme have different advantages: 1) No idle slots wasted when there are no new stations; i.e. no need to stop active transmissions. 2) Fairness to new stations as they can contend for the channel directly (like in DCF) without long delays as contention cost is much smaller. 3) Stations transmit in random order without the need for slotted channel, reserved periods, time synchronization, central control, or knowledge of number of active users.

Finally, Just like 802.11 DCF, HDCF stations may adapt their transmissions according to network and channel characteristics using different techniques used for the 802.11 like RTS threshold, fragmentation, link adaptation, and the use of RTS/CTS for hidden nodes.

2.5 Performance Analysis

In this section, excluding subsection 2.5.5, the same assumptions and the analysis model described in (2) are used for simplicity in analysis and discussion. There are n stations with each station always has a packet to transmit. In addition, all stations can overhear each other transmission, i.e., there are no hidden terminals. A DCF network of greedy stations is modeled using a nonlinear system of equations that can be solved by means of numerical techniques. To summarize the analysis model, let τ be the probability that a station transmits in any slot time. The value τ can be found by solving the following non-linear system:

$$\tau = \frac{2(1-2\rho)}{(1-2\rho)(W+1)+W\rho(1-(2\rho)^m)} \tag{2.2}$$

$$\rho = 1-(1-\tau)^{n-1} \tag{2.3}$$

Where ρ is the probability that the transmitted packet will collide. W is equivalent to CW_{min}, and m is the maximum backoff stage where $CW_{max} = 2^m CW_{min}$. Moreover, let P_{tr} be the probability of transmission, P_s the probability of a successful transmission, and P_c the probability of collision.

$$P_{tr} = 1-(1-\tau)^n \tag{2.4}$$

$$P_s = \frac{n\tau(1-\tau)^{n-1}}{1-(1-\tau)^n} \tag{2.5}$$

$$P_c = 1-P_s \tag{2.6}$$

$$P_{idle} = 1-P_{tr} \tag{2.7}$$

Then, the expected number of idle slots can be calculated by:

$$E[\text{Number of idle slots}] = \frac{1}{P_{tr}} - 1 \tag{2.8}$$

The following subsections discuss special performance issues in HDCF compared to other schemes.

2.5.1 How to Handle New Arrivals

For DCF, if x more stations become ready to transmit, then there is a need only to replace n by $n+x$. The reason is that, under DCF all stations are contending for the channel with equal opportunities. For HDCF, the situation is different since only the new stations will contend for the channel. Starting with n active stations, the transmission probability is 1 and collision

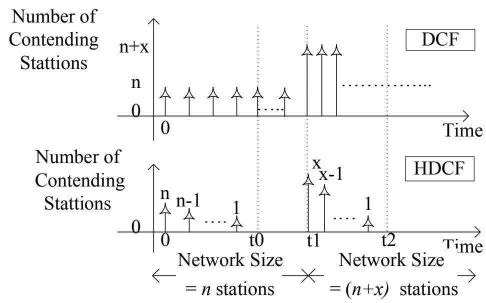


Figure 2.9 New arrivals effect on contention level

probability is 0. If x new stations become ready to transmit, then equations (2.2) and (2.3) can be used by replacing n with x. Only the new x stations will be contending for the channel. Once a new station becomes active, the contention is reduced to be among $x-1$ stations. This is repeated until all x stations become active, and therefore, the collision goes back to zero. In other words, stations go back into active mode after being in contending mode.

Fig. 2.9 explains this behavior for both DCF and HDCF. The network size from 0 to $t1$ is n, and $n+x$ after $t1$. For DCF, all existing stations contend for the channel. On the other hand, the number of contending stations is variable for HDCF case. At $t=0$, n stations start contending for the channel. Once a station becomes active, it will not contend for the channel. Hence, number of contending stations drops by one after every successful transmission until all nodes become active at $t0$. There is no contention from $t0$ to $t1$ since all n stations are active. However, x new arrivals occur at $t1$ and all of them will become active. Hence, contention from $t1$ to $t2$ is only among the new x arrivals. Once a station successfully transmits and joins the active list, it will no longer contend for the channel using DCF rules. This is repeated until all x stations become active, and therefore, no more contention occurs. The next winner of the channel will be determined by the transmitting station using the uniform random distribution described in section 2.4. Fig. 2.10 shows a comparison between the probability of collision of DCF and HDCF. The x-axis in the figure shows steps of collision resolution, and the y-axis is the probability of collision. The comparison is made for a system that starts with 5 active stations. After some time, 10 new stations are added to system. Under DCF, the probability of collision increases to that of 15 contending stations. The probability of collision stays at

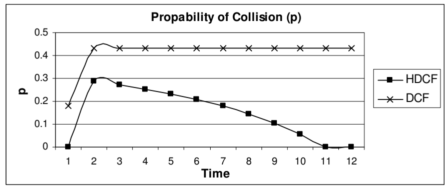


Figure 2.10 Probability of collision comparison

that level. On the other hand, HDCF starts with a probably of collision of 10 stations. After that, the probability drops to that of 9 stations. The process is repeated until all stations are active, and the probability of collision becomes 0.

To understand the importance of reducing collision probability, the expected number of backoff slots a station will experience per packet can be expressed:

$$\begin{aligned} W_{backoff} &= (1-\rho)\,\frac{2^0 W}{2} + \rho\,(1-\rho)\,\frac{2^1 W}{2} \\ &\quad +........ + \rho^m(1-\rho)\frac{2^m W}{2} \\ &= (\frac{1-\rho-\rho(2\rho)^m}{1-2\rho})(\frac{W}{2}) \end{aligned} \tag{2.9}$$

Again, ρ is the probability that a packet transmitted will collide, W is CW_{min}, and m is the maximum backoff stage. Hence, the number of backoff slots increases when more stations are contending for the channel. HDCF mitigates the problem during contention by reducing number of contending stations linearly with every successful transmission. The result would be increasing the throughput and reducing the delay seen by different stations.

2.5.2 How Contentions Are Resolved

Compared to DCF, HDCF is designed to achieve a higher performance while maintaining an important feature provided by DCF. In DCF, every station has equal opportunity to access the channel. This results in throughput-based fairness property. DCF operation consists of cycles such that on average, each cycle includes a transmission by each user in the network. Using the results of (2), it can be proved that such a cycle is reached by stations. Let n be the number of active stations in the network and υ be the probability that a given station transmits successfully for the next slot following DCF rules. Also, let X be the number of stations transmitting between two consecutive transmissions of a given station. The random variable X follows a geometric distribution:

$$\upsilon = \frac{\tau(1-\tau)^{n-1}}{n\tau(1-\tau)^{n-1}} = \frac{1}{n} \tag{2.10}$$

$$P[X=k] = \upsilon(1-\upsilon)^k = \frac{1}{n}(1-\frac{1}{n})^k \tag{2.11}$$

$$E[X] = \frac{1}{\upsilon} - 1 = n-1 \tag{2.12}$$

The $E[X]$ value implies that on average each station transmits once in a cycle consisting of transmissions from all stattions. On the other hand, using HDCF, it can also be proved that a cycle exits such that every station takes turn to transmit. Now, let υ be the probability that a given station transmits following HDCF rules. The random variable X also follows a geometric distribution:

$$\upsilon = \frac{1}{n} \tag{2.13}$$

$$P[X=k] = \upsilon(1-\upsilon)^k = \frac{1}{n}(1-\frac{1}{n})^k \tag{2.14}$$

$$E[X] = \frac{1}{\frac{1}{n}} - 1 = n-1 \tag{2.15}$$

The difference between HDCF and DCF is that DCF achieves this property with the cost of idle slots and collisions. On the other hand, HDCF is free of such overheads, and thus is expected to enhance the fairness property (verified by simulation results, section 2.6). In DCF, equation (2.10) is equivalent to the probability that a station successfully transmits while all others do not, and equation (2.11) accounts for a variable number of collided and idle slots before such a transmission occurs. However, equation (2.13) of HDCF is equivalent to the probability that a station successfully transmits after being selected using a random uniform distribution with n distinct outputs, and equation (2.14) accounts for a variable number of successful transmissions before such a transmission occurs.

Throughput-based fairness is proper for a single-rate network. However, in a multi-rate wireless network where users have different rates, throughput-based fairness degrades the overall network performance and the higher rate stations performance. The reason is that stations with slower rates occupy the channel for longer times. In such an environment, time-based fairness is desired. Time-based fairness allocates same amount of resources, time, to all users regardless of their data rates. The same techniques used in a DCF network to achieve time-based fairness can also be used for HDCF. For example, in OAR mechanism (62), a station may transmit a number of packets in proportion to its data rate once it wins the contention.

2.5.3 Maximum Achieved Throughput

The maximum saturation throughput of a DCF network can be approximated by:

$$S_{DCF} = \frac{E[L]}{DIFS + SIFS + \frac{CW_{min}}{2}\sigma + T_{ack} + T_{data}} \tag{2.16}$$

Here, σ is one slot time, T_{data} is the time needed to send one data packet, T_{ack} is the time needed to send an ACK, and $E[L]$ is the average packet size.

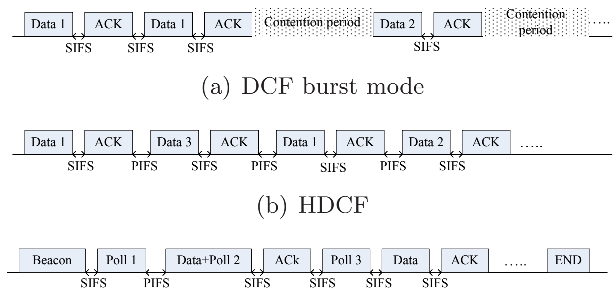
On the other hand, we can approximate the maximum saturation throughput that can be achieved using HDCF:

$$S_{HDCF} = \frac{E[L]}{PIFS + SIFS + T_{data} + T_{ack}} \tag{2.17}$$

Note that the time needed by stations to join the active mode is ignored. Using these formulas, one can expect a high gain by using HDCF. The simulation results, section 2.6, show that HDCF outperforms DCF and provides an efficient performance.

2.5.4 Packets Transmission Differences

This section explains the differences in how packets are transmitted in different schemes compared to HDCF. Fig. 2.11(a) explains the operation of DCF with burst mode. A station is allowed to transmit more than one packet after winning a contention using DCF rules. The contention period includes DIFS and backoff timer. Fig. 2.11(c) shows the operation of PCF, a polling-based scheme which requires the existence of a PC (Point Coordinator) which usually is at the AP. The PC assigns the right of accessing the channel to different stations by the use of polling messages. In general, PCF is not an attractive method because it is centralized and it introduces the overhead of polling. Refer to (3) and (64) for more information about polling schemes. Finally, Fig. 2.11(b) shows packets transmission in HDCF. Notice that it is fairer compared to other schemes in Fig. 2.11, and at same time has no collisions or idle slots when stations are all active.



(a) DCF burst mode

(b) HDCF

(c) PCF

Figure 2.11 Different modes of transmissions

2.5.5 Approximate Analysis

Now we provide an approximate analysis for the system at a given state. Assume a Poisson arrival process with λ packets per second at each station. Consider the system's state where there are m active stations, and $n-m$ stations are not active. Let γ be the probability that a station has a packet at the end of the last active transmission, and X be a random variable that represents the number of stations that would jam after the last active transmission. The probability γ is equivalent to the probability that at a given station, there was at least one arrival during the service time of an active transmission T_a $(PIFS + SIFS + T_{data} + T_{ack})$. If N is a random variable representing the number of packets at a station after T_a, then:

$$\begin{aligned} \gamma &= P[N(T_a) \geq 1] \\ &= 1 - P[N(T_a) = 0] \\ &= 1 - e^{-\lambda T_a} \end{aligned} \tag{2.18}$$

$$P[X = x] = \binom{n-m}{x} \gamma^x (1-\gamma)^{n-m-x} \tag{2.19}$$

$$E[X] = (n-m)\gamma \tag{2.20}$$

Hence, the throughput can be approximated by:

$$S_{HDCF|m} = \frac{(1 + E[X])L}{T_a + \frac{1}{\mu_{E[X]}}} \tag{2.21}$$

Here, $\mu_{E[X]}$ is the service rate of DCF given that there are $E[X]$ stations with data to transmit, and L is the packet size. Equation (2.21) converges to (2.17) when there are no interrupting stations, and to (2.16) when some or all stations always contend for the channel but with one packet per station at each time.

2.6 Simulation

This section presents the simulation we used to evaluate the performance of HDCF and compare it to that of the IEEE 802.11 DCF. We utilized the commercial Opnet Modeler 11.5.A, (1), to implement HDCF by modifying the 802.11 model.

The simulations are performed for networks using two different PHYs, 802.11b and 802.11g. Table 2.1 shows the parameters used by the Opnet 802.11 model. We consider different scenarios to study the performance of HDCF and compare it to that of DCF. First, we assume a fully-connected network with no channel errors; collisions are the only source of errors. Here, we start with a saturated scenario to provide a reference and an understanding of the maximum achievable performance. Second, we study the performance under different loads, or a non-saturated network. We also consider CBR (Constant Bit Rate) and VBR (Variable Bit Rate) traffic sources. Finally, we study the performance under different noise levels (channel conditions). To implement noise, we used a jammer node provided by Opnet.

2.6.1 Performance Metrics

For performance measurements, we use the following metrics:

1) Throughput: the total data transmitted per the simulation period. The simulation considered the throughput versus different sizes of packets, different number of stations, and different offered loads. We used normalized throughput defined as $Throuput/Rate_{MAC}$.

2) Fairness Index: we used Jain Index, (12; 65), defined by (2.22):

$$JF = \frac{(\sum_{i=1}^{n} S_i)^2}{n \sum_{i=1}^{n} S_i^2} \tag{2.22}$$

Where n is number of stations and S_i is the throughput of station i. The closer the value of FI to 1, the better the fairness provided. We provide results for different simulation periods

Table 2.1 Network Parameters

Parameter	**802.11g**	**802.11b**	**Parameter**	**802.11g**	**802.11b**
$aSlotTime$	$20\mu s$	$20\mu s$	$SIFS$	$10\mu s$	$10\mu s$
$PIFS$	$30\mu s$	$30\mu s$	$DIFS$	$50\mu s$	$50\mu s$
CW_{min}	15	31	CW_{max}	1023	1023
$PLCP\ Overhead$	$20\mu s$	$192\mu s$	$DCF\ Overhead$	$28\ Bytes$	$28\ Bytes$
$MAC\ ACK\ Size$	$14\ Bytes$	$14\ Bytes$	$HDCF\ Overhead$	$34\ Bytes$	$34\ Bytes$
$Data\ Rate$	$54Mbps$	$11Mbps$	$ControlRate$	$24Mbps$	$1Mbps$

to have better conclusions about both long and short term fairnesses.

2.6.2 Saturated Stations

This section provides a comparison between DCF and HDCF for a network of fully connected and saturated stations, i.e stations that have data packets to transmit at all times. Fig. 2.12 compares the normalized throughput between HDCF and DCF for 50 contending stations as a function of the packet size, which changes from 50 bytes to 2304 bytes. For HDCF, the normalized throughput reaches values of about 72.7% for 802.11b and 74.4% for 802.11g networks. For DCF, the values are about 48% and 33.8% for 802.11b and 802.11g networks respectively. The gain goes from 45.7% to 64% for 802.11b, and from 119.8% to 282.5% for 802.11g. One can see that the normalized throughput increases with the packet size for both protocols. However, the gain increases as the packet size gets smaller. This is due to the fact that collision's cost is higher as it takes longer time for larger packets. In addition, the figure shows the maximum normalized throughput values estimated by equations (2.17) for HDCF and (2.16) for DCF. While HDCF almost achieves the maximum performance, DCF performance is always lower than the maximum possible values.

Fig. 2.13 compares the normalized throughput between HDCF and DCF as a function of the network size, number of stations. Here, the packet size is fixed at 1000 bytes. Fig. 2.13 shows a higher stability performance of HDCF; the number of stations has small effect on the performance of HDCF. On the other hand, DCF performance degrades as the number of

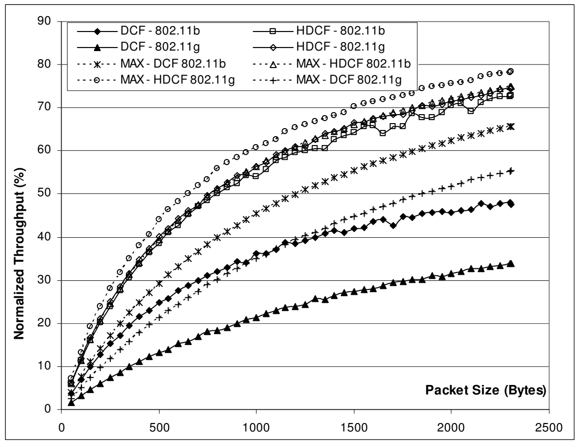


Figure 2.12 Throughput vs. packet size

stations gets larger. The main reason is that, the probability of collisions in DCF increases exponentially when number of stations increases. Moreover, all stations contend for the channel all times. However, HDCF reduces the number of contending stations linearly, and remove any unnecessary idle slots. Hence, collision probability and overheads are much reduced by HDCF allowing a higher stability.

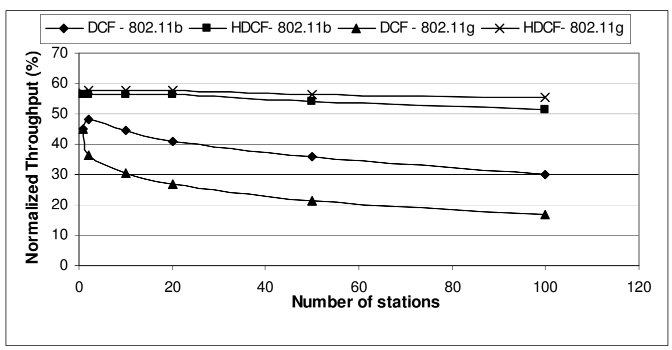


Figure 2.13 Throughput vs. network size

In Fig 2.14, we summarize results from different simulations we conducted. The figure shows the minimum and maximum gains of normalized throughput for different network sizes. Again, packet size is changed from 50 to 2304 bytes for each simulation. The least gain is 10.5%, and the greatest is 391.2%. In all cases, the gain increases when the number of stations increases.

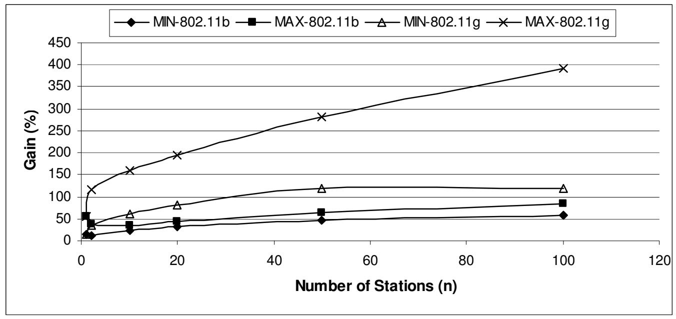


Figure 2.14 Minimum and maximum gain of throughput

Figures 2.12, 2.13, and 2.14 show that the higher the rate, the lower the throughput of a DCF network. In general, this is due to the augment of the overhead ratio. On the contrary, HDCF has a slightly higher performance when the rate increases because of the reduction of contention level and unnecessary idle slots.

Figures 2.15, and 2.16 illustrate that HDCF provides a higher short-term and long-term fairness among all stations. For these simulations, we used an average packet size of 1000 bytes. For the 1 second simulation, the fairness index of HDCF is always above 0.84. For the 3 seconds simulation, the fairness index is almost 1 for all sizes from 1 to 100 stations. On the other hand, the fairness index in DCF continues to decrease as the number of stations increases for both scenarios. The index reaches values of 0.49 and 0.74, respectively. For 802.11b, the gains are up to about 31.1% for long-term fairness, and 86.7% gain for short-term fairness. Correspondingly, the gains are up to 10.1% and 26.8% for 802.11g. The smaller gains in 802.11g are simply because of the lesser time required to transmit a packet using the higher rate.

2.6.3 Non-saturated stations, CBR Traffic

In this section, we simulated a network of non-saturate stations to study the performance of HDCF under different offered loads. We fix number of stations in the network to 50, set packet size to 1000 bytes, and vary the offered load at every station from 1 to 400 packets per second. Each simulation is run for a 50 second period. Figures 2.17 and 2.18 illustrate that while providing better fairness levels, HDCF could achieve up to 56.6% for 802.11b and

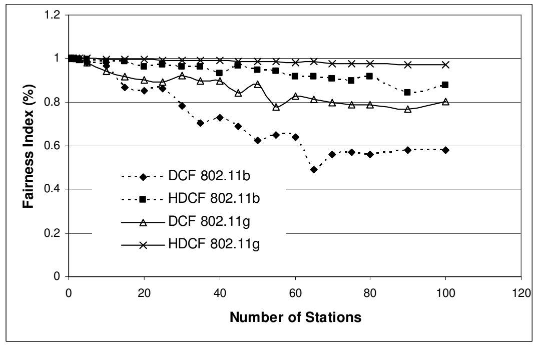


Figure 2.15   Fairness Index for 1 second period

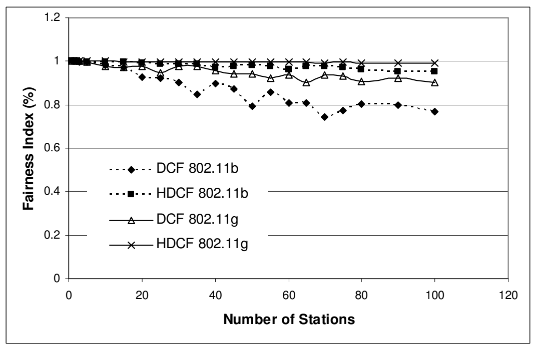


Figure 2.16   Fairness Index for 3 seconds period

168.3% for 802.11g more throughput than that of DCF.

From the previous experiments, we select a load rate of 20 packets per second and vary network size from 1 to 100 stations. Figures 2.19 and 2.20 show that HDCF concurrently improves fairness and provides throughput gains as high as 81.8% and 71.7% for the 802.11b and 802.11g networks respectively.

Figures 2.17 and 2.18, 2.19, and 2.20 also explain that once the network load, or the network size, gets beyond a certain level, DCF performance starts to degrade due to higher contention levels. Alternatively, HDCF enhances the performance because it reduces number of contending stations and unnecessary idle slots. In addition, HDCF performs just like DCF when the network load is light or when the network size is small. In such cases, DCF is proved to be highly efficient. Consequently, HDCF achieves higher performance and adapts better to different loads and network sizes.

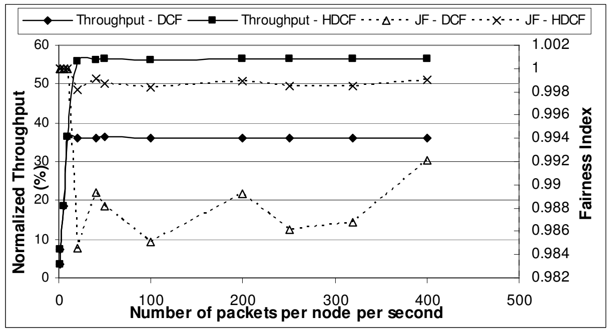


Figure 2.17 Throughput vs. load, 802.11b

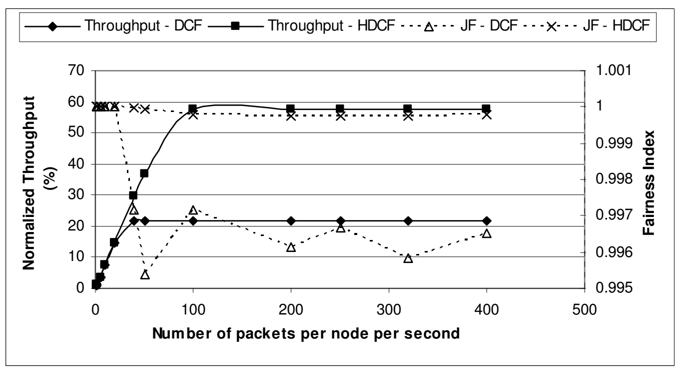


Figure 2.18 Throughput vs. load, 802.11g

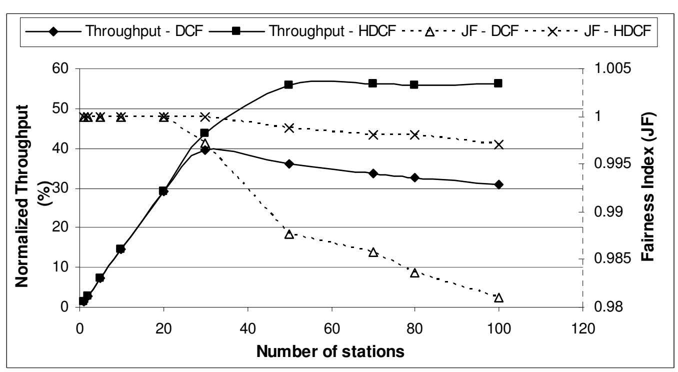


Figure 2.19 Throughput vs. number of users, 802.11b

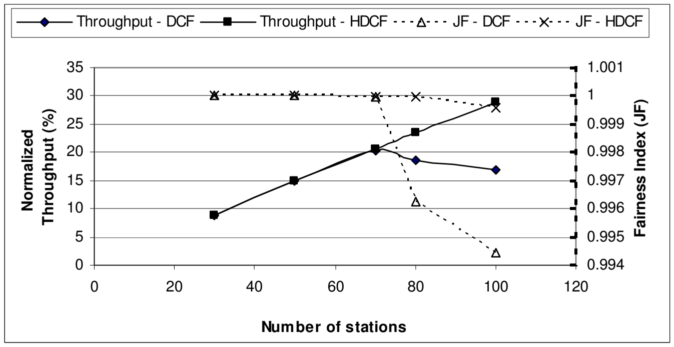


Figure 2.20 Throughput vs. number of users, 802.11g

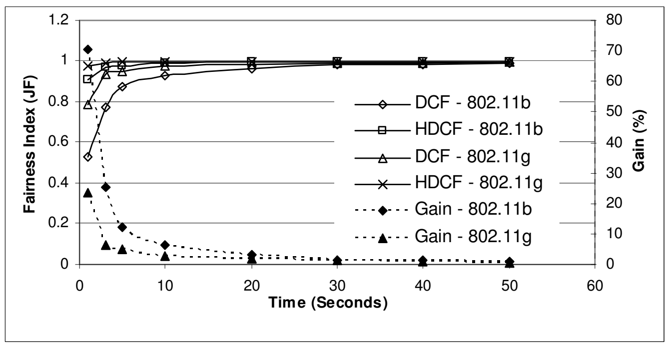


Figure 2.21 Fairness Index vs. time

Finally, we provide results for different simulation's periods to have better conclusions about short-term fairness. Fig. 2.21 presents fairness index vs. duration for a network of 80 stations, and a constant packet interarrival time of 0.01 second at each station. The gains can reach values up to 70.6% and 23.6% for 802.11b and 802.11g respectively. Since the gain increases for shorter periods, HDCF improves the short-term fairness of the network. Such enhancement is related to throughput's improvements as explained above, and how the next station is selected (see section 2.4.2).

2.6.4 Non-saturated stations, VBR Traffic

Here, we consider a network of 50 users, and 500 bytes per packet. Simulation period is 50 seconds. Instead of using CBR traffic, packets are generated at each user following the distribution $Exponential(\lambda)$, where λ is the mean interrival time (in seconds). Since packets' generation does not follow a CBR distribution, different users may have different loads. Hence, we redefine JF:

$$JF = \frac{(\sum_{i=1}^{n} \frac{S_i}{L_i})^2}{n \sum_{i=1}^{n} (\frac{S_i}{L_i})^2} \tag{2.23}$$

where L_i is the total normalized load of user i.

Figures 2.22 and 2.23 show the throughput and fairness index as a function of λ. It is clear that HDCF provides always the same or better fairness levels. In addition, HDCF outperforms DCF when considering the total throughput with gains up to 58.7% (169.7%) for 802.11b (802.11g). The figure explains that beyond some threshold of normalized load (about 22.7% (9.2%) at $\lambda = 80m$ ($40m$) second for 802.11b (802.11g)), HDCF adapts better to different loads at different users and therefore enhances the performance of the network.

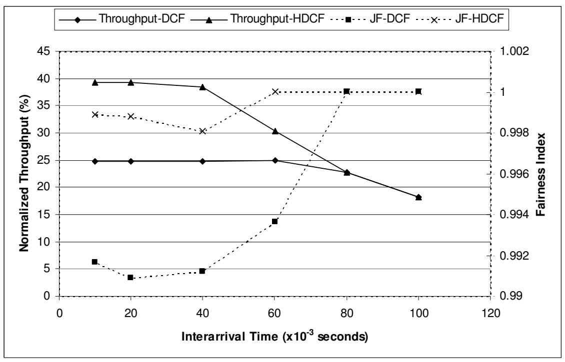


Figure 2.22 Network with VBR traffic, 802.11b

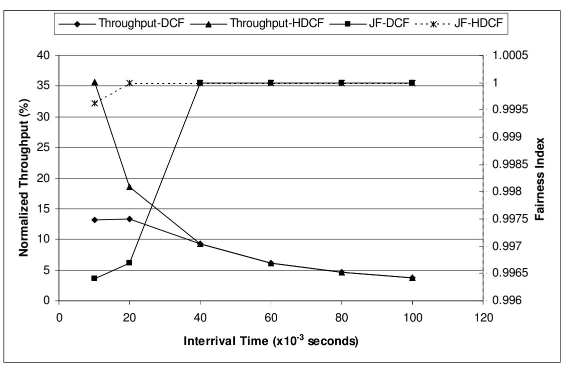


Figure 2.23 Network with VBR traffic, 802.11g

Finally, we modify network configuration so that 10 users generate CBR traffic with 40m seconds interrival time, and 40 users generate VBR traffic as described above. Figures 2.24 and 2.25 show that while providing slightly a higher fairness level, HDCF achieves higher throughput with gains up to about 57.5% (130.3%) for 802.11b (802.11g). Again, HDCF adapts better to different users' loads.

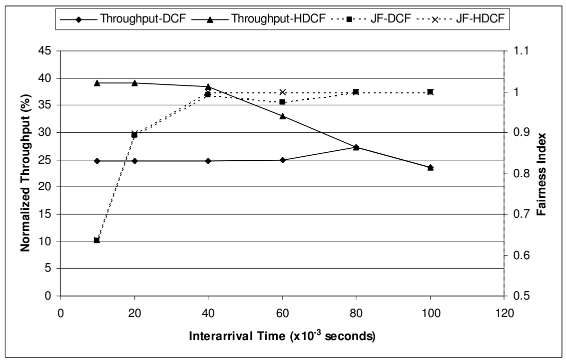


Figure 2.24 Network with both CBR and VBR traffic, 802.11b

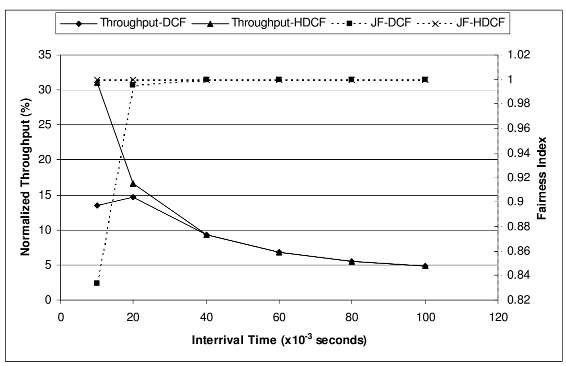


Figure 2.25 Network with both CBR and VBR traffic, 802.11g

As illustrated in CBR scenarios, HDCF performs just like DCF when the network load is light where DCF is proved to be highly efficient. However, HDCF outperforms DCF under higher loads. In other words, HDCF achieves higher performance and adapts better to different loads and network sizes.

2.6.5 Channel Noise

The results provided are for a network of 50 stations with CBR traffic of 50 packets per second. Each packet is 1000 bytes, and each simulation is run for 100 seconds. To implement noise, we used a jammer node provided by Opnet. The jammer was configured to produce noise signals with a constant length of 1024 bits/signal at a constant rate varied for different runs. Fig. 2.26 shows the performance measures vs. the number of noise bits per second for an 802.11g network. In this figure, the x-axis is log-scaled. The figure show that the throughput gain increases up to about 93.7%, and then starts to decrease until there is no gain when the channel errors are very high. Furthermore, HDCF provides higher throughput and fairness for all tested noise levels.

For both protocols, stations defer their access to the channel whenever sensed busy. However, the number of contending stations would increase when more noise is introduced. As a result, the performance of DCF degrades since more stations are contending; a higher collision rate and backoff slots. Quite the opposite, HDCF would reduce number of contending stations and unnecessary idle slots. Moreover, active stations would always recover from channel errors as explained in section 2.4.6. Thus, HDCF performance is steady as long as errors are not severe. Once the noise reaches a very high level, both protocols are severely affected.

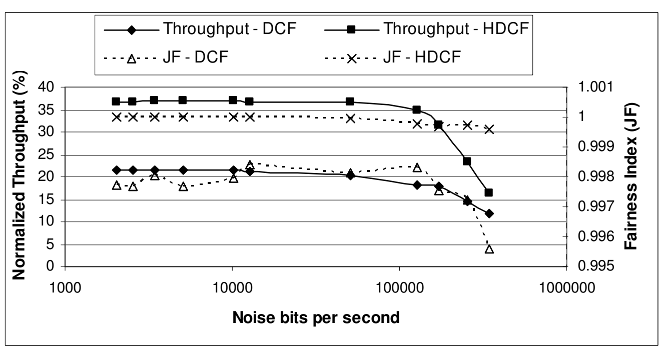


Figure 2.26 Throughput vs. noise level, 802.11g

2.7 Conclusions

In IEEE 802.11 wireless networks, DCF is the basic channel access scheme. However, the performance of DCF degrades when the network size (number of users) or offered loads get larger because of higher contention levels, and so more idle slots and higher collision rates. In this chapter, we proposed a new high-performance DCF (HDCF) MAC protocol to address the problem of wasted time in contention resolution in DCF. HDCF eliminates the need for unnecessary contention and idle slots by allowing transmitting stations to select the next user to transmit. To assure fairness, next station is selected in a random uniform fashion. In addition, new stations utilize an interrupt scheme to contend directly without delays. Thereafter, active stations would stop their active transmissions and only new stations would compete for the channel using DCF. As a result, HDCF reduces the number of contending stations, and so collision rates, and backoff slots. Also, HDCF is designed so that stations transmit in a uniform random order without the need for slotted channel, reserved periods, time synchronization, central control, or knowledge of number or order of active users.

We provided an analytical model to show the effectiveness of HDCF compared to DCF. Furthermore, we presented a simulation study using Opnet Modeler. Simulation results illustrated that HDCF significantly improves the performance as it achieves higher throughput and fairness levels for both saturation and non-saturation scenarios. For 802.11g, the gains can be up to 391.2% of throughput and 26.8% of fairness index. For example, HDCF provides gains of about 164.7% of normalized throughput, and 5.6% of long term and 11.6% of short-term fairness levels when using the IEEE 802.11g specifications for a network of 50 saturated stations, a packet size of 1000 bytes, and no channel errors. Future work includes evaluating HDCF's performance with the existence of hidden nodes.

CHAPTER 3. A New ACK Policy To Mitigate the Effects of Coexisting IEEE 802.11/802.11e Devices

Submitted to the IEEE Transactions on Vehicular Technology (TVT)

Haithem Al-Mefleh [1,3], J. Morris Chang [2,3]

3.1 Abstract

The 802.11e standard is designed to be backward compatible with the 802.11. As a result, wireless networks are expected to have a combination of both EDCA (802.11e Enhanced Distributed Channel Access) and legacy DCF (802.11 Distributed Coordination Function) users. Typically, the 802.11e users who have QoS requirements are supposed to get a higher priority service than that of legacy users. However, the EDCA users' performance may be degraded because of the existence of legacy DCF users, and therefore would get a lower priority service. The main reason for such effects is due to the following fact: EDCA users are controlled through the use of different contention parameters ($AIFS$, CW_{min}, CW_{max}, $TXOP$) that are distributed via the beacon frames. Nevertheless, there is no control over legacy users because their contention parameters ($DIFS$, CW_{min}, CW_{max}) are PHY dependent, i.e. they have constant values. In this chapter, we discuss different aspects of the legacy DCF and EDCA users coexistence. Moreover, we propose a simple distributed management scheme (called NZ-ACK) that mitigates the influence of legacy DCF on EDCA performance in networks that consist of both types of users. Finally, we use *Opnet* simulation to evaluate the performance of the proposed scheme and compare it to 802.11 and ACKS. The results show that NZ-ACK outper-

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forms the other two approaches in terms of enhancing the overall network performance, and maintaining the priority of service and delay bounds of EDCA users while providing acceptable throughput for legacy users.

3.2 Introduction

The IEEE 802.11 standard is the most popular MAC protocol used in today's wireless local area networks (WLANs). The 802.11 standard is widely deployed because it was designed to be simple, and to support best effort traffic while providing all users an equal opportunity to access the wireless medium. However, the 802.11 standard is not suitable for applications that require QoS support where not every user requires the same amount of bandwidth, and long delays are intolerable by real-time applications such as voice and video. Therefore, the IEEE Std 802.11e-2005 standard was developed to provide different mechanisms to meet the growing demand of users for real-time applications.

The 802.11 standard defines two modes of operation: DCF (Distributed Coordination Function), and PCF (Point Coordination Function). Alternatively, the new Hybrid Coordination Function (HCF) is introduced in the 802.11e. HCF includes two modes of operation: Enhanced Distributed Coordination Access (EDCA), and HCF Controlled Access (HCCA).

PCF and HCCA are centralized controlled access methods that exist at a coordinator node, the access point (AP). The AP uses polling to assign the right to access the channel following a predetermined schedule. Both operations have the drawbacks of requiring a coordinator node, and adding the overhead of polling messages that are usually transmitted using lower physical rates. On the other hand, DCF and EDCA are distributed contention-based access functions in which the right to access the wireless channel is determined by different local contention parameters used by every user. Extending DCF, EDCA introduces different QoS mechanisms like priority levels and transmission time bounds.

The 802.11e standard is designed to be backward compatible with the 802.11. As a result, wireless networks are expected to have a combination of both EDCA (802.11e) and legacy DCF (802.11) users. Typically, the 802.11e users who have QoS requirements are supposed to get a higher priority service than that of legacy users. However, the EDCA users' performance may be degraded because of the existence of legacy users, and therefore would get a lower priority service. The main reason for such effects is due to the following fact: *EDCA users are controlled through the use of different contention parameters ($AIFS$, CW_{min}, CW_{max}, $TXOP$) that are distributed via the beacon frames. Nevertheless, there is no control over legacy DCF users because their contention parameters ($DIFS$, CW_{min}, CW_{max}) are PHY dependent, i.e. they have constant values.*

To give an example, consider a simple scenario where 802.11b PHY is used and all EDCA users are using voice access category with a CW_{min} of 8 and $AIFS$ of 50μ seconds. In addition, any existing legacy DCF users use CW_{min} of 32 and $DIFS$ of 50μ seconds. Due to an increase in the number of EDCA users, the QAP (QoS access point) broadcasts new values of CW_{min} of 32. The AIFS cannot be reduced since 50μ seconds is the smallest value allowed for non-QAP users. Moreover, legacy DCF users' parameters are fixed. Hence, coexisting EDCA and legacy DCF users would have the same priority to access the channel, and so the performance of EDCA users could be affected.

In this chapter, we discuss different reasons that result in the performance degradation when EDCA and DCF users coexist, and provide general desirable features for any mitigation solution. Based on these features, we propose a simple distributed management scheme to mitigate the influence of legacy DCF on EDCA performance in networks that consist of both types of users. The proposed scheme is based on the following common behavior of EDCA and DCF: when a frame is received, the included duration in that frame is used by each user to update the local NAV (Network Allocation Vector) counter. The NAV value is used to defer access to the channel unless the user is the destination and is required to send back a response frame. In addition, the duration of the last ACK frame in a transmission exchange (i.e. data frame and its ACK, or all data frames and their ACKs if more than one frame or fragment are transmitted) is set to zero. Accordingly, all EDCA and DCF users are allowed to start contending for the channel directly after the last ACK frame.

In our proposed mechanism, the QAP is allowed to set the duration of the last ACK frame in a transmission exchange to a non-zero value; hence we call these frames NZ-ACK frames, and we call the proposed mechanism NZ-ACK scheme. Upon receiving an NZ-ACK frame, an EDCA user sets its local NAV counter to zero just as if a zero duration ACK frame is received, and thus would start directly to contend for the channel. On the other hand, a legacy DCF user does not recognize any difference between a normal ACK frame and a NZ-ACK frame. As a result, DCF users will set their NAV counters according to the non-zero duration value included in the received NZ-ACK frame, and use that duration to defer their access to the channel.

For an efficient performance, the QAP requires deciding the following two challenging issues: when to issue NZ-ACK frames, and the duration value of an issued NZ-ACK frame. We address these issues with the objective of mitigating the coexisting effects while utilizing bandwidth efficiently, and without starving the legacy DCF users.

In addition to being simple and distributed, the proposed scheme has the following features:

1. *Full transparency to legacy DCF users:* no modifications are required to legacy DCF users; they would not recognize any difference between normal ACK and NZ-ACK frames. Hence, full backward compatibility is kept.

2. *Minimal modification added to EDCA users:* NZ-ACK requires minimal modification to the 802.11e standard; while the processing is at the QAP, non-QAP EDCA users are only required to recognize the new ACK policy used with the NZ-ACK frame.

3. *No changes to the 802.11e standard frames' formats:* NZ-ACK does not add any overhead bits to any frame, and does not define any new messages.

4. *Adaptive control of legacy DCF users:* NZ-ACK controls legacy DCF users by having them defer their access to the channel adaptively according to network status (number of users of both types and available QoS traffic from EDCA users) to maintain EDCA users' priority of service. In addition, the overall network performance is enhanced when considering throughput, delay, and retransmission attempts. The performance gain is due to fact that NZ-ACK reduces the number of contending users, and thus collision

rates, when issuing non-zero duration NZ-ACK frames; only EDCA users are competing for the channel when DCF users are yielding.

The rest of this chapter is organized as follows. Section 3.3 provides background information about both DCF and EDCA. In section 3.4, we provide an insight on the effects of coexisting DCF and EDCA devices, and present general desirable features for any proposed solution. In section 3.5, different related works are summerized. We discuss the details of our proposed mechanism in section 3.6, and present its evaluation via *Opnet* simulation in section 3.7. Finally, conclusion remarks are provided in section 3.8.

3.3 IEEE 802.11 Background

3.3.1 Distributed Coordination Function (DCF)

The IEEE 802.11 standard (3; 4; 66) defines two mechanisms for DCF which are based on CSMA/CA. In basic operation, a station that has a packet to transmit will do so if the medium is sensed idle for a period of distributed interframe space (DIFS). Otherwise, the station will go into backoff where the Binary-Exponential-Backoff (BEB) procedure is used. The station chooses a number of time slots to wait before trying to transmit again. The number, or the backoff counter, is selected from the range $[0, CW]$, where CW is called the contention window and is initially set to CW_{min}. The station decrements its backoff counter by one for every slot time the medium is sensed idle. When the backoff counter reaches zero, the station transmits its packet. Upon receiving a data frame, the destination responds by sending back an acknowledgment (ACK) frame after a short interframe space (SIFS) time. The ACK frame has a higher priority because SIFS is the shortest interframe space (IFS) used in DCF. The packets transmitted carry the time needed to complete the transmission of a packet and its acknowledgement. This time is used by all other stations to defer their access to the medium and is called NAV, Network Allocation Vector. Collisions occur when two or more stations are transmitting at the same time, or when the ACK frame is not received after a timeout period. With every collision, the transmitting station will double its CW unless it reaches a maximum limit CW_{max}, and selects a new backoff counter from the new range. The process is repeated until the packet is successfully transmitted or is dropped because a retry limit is reached. In RTS/CTS operation, a station uses control packets to contend for the channel before transmitting data frames, data frames are free of collision.

Other than being simple and distributed, DCF is most popular because it assures long-term fairness where each station has the same opportunity to access the channel. However, DCF is not suitable for applications that require QoS support due to highly possible long delays that are intolerable by real-time applications like voice and video. As a result, the newer IEEE 802.11e standard provides an enhanced version of DCF, i.e. EDCA that we introduce in the next subsection.

3.3.2 Enhanced Distributed Channel Access (EDCA)

The IEEE 802.11e standard (13) defines EDCA that provides better service to real-time traffic by differentiating traffic using different priority levels. As shown in Fig. 3.1, EDCA classifies data frames into four different access categories (ACs) according to the user priority (UP) provided by the above layers. Each AC constitutes an enhanced distributed channel access function (EDCAF) that works exactly the same as DCF. However, the contention parameters for each EDCAF could be different and are announced in the beacon frames. Each AC is categorized by different contention parameters including the arbitration interframe space (AIFS), CW_{min}, and CW_{max}. AIFS is the amount of time the medium should be sensed idle first. Moreover, EDCA introduces a new concept, the transmission opportunity (TXOP) limit which indicates the maximum amount of time that the user should use when winning the right to transmit data frames. An internal collision occurs when two or more EDCAFs win the contention at the same time and the same user. The AC with the higher priority is allowed to start transmitting data frames, and all others go into backoff as if an actual collision has occurred.

In summary, the $EDCAF_i$ for each AC_i ($i = 0, \ldots, 3$) is defined by $AIFS[i]$, $CW_{min}[i]$, $CW_{max}[i]$, and $TXOPLimit[i]$. An AC with a smaller AIFS value, smaller CW_{min}, and smaller CW_{max} has a higher priority to access the channel as explained in Fig. 3.2.

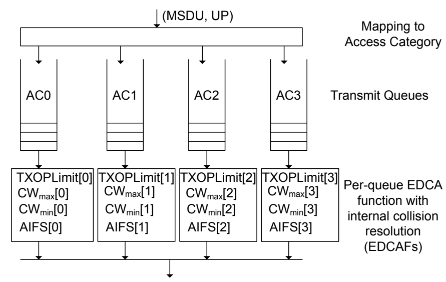

Figure 3.1 EDCA Transmission Queues and EDCAFs

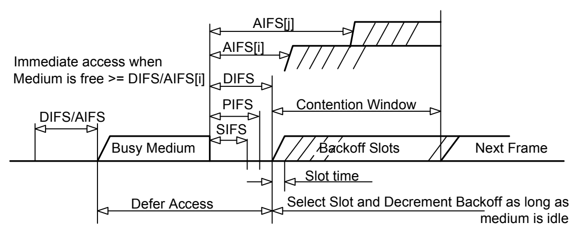

Figure 3.2 Interframe Space Relations

3.4 IEEE 802.11 DCF/EDCA Coexistence

3.4.1 Problem Statement

The 802.11e standard is designed to be backward compatible with the 802.11. As a result, wireless networks are expected to have a combination of both EDCA (802.11e) and legacy DCF (802.11) users. Typically, the 802.11e users are supposed to get a higher priority service than that of legacy users. However, the EDCA users' performance may be degraded because of the DCF users, and therefore would get a lower priority service. We summarize the reasons for this degradation in the following:

1. EDCA users are controlled through the use of different contention parameters (AIFS, CWmin, CWmax, TXOP) that are distributed via the beacon frames. On the other hand, there is no control over DCF users because their contention parameters (DIFS, CWmin, CWmax) are PHY dependent, i.e. they have constant values.

   Because of this difference in control of the contention parameters, the following scenarios may arise. Fisrt, when the total number of users increases, the CW values of EDCA clients may be adjusted to reduce collision rates. As a result, DCF users would get a higher priority and hence may degrade the service provided to EDCA users. Second, the collision rate due to legacy stations would affect the EDCA performance specially when there are a large number of contending DCF users.

2. The smallest AIFS value allowed for non-QAP EDCA users is equivalent to the defer value used in DCF, i.e. DIFS. Since a smaller AIFS leads to a higher priority, DCF users probably will get a higher priority than some or all access categories of EDCA including real-time ones.

3. To grant EDCA users a higher priority, one may assign them smaller CW values than that of DCF users. However this leads to a higher collision rate as seen by EDCA users, and so the overall collision rate of the network. The situation gets worse as the number of EDCA stations increases. Hence, the QoS support and overall performance are degraded.

4. In EDCA, the transmission time is controlled via the TXOP feature in order to provide QoS guarantees. Such control is not applied by legacy users. Therefore, transmissions from DCF users may overlap with TBTT (Target Beacon Transmission Time), and may occupy most of the channel time when using lower physical data rates. Hence, the performance of EDCA users would be degraded.

3.4.2 Desirable Features

Apparently, introducing new mechanisms is essential to mitigate the influence of legacy DCF on EDCA performance in networks that consist of both types of users. For the design

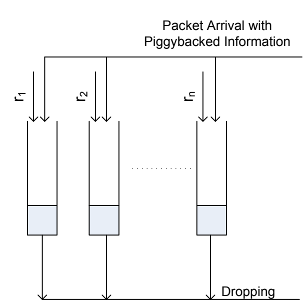

Figure 3.3 Virtual real-time queues at the QAP

of such techniques, we argue that the following considerations are important for an effective performance and practical issues:

1. *No change to the legacy stations:* for compatibility issues, no modification should be introduced to legacy users. Therefore, the new changes are to be implemented on the QoS part, i.e. the new 802.11e devices specially the QAPs in infrastructure networks.

2. *Provide control over legacy stations:* the behavior of legacy users should be controlled to provide EDCA users with a higher priority as expected, and to mitigate the performance degradation.

3. *Utilize bandwidth efficiently:* the control over DCF stations should not waste bandwidth unnecessarily. For example, there is no need to prevent DCF users from accessing the wireless medium if there is no EDCA traffic.

4. *Minimal overhead:* a new mechanism should not require complex computations or processing by the non-AP users, and should not alter the 802.11e/802.11 frames' formats. Such overheads may not be feasible or easy to implement.

5. *Contention-based operation:* because polling is not an attractive solution, new mechanisms should be working with the contention operation.

6. *Fairness:* the influence of any new technique should be the same for all DCF users.

In section 3.6, we present NZ-ACK, our proposed mechanism that addresses the impact of legacy DCF on EDCA users, and satisfies all these requirements.

3.5 Related Work

In (31), the authors evaluate using simulation analysis the effect of different contention parameters on the network performance when 802.11e EDCA and 802.11b DCF users coexist. They show that AIFS is the best for delay performance, but would result in throughput starvation for legacy users. They conclude that to achieve fairness, both CW_{min} and $AIFS$ should be adapted with the mix of 802.11e and 802.11b priority users. In addition to these results, its demonstrated in (32) that the increase of collisions due to small CW values reduces the difference between EDCA and legacy DCF users.

In (33), the authors suggest a scheme to improve the performance of the legacy users assuming they have multimedia traffic. A Hierarchical Token Bucket (HTB) discipline between the IP layer and Layer 2 at the legacy users is used to classify, police, and schedule and shape the incoming traffic. The presented solution requires modifications to legacy users, and does not show how to solve the coexistence effects.

In ACKS (35), the authors proposed that the QAP should skip sending back an ACK frame to a DCF station with some probability δ. Skipping ACK frames results in a waste of bandwidth for all stations, regardless of the fact they are using DCF or EDCA. The time wasted is equivalent to the total time needed to contend for the channel, and to transmit all data fragments, and corresponding ACK frames. In addition, dropping a data frame that already has been successfully transmitted is not a good solution in a wireless network that is noisy. As a result, ACKS may result in unfairness among DCF stations. Finally, ACKS is proposed for a saturated network to achieve weighted throughput guarantees by fixing AIFS to DIFS and adapting the CW_{min} for all users. Consequently, as explained in (36), although the weighted throughput ratios are met, the QoS requirements of EDCA users would be affected when legacy users transmit at lower physical rates since they do not deploy the TXOP limit feature.

In (34), a mechanism is used to prevent a legacy user from starting a data transmission if its transmission would overlap with the TBTT (Target Beacon Transmission Time). Using the beacon frame, the QAP broadcasts a factor that is used by legacy users to determine when such an overlap may occur. Accordingly, the time is divided into two periods: the first is used by all stations to contend for the channel, and then followed by the second period during which only EDCA users do contend for the channel. The proposed mechanism requires modifications to the legacy users, does not reduce the coexistence effects during the first period but may increase it because of the accumulation of the DCF users' contention into only the first period, and may waste bandwidth unnecessarily during the second period when not used by any of the EDCA users.

3.6 NZ-ACK Details

We propose a simple distributed management scheme, called NZ-ACK, that mitigates the influence of the legacy IEEE 802.11 DCF users on the IEEE 802.11e EDCA users in an infrastructure network via introducing a new policy of ACK frames. The design of NZ-ACK satisfies all features described in subsection 3.4.2.

NZ-ACK controls legacy DCF users by having them yield the channel to EDCA users adaptively according to number of users of each type and available EDCA QoS traffic to maintain the priority of service of EDCA users. In addition, the overall performance is enhanced when considering throughput, delay, and retransmission attempts. The performance improvement is because NZ-ACK reduces the number of contending users, and thus collisions, when issuing non-zero duration NZ-ACK frames; only EDCA users are competing for the channel when DCF users are yielding.

In this section, we explain the basic idea, the implementation, and different operations and challenges of NZ-ACK scheme.

3.6.1 An Overview

Fig. 3.4 explains the basic principal of NZ-ACK scheme, and shows how users behave in a network with or without NZ-ACK being employed. As explained in *Part* 1 of Fig. 3.4, competing users would set their local NAV counters according to the duration value included in the header of the received frame. Following the EDCA or DCF rules, the NAV value of the last ACK frame in the current transmission exchange is set to zero, *ACK* 1 in *Part* 1 of Fig. 3.4. Accordingly, all EDCA and DCF users are allowed to start contending for the channel directly after the last ACK frame. Before starting the backoff period, each user must first sense the channel to be idle for a specific period, i.e. AIFS for EDCA users, and DIFS for legacy DCF users.

To mitigates the impact of the legacy DCF users on the EDCA users, we propose a man-

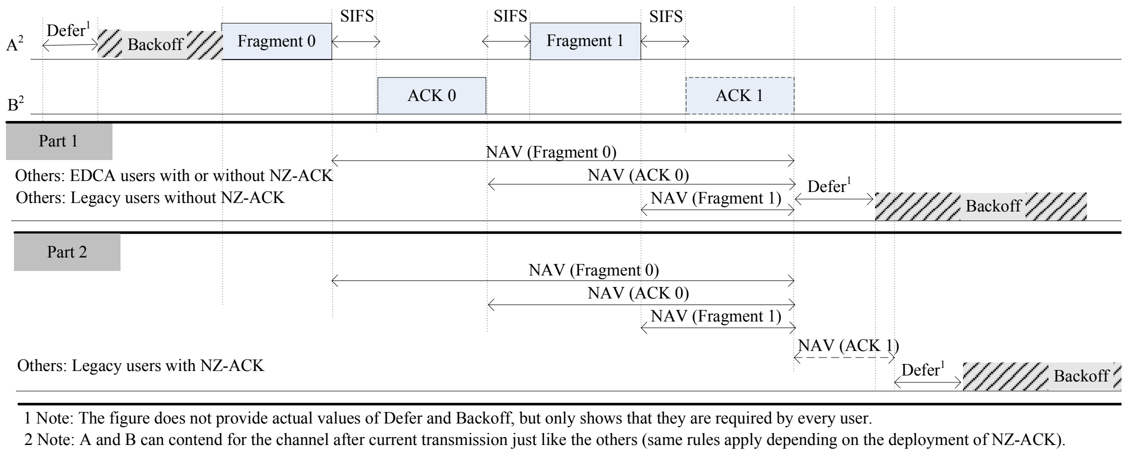

Figure 3.4 An example of NZ-ACK operation with a transmission of two fragments

agement scheme that increases the defer value of legacy users as required. We introduce a new type of ACK frames that are called Non Zero ACK (NZ-ACK) frames. Thus the proposed scheme is called NZ-ACK.

A NZ-ACK frame is simply the last ACK frame of the ongoing transmission, *ACK* 1 in Fig. 3.4. Moreover, a NZ-ACK frame can be sent in response to a data frame sent by a legacy user or an EDCA user. As explained in *Part* 2 of Fig. 3.4, when *ACK* 1 is used as an NZ-ACK frame, the legacy DCF users would simply update their NAV values using the duration of *ACK* 1. On the contrary, EDCA users would start directly their contention by deferring using their AIFS values as shown in *Part* 1 of Fig. 3.4.

In an infrastructure network, only the QAP may determine to transmit NZ-ACK frames. Hence, all users would be able to receive the frame unless lost due to channel errors. The QAP sets the duration filed of a NZ-ACK frame to a nonzero value. Nevertheless, an EDCA user should set its local NAV counter to zero upon receiving the NZ-ACK frame. On the other hand, NZ-ACK scheme is designed so that a legacy DCF user does not recognize any difference between a normal ACK frame and a NZ-ACK frame. As a result, DCF stations will set their NAV counters according to the duration value included in the received NZ-ACK frame.

When there are EDCA users with NZ-ACK not implemented, these users would treat NZ-ACK frames the same way the legacy DCF users do. As a result, NZ-ACK is fully backward compatible but with NZ-ACK EDCA users having a higher priority than EDCA users with NZ-ACK not implemented. Nonetheless, there would still be performance gain because of the separation of competitions of different types of users.

Consequently, NZ-ACK allows the QAP to increase the defer periods of the legacy stations in an adaptive way using the ACK frames that are common to all users. In addition, the QAP would be able to respond faster to different changes in users' behaviors because the ACK frame is a part of any data frame transmission; for example, a legacy user may adjust to a lower physical rate. Although there is an exception when direct link is used by EDCA users, legacy users always require receiving the ACK frames. In other words, more resources for the EDCA users could be reserved in a dynamic, reactive, and distributed fashion.

3.6.2 Operations of NZ-ACK

Before starting a QoS flow, an EDCA user first sends a request to the QAP with the QoS requirements of the flow including the average data rate, peak data rate, and nominal packet size. The QAP would send a response back to the user. If admission control is not implemented, the QAP would always accept flows. Then the QAP can determine the required utilization of these users, U_{EDCA}. Hence the rest of the channel utilization (U) can be used by all legacy users, U_{DCF}.

$$U_{EDCA} = \sum_{i=1}^{n_{EDCA}} \frac{r_i}{l_i} T_s \tag{3.1}$$

$$U_{DCF} = U - U_{EDCA} \tag{3.2}$$

Where n_{EDCA} is the total number of EDCA users with QoS requirements, r_i is the data rate (the rate at which packets are generated), l_i is the packet size, and T_s is the time needed to successfully transmit the packet ($T_s = AIFS + SIFS + T_{ACK} + T_{DATA}$) for every EDCA user i with QoS requirements.

We also use the concept of virtual EDCA queues, Fig. 3.3. The QAP generates a virtual queue for each admitted flow i, and adds a virtual packet to the queue every $1/r_i$ seconds. In order to maximize bandwidth utilization, we use the maximum possible interarrival time; r_i is set to the average rate for VBR sources. For CBR, r_i is the average rate which is also the same as the peak rate. Moreover, a virtual packet is added to an empty virtual queue when a received frame indicates more data buffered at the user. Also, all queues would be arranged according to the rate; i.e. the smaller the rate, the higher the priority. Finally, virtual packets are dropped in different cases:

1. An EDCA user would drop a packet when its waiting time becomes longer than the delay requirement of the flow to which the packet belongs. Therefore, a virtual packet also would be dropped by the QAP for the same reason.

2. The virtual packet is the reason to issue the NZ-ACK frame (explained in section 3.6.4).

3. Virtual packets of a flow for which a data frame is received indicating no more data buffered.

From the virtual queues, the QAP estimates the number of active EDCA stations, or stations that have data frames, ($\hat{n}_{EDCA}$) by the total number of nonempty virtual queues.

In the following two subsections, we address how the QAP determines when to issue NZ-ACK frames, and how long is the duration of a NZ-ACK frame.

3.6.3 When to Issue NZ-ACK Frames

NZ-ACK frames should not be issued all the times but depending on the network status. Apparently, NZ-ACK frames should not be issued when there are no data frames to be sent by EDCA users. Otherwise, the time that is used to defer the legacy DCF users would be wasted. Moreover, when the number of EDCA users is significantly greater than that of DCF users, we might want to reduce the probability of issuing a NZ-ACK frame since the DCF users might be deferred indefinitely, and so might be starved. From these observations, we propose to issue NZ-ACK frames only when $\hat{n}_{EDCA} > 0$ with a probability based on the ratio between the number of DCF stations and active EDCA stations:

$$\rho = \frac{n_{DCF}}{n_{DCF} + \hat{n}_{EDCA}} \tag{3.3}$$

Where n_{DCF} is the total number of legacy stations. First, when n_{DCF} is much greater then $\hat{n}_{EDCA}$, there is a high probability of issuing NZ-ACK frames. In addition, when the number of active EDCA stations is constant, a small increase in the number of legacy users results in a faster increase of the probability. In general, the higher the number of DCF users in the network, the higher the need for NZ-ACK frames to protect EDCA users. Second, when the number of DCF users gets very small compared to that of EDCA, the probability approaches 0. This is accepted since the effect of legacy users would be much smaller. Therefore, in such scenario we rely mostly on contention parameters so that DCF users will have a chance to compete with EDCA users without being starved. However, the probability is high when on average $\hat{n}_{EDCA}$ is small compared to n_{DCF}. Therefore, to protect DCF users, we add the following condition

$$U_{DCF\_Measured} >= U_{DCF} \tag{3.4}$$

where $U_{DCF\_Measured}$ is the utilization of DCF users measured by the QAP with all added NZ-ACK's durations considered as a part of $U_{EDCA\_Measured}$. To summarize, we maintain the service priority of EDCA users while allowing DCF users to use the rest of bandwidth under different network conditions.

3.6.4 How Long is the Duration of an NZ-ACK Frame

One could use values as high as possible to guarantee that QoS traffic is always transmitted before DCF users' traffic. For example, we may use $CW_{max}[voice]$ or higher as the duration value to guarantee that all voice traffic is transmitted first. However, this could result in wasting bandwidth unnecessarily and possibly starving legacy DCF users. Therefore, we add values depending on the utilization required by EDCA users while attempting to allow legacy users to utilize the remaining bandwidth.

Let u_c ($u_c = \frac{r_c}{l_c} T_s$) be the utilization of the virtual packet at the head of line (HOL) of the first non-empty virtual queue; i.e. the virtual queue with the lowest rate among all non-empty queues. Then we find the value to be used as the duration of current NZ-ACK frame by $d_c = u_c T$, where T is a predetermined period. The QAP maintains two parameters that are updated every T seconds: the time used by EDCA users with QoS requirements (t_{EDCA}), and the time used by legacy DCF users (t_{DCF}). Then we define the utilized time

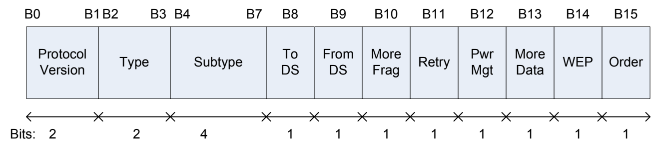

Figure 3.5 Frame Control field

by $t_{used} = t_{EDCA} + t_{DCF}$, and the remaining time by $t_r = T - t_{used}$. The NZ-ACK frames is issued as long as the following condition applies:

$$\frac{t_{DCF} + t_r - d_c}{T} \geq U_{DCF} \tag{3.5}$$

This condition assures that more time, i.e d_c, is reserved for EDCA users if such reservation would not deplete U_{DCF}. Once the condition is not met, normal ACK frames are used.

3.6.4.1 Saturated Users

When users always have frames to transmit, delay requirements can not be guaranteed. In such scenario, we add the following changes. First, virtual queues are not used since they are not useful any more; all stations are active all the time. Second, since no delay requirements can be met and users are all active all times, NZ-ACK frames are issued only when the ACK frame is a response to a legacy user with the probability $\rho = \frac{n_{DCF}}{n_{DCF}+n_{EDCA}}$. Finally, one time slot is used for the duration of an NZ-ACK frame.

3.6.5 Implementation

All processing performed by NZ-ACK is implemented at the QAP. This includes determining when to use NZ-ACK frames, and what values to be used for the duration field of the NZ-ACK frames. For the QAP to recognize the last fragment from a legacy user, the $morefragments(B10)$ bit of Frame Control field, Fig. 3.5, can be used since only one packet is allowed. On the other hand, an EDCA user is allowed to transmit more than one packet within the TXOP. In such case, the QAP can recognize last fragment or packet whenever the duration included is not enough (less than or equal to SIFS is used in our implementation) to start a new transmission from the same user.

EDCA users are required to distinguish between a regular ACK and a NZ-ACK. At the same time, a legacy DCF user must recognize no difference between both ACK and NZ-ACK frames (NZ-ACK must be seen as an ACK).

To distinguish between ACK and NZ-ACK frames, we used the fact that all bits $B8$ to $B15$ except for $B12$ in the Frame Control field, Fig. 3.5, of control frames are always set to $'0'$. In our implementation, we selected $B10$. An EDCA user would recognize an ACK frame as a NZ-ACK frame when $B10$ is set to $'1'$, and as a normal ACK otherwise.

Because no change is made to *Type* and *Subtype* fields of Frame Control field, legacy DCF users would still understand NZ-ACK frames as normal ACKs. In other words, such users would not try to interpret the bit $B10$ of Frame Control in control frames including ACK, RTS, and CTS. Consequently, NZ-ACK scheme requires no changes to the legacy users' implementations.

Finally, NZ-ACK does not add any overhead bits to the ACK frames, and does not require any extra messages other than those found in the IEEE 802.11e standard. The ADDTS requests, ADDTS responses, and DELTS frames are used to convey QoS requirements between the QAP and EDCA users.

3.7 Evaluation

This section presents the simulation we used to evaluate the performance of NZ-ACK (802.11 EDCA/DCF with NZ-ACK) and compare it to that of 802.11 (802.11 EDCA/DCF without NZ-ACK or any other modification), and ACKS (35) (802.11 EDCA/DCF with ACKS) which we discussed in related work section 3.5. We utilized the commercial Opnet Modeler 11.5.A, (1), to implement NZ-ACK and ACKS by modifying the *Opnet* 802.11e models.

In each simulation experiment, we consider an infrastructure network that consists of stations that share a single wireless channel. We also assume a fully connected network; each station can listen to every other one in the network. Moreover, there are no channel errors; collisions are the only source of errors.

3.7.1 Performance Metrics

For performance analysis, we use the following metrics:

1. *Throughput:* the total data bits successfully transmitted per the simulation time. We look at overall network throughput, EDCA throughput (throughput per EDCA ACs), DCF throughput (throughput per DCF), and throughput ratio ($\frac{EDCA\ Throughput}{DCF\ Throughput}$).

2. *Fairness Index (FI):* we used Jain Index (12; 65) defined by (3.6):

$$FI = \frac{(\sum_{i=1}^{n} S_i)^2}{n \sum_{i=1}^{n} S_i^2} \tag{3.6}$$

   Where n is number of stations and S_i is the throughput of station i. The closer the value of FI to 1, the better the fairness provided. We use FI to find how fair a scheme is to different DCF users.

3. *Delay:* the delay for each packet is measured from the moment that packet arrives at the MAC layer until its ACK response is received correctly. We report the total deal of every packet, and delay of EDCA packets.

4. *Retransmission attempts (ReTx):* a higher average number of retransmission attempts indicates a higher collision rate.

3.7.2 Saturated Network

We evaluate NZ-ACK performance in a saturated network where each user always has a data frame to transmit, and compare it to that of ACKS (35) (we selected ACKS from related work since, like NZ-ACK, it requires no modification to legacy users) and to the 802.11 DCF/EDCA with no modification. In addition, we report results for a special case, called *OneSlot*, where a NZ-ACK frame is always transmitted with a duration value of one slot.

For this subsection, the 802.11g PHY is used with a data rate of $54Mbps$ and control rate of $24Mbps$. For NZ-ACK, and 802.11 EDCA/DCF, we consider two different settings of CW parameters; $NZ\text{-}ACK_i$ and 802.11_i with $i = 1, 2$ as summarized in Table 3.1. For a fair comparison, we took one scenario from ACKS work in which there are 50 users of EDCA of

Table 3.1 Saturation Results - 1

Scheme	CW_{min}/CW_{max}	FI	*Total Throughput*	*Delay*
$ACKS$	196/196	0.915631	18700800 *bps*	1.011368 *sec*
NZ-ACK_1	63/1023	0.952897	19244800 *bps*	0.995508 *sec*
NZ-ACK_2	63/511	0.949557	19969600 *bps*	0.945712 *sec*
802.11_1	63/1023	0.955616	18040000 *bps*	1.080050 *sec*
802.11_2	63/511	0.946775	18491200 *bps*	1.046805 *sec*
OneSlot	63/1023	0.914584	22313600 *bps*	0.812835 *sec*

Table 3.2 Saturation Results - 2

Scheme	*DCF Throughput*	*EDCA Throughput*	*Delay*	*Throughput Ratio*	*ReTx*
$ACKS$	4244800 *bps*	14456000 *bps*	0.770447 *sec*	3.405579	0.812201
NZ-ACK_1	4924800 *bps*	14320000 *bps*	0.748646 *sec*	2.907732	0.723645
NZ-ACK_2	3969600 *bps*	16000000 *bps*	0.702277 *sec*	4.030633	0.669818
802.11_1	6049600 *bps*	11990400 *bps*	0.863025 *sec*	1.982015	0.847450
802.11_2	4840000 *bps*	13651200 *bps*	0.788217 *sec*	2.820496	0.810678
OneSlot	3201600 *bps*	19112000 *bps*	0.581907 *sec*	5.969515	0.490320

the same access category and 50 legacy users, and the EDCA users are assigned a throughput weight of 3 times that of DCF users; i.e. the throughput ratio is 3. In ACKS, all EDCA users set their CW_{max} equal to CW_{min} and use $DIFS$ for long inter-frame spacing, and no modification is applied to legacy DCF users. Using the provided optimal value of δ (0.489 based on ACKS (35)), we solved the given nonlinear equations to get a CW_{min} of 196 for EDCA users. For NZ-ACK and the 802.11 scenarios, $DIFS$ is used, and the PHY CW_{min}/CW_{max} are 16/1024 which are used by legacy users. T is set to the beacon interval. Finally, the average results of conducted simulations are summarized in Tables 3.1 and 3.2 on which we base the following discussion.

We first explain why not to issue NZ-ACK frames all the time by looking at *OneSlot* scenario. In this scheme, a NZ-ACK frame with a duration of one slot is alwayes issued. Note that this scheme provides the best performance for EDCA users compared to DCF users (the lowest possible delays, highest throughputs, and highest ratio of EDCA Throughput to that of DCF). However, it also results in high degradation in performance for legacy users. For example, DCF throughput is at least 20% lower than that achieved using other schemes. Also, *OneSlot* scenario has the lowest FI value. Hence, as the number of EDCA users increases, this scheme may result in DCF users' starvation.

On the other hand, compared to 802.11 scenarios (we compare scenarios with same contention window parameters, i.e. $NZ\text{-}ACK_i$ with 802.11_i, $i = 1, 2$ as shown in Tables 3.1 and 3.2), NZ-ACK provides the highest total throughput (about 6.67% and 7.99%of gain), the highest EDCA throughput (about 17.2% and 19.4% of gain), the lowest average total delay (at least 7.82% and 9.65% lower), and lowest average EDCA delay (about 10.9% and 13.2% lower). This is because the legacy users have higher effects on EDCA users in the 802.11 scenarios, which can be seen by the higher DCF throughput in these scenarios and the higher retransmission attempts.

ACKS attempts to achieve throughput weighted fairness by having the access point skipping some of the ACK frames of DCF users. However, the result is wasting the time required to transmit the data frame and its skipped ACK since all users in the network would defer their access to the channel using the duration of that data frame. On the opposite, NZ-ACK mitigates the effects of DCF users by having them yield the channel to EDCA users when necessary. Hence, the average delay and delay per EDCA users are lower than that of ACKS; for example, $NZ\text{-}ACK_2$ achieves about 6.7% and 8.8% lower than that of ACKS for both delays respectively. At the same time, NZ-ACK provides DCF users with an acceptable performance as seen by the throughput and throughput ratio that are close to that of ACKS (about 3 and 4 with both NZ-ACK scenarios compared to that of about 3.4 with ACKS).

Both NZ-ACK variants provide a higher fairness index (FI) than that of ACKS, and almost

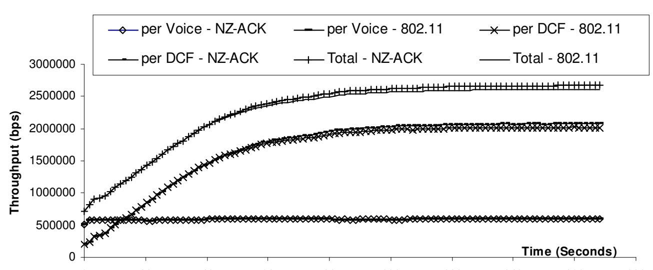

Figure 3.6 Throughput

the same FI values as that achieved by both 802.11 scenarios. Hence, NZ-ACK does provide fair access among all DCF users; the effect of NZ-ACK is the same for all DCF users. This is explained by the fact that the used NZ-ACK frames are sent by the QAP and thus are seen by all DCF users. Moreover, the retransmission attempts, and so the collision rates, in NZ-ACK are lower by at least 14% than that of ACKS and 802.11 becuase it reduces the number of contending stations when issuing non-zero duration NZ-ACK frames; only EDCA users are competing for the channel when DCF users are yielding.

Finally, the overall network performance with NZ-ACK is higher compared to that of 802.11 and ACKS scenarios. NZ-ACK achieves the highest total throughput, lowest average packet delay, and lowest retransmission attempts.

3.7.3 Non-saturated Networks

Here, we evaluate the performance of 802.11 with NZ-ACK deployed in a non-saturated network and compare it to that of 802.11 with no modification. We consider an 802.11b PHY network with $11Mbps$ data rate and $1Mbps$ control rate, and CW_{min}/CW_{max} are 32/1024 (these are used by legacy users). There are 18 voice EDCA users with CW_{min}/CW_{max} of 31/63. Each voice source is modeled by an NO/OFF model with the ON and OFF periods are both exponential (0.352 seconds), and uses G.711 (silence) encoder with $64kbps$ coding rate and 160 bytes per one packet. For legacy DCF users, the simulation starts with one user, and every 3 seconds another DCF user is added with no more than 50 DCF users are added. Each legacy users generates traffic with an inter-arrival rate of exponential ($40ms$), and 1000 bytes

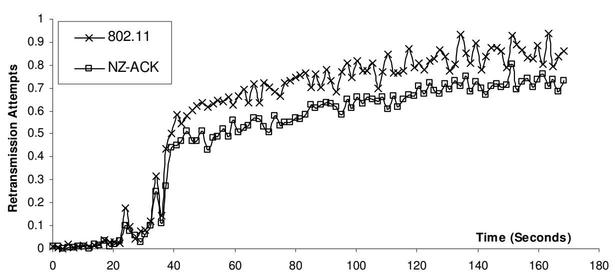

Figure 3.7 Retransmission attempts

per packet. DIFS of 50μs seconds is used by all users. Finally, the simulation is conducted for 170 seconds, and T is set to the beacon interval and delay used for dropping virtual packets is 0.1 seconds.

Fig. 3.6 shows the average total network throughput, average throughput per voice, and average throughput per DCF users. Throughputs per voice are the same for NZ-ACK and 802.11, which is also equivalent to the total voice load (not shown because it is the same value). However, there is a very small enhancement of the total throughput and throughput per DCF when using NZ-ACK. The slight enhancement starts after 40 seconds, i.e. when there are at least about 14 DCF users. In addition, Fig. 3.7 shows the retransmission attempts. The figure explains that NZ-ACK reduces the retransmission attempts, and thus number of collisions, with time as more legacy users are added to the network. This is expected because NZ-ACK reduces number of contending users during the periods where DCF users are deferred by the NZ-ACK frames.

In figures 3.9 and 3.8, the packet delay for voice packets is illustrated. For the 802.11, Fig. 3.9 shows that the delay is maintained very small as long as the number of DCF users is less than about 14 (at about 40 seconds). After that, the delay starts to increase and reach values up to 0.2 seconds. Moreover, the figure shows that the delay variation increases. On the other hand, NZ-ACK protects the voice traffic and keeps the delay and delay variation very small. Fig. 3.8 illustrates the CDF of packet delay; the probability of having a delay lower than a given value. While with NZ-ACK all delays are less than 0.026 seconds, there are chances of more than 0.2 that the delay is higher than 0.1 seconds for the 802.11.

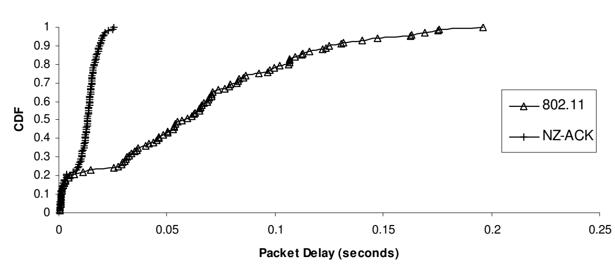

Figure 3.8 Packet Delay, CDF

3.8 Conclusions

The 802.11e standard is designed to be backward compatible with the 802.11. As a result, wireless networks are expected to have a combination of both EDCA (802.11e Enhanced Distributed Channel Access) and legacy DCF (802.11 Distributed Control Function) users. Typically, the 802.11e users who have QoS requirements are supposed to get a higher priority service than that of legacy users. However, the EDCA users' performance may be degraded because of the existence of legacy users, and therefore would get a lower priority service. The main reason for such effects is due to the fact that EDCA users are controlled through the use of different contention parameters (AIFS, CWmin, CWmax, TXOP) that are distributed via the beacon frames. In contrast, there is no control over legacy users because their contention parameters (DIFS, CWmin, CWmax) are PHY dependent, i.e. they have constant values. As a result, depending on the network status like the number of DCF/EDCA users, DCF users could achieve a higher priority and could result in high collision rates, and thus degrade the performance of EDCA users.

In this chapter, we discussed different aspects of the legacy DCF and EDCA coexistence and provided general desirable features for any mitigation solution. Based on those features, we proposed a simple distributed management scheme, called NZ-ACK, to mitigate the influence of legacy DCF on EDCA performance in networks that consist of both types of users. NZ-ACK controls legacy users by introducing a new ACK policy in which the QAP is allowed to set the duration of the last ACK in a transmission exchange to a non-zero value.

In addition, we presented strategies to determine when to issue such NZ-ACK frames, and

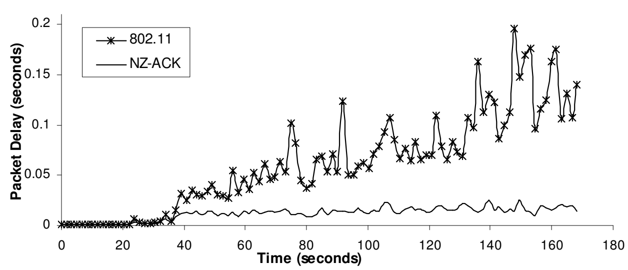

Figure 3.9 Packet Delay

the non-zero duration value of a NZ-ACK frame. All the processing of NZ-ACK scheme is implemented at QAP. However, non-QAP EDCA users only are required to distinguish the new ACK policy in order to ignore the non-zero value duration included in a NZ-ACK frame. On the other hand, NZ-ACK requires no modification (i.e. fully transparent) to legacy users. Thus, NZ-ACK maintains backward compatibility.

The proposed scheme allows EDCA users to start competing directly after NZ-ACK frames. However, DCF users would defer their access to the channel according to the non-zero duration of NZ-ACK frame. Moreover, when to issue NZ-ACK frames and their duration values are determined adaptively according to network status. Thus, more resources for the EDCA users are reserved in a dynamic and distributed fashion to maintain their priority. The performance gain is due to the fact that NZ-ACK reduces the number of contending users when issuing non-zero duration NZ-ACK frames; only EDCA users are competing for the channel when DCF users are yielding. As a result, lower collision rates for both types of users are expected and thus higher throughputs, fairness levels, and lower delays.

Finally, we used $OpnetModeler$ to evaluate NZ-ACK and compare its performance to that of 802.11 and ACKS. The results show that NZ-ACK outperforms the other two approaches in terms of maintaining the priority of service and delay bounds of EDCA users while providing acceptable throughput for legacy users.

CHAPTER 4. Turning Hidden Nodes into Helper Nodes in IEEE 802.11 Wireless LAN Networks

Submitted to the IEEE Transactions on Mobile Computing (TMC)

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4.1 Abstract

To enhance the performance of IEEE 802.11 WLANs in the presence of hidden terminal problem, we propose a protocol that allows non-hidden stations to help each other retransmit faster whenever possible. Opposite to other approaches, the new protocol benefits from the hidden terminal problem to improve the performance of DCF, which is the basic operation of IEEE 802.11. The proposed protocol is compatible with IEEE 802.11, and works with the same PHY of IEEE 802.11. We also provide an analytical model to evaluate the throughput of the new scheme and compare it to that of DCF. The model is validated via Opnet simulation. Using Opnet simulation, results show that the proposed scheme improves throughput, delay, packet drop, retransmissions, and fairness with small trade-off regarding fairness depending on the network topology.

4.2 Introduction

The IEEE 802.11 (3; 4; 13) wireless networks are widely deployed. Therefore, many challenges of the wireless medium are addressed by research especially to improve the performance of the IEEE 802.11 DCF (Distributed Coordination Function), which is the basic operation of

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the medium access control (MAC) defined in IEEE 802.11. One major challenge is the hidden terminal problem which significantly degrades the performance of DCF because it results in high collision rates.

When a collision occurs, some stations other than the destination may be able to successfully receive one of the collided packets. Reasons include the capture effect and hidden terminal problem because of different locations of stations, existing obstacles like walls and doors, and interferences. Accordingly, and different than other proposals, we would like to investigate whether non-hidden stations could help each other retransmit faster whenever possible to enhance the performance of the 802.11 wireless local area networks (WLANs). In this chapter, we propose a new simple protocol that modifies 802.11 DCF, is backward compatible, and works over the 802.11 PHY to achieve such goal. We present an analytical model to study the throughput performance of the new scheme and validate that model via simulation using Opnet Modeler. We also evaluate the new scheme using Opnet with and without capture effect for different topologies. Results show gains of retransmissions, throughput, fairness, delay, and packet drops with a small trade-off regarding fairness in some scenarios.

The rest of the chapter is organized as following. In section 4.3 we provide background information about the IEEE 802.11 DCF and hidden terminal problem, and then related works are discussed in section 4.4. In section 4.5, we provide detailed description of the proposed protocol. We then present an analytical model to analyze the throughput performance of the proposed scheme in section 4.6. Simulation results are given in section 4.7 to validate the analysis model and provide performance evaluation of the proposed scheme. Finally, conclusions are in section 4.8.

4.3 Background

In this section, we first introduce the IEEE 802.11 DCF, and then we discuss the the hidden terminal problem.

4.3.1 IEEE 802.11 DCF

The IEEE 802.11 standard defines two mechanisms for DCF which are based on Carrier Sense Multiple Access/Collision Avoidance (CSMA/CA). In basic operation, a station that has a packet to transmit will do so if the medium is sensed idle for a period of distributed interframe space (DIFS). Otherwise, the station will go into backoff where the Binary-Exponential-Backoff (BEB) procedure is used. The station chooses a number of time slots to wait before trying to transmit again. The number, or the backoff counter, is selected from the range $[0, CW]$, where CW is called the contention window and is initially set to CW_{min}. The station decrements its backoff counter by one for every slot time the medium is sensed idle. When the backoff counter reaches zero, the station transmits its packet. Upon receiving a data frame, the destination responds by sending back an acknowledgment (ACK) frame after a short interframe space (SIFS) time. The ACK frame has a higher priority because SIFS is the shortest interframe space (IFS) used in DCF. The packets transmitted carry the time needed to complete the transmission of a packet and its acknowledgement. This time is used by all other stations to defer their access to the medium and is called NAV, Network Allocation Vector. Collisions occur when two or more stations are transmitting at the same time, or when the ACK frame is not received after a timeout period. With every collision, the transmitting station will double its CW unless it reaches a maximum limit CW_{max}, and selects a new backoff counter from the new range. The process is repeated until the packet is successfully transmitted or is dropped because a retry limit is reached.

In RTS/CTS operation, a station uses control frames to contend for the channel before transmitting data frames, i.e. data frames are free of collision. When the backoff counter reaches zero, the transmitter starts by sending RTS frame to the receiver who then replies with CTS if RTS frame is received successfully. The durations of RTS and CTS frames are used to reserve the channel long enough to exchange the following data frame and its acknowledgement. Fig. 4.1 illustrates the RTS/CTS operation in a fully connected (no hidden nodes) network WLAN.

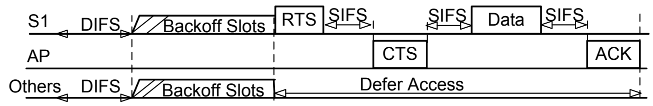

Figure 4.1 RTS/CTS operation without hidden nodes

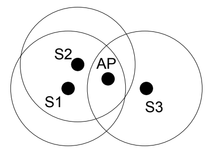

Figure 4.2 Hidden terminal problem

4.3.2 Hidden Terminal Problem

Using the wireless medium, a station is not able to hear frames transmitted by another station when they are out of range. Such phenomenon is referred to as the hidden terminal problem, and significantly degrades the performance of 802.11 DCF because it results in high collision rates. An example is shown in Fig. 4.2 where S1 and S2 are within range, and are hidden from S3. Just like when all stations are within range, collisions occur because of equal backoff values used by different nodes. However, the hidden terminal problem adds another type of collisions as shown in Fig. 4.3. Here, S1 and S3 are contending for the channel with S1 backoff value is smaller than that of S3. Accordingly, S1 starts to transmit its RTS frame to the AP (access point). Unfortunately, S3 is unaware of S1's transmission and thus does not freeze its backoff counter. S1's RTS frame would not experience a collision only if S3's backoff counter reaches zero after the start of the response frame, i.e. a CTS frame from the AP. However, here S3 backoff counter reaches zero sometime before the end of S1's transmission, and thus S3 starts transmitting its RTS frame. As a result, a collision occurs at the AP and both station S1 and S3 would timeout and then double their contention windows.

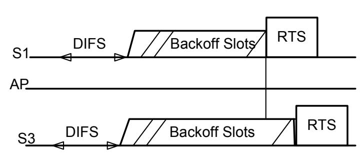

Figure 4.3 Collision due to hidden terminal problem

A special situation occurs when S3 starts transmitting RTS frame at the same time the AP may start transmitting CTS frame. Accordingly, S1 would start transmitting a data frame. However, S3 would time out and begin backoff procedure. As a result, S3 may attempt to retransmit while S1 is transmitting the data frame resulting in a collision of the data frame. Consequently, data frames are not collision-free with RTS/CTS operation when hidden terminal problem exists.

4.4 Related Work

There are few analytical models for wireless networks with hidden terminals like (14; 15; 16). In (17), the authors analyze the effect of hidden terminal on the throughput of a WLAN AP. They find that hidden terminal problem reduces the network throughput by 50%, and the capture effect (receiving one of the collided frames correctly under some conditions (18; 19; 20; 21; 22)) can enhance the performance by $(10 - 15)\%$. Capture effect adds to the complexity and cost of wireless devices, and thus is mostly not implemented. In (23), a study of the effect of hidden terminal problem in a multi-rate 802.11b network for both basic and RTS/CTS methods is provided. The study shows that although RTS/CTS method does not help against hidden nodes for rates higher than $1Mbps$ and $2Mbps$, it is recommended for all rates since it alleviates the packet collisions.

Different approaches are proposed to reduce the effect or/and the number of hidden nodes. First, in many protocols like 802.11 DCF (3), the RTS/CTS exchange is used to mitigate the hidden terminal problem. Second, the use of centralized scheduling like 802.11 PCF (3) would help. However, scheduling is not attractive because of its higher complexity, centralized control, and overhead of control packets which increase latency and reduce throughput. Third, increasing the transmission power or the carrier-sensing range may reduce the number of hidden nodes. In (22), the authors define a set of conditions for removing the hidden terminal problem for 802.11 ad-hoc and infrastructure networks: 1) the use of a sufficiently large carrier-sensing range, and 2) the use of capture effect which is referred to as the "Restart Mode" of the receiver. The authors show that one of these conditions alone is not sufficient; both conditions are required. Moreover, the work assumes that there are no significant obstructions in the network. In general, such approaches could be undesirable for energy-efficiency reasons, and would increase the exposed nodes problem in overlapping WLANs and ad-hoc networks. In addition, it may not be feasible due to different limits like available power levels, obstacles, and regulations. On the contrary, power control schemes (24; 25) could result in increasing the number of hidden nodes. Fourth, multi-channel approaches (26) mitigate the effect of hidden stations. These approaches require more transceivers and channels, and more complex MAC protocols. Fifth, busy tone protocols (27; 28) require a central node or a separate channel to transmit a special signal to inform other nodes that there is an ongoing data transmission. Finally, using new MACs and backoff algorithms, adapting the contention parameters, and broadcasting helpful information are used (many of which do not consider the hidden-terminal problem). In (29), each station broadcasts its backoff time in data frames to achieve fairness with its neighbors, and a multiplicative increase/linear decrease backoff algorithm is used. In (30), an impatient backoff algorithm is proposed to enhance the fairness level toward the nodes in the middle of an ad-hoc network. In contrast to all existing approaches, impatient nodes decrease their backoff upon a collision or losing contention, and increase it upon a successful transmission using an exponential instead of a uniform random backoff. The authors assume slotted system where synchronization is achieved, and propose to use reset messages to address the issues of small backoff values when there are many collisions and high backoff values when there are no collisions.

4.5 The Proposed Scheme

In the following, we first explain the motivation behind the proposed scheme. Then we show the details of the new protocol, and discuss implementation issues.

4.5.1 Motivation

With the IEEE 802.11's distributed operation of DCF, stations compete for the channel using a random access scheme. Hence, there are always collisions whose level increases with the number of contending stations, and the existence of hidden terminal problem. Different approaches were proposed to enhance DCF by adjusting contention parameters and the backoff procedure. However as discussed in related work (Section 4.4), they do not eliminate the hidden terminal problem, or even do not consider it.

When a collision occurs because of hidden terminal problem, some stations other than the destination may be able to successfully receive one of the collided packets. The same scenario may occur if there is a bad channel between the transmitter and the destination, like existing noise at destination, while there is a good channel between the transmitter and some stations other than the destination. In the presence of hidden nodes, we would like to investigate if non-hidden stations could help each other for retransmitting collided frames to enhance the performance of infrastructure WLANs. Such cooperative retransmission is expected to be faster since with DCF a non-collided station mostly transmits earlier than collided stations that double their CW. First, we propose a new simple protocol that modifies 802.11 DCF, is backward compatible, and works over the 802.11 PHY to achieve such goal. Then, we evaluate the proposed protocol via simulation.

4.5.2 Description of the New Scheme

We distinguish between two types of transmission opportunities (TXOPs) as shown in Fig. 4.4. First, a normal TXOP (NTXOP) occurs when a stations starts to transmit a data frame after the required DIFS, or EIFS, and backoff periods. Second, a compensating TXOP (CTXOP) occurs when a station starts to transmit after the current NTXOP by SIFS period. Also, each station maintains locally a table, called CTABLE, of other stations that may need to be assigned CTXOPs. When a station (say S2) overhears an RTS frame or a data from another station (say S1) sent to the AP, it adds an entry (the MAC address) of the frame transmitter (S1) to its CTABLE if no such entry exists. A station (S2) drops an existing entry from local CTABLE when overhearing an ACK frame sent to another station (S1) whose MAC address is equal to that entry. Note that a station is not required to wait for ACK frames after RTS or data frame to add/remove an entry to/from its CTABLE.

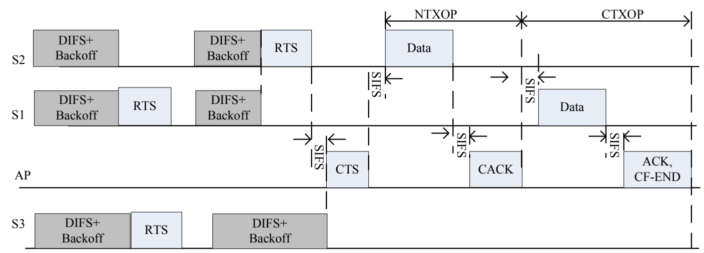

Figure 4.4 The proposed scheme

Fig. 4.4 illustrates the new scheme. Here, only S3 is hidden from both S1 and S2. After DIFS and backoff periods following DCF operation, both S1 and S3 transmissions overlap resulting in a collision. Since S2 overheard S1's data frame, it adds S1's MAC address to its CTABLE. After backoff, S2 transmits without interference, and at the same time informs the AP that S1 has a collided packet to transmit by including S1's MAC address in the transmitted data frame. The AP responds by sending back an ACK to S2 while piggybacking the AID of S1 in this ACK frame (CACK frame in Fig. 4.4). Upon receiving the ACK frame, S2 remove the entry of S1 from its CTABLE, S1 removes the entry of S2 from its CTABLE if exists, and S1 recognize that it is assigned a CTXOP. Thereafter, S1 sends a data frame after a period of SIFS to the AP who then replies with an ACK (last frame in Fig. 4.4). When overhearing the ACK, S2 removes S1's MAC address from its CTABLE, and all stations continue their contention for the channel.

For reasons like power saving (energy will be consumed for every bit transmitted or received), an 802.11 station first receives the MAC header of a frame and then receives the payload only if the frame is destined to that station. This behavior is not changed by the new scheme as only headers information is needed. Also, the helping station does not reserve the channel for a CTXOP, but the AP does so using the duration value of the CACK frame. Since duration is not known in advance, it is set to the time required to transmit a frame with maximum possible length and lowest rate. If needed (duration reserved is longer than CTXOP), the AP sends an ACK+CF-END frame instead of ACK frame in the CTXOP so that all stations reset their NAV values to start contention. On the other hand, the AP sends a CF-END if the helped station did not start transmitting after PIFS. Finally, when a station

Octets:	2	2	6	4
	Frame Control	Duration	RA	FCS

Figure 4.5 ACK frame

Octets:	2	2	6	2	4
	Frame Control	Duration	RA	CAID	FCS

Figure 4.6 CACK frame

gets a CTXOP, it does not reset its CW value and it uses its current backoff counter for the next frame in order to maintain adapting to congestion levels.

4.5.3 Capture Effect

Capture effect (18; 19; 20; 21; 22) allows receiving one of the collided frames correctly under some conditions, and thus would enhance the throughout of the network while decreasing the fairness level. Our scheme is expected to improve the performance of WLANs with or without the hidden terminal problem when capture effect is enabled since more than one station is included in a collision; using the proposed scheme, those transmissions not captured still can be helped as different stations would capture different frames depending on the distance and environment between each receiver and different transmitters.

4.5.4 Implementation Issues

4.5.4.1 CACK frame

A CACK is a new ACK type with a format shown in Fig. 4.6, and adds only one field, named CAID, to that standard ACK frame shown in Fig. 4.5. CAID represents the AID of the station that is assigned a CTXOP following the current NTXOP. The 16-bit AID is used because of its smaller size compared to that of the 48-bit MAC address, and thus reducing the extra time required.

To distinguish between ACK and CACK frames, we used the fact that all bits $B8$ to $B15$ except for $B12$ in the Frame Control field of IEEE 802.11 control frames are always set to $'0'$. In our implementation, we selected $B10$ to be set to $'1'$ for CACK. Note that a CTS frame also can be used with the same modifications to implement a CACK. The new scheme is fully backward compatible since CACK is of known type and subtype, and will not be used to acknowledge data frames from stations that do not implement the new scheme.

4.5.4.2 Data frames

Data frames are not changed. AIDs cannot be used here becuase a non-AP station maintains only its own AID. Hence, the 48-bit "Address4" of the IEEE 802.11 data frame's header can be used by a station to inform the AP about a collided station.

4.6 Analysis

Here, we present an analytical model for an infrastructure network where hidden terminal problem exists. First, we provide general analysis for the network throughput, and then focus on a specific topology.

In our analysis, we assume saturation conditions where users always have some data packets to transmit. In addition, we assume ideal channel with collisions being the only source of errors. Finally, we consider RTS-CTS operation of DCF.

4.6.1 Channel State

As shown in Fig. 4.7, generally the channel can be in one of three states: *idle*, *success*, and *collision*. Idle state refers to the time spent in backoff; i.e. no transmission from any user. On the other hand, a transmission can be a collision or a successful one. While the success state refers to the time used for sending a data frame correctly, the collision state refers to the time needed to send a data frame incorrectly due to overlapping with at least another transmission.

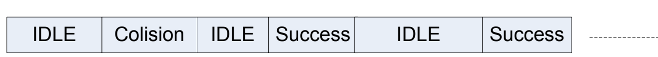

Figure 4.7 Channel state

Each idle period consists of a number of time slots. Fig. 4.8 illustrates the time spent in a collision or a successful transmission in IEEE 802.11 DCF with RTS-CTS enabled. A successful state occurs when there is only one user transmitting at a time, or when there is a collision with one RTS frame is captured. However, a collision happens when more than one user are transmitting at the same time and none of the transmitted RTS frames can be captured.

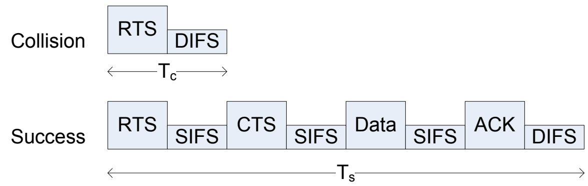

Figure 4.8 Success and collision times in DCF with RTS-CTS

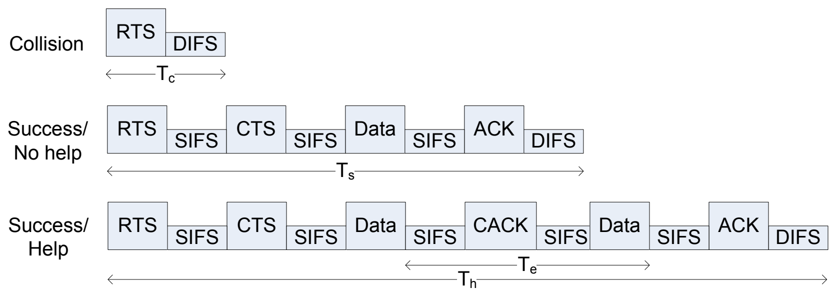

Figure 4.9 Success and collision times in DCF with RTS-CTS when the new scheme is enabled

When the proposed scheme is enabled, the channel also can be in one of the three states as discussed above. However, a successful transmission can be of two possible lengths. The successful transmission can be of the same length as that in a normal DCF operation, or may occur as shown in Fig. 4.9. The normal operation occurs when the transmitting user has an empty CTABLE (i.e. no one to help), and the new type happens when the transmitting user has a non-empty CTABLE (i.e. at least one other user to help).

4.6.2 Throughput and Throughput Gain

Consequently, the average throughput of DCF (S_{DCF}) can be described by the formula:

$$S_{DCF} = \frac{P_{success}L}{P_{idle}\sigma + P_{success}T_s + P_{collision}T_c} \tag{4.1}$$

where L is the average data frame size, $P_{success}$ is the probability of a successful transmission, $P_{collision}$ is the probability of collision, and P_{idle} is the probability of idle periods, σ is the time slot length, T_s is the time of a successful transmission, and T_c is the time of a collision.

Since the new scheme does not change the backoff counter after a CTXOP, the average throughput of the new scheme (S_H) is given by equation (4.2) where P_{help} is the probability

$$
\begin{aligned}
S_H &= \frac{P_{success}P_{help}(2L) + P_{success}(1 - P_{help})L}{P_{idle}\sigma + P_{success}(1 - P_{help})T_s + P_{success}P_{help}T_h + P_{collision}T_c} \\
&= \frac{P_{success}(1 + P_{help})L}{P_{idle}\sigma + P_{success}(1 - P_{help})T_s + P_{success}P_{help}T_h + P_{collision}T_c} \qquad (4.2)
\end{aligned}
$$

that the transmitting station would help another user, and T_h is the time of a successful transmission when a help occurs. Therefore, $P_{success}P_{help}$ is the probability of a successful transmission with help taking place, and $P_{success}(1 - P_{help})$ is the probability of a successful transmission without any help.

The following equations define different times, see Fig. 4.8 and Fig. 4.9.

$$T_s = T_{RTS} + T_{CTS} + T_{Data} + T_{ACK} + 3SIFS + DIFS \qquad (4.3)$$

$$T_c = T_{RTS} + DIFS \qquad (4.4)$$

$$
\begin{aligned}
T_h = T_{RTS} + T_{CTS} + T_{Data} + T_{CACK} + T_{Data} + \\
T_{ACK} + 5SIFS + DIFS \\
= T_s + T_{Data} + T_{CACK} + 2SIFS \qquad (4.5)
\end{aligned}
$$

We also define T_e as the extra time when a successful transmission with help occurs.

$$T_h = T_s + T_e \qquad (4.6)$$

i.e. $T_e = T_{Data} + T_{CACK} + 2SIFS$.

Then, S_H becomes:

$$S_H = \frac{P_{success}(1 + P_{help})L}{P_{idle}\sigma + P_{success}T_s + P_{success}P_{help}T_e + P_{collision}T_c} \qquad (4.7)$$

From equations (4.1) and (4.7), we can find the throughput ratio R (i.e. $\frac{S_H}{S_{DCF}}$) and the throughput gain defined by $G \times 100\%$ where G is $\frac{S_H - S_{DCF}}{S_{DCF}}$ or $R - 1$

$$R = \frac{(1 + P_{help})T}{T + P_{success}P_{help}T_e} \qquad (4.8)$$

$$G = \frac{P_{help}(T - P_{success}T_e)}{T + P_{success}P_{help}T_e} \qquad (4.9)$$

where $T = P_{idle}\sigma + P_{success}T_s + P_{collision}T_c$.

From equations (4.8) and (4.9), it follows that:

1. Since $(T - P_{success}T_e) > 0$, then $G \geq 0$. In other words, there is no degradation of the network throughput; i.e. the throughput of the network is at least equal to that of DCF without the new scheme ($R \geq 1$).

2. There is no gain when help is not possible; i.e. $G = 0$ and $R = 1$ when $P_{help} = 0$. Such scenario occurs when the topology of the network does not allow for nodes to help each other. An example is a network of fully connected nodes with capture effect not utilized.

3. Note that $(T - P_{success}T_e)P_{help} < (T + P_{success}P_{help}T_e)$. Accordingly, the throughput gain is always less than 100% (i.e. $0 \leq G < 1$, and $1 \leq R < 2$).

4.6.3 Throughput Analysis

To find the network throughput, we need to solve different probabilities (P_{idle}, $P_{success}$, $P_{collision}$, and P_{help}). We use the analysis model of (2) as it is known to be simple and correct. To summarize, the analysis model is based on solving the non-linear system of equations (4.10) and (4.11).

$$\tau = \frac{2(1-2\rho)}{(1-2\rho)(W+1) + W\rho(1-(2\rho)^r)} \tag{4.10}$$

$$\rho = 1 - (1-\tau)^{n-1} \tag{4.11}$$

where n is the total number of contending users, W is CW_{min}, r is the maximum backoff stage ($CW_{max} = 2^r W$), τ is the probability that a station transmits in any slot time, and ρ is the conditional probability that the transmitted packet will collide. Accordingly, P_{tr} is the probability of a transmission, and P_s is the probability that a transmission is successful given that there is exactly one transmitter

$$P_{tr} = 1 - (1-\tau)^n \tag{4.12}$$

$$P_s = \frac{n\tau(1-\tau)^{n-1}}{P_{tr}} \tag{4.13}$$

Then

$$P_{success} = P_{tr}P_s \tag{4.14}$$

$$P_{idle} = 1 - P_{tr} \tag{4.15}$$

$$P_{collision} = P_{tr}(1 - P_s) \tag{4.16}$$

Now we consider an infrastructure network with one AP and a number of stations that are positioned to be in two groups as shown in Fig. 4.10. In this topology, stations of different groups are hidden from each other, and stations of the same group are non-hidden. There are n stations in the first group and m stations in the second group.

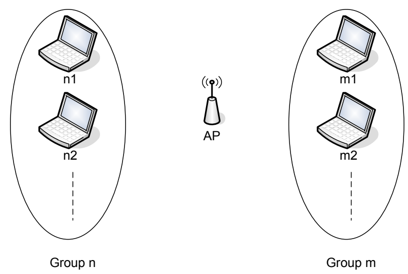

Figure 4.10 Two groups of hidden nodes

Consequently, each group may have different probabilities and therefore different performance. In the following we will be using terms like $P_{success,n}$ for the probability of a successful transmission of stations in group n, and so on (the same way will be used for all other parameters like $P_{success,n}$, $P_{help,n}$, $S_{DCF,m}$, $S_{H,n}$, and so on).

The analysis model described in (2) cannot be applied directly. This is because equation (4.11) is accurate only for a fully connected network. For the model we are considering in Fig. 4.10, we use two Markov chain models (one for each group) with the the following equations:

$$\tau_n = \frac{2(1 - 2\rho_n)}{(1 - 2\rho_n)(W + 1) + W\rho_n(1 - (2\rho_n)^r))} \tag{4.17}$$

$$\rho_n = 1 - (1 - \tau_n)^{n-1}(1 - \tau_m)^{mT_1} \tag{4.18}$$

$$\tau_m = \frac{2(1-2\rho_m)}{(1-2\rho_m)(W+1)+W\rho_m(1-(2\rho_m)^r))} \tag{4.19}$$

$$\rho_m = 1-(1-\tau_m)^{m-1}(1-\tau_n)^{nT_1} \tag{4.20}$$

where T_1, shown in Fig. 4.11, is the time, normalized to the number of time slots in the equations or $\frac{2T_{RTS}+2SIFS}{\sigma}$, during which a transmission from a hidden node may start during the ongoing transmission. We consider that a collision occurs whenever there is an overlap between transmitted frames from different stations. The non-linear system of equations (4.17)-(4.20) can be solved numerically.

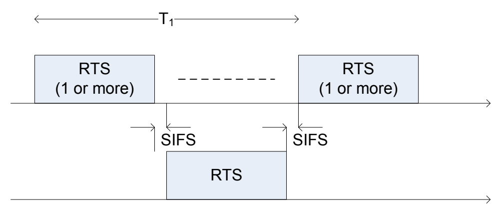

Figure 4.11 Time during which a collision due to hidden stations may occur

Accordingly, we can find the following probabilities

$$P_{tr,n} = 1-(1-\tau_n)^n \tag{4.21}$$

$$P_{tr,m} = 1-(1-\tau_m)^m \tag{4.22}$$

$$P_{s,n} = \frac{n\tau_n(1-\tau_n)^{n-1}(1-\tau_m)^{mT_1}}{P_{tr,n}} \tag{4.23}$$

$$P_{s,m} = \frac{n\tau_m(1-\tau_m)^{m-1}(1-\tau_n)^{nT1}}{P_{tr,m}} \tag{4.24}$$

Hence,

$$P_{idle,n} = 1-P_{tr,n} \tag{4.25}$$

$$P_{success,n} = P_{tr,n}P_{s,n} \tag{4.26}$$

$$P_{collision,n} = P_{tr,n}(1-P_{s,n}) \tag{4.27}$$

$$P_{idle,m} = 1-P_{tr,m} \tag{4.28}$$

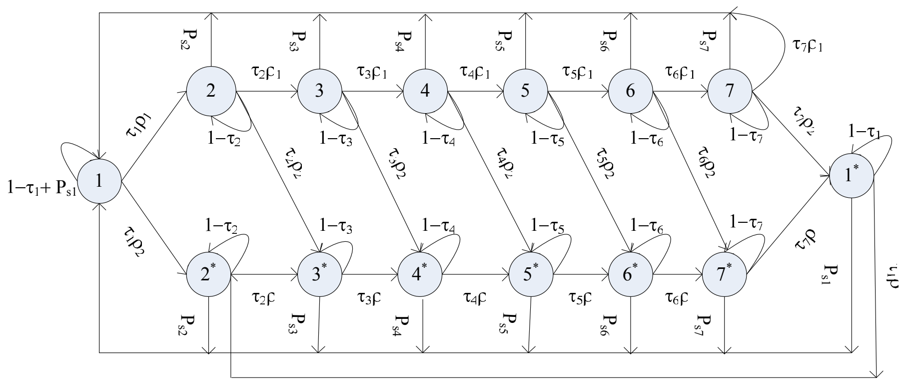

Figure 4.12 Markove chain model

$$P_{success,m} = P_{tr,m} P_{s,m} \tag{4.29}$$

$$P_{collision,m} = P_{tr,m}(1 - P_{s,m}) \tag{4.30}$$

Therefore, the throughput per group can be found using equation (4.1) for DCF with the new scheme disable ($S_{DCF,n}$, and $S_{DCF,m}$), and equation (4.7) for DCF with the new scheme enabled ($S_{H,n}$, and $S_{H,m}$). Then, the total network throughput for DCF without the new scheme is found by

$$S_{DCF} = S_{DCF,n} + S_{DCF,m} \tag{4.31}$$

and the total network throughput for DCF with the new scheme enabled is found by

$$S_H = S_{H,n} + S_{H,m} \tag{4.32}$$

In the following subsection, we compute the probability of help which is needed for the computation of $S_{H,n}$ and $S_{H,m}$, and hence S_H.

4.6.4 $P_{help,n}$, and $P_{help,m}$

We model each DCF station using a Markov chain as shown in Fig. 4.12. Again, a different model is used for stations of different groups. In this model, we attempt to separate collisions due to transmissions of the same group from collisions due to transmissions of the hidden group.

In the figure, the state numbers (states are represented by circles) represent the backoff stages. The first backoff stage is the first attempt to transmit a packet, or when CW is drawn from the range $[0, CW_{min} - 1]$. After each collision, the station moves to the next backoff stage with the new range $[0, 2^i CW_{min}]$ where i is the number of the backoff stage. When i is more than r (the maximum allowed backoff stage) where no change is made to the contention window's range. Here, we assume $r = 5$, and the maximum number of retries is 7. Also in Fig. 4.12, τ_i is the probability of transmission when the station is at state i, P_{si} is the probability of successful transmission when the station is at state i, ρ_1 is the conditional probability of collisions due transmissions of the same group, and ρ_2 is the conditional probability of collisions due to transmissions of the hidden group. Finally, we use two different types of each backoff stage. A station cannot be helped if it is in state $i \in (1, 2, ..., 7)$. On the other hand, the station may be helped if it is in state $i \in (1^*, 2^*, ..., 7^*)$.

$$\tau_i = \frac{2}{1 + 2^{i-1} CW_{min}} \tag{4.33}$$

$$P_{si} = \tau_i (1 - \rho) \tag{4.34}$$

$$\rho_{n,1} = 1 - (1 - \tau_n)^{n-1} \tag{4.35}$$

$$\rho_{n,2} = (1 - (1 - \tau_m)^{mT_1})(1 - \tau_n)^{n-1} \tag{4.36}$$

$$\rho_{m,1} = 1 - (1 - \tau_m)^{m-1} \tag{4.37}$$

$$\rho_{m,2} = (1 - (1 - \tau_n)^{nT_1})(1 - \tau_m)^{m-1} \tag{4.38}$$

Note that $\tau_{i^*} = \tau_i$, $P_{si} = P_{si^*}$, $\rho_m = \rho_{m,1} + \rho_{m,2}$, and $\rho_n = \rho_{n,1} + \rho_{n,2}$. Also, we already can calculate these probabilities using the values of τ_n and τ_m found in subsection 4.6.3.

We now solve the Markov chain model in Fig. 4.12 in general for both groups. By letting π_i be the probability of being in state i, the following holds in the Markov chain model

$$\sum_{i=1}^{7} (\pi_i + \pi_{i^*}) = 1 \tag{4.39}$$

$$\begin{aligned} \pi_i &= \pi_{i-1} \tau_{i-1} \rho_1 + \pi_i (1 - \tau_i) \\ &= \frac{\tau_1 \rho_1^i}{\tau_i} \pi_1, i = 2, 3, ..., 7. \end{aligned} \tag{4.40}$$

$$\begin{aligned}\pi_{i^*} &= \pi_{i^*-1}\tau_{i-1}\rho + \pi_{i-1}\tau_{i-1}\rho_2 + \pi_{i^*}(1-\tau_i) \\ &= \frac{\pi_{i^*-1}\tau_{i-1}\rho + \pi_{i-1}\tau_{i-1}\rho_2}{\tau_i}, i = 2^*, 3^*, ..., 7^*.\end{aligned} \tag{4.41}$$

$$\begin{aligned}\pi_1 &= \sum_{i=2}^{7}(\pi_i P_{si} + \pi_{i^*} P_{si}) + \pi_{7^*} P_{s7} + \pi_7 \tau_7 \rho_1 \\ &\quad +\pi_1(1-\tau_1) \\ &= \frac{\sum_{i=2}^{7}(\pi_i P_{si} + \pi_{i^*} P_{si}) + \pi_{7^*} P_{s7} + \pi_7 \tau_7 \rho_1}{\tau_1 - P_{s1}}\end{aligned} \tag{4.42}$$

$$\begin{aligned}\pi_{1^*} &= \pi_{7^*}\tau_7\rho + \pi_7\tau_7\rho_2 + \pi_{1^*}(1-\tau_1) \\ &= \frac{\pi_{7^*}\tau_7\rho + \pi_7\tau_7\rho_2}{\tau_1}\end{aligned} \tag{4.43}$$

Solving the above equations

$$\pi_{1^*} = \alpha\pi_1 \tag{4.44}$$

where $\alpha = \rho_2 \frac{\rho^6+\rho^5\rho_1+\rho^4\rho_1^2+\rho^3\rho_1^3+\rho^2\rho_1^4+\rho\rho_1^5+\rho_1^6}{1-\rho^7}$. Hence, the probability of every state can be expressed as a function of π_1 using equations (4.41) and (4.40). Thus we first find π_1 using equation (4.39) or (4.42), and then we calculate the probability of all other states.

Now we illustrate how to estimate the probability of help. Assume that a station of group n can help another station within the same group with the probability η_n. A station can be helped only when all the following conditions are true: 1) the station is in state $i \in 1^*, 2^*, ..., 7^*$, 2) the station was not helped while in the current state, and 3) the station was not helped in another state $j \in 1^*, 2^*, ..., 7^*$ and moved to the current state $i \neq j$ after collisions due to only transmissions from stations of the same group. Then assuming a constant probability of help and for given state i^*, we approximate η_n by π_i^* - $\pi_i^*\eta_n$ - $\sum_{j=1, j\neq i}^{7} \pi_j^*\eta_n$ (the third part of this expression is the approximated one). Then summing up over all states, we get

$$\begin{aligned}\eta_n &= \sum_{i=1}^{7} \eta_{n,i^*} \\ &= 1 - \frac{\sum_{i=1}^{7} \eta_{n,i}\pi_{i^*}}{1+\sum_{i=1}^{7} 7\pi_{i^*}}\end{aligned} \tag{4.45}$$

Then

$$P_{help,n} = 1-(1-\eta_n)^{n-1} \tag{4.46}$$

Table 4.1 Network Parameters

Parameter	Value	Parameter	Value
Slot Time	$20\mu s$	*MAC ACK Size*	14 *Bytes*
SIFS	$10\mu s$	*MAC CTS Size*	14 *Bytes*
DIFS	$50\mu s$	*MAC RTS Size*	20 *Bytes*
CW_{min}	32	*PLCP Overhead*	$192\mu s$
CW_{max}	1023	*DCF MAC Overhead*	28 *Bytes*
Control Rate	$1Mbps$	*Short Retry Limit*	4
Data Rate	$11Mbps$	*Long Retry Limit*	7

where η_{n,i^*} is the probability of help given state i^*, and $(1-\eta_n)^{n-1}$ is the probability that all other $n-1$ stations cannot be helped. Note that there is no need to include the hidden nodes, i.e. stations in group m, since they cannot be helped by any station in group n.

Similarly, we can estimate η_m and then

$$P_{help,m} = 1 - (1-\eta_m)^{m-1} \tag{4.47}$$

4.7 Performance Evaluation

This section presents the simulation we used to evaluate the performance of the proposed scheme and compare it to that of 802.11 DCF. We implemented the new scheme with the commercial Opnet Modeler 11.5.A (1) by modifying the Opnet 802.11 models. We consider an infrastructure network which consists of one AP and a number of stations that share a single wireless channel. Moreover, there are no channel errors; collisions are the only source of errors. For each scenario, the results are the average of 100 different runs with a different seed, which is used for the random generator, for each run. Finally, 802.11b and RTS/CTS operation are used with the parameters shown in Table 4.1.

4.7.1 Performance Metrics

For performance analysis, we use the following metrics:

1. *Throughput (S):* the total data bits transmitted successfully per the simulation time.

2. *Fairness Index (FI):* we used Jain Index (12) defined by ($FI = \frac{(\sum_{i=1}^{n} S_i)^2}{n \sum_{i=1}^{n} S_i^2}$), where n is number of stations and S_i is the throughput of station i. The closer the value of FI to 1, the better the fairness provided. We use FI to find how fair a scheme is to different DCF users.

3. *Average Delay:* the delay of a data packet is measured from the moment it was generated until it is successfully received. Only successfully transmitted packets are considered for finding the average delay.

4. *Packet drop:* number of data packets dropped due to buffers overflow, and due to reaching a retry limit.

5. *Retransmissions:* the number of retransmission attempts of each packet before it is successfully transmitted or dropped.

4.7.2 Hidden Groups without Capture Effect

Here, each scenario is an infrastructure network with one AP and a number of stations that are positioned to be in two groups, see Fig. 4.10. Stations of different groups are hidden from each other, and stations of same group are non-hidden. Each scenario is referred to as n-m, with n stations in the first group and m stations in the second group, and n is fixed while m is variable. Then we test with scenarios referred to as n-m-c, where a third group (group c) of 5 stations, which are not hidden from each other, is added to each of the previous n-m networks. However, stations of group c are arranged as following: 1) c_1 and c_2 are non-hidden from all stations in network. 2) $c_3 - \{n_2\}$. 3) $c_4 - \{n_1, m_1, m_5, m_9, m_{10}\}$. 4) $c_5 - \{m_1, m_5, m_6, m_7, m_9, m_{10}\}$. Here, x_i is station i in group x, and $x_i - \{x_j\}$ means that x_i and x_j are hidden from each other. Also, $|x|$ is used to refer to the number of stations in group x. These scenarios include a general topology of a wireless network. Results are provided in Fig. 4.13 to Fig. 4.22. In all figures, the letter "d" ("e") is used if the new scheme is disabled (enabled).

For the n-m scenarios, different measures follow the same trend for DCF with the new

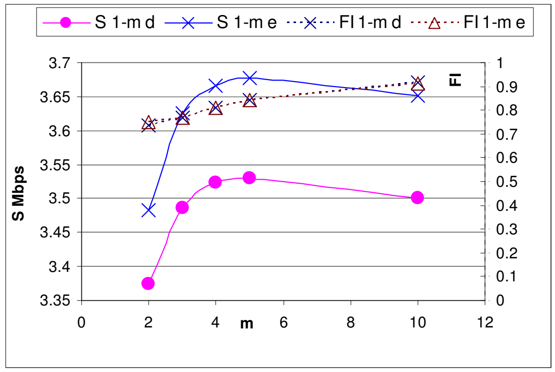

Figure 4.13 Scenario 1-m

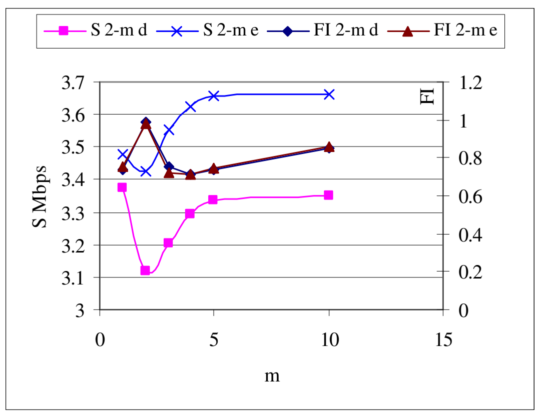

Figure 4.14 Scenario 2-m

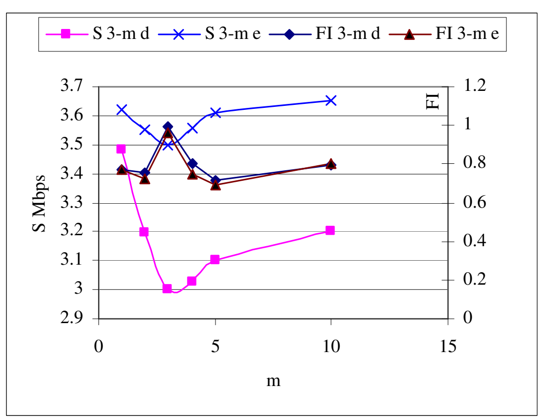

Figure 4.15 Scenario 3-m

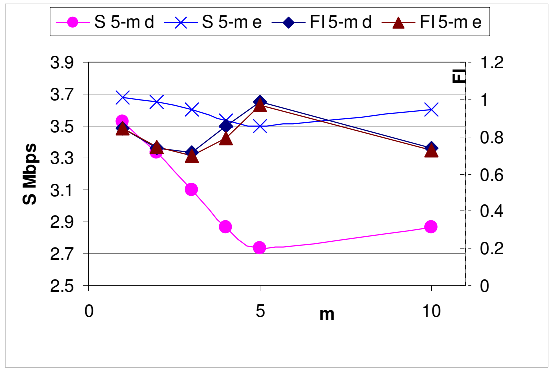

Figure 4.16 Scenario 5-m

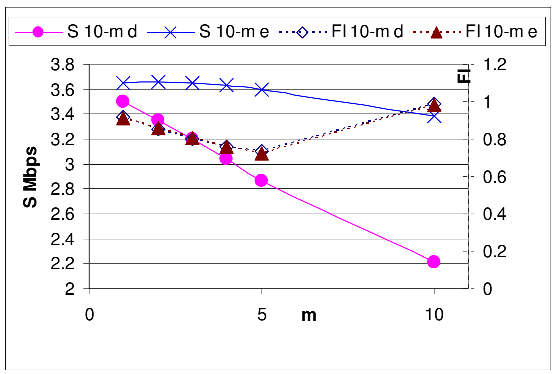

Figure 4.17 Scenario 10-m

scheme enabled or disabled; we show this for fairness and throughput in Fig.4.13 to Fig.4.17. This can be explained by the fact that CW resetting and backoff counters are unchanged after a CTXOP. Fig. 4.13 to Fig. 4.17 also show a trade-off between throughput and fairness for the n-m scenarios. The fairness gets smaller for some cases when the new scheme is enabled. This is because collided stations may retransmit before being helped due to random backoff values. However, the difference is small and FI of the new scheme is always above 0.7, and almost is the same as that of DCF for the 1-m, and 10-m scenarios. On the other hand, fairness is always enhanced for the n-m-c scenarios, Fig. 4.18 and Fig. 4.19, where there is higher probability of being helped before retransmitting using contention due to more general relations (not just two groups).

Fig. 4.13 to Fig. 4.19 illustrate that throughput is always enhanced. The minimum (maximum) gains (%) are about 3.2 (4.3), 3.1 (11.1), 3.9 (17.4), 4.1 (27.8), 4.3 (52.7), 1.8 (6.4),

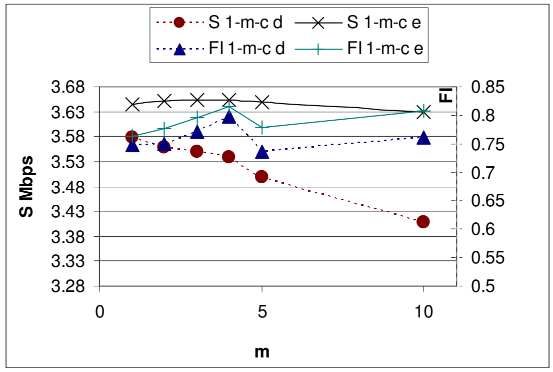

Figure 4.18 Scenario 1-m-c

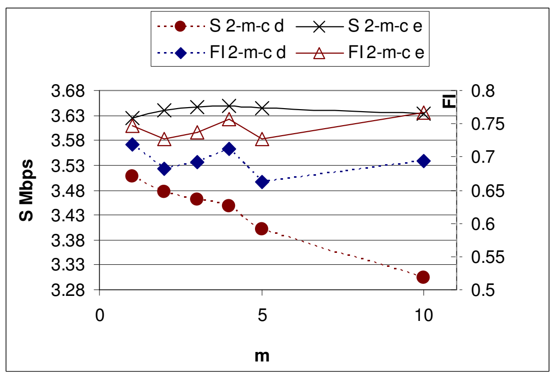

Figure 4.19 Scenario 2-m-c

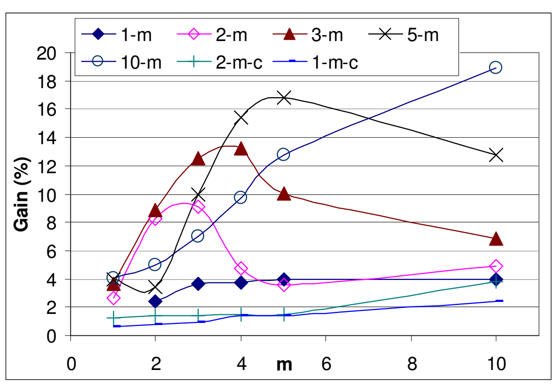

Figure 4.20 Delay

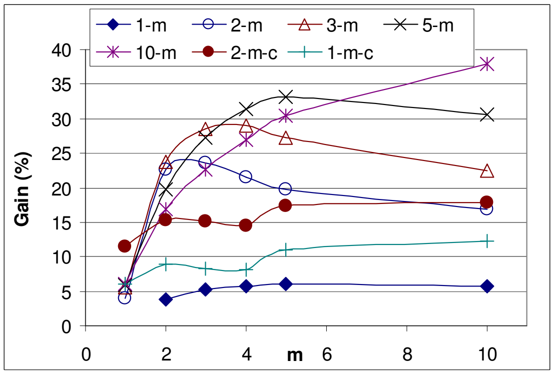

Figure 4.21 Retransmissions

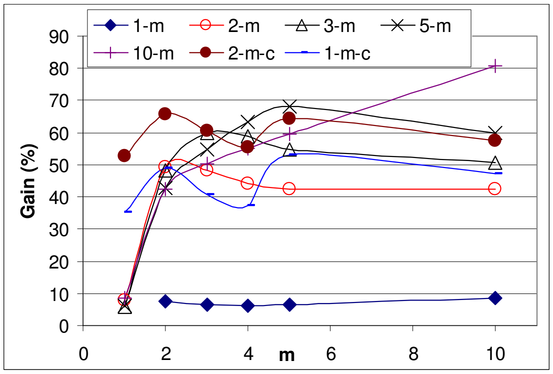

Figure 4.22 Packet drop

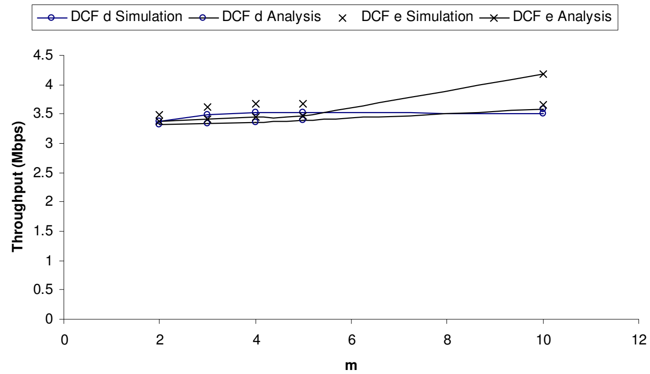

Figure 4.23 Scenario 1-m

3.3 (9.9) for the 1-m, 2-m, 3-m, 5-m, 10-m, 1-m-c, and 2-m-c scenarios respectively. Also, the throughput is always above 3.4$Mbps$ when the new scheme is enabled, and may reach 2.2$Mbps$ otherwise for the n-m networks. In addition, throughput of n-m-c networks is always above 3.6$Mbps$ with the new scheme but keeps decreasing otherwise. Delay, retransmissions, and drops are also enhanced in all scenarios, Fig. 4.20 to Fig. 4.22. Gains come from the fast retransmissions as shown in Fig. 4.21. The performance of the new scheme is affected by number of stations in each group. For the n-m networks, the gain (all measures except FI) increases until a maximum value, and then decreases until it reaches a saturated value. This is explained by the fact that the probability of collisions due to hidden nodes decreases when $|n|$ is small compared to $|m|$ (or $|m|$ is small compared to $|n|$).

4.7.3 Analysis Model Validation

In this subsection, we validate the analysis model presented in section 4.6. We compare simulation results of different n-m scenarios explained in subsection with results we get from equations of our analysis model in Fig. 4.23-Fig. 4.27. As it can be seen in these figures, our analysis model can predict the throughput performance for different scenarios with the new scheme enabled or disabled.

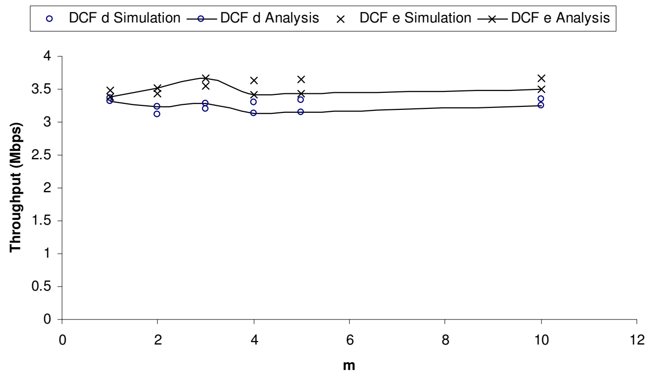

Figure 4.24 Scenario 2-m

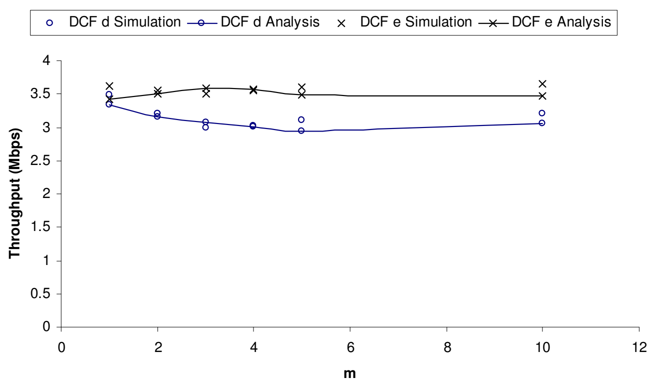

Figure 4.25 Scenario 3-m

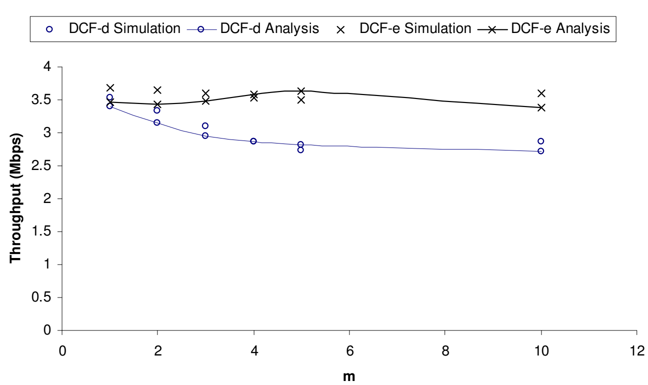

Figure 4.26 Scenario 5-m

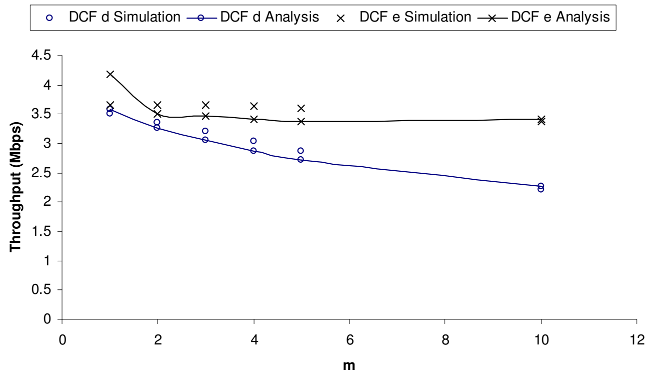

Figure 4.27 Scenario 10-m

4.7.4 Random Scenario with Capture Effect

When considering capture effect for scenarios in previous subsection, collected results showed similar gains but higher values of different measures in both schemes. Therefore, we do not show those results. We also randomly generated a network of 30 stations positioned around the AP which is in center of an area of $420 \times 420m^2$. In addition, a signal can be captured if received power is at least 10 times greater than received power of any other one, and SNR requirement is met according to the model used in Opnet. Also, each station follows an ON/OFF model: each period is $Exponential$(0.375 seconds), traffic is generated during the ON period with $Exponential$(r seconds), and a packet is 1024 bytes. Changing r allows for testing the network with different loads.

Results are given in Fig. 4.28. For very small loads, there are almost zero collisions and the number of transmitters, and so helpers, is smaller. Thus no improvement is seen for such loads. However, improvements start at about loads of 14% for throughput and fairness, and at about 5.3% for all other measures. The gains (except FI which continues to increase) increase with load until a maximum value, and then start to decrease. The decrease is because when loads are higher, collisions due to hidden and non-hidden nodes also gets higher (our proposed algorithm does not change collisions), and also more packets are buffered at different stations (more delays and drops due to long waiting).

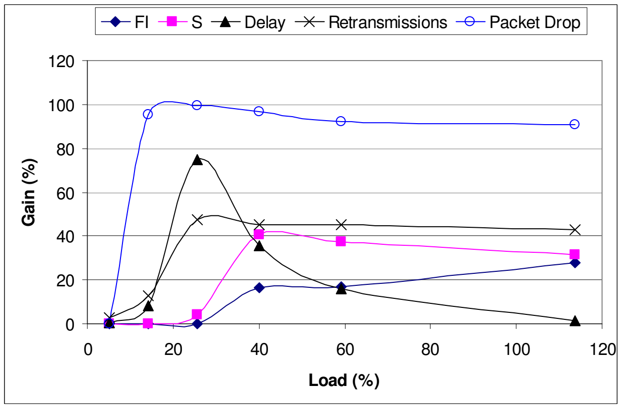

Figure 4.28 Performance gain for random scenario with capture effect

4.8 Conclusions

We proposed a new protocol for 802.11 WLANs to take advantage of the hidden terminal problem by allowing non-hidden stations to assist each other retransmit faster whenever possible. The new scheme is a modification to DCF, is backward compatible, and works over the 802.11 PHY. We also presented an analysis model to calculate the saturation throughput of the new scheme and compare it to that of DCF. We evaluated the proposed scheme and validated the analytical model via simulation which was conducted using Opnet Modeler for different scenarios. Results showed that the new scheme improves the throughput, delay, packet drop, fairness, and retransmissions. The performance gain comes from cooperative retransmissions that are faster than that used in DCF where a collided station doubles its CW. In addition, results showed a trade-off between throughput and fairness only in some scenarios. Further work includes investigating performance enhancements using different design issues like having the AP decide when not to allow stations to assist each other, and using help information to update backoff counters and CW.

CHAPTER 5. Maintaining Priority among IEEE 802.11/802.11e Devices

Modified from a paper submitted to the IEEE Transactions on Mobile Computing (TMC)

Haithem Al-Mefleh [1,3], J. Morris Chang [2,3]

5.1 Introduction

In chapter 3, we proposed a simple distributed management scheme, called Non-Zero Acknowledgement (NZ-ACK), to mitigate the influence of legacy DCF on EDCA performance in networks consisting of both types of users without any modifications to legacy users. In chapter 3, we used an intuition approach to address the main challenges of NZ-ACK: when to issue NZ-ACK frames, and how long should be the duration of a NZ-ACK frame. In such approach, we only considered the number of users of each type and the utilization required by each EDCA user. Also, the approach presented in chapter 3 requires that the AP keeps a buffer (called virtual buffer) for every EDCA user and maintains that buffer according to the user requirements. Maintenance of buffers includes adding and dropping virtual packets.

In this chapter, we revise and extend the work presented in chapter 3. We modify NZ-ACK protocol. First, virtual buffers are no longer needed. Second, we provide a model for solving the main challenges of NZ-ACK such that we maintain the priority of EDCA users. We include contention parameters (the contention window), number of users, and transmission activities of both types of users. Third, we consider different ratios of DCF and EDCA users in evaluation.

This chapter starts with the analytical model in section 5.2. Then, evaluation of the new approach is provided in 5.3. Finally, 5.4 gives collision remarks. Refer to chapter 3 for

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background information and problem statement.

5.2 Analysis

In this section, we address how the QAP determines when to issue NZ-ACK frames, and how long is the duration of a NZ-ACK frame.

Assume there are n legacy DCF users in the network. For every user i ($i = 1, ..., n$), let X_i be a discrete random variable representing the number of backoff slots selected following a uniform distribution in the range from 0 to $W - 1$ (i.e. $X_i \sim U[0, W-1]$, and W is the minimum CW for DCF). Note that:

$$f_{X_i}(x) = P(X_i = x) = \frac{1}{W};\ x = 0, 1, ..., W-1 \tag{5.1}$$

In addition, note that all X_i random variables are identical and independent.

$$f_{X_1 X_2 ... X_n}(x_1, x_2, .., x_n) = f_{X_1}(x_1) f_{X_2}(x_2) ... f_{X_n}(x_n) \tag{5.2}$$

The minimum backoff value selected by any DCF user is also a random variable X_{DCF}:

$$X_{DCF} = min(X_1, X_2, ..., X_n) \tag{5.3}$$

The distribution function $f_{X_{DCF}}(x)$ can be found by finding the probability $P(X_{DCF} = x)$, for each $x = 0, 1, 2, ..., W - 1$.

$$\begin{aligned}
&P(X_{DCF} = x) = \\
&\quad P(X_1 = x, X_2 \geq x, X_3 \geq x, X_4 \geq x, ..., X_n \geq x) \\
&+P(X_1 > x, X_2 = x, X_3 \geq x, X_4 \geq x, ..., X_n \geq x) \\
&+P(X_1 > x, X_2 > x, X_3 = x, X_4 \geq x, ..., X_n \geq x) \\
&\quad +... \\
&+P(X_1 > x, X_2 > x, X_3 > x, X_4 > x, ..., X_n = x)
\end{aligned} \tag{5.4}$$

After some processing of equations (5.1), (5.2), and (5.4), it can be found that:

$$P(X_{DCF} = x) = \begin{cases} \frac{g^n - y^n}{W^n} & ;\ x = 0, 1, ..., W-1 \\ 0 & ;\ otherwise \end{cases} \tag{5.5}$$

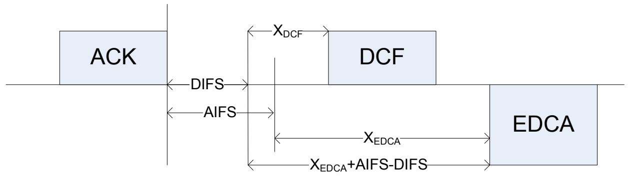

Figure 5.1 Expected CW_{min}

where $g = W - x$, and $y = W - x - 1$. Therefor, the expected value of X_{DCF} can be calculated:

$$\begin{aligned} E[X_{DCF}] &= \sum_{x=0}^{W-1} xP(X_{DCF} = x) \\ &= \frac{\sum_{x=1}^{W-1} xg^n - \sum_{x=1}^{W-1} xy^n}{W^n} \\ &= \frac{1}{W^n} \sum_{x=1}^{W-1} x^n \end{aligned} \tag{5.6}$$

These formulas also can be used to find X_{EDCA_j} for each EDCA access category j ($j = 0, 1, 2, 3$; where access category 3 has the highest priority).

$$E[X_{EDCA_j}] = \frac{1}{W_j{}^n} \sum_{x=1}^{W_j - 1} x^{n_j} \tag{5.7}$$

where W_j is the minimum CW for access category j. Accordingly, we define $E[X_{EDCA}]$ as the maximum of all $E[X_{EDCA_j}]$, $AIFS$ as the corresponding $AIFS_j$, and W^* as the corresponding W_j. Since we are interested in maintaining service guarantees to the real time (voice and video) access categories, we only consider access categories 3 and 2.

Consequently, we set the duration of an NZ-ACK frame to:

$$d = \begin{cases} 0 & ;\ D \leq -1 \\ 1 & ;\ -1 < D < 1 \\ \lceil D \rceil & ;\ D < W^* \\ W^* & ;\ D \geq W^* \end{cases} \tag{5.8}$$

where

$$D = E[X_{DCF}] - E[X_{EDCA}] + DIFS - AIFS \tag{5.9}$$

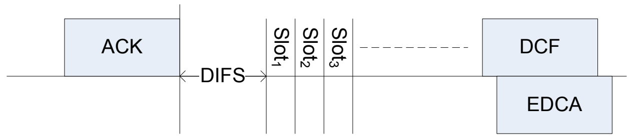

Figure 5.2 Channel access after an ACK with AIFS=DIFS

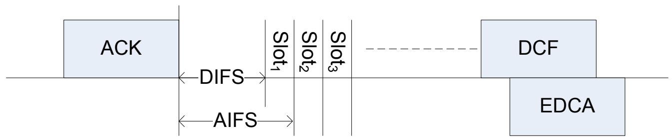

Figure 5.3 Channel access after an ACK with AIFS>DIFS

This guarantees that EDCA users access the channel before DCF users, and is explained in Fig. 5.1.

NZ-ACK frames are issued with the probability that a DCF user access the channel as early as any of EDCA users. We refer to this probability as Φ. To find Φ, we use λ_{DCF} to represent the probability that no DCF user access the channel in a slot after an ACK. Also, we use λ_{EDCA_j} to refer to the probability that no EDCA user of access category j access the channel in a slot after an ACK.

$$\lambda_{DCF} = (1 - \frac{1}{W})^n \tag{5.10}$$

$$\lambda_{EDCA_j} = (1 - \frac{1}{W_j})^{n_j} \tag{5.11}$$

Accordingly, let α_{DCF} be the probability that at least one DCF user access the channel in a slot after the ACK frame, and α_{EDCA_j} be the same probability but for EDCA user of access category j. In addition, let σ be the probability that no user access the channel in a given slot after the ACK frame.

$$\alpha_{DCF} = 1 - \lambda_{DCF} \tag{5.12}$$

$$\alpha_{EDCA_j} = 1 - \lambda_{EDCA_j} \tag{5.13}$$

$$\sigma = \lambda_{DCF}\lambda_{EDCA_j} \tag{5.14}$$

Assuming the case where AIFS=DIFS as shown in Fig. 5.2, Φ can be found by:

$$
\begin{aligned}
\Phi &= \alpha_{DCF^1}(\lambda_{EDCA_j^1} + \alpha_{EDCA_j^1}) \\
&+ \alpha_{DCF^2}(\lambda_{EDCA_j^2} + \alpha_{EDCA_j^2})\sigma_1 \\
&+ \alpha_{DCF^3}(\lambda_{EDCA_j^3} + \alpha_{EDCA_j^3})\sigma_1\sigma_2 \\
&+ \ldots \\
&+ \alpha_{DCF^i}(\lambda_{EDCA_j^i} + \alpha^i_{EDCA_j})\sigma_1\ldots\sigma_{i-1} \\
&+ \ldots \\
&+ \alpha_{DCF^k}(\lambda_{EDCA_j^k} + \alpha_{EDCA_j^k})\sigma_1\ldots\sigma_{k-1} \\
&= \alpha_{DCF} + \alpha_{DCF}\sigma + \ldots.. + \alpha_{DCF}\sigma^{k-1} \qquad (5.15)
\end{aligned}
$$

where i refers to the slot number, and k is the minimum of W and W_j. The first term in equation (5.15) is the probability that a DCF user access the channel in the first slot after an ACK frame. The second term in the equation is the probability that a DCF user access the channel in the second slot after the ACK frame given the channel was not accessed before slot 2. This is repeated for k slots because after that the probability of accessing the channel from EDCA or DCF users becomes zero.

Processing equation (5.15) results in:

$$
\Phi = \alpha_{DCF}\frac{\sigma^k - 1}{\sigma - 1} \qquad (5.16)
$$

Now we consider the case where AIFS>DIFS as shown in Fig.5.3. The same approach used above can be followed to find Φ. Assuming AIFS is more than DIFS by one slot.

$$\begin{aligned}
\Phi &= \alpha_{DCF^1} \\
&+ \alpha_{DCF^2}(\lambda_{EDCA_j^2} + \alpha_{EDCA_j^2})\lambda_{DCF^1} \\
&+ \alpha_{DCF^3}(\lambda_{EDCA_j^3} + \alpha_{EDCA_j^3})\lambda_{DCF^1}\sigma_2 \\
&+ ... \\
&+ \alpha_{DCF^i}(\lambda_{EDCA_j^i} + \alpha^i_{EDCA_j})\lambda_{DCF^1}\sigma_2...\sigma_{i-1} \\
&+ ... \\
&+ \alpha_{DCF^k}(\lambda_{EDCA_j^k} + \alpha_{EDCA_j^k})\lambda_{DCF^1}\sigma_2...\sigma_{k-1} \\
&= \alpha_{DCF} + \alpha_{DCF}\lambda_{DCF} + \alpha_{DCF}\lambda_{DCF}\sigma + ... + \\
&\quad \alpha_{DCF}\lambda_{DCF}\sigma^{k-2} \qquad (5.17)
\end{aligned}$$

Note that the difference from equation (5.15) is that EDCA users may not access the channel in slot 1. Hence, $\sigma_1 {=} \lambda_{DCF^1}$.

After processing equation (5.19), we get:

$$\Phi = \alpha_{DCF}(1 + \lambda_{DCF}\frac{\sigma^{k-1} - 1}{\sigma - 1}) \qquad (5.18)$$

Finally, combining both cases results in:

$$\begin{aligned}
\Phi &= \alpha_{DCF} + \alpha_{DCF}\lambda_{DCF}\lambda_{EDCA_j} + \alpha_{DCF}\lambda_{DCF}\lambda_{EDCA_j} \\
&\quad +\alpha_{DCF}\lambda_{DCF}\lambda_{EDCA_j}\sigma + ... \\
&\quad +\alpha_{DCF}\lambda_{DCF}\lambda_{EDCA_j}\sigma^{k-2} \\
&= \alpha_{DCF}(1 + \lambda_{DCF}\lambda_{EDCA_j}\frac{\sigma^{k-1} - 1}{\sigma - 1}) \qquad (5.19)
\end{aligned}$$

where j is access category of the case of $AIFS{=}DIFS$, i is access category of the case of $AIFS{=}OneSlot + DIFS$, and $\sigma = \lambda_{DCF}\lambda_{EDCA_j}\lambda_{EDCA_i}$.

Now we use simulation analysis to find how to set contention window parameters for EDCA users. We use Opnet simulator for a network of IEEE 802.11b with EDCA and DCF users, all users saturated, and 1000 bytes per packet. We have conducted different scenarios with different ratios of EDCA and DCF users. Here, we show only few scenarios since the same

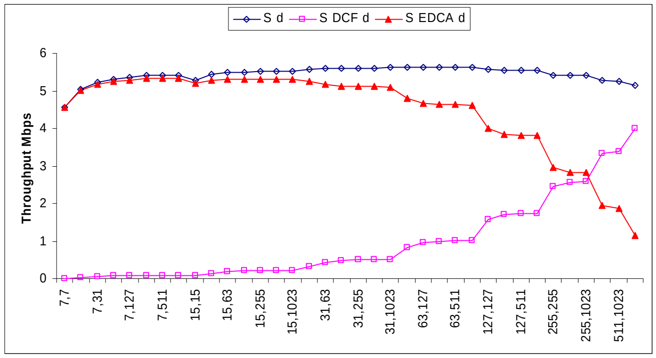

Figure 5.4 Throughput Disabled; 5 EDCA, 1 DCF

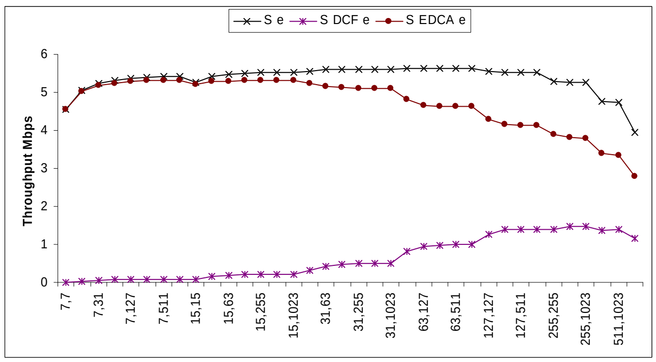

Figure 5.5 Throughput Enabled; 5 EDCA, 1 DCF

results apply. We run simulation for all possible combinations of CW_{min} and CW_{max} following the standard values.

Fig. 5.4 and Fig. 5.5 provides throughput results for a network of a number of EDCA users and a small number of DCF users. On the other hand, Fig. 5.6 and Fig. 5.7 give results of same network but with a very high number of DCF users (5 EDCA and 50 DCF users). Moreover, delay results of both networks are provided in Fig. 5.8 and Fig. 5.9. In these figure, "d" refers to NZ-ACK disabled and "e" refers to NZ-ACK enabled. Also the x-axis represents $CW_{min,EDCA}/CW_{max,EDCA}$ values.

When NZ-ACK is disabled, results illustrate that DCF users indeed affect the performance of EDCA users. A higher number of DCF users degrades both throughput and delay performance of EDCA users. In addition, a small $CW_{min,EDCA}$ degrades throughput of DCF and EDCA users as smaller $CW_{min,EDCA}$ results in a higher collision level. Also, a higher

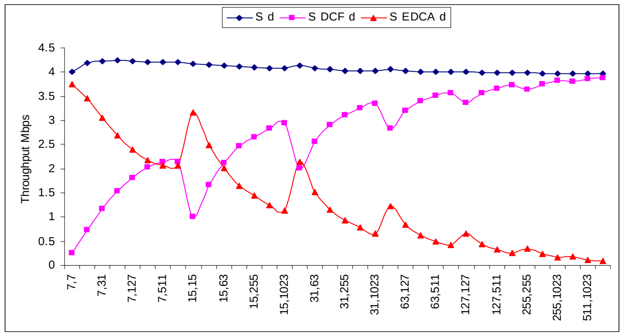

Figure 5.6 Throughput Disabled; 5 EDCA, 50 DCF

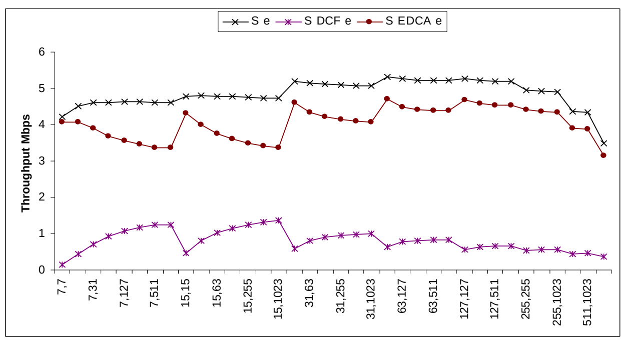

Figure 5.7 Throughput Enabled; 5 EDCA, 50 DCF

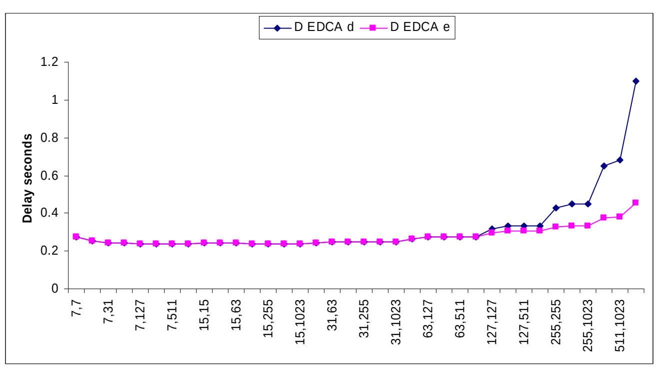

Figure 5.8 Delay; 5 EDCA, 1 DCF

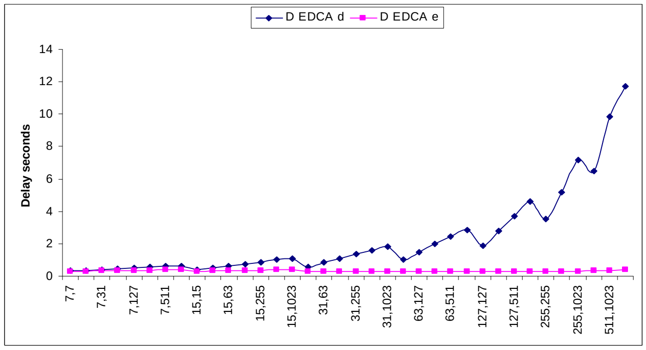

Figure 5.9 Delay; 5 EDCA, 50 DCF

$CW_{min,EDCA}$ degrades throughput and delay performance of EDCA users as they have to wait longer backoff times, and they gets a lower priority than DCF users.

On the other hand, DCF users are controlled when NZ-ACK is enabled. Also, NZ-ACK adapts to different numbers of EDCA and DCF users. As seen in these figures, we also find that the delay of EDCA is kept low and almost the same for all combinations of $CW_{min,EDCA}/CW_{max,EDCA}$ when NZ-ACK is enabled. Also, a prober $CW_{min,EDCA}$ can be selected based on number of EDCA users, and then a higher $CW_{max,EDCA}$ can be selected to provide DCF users with a higher throughput.

In summary, NZ-ACK adapts to different network conditions including number of users of each type, contention window, and AIFS/DIFS values. In addition, NZ-ACK depends on the transmission probability because NZ-ACK frames are ACK frames, i.e. NZ-ACK frames are issued only when users transmit.

5.3 Evaluation

This section presents the simulation we used to evaluate the performance of NZ-ACK (802.11 EDCA/DCF with NZ-ACK) and compare it to that of 802.11 (802.11 EDCA/DCF without NZ-ACK or any other modification). We utilized the commercial Opnet Modeler 11.5.A (1) to implement NZ-ACK by modifying the Opnet 802.11e models.

In each simulation experiment, we consider an infrastructure network that consists of stations that share a single wireless channel. We also assume a fully connected network; each station can listen to every other one in the network. Moreover, there are no channel errors; collisions are the only source of errors. In all figures, d refers to NZ-ACK being disabled and e refers to NZ-ACK being enabled.

5.3.1 Performance Metrics

For performance analysis, we use the following metrics:

1. *Throughput (S):* the total data bits successfully transmitted per the simulation time. We look at overall network throughput, EDCA throughput (throughput per EDCA), DCF throughput (throughput per DCF).

2. *Fairness Index (JF):* we used Jain Index (12; 65) defined by (5.20):

$$JF = \frac{(\sum_{i=1}^{n} S_i)^2}{n \sum_{i=1}^{n} S_i^2} \tag{5.20}$$

   Where n is number of stations and S_i is the throughput of station i. The closer the value of JF to 1, the better the fairness provided. We use JF to find how fair a scheme is to different DCF users.

3. *Delay (D):* the delay for each packet is measured from the moment that packet arrives at the MAC layer until its ACK response is received correctly. We report the average delay of EDCA packets.

5.3.2 Saturated Network

We evaluate NZ-ACK performance in a saturated network where each user always has a data frame to transmit. For this subsection, the 802.11b is used with a data rate of $11Mbps$ and 1000 bytes per packet. Results are provided in Fig. 5.10 to Fig. 5.13. We show results of two scewnarios with 5 and 10 voice EDCA users with $CW_{min,EDCA}/CW_{max,EDCA}$ of 55/511, and 117/117. The reason for selecting $CW_{min,EDCA}$ is they are optimal values for EDCA (67).

Results show that NZ-ACK controls DCF users. Thus provide higher throughputs and lower delays for EDCA no matter what is the number of DCF users. In addition, the overall

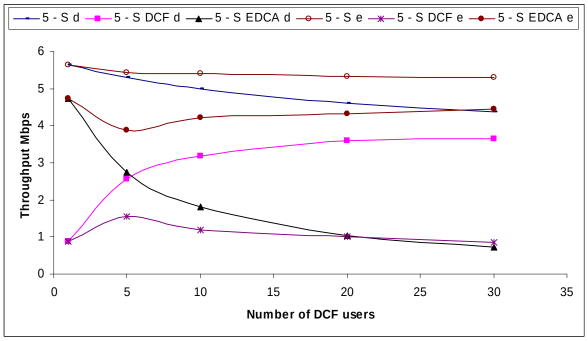

Figure 5.10 Throughput v.s. number of DCF users; 5 EDCA

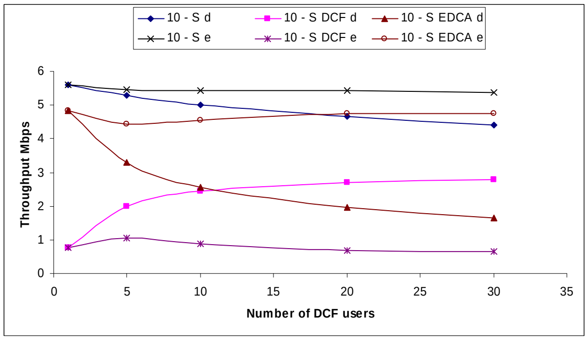

Figure 5.11 Throughput v.s. number of DCF users; 10 EDCA

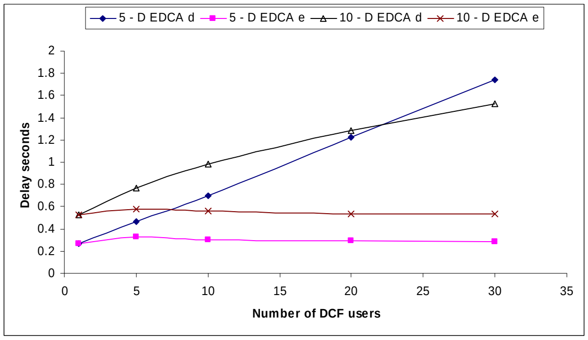

Figure 5.12 Delay v.s. number of DCF users; 5,10 EDCA

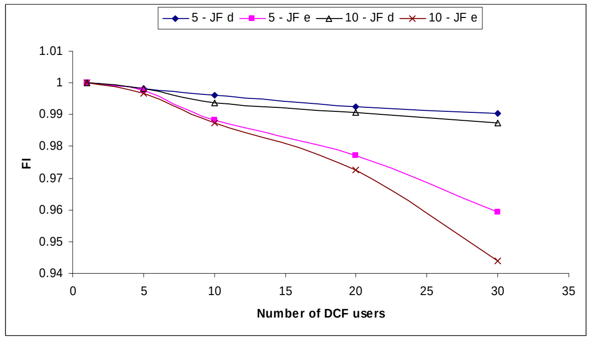

Figure 5.13 Fairness Index v.s. number of DCF users

throughput performance of the network is enhanced. Also, the throughput DCF becomes almost fixed when the number of DCF users gets very high. Actually, NZ-ACK first serve EDCA users, and the remaining bandwidth is shared among DCF users. Finally, JF results show that NZ-ACK lowers JF values as the number of DCF users increases. The reason is that NZ-ACK reserves some time for EDCA users, and that time is not used by DCF users. However, it is still considered fair enough as values of JF are always above 0.94.

5.3.3 Non-Saturated Network

Here, we evaluate the performance of 802.11 with NZ-ACK deployed in a non-saturated network and compare it to that of 802.11 with no modification. We consider an 802.11b PHY network with $11Mbps$ data rate, and CW_{min}/CW_{max} are 32/1024 (these are used by legacy users). There are 15 voice EDCA users with CW_{min}/CW_{max} of 63/511. Each voice source is modeled by an NO/OFF model with the ON and OFF periods are both exponential (0.352 seconds), and uses G.711 (silence) encoder with $64kbps$ coding rate and 160 bytes per one packet. For legacy DCF users, number of users is varied. Each legacy user is saturated with traffic of 1000 bytes per packet. DIFS of $50\mu s$ seconds is used by all users.

Fig. 5.14 shows the average total network throughput, average throughput per voice, and average throughput per DCF users. Throughputs per voice are the same for NZ-ACK and 802.11, which is also equivalent to the total voice load (not shown because it is the same value). However, DCF throughput is slightly lowered with NZ-ACK enabled. Thus the

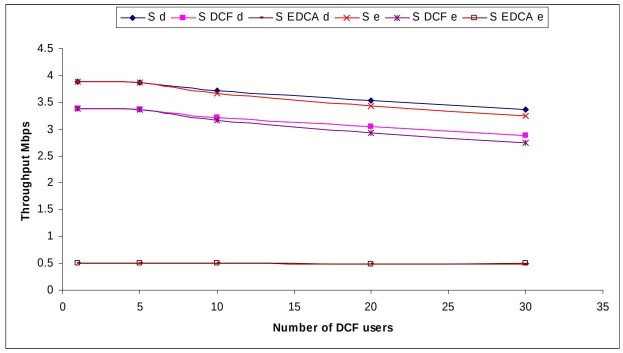

Figure 5.14 Throughput; 15 EDCA - on-off model

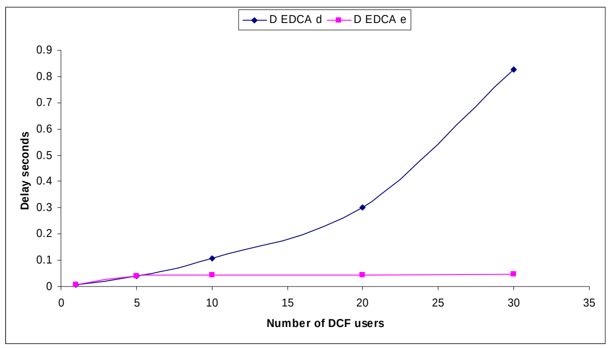

Figure 5.15 Delay; 15 EDCA - on-off model

total throughput is also slightly decreased (no change to EDCA throughput and lower DCF throughput).

In Fig. 5.15, the packet delay for voice packets is illustrated. Results show that the delay is maintained very small (a max value of 0.047761 seconds) for any number of DCF users with NZ-ACK. The performance gain is due to the fact that NZ-ACK reduces the number of contending users when issuing non-zero duration NZ-ACK frames; only EDCA users are competing for the channel when DCF users are yielding.

5.4 Conclusions

The 802.11e standard is designed to be backward compatible with the 802.11. As a result, wireless networks are expected to have a combination of both EDCA (802.11e Enhanced Distributed Channel Access) and legacy DCF (802.11 Distributed Control Function) users.

Typically, the 802.11e users who have QoS requirements are supposed to get a higher priority service than that of legacy users. However, the EDCA users' performance may be degraded because of the existence of legacy users, and therefore would get a lower priority service. The main reason for such effects is due to the fact that EDCA users are controlled through the use of different contention parameters (AIFS, CWmin, CWmax, TXOP) that are distributed via the beacon frames. In contrast, there is no control over legacy users because their contention parameters (DIFS, CWmin, CWmax) are PHY dependent, i.e. they have constant values. As a result, depending on the network status like the number of DCF/EDCA users, DCF users could achieve a higher priority and could result in high collision rates, and thus degrade the performance of EDCA users.

In this chapter, we discussed different aspects of the legacy DCF and EDCA coexistence and provided general desirable features for any mitigation solution. Based on those features, we proposed a simple distributed management scheme, called NZ-ACK, to mitigate the influence of legacy DCF on EDCA performance in networks that consist of both types of users. NZ-ACK controls legacy users by introducing a new ACK policy in which the QAP is allowed to set the duration of the last ACK in a transmission exchange to a non-zero value.

In addition, we presented strategies to determine when to issue such NZ-ACK frames, and the non-zero duration value of a NZ-ACK frame. NZ-ACK adapts to number of users, activity or load level, and contention windows of both EDCA and DCF. All the processing of NZ-ACK scheme is implemented at QAP. However, non-QAP EDCA users only are required to distinguish the new ACK policy in order to ignore the non-zero value duration included in a NZ-ACK frame. On the other hand, NZ-ACK requires no modification (i.e. fully transparent) to legacy users. Thus, NZ-ACK maintains backward compatibility.

The proposed scheme allows EDCA users to start competing directly after NZ-ACK frames. However, DCF users would defer their access to the channel according to the non-zero duration of NZ-ACK frame. Moreover, when to issue NZ-ACK frames and their duration values are determined adaptively according to network status. Thus, more resources for the EDCA users are reserved in a dynamic and distributed fashion to maintain their priority. The performance gain is due to the fact that NZ-ACK reduces the number of contending users when issuing non-zero duration NZ-ACK frames; only EDCA users are competing for the channel when DCF users are yielding. As a result, lower collision rates for both types of users are expected and thus higher throughputs, and lower delays for EDCA.

Finally, we used Opnet Modeler to evaluate NZ-ACK and compare its performance to that of IEEE 802.11. The results show that NZ-ACK outperforms IEEE 802.11 in terms of maintaining the priority of service and delay bounds of EDCA users while providing acceptable throughput for legacy users.

CHAPTER 6. Enhancing Bandwidth Utilization for the IEEE 802.16e

Submitted to the IEEE Transactions on Wireless Communications (TWC)

Haithem Al-Mefleh [1,3], J. Morris Chang [2,3]

6.1 Abstract

The IEEE 802.16 provides a promising broadband wireless access technology, and thus its efficiency is of high importance. We investigate encouraging ertPS (enhanced real time Polling Service) connections to benefit from contention, and aims at improving the network performance without violating any delay requirements of voice applications. Instead of always allocating bandwidth to ertPS connections, we propose an algorithm that adaptively uses a mix of contention and unicast polling. Moreover, as there is no differentiation between different classes in contention in the current standard, a problem occurs when ertPS connections compete with many BE (Best Effort) connections within a contention region. This would cause more delays to get the required bandwidth of ertPS. Therefor, we also propose to implement a mechanism at the SS's scheduling side to maintain the priority of the delay-sensitive ertPS connections in contention. We apply the new scheme to voice applications using the well-known ON-OFF model. Finally, we use Qualnet Modeler for the performance evaluation. Results show that the proposed scheme improves the jitter measures (with gains around 60%) and the throughput performance (about 2% to 155% of gain) without violating any latency requirements.

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6.2 Introduction

The IEEE 802.16 (37; 38) provides a promising broadband wireless access technology. Using advanced communication technologies such as OFDM/OFDMA and MIMO, the IEEE 802.16 is capable of supporting higher transmission rates, provides strong QoS mechanisms, and extends the service ranges. Moreover, the IEEE 802.16 is evolving toward supporting nomadic and mobile users (38), and using relay devices (68). Supported by these modern technologies, WiMAX (Worldwide Inter-operability for Microwave Access) is able to provide a large service coverage, a high speed data rate and QoS guaranteeing services. As a result, it may become the last mile access in suburban areas replacing DSL and cable.

IEEE 802.16 defines both the MAC (medium access control) and PHY (physical) layers of a broadband wireless network. The IEEE 802.16's MAC is a connection-oriented reservation scheme in which the subscriber stations (SSs) have to reserve any required bandwidth for transmissions. The BS (base station) coordinates reservations for all transmissions and receptions. A connection is used to uniquely identify a flow from, or to, a SS. Hence, the standard also specifies bandwidth request/allocation mechanisms for different traffic service types. Accordingly, efficient bandwidth requests, bandwidth allocations, scheduling at both BS and SSs sides, QoS architectures, admission control, and traffic's classifications are essential for 802.16 networks.

The IEEE 802.16 introduced different QoS classes which characterize different QoS requirements including UGS (Unsolicited Grant Services), rtPS (Real Time Polling Services), nrtPS (Non Real Time Polling Services), and BE (Best Effort). The IEEE 802.16e-2005 added the ertPS class as an enhancement for UGS and rtPS. Hence, it is expected that different real-time applications will be using ertPS class instead of UGS and rtPS classes. Since UGS is allocated unsolicited bandwidth and rtPS is polled periodically with higher priority, they are not affected, and thus not considered, by our study. On the other hand, different applications are using BE and nrtPS connections. For ertPS, the BS allocates bandwidth based on the negotiated characteristics. However, when used for VBR (variable bit rate) applications, such allocation may not be fully used due to the variability of traffic at a SS side. Hence, the total efficiency or utilization of the network may be degraded.

In this chapter, we consider the performance of an IEEE 802.16 network with ertPS because it is critical for VoIP applications. Thus, our work focuses on ertPS for voice applications using the well-known ON-OFF model. Such model has proven to be practical and accurate. Our main objective is to improve the network performance without violating the delay requirements of voice applications. The improvement of throughput is due to the fact that the bandwidth is allocated only when requested via bandwidth requests for an ertPS connection, and thus the wasted bandwidth is much reduced and is given to other connections that actually need it. Also, delay improvements is due to the proper use of unicast polling and the introduction of service differentiation in contention.

Since the IEEE 802.16 allows ertPS to use both contention and unicast polling, we investigate encouraging ertPS connections to benefit from contention. Instead of always allocating bandwidth to ertPS connections, we propose an algorithm that adaptively uses a mix of contention and polling. However, as there is no differentiation between different classes in contention in the current standard, a problem occurs when ertPS connections compete with many low priority connections within a contention region. This would cause more collisions, idle slots, and delays to get the required bandwidth. To overcome this problem, we propose to implement a mechanism at the SS's UL scheduler of bandwidth requests to maintain the priority of the delay-sensitive ertPS connections in contention. While UGS connections are granted bandwidth without any request, rtPS connections are polled periodically to request bandwidth, and nrtPS connections are polled but less frequently than rtPS. On the other hand, BE connections will be using contention most of the time as they are provided with no guarantees. Hence, we consider the performance of ertPS and BE connections in an IEEE 802.16e network. Finally, we use Qualnet Modeler (69) for the performance evaluation. Results show that the proposed scheme improves the jitter measures (with gains around 60%) and the throughput performance (about 2% to 155% of gain) without violating any latency requirements.

The rest of the chapter is organized as following. First, section 6.3 gives an overview of

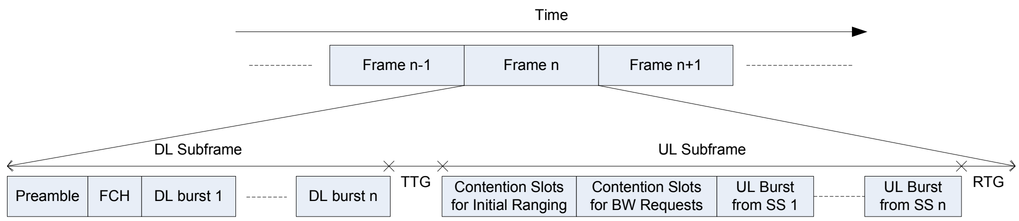

Figure 6.1 An example of IEEE 802.16 frame structure with TDD

IEEE 802.16 focussing on operations related to our work. Related work is discussed in section 6.4. Then, we illustrate the problem addressed, and we explain details of the proposed solution in section 6.5. We also evaluate our work via Qualnet 4.5 simulation in section 6.6. Finally, conclusions are provided in section 6.7.

6.3 An Overview of the IEEE 802.16 MAC

A reservation scheme is used in IEEE 802.16 networks to allow the SSs to reserve their required bandwidth from the BS. In addition, IEEE 802.16 is connection-oriented. In other words, the bandwidth requests are made based on the connection IDs (CID) which are used for identifying the traffic flows between different SSs and the BS. Therefore, SSs must establish connections with BS before transmitting any data. The BS can grant or reject the requests based on the available bandwidth and scheduling policy.

There are two types of operational modes defined in the IEEE 802.16 standard: point-to-multipoint (PMP) mode and mesh mode. An IEEE 802.16 PMP (point-to-multipoint) network, which we consider in this chapter, includes a base station (BS) and a number of subscribers (SSs) controlled by the BS. Also, there are two transmission modes: Time Division Duplex (TDD) and Frequency Division Duplex (FDD). Both UL and DL transmissions can not be operated simultaneously in TDD mode but in FDD mode. The time is divided into frames, and each frame is divided into a DL (downlink) and an UP (uplink) subframes. DL subframe is used for transmissions from the BS to SSs, and UL subframe is used for transmissions from SSs to the BS. The BS controls the network by starting each frame with maps (UL MAP and DL MAP) to indicate which and when SSs are transmitting or receiving. Fig. 6.1 shows an overview of an 802.16's frame structure operating in the Time Division Duplex (TDD) mode. The preamble is used to synchronize the BS and SSs, and time gaps (TTG and RTG) are used to give the BS and SSs enough time for the transition between transmitting and receiving operations. In addition, SSTGs (subscriber station transition gaps) separate the transmissions of the various SSs during the uplink subframe.

Consequently, the standard introduces different QoS classes which characterize different connections:

1. Unsolicited Grant Services (UGS): a fixed amount of bandwidth is granted periodically. This is proper for CBR (Constant Bit Rate) traffic.

2. Real Time Polling Services (rtPS): the SS must first request bandwidth, and the BS provide a periodic bandwidth request opportunities. This is used for VBR (Variable Bit Rate) with delay sensitivity characteristics.

3. Extended Real Time Polling Services (ertPS): a new class that was added by IEEE 802.16e to enhance the efficiency of UGS and rtPS.

4. Non Real Time Polling Services (nrtPS): like rtPS but for applications without delay requirements.

5. Best Effort (BE): no guarantees are provided for best effort users.

In addition, the IEEE 802.16 standard defines different bandwidth request and allocation schemes:

1. Unicast Polling: the BS allocates enough time in the UL for a SS to request bandwidth.

2. Piggybacking: a SSs can append the bandwidth request subheader to the regular MAC header for requesting more bandwidth.

3. Bandwidth Stealing: If SSs grant the bandwidth for pervious bandwidth requests, they can use the granted bandwidth to send another bandwidth request messages instead of transmitting data.

4. Contention: SSs contend for the channel to send bandwidth requests using a backoff procedure.

5. Unsolicited Grant: an amount of bandwidth is always granted for a connection without any bandwidth request.

Table 6.1 summarizes the poll and grant options for each scheduling service according to the IEEE 802.16/802.16e standards.

In this work, we consider the performance for SSs that are allowed to use both the contention-based polling and the unicast polling. Contention-based polling (or simply contention) follows a truncated exponential backoff procedure, and is controlled by start/end backoff values and a number of the request transmission opportunities which are broadcast by the BS. The SS transmits a bandwidth request when its backoff counter reaches zero. The SS doubles its contention window and reattempt to send the bandwidth request if no grant is received within a timeout value. Contention can be used only by BE, nrtPS, and ertPS connections and has two types. Broadcast polling is where all SSs contend for the channel, and multicast polling is where a group of SSs can participate in contention. On the other hand, the BS allocates enough bandwidth for a SS to request bandwidth using unicast polling. Such allocation simplifies the MAC operations, and provide delay guarantees. The BS provides periodic unicast polling for a connection with the period being determined using the negotiated requirements at the setup time.

6.4 Related Work

Many studies of IEEE 802.16 have been conducted via simulation and analytical modeling. These studies include the performance of different classes, optimization of contention parameters, bandwidth requests and allocation, quality of service (QoS) architectures, scheduling, and many other features of 802.16. An overview of IEEE 802.16 WiMAX is provided in (70; 71). In the following, we summarize research work relevant to our work.

In (39), a simulation study is given for rtPS, nrtPs, and BE connections in the UL. Here, it is shown that nrtPS and BE have almost the same performance, and rtPS outperforms

Table 6.1 Poll/grant options for each scheduling service (37; 38)

Scheduling service	Piggyback grant request	Bandwidth stealing	Unicast polling	Contention-based polling
UGS - Unsolicited Grant Service	Not Allowed	Not Allowed	Poll-ME bit for non UGS	Not allowed
ertPS - extended rtPS	Extended piggyback	Allowed	Allowed	Allowed
rtPS - Real Time Polling Service	Allowed	Allowed	Allowed	Not allowed
nrtPS - Non Real Time Polling Service	Allowed	Allowed	Allowed	Allowed
BE - Best Effort	Allowed	Allowed	Allowed	Allowed

nrtPS. It is also demonstrated that throughput decreases with a larger number of SSs due to the overhead of preambles and headers of transmissions. It is also illustrated that the longer the frame, the higher the average delay. Finally, it is shown that piggybacking is an efficient mechanism, and can be done when there are multiple traffic sources in the same SS. In (40), delay is analytically studied for three different simple polling schemes. However, the authors assume that all SSs are polled sequentially in every UL subframe.

In WiMAX, scheduling and QoS architectures are essential to guarantee the demanded QoS (41; 42; 43; 44). In general, scheduling and QoS architecture define the queues of data and control packets, and how they are served (frequency, priority, and weight). In (44), for example, UGS flows are served first. Then, rtPS and nrtPS connections are served with a max-min allocation of bandwidth when there are no sufficient bandwidth. If any bandwidth remains, it is given to BE connections in a round robin fashion.

The amount of bandwidth to allocate or to request is another interesting subject in 802.16 (45; 46; 47). In (45), a queue based scheduling scheme is used for rtPS and/or nrtPS connections. The allocated amount of bandwidth for a connection is based on number of packets in the buffer of that connection. Then, a delay feedback is used to predict required bandwidth during the next frame to prevent the buffer's overflow. In (46), a similar approach is proposed with two feedback parameters to calculate the bandwidth amount that should be requested. In (47), simple formulas are used to find the amount of bandwidth allocated by the BS. The allocated bandwidth is based on the connection parameters (like minimum bandwidth required), amount of bandwidth requested, and the service type (like BE and ertPS). For example, a UGS connection is always allocated all its required bandwidth without any request. Another example is to allocate the needed time to send a bandwidth request message for an rtPS (or ertPS) connection when its requested bandwidth is zero. However, this may degrade the performance because of that allocated slots that are not used while they could have been assigned to other SSs.

Contention operations are also studied for IEEE 802.16 (48; 49; 50; 51; 52; 53; 54; 55; 56; 57). In (48), an optimal number of contentions slots is derived based on objective functions to reduce the access delay. In (51), the start contention window is calculated in order to minimize a cost function. In (50), the authors extended the Markov chain in (2) to model random access in 802.16. Based on that model, an optimal fixed backoff window size is determined to minimize the delay assuming a fixed number of active SSs. Performance analysis using a Markov chain model for multicast and broadcast polling is provided in (55), and the ratio of successfully transmitted requests is derived based on the given model. In (57), It is shown that VoIP connections can have better performance when they are grouped into different multicast groups. Unicast polling is also considered. For example, (72) provides an adaptive polling scheme to increase the bandwidth utilization. Such adaptiveness would reduce the overhead of polling a user with no data; i.e. a user who will waste the bandwidth given for polling. In general, overheads include preambles, headers, unused bandwidth, and frequent signaling.

Different than other schemes, we utilize both polling and contention. In addition, we propose to use a mechanism to differentiate different service types in contention, and thus maintain the priority of delay-sensitive types while enhancing the network performance.

6.5 Details of the Proposed Solution

In this section, we first discuss the problem addressed. Then we provide details of the proposed scheme including the algorithms proposed at the BS and SSs sides.

6.5.1 Problem Statement

The IEEE 802.16e-2005 added the ertPS class which allows for unsolicited grants like UGS to save overheads of bandwidth requests. In addition, ertPS is allowed to dynamically adjust the size of the bandwidth grant like rtPS to maximize the utilization. Moreover, ertPS is allowed to use contentionbased polling to decrease the access delay and increase utilization. It is expected that different real-time applications, like VoIP, will use ertPS class instead of rtPS and UGS.

While UGS connections are granted bandwidth without any request, rtPS connections are polled periodically to request bandwidth, and nrtPS connections are polled but less frequently than rtPS. On the other hands, a large number of users are expected to use the BE class specially when considering WEB users. Moreover, a significant number of BE users would be using contention bandwidth requests most of the time since they are not guaranteed any bandwidth. Consequently, we consider the performance of an IEEE 802.16 network with ertPS and BE connections. The BS allocates bandwidth to ertPS connections based on the negotiated characteristics. For VBR (variable bit rate) applications, such allocation may not be fully used due to the variability of traffic at SSs side. Moreover, the unused bandwidth allocated to ertPS connections can be used by other connections including ertPS and BE. Hence, the performance of BE and ertPS connections, and the total efficiency or utilization of the network may be degraded. For our study, we assume that the ertPS class is used for voice applications with the well known ON-OFF model (an applicable model for voice and used by many works. More information and references can be found in [24]). Finally, Qualnet Modeler [26] will be used for the simulation study.

6.5.2 Description

The IEEE 802.16 allows ertPS to use both contention and unicast polls. Accordingly, we summarize our proposal in the following:

1. The BS side: Instead of allocating data grants, or unicast polls, to ertPS connections, we propose to have ertPS connections use contention while unicast polling them when necessary, so SSs may request bandwidth. Therefore, we propose an algorithm at the BS side to allocate bandwidth to ertPS connections to enhance the network performance without violating the maximum latency requirements. Such algorithm allows dynamically choosing the proper polling scheme, unicast polling or contention, for each SS depending on the network conditions like the number of SSs of each class.

2. The SS side: A problem occurs when ertPS connections compete with many BE connections within a contention region due to the fact that there is no differentiation between different classes in contention (all connections have the same start and end contention windows, and follow the same backoff procedure) in the current standard. This would

cause more collisions, idle slots, and delays to get the required bandwidth. Thus, the delay and throughput performance in the network may degrade. Accordingly, we need to introduce a new mechanism to increase the priority of ertPS connections in contention.

In IEEE 802.16 standard, each SS has an UL scheduler for BW requests. Therefore, in our scheme we propose that a SS schedules the start time of a bandwidth request of a BE connection one contention slot after the beginning of the contention region. The reason of using one slot is that it can be equivalent to doubling contention window (73), and that more than one slot may result in starving BE connections. On the other hand, note that there is no overhead added since no change is made to the UL map; i.e. no new information is added to the UL map. Finally, pigyybacking should be used by a SS when possible. Hence, the SS does not have to contend or wait for a unicast poll when more data is available at the SS side.

6.5.3 Utilization

In the following, we assume that:

- N is the total number of SSs.
- t_{poll} is the polling period for a connection (negotiated at connection setup time).
- M is the size of contention region in number of transmission opportunities (slots).
- t_f is the frame length.
- W is the initial contention window ($W = 2^S$, S is the backoff start power value).
- T is a predefined period which is equal to an integer number of frames.
- n_f represents a number of frames.

A simple way to find the utilization of polling and contention: $Utilization = Total\ Bandwidth - Wasted\ Bandwidth$. In the following, we find the wasted bandwidth for polling (W_p) and contention (W_c). For the traffic model for a connection at SS_i, we are assuming

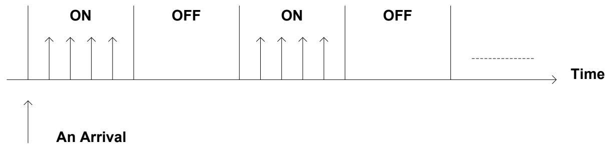

Figure 6.2 ON-OFF Model

an ON-OFF model (Fig. 6.2). The ON and OFF periods (i.e. T_{ON}, and T_{OFF}) are both exponentially distributed with mean rates of α and β respectively. In addition, packets are generated only during the ON period with a constant mean interarrival rate of λ. T_g represents the interarrival time of packets.

$$T_{ON} \sim Exp(\alpha) \tag{6.1}$$

$$T_{OFF} \sim Exp(\beta) \tag{6.2}$$

Accordingly, the average ON time t_{on} is $\frac{1}{\alpha}$ seconds, the average OFF time t_{off} is $\frac{1}{\beta}$ seconds, and the average interarrival time t_g is $\frac{1}{\lambda}$ seconds. We also define the ratio of ON time as $\zeta = \frac{t_{on}}{t_{on}+t_{off}}$, and the ratio of OFF time as $\eta = \frac{t_{off}}{t_{on}+t_{off}}$.

A polling is wasted when it is not used to request bandwidth because no data exists at the SS at the time of polling. The total ON time within a polling period can be estimated by ζt_{poll}, and the OFF time within a polling period is ηt_{poll}. Then the probability that a SS will make no bandwidth requests when polled can be found by calculating the probability of having no data arrivals within t_{poll}. First, the probability of a SS having no data within one ON period is:

$$\begin{aligned} P[T_{ON} <= t_g] &= \int_0^{t_g} \frac{1}{t_{on}} e^{\frac{t}{t_{on}}}\, dt \\ &= 1 - e^{\frac{t_g}{t_{on}}} \end{aligned} \tag{6.3}$$

hence, the probability the SS has data is φ=$e^{\frac{t_g}{t_{on}}}$. Then using $\frac{t_{poll}}{t_{on}+toff}$, the average number of ON periods within a t_{poll}, we approximate the probability that a SS has data within the polling period using:

$$\begin{aligned} \pi &= 1-(1-\varphi)^{\frac{\varsigma t_{poll}}{t_{on}}} \\ &= 1-(1-\varphi)^{\frac{t_{poll}}{t_{on}+t_{off}}} \end{aligned} \tag{6.4}$$

Thus, $N\pi$ is the total number of SSs that are expected to have data. In other words, W_p=$N(1-\pi)$ SSs are expected to waste their unicast polls.

Now we need to find an estimate of the bandwidth wastage in the contention scheme. The waste in contention is simply the idle and collided slots. Since each user selects a backoff value from the range $[0, W-1]$ using a uniform distribution, then we can find the probability that a SS transmits in a slot (τ), the probability of a successful transmission in a slot (P_s), and the probability of wasting a slot (υ).

$$\tau = \frac{1}{W} \tag{6.5}$$

We use the following can be used as a worst-case estimation (54).

$$P_s = N\frac{1}{W}(1-\frac{1}{W})^{N-1} \tag{6.6}$$

$$\upsilon = 1 - P_s \tag{6.7}$$

Consequently, the number of wasted slots can be defined as a binomial random variable X.

$$P[X = x] = \begin{pmatrix} M \\ x \end{pmatrix} \upsilon^x(1-\upsilon)^{M-x} \tag{6.8}$$

Hence, the expected number of wasted slots is W_C=υM.

6.5.4 Algorithm at BS

In this subsection, we provide details of the algorithm part to be run by the BS. In general, the BS first allocates the data grants for ertPS connections. Then, we need to compensate for contention. In other words, the algorithm maintains the probability of successful transmission of BW requests in the contention region. Thereafter, the BS allocates unicast polls of a ertPS connection if maximum delay requirement is to be violated. Finally, data grants of the low priority BE connections are allocated. In the following, a detailed description of the algorithm is provided.

The algorithm is summarized in Algorithm 1, and discussed in the following. In the algorithm, the following terms are used:

- BWR_i is the bandwidth requested by an ertPS connection.
- C_i is an an ertPS connection.
- T_c is current time.
- t_f is the frame time.
- T_i is the last time a bandwidth request is received for C_i
- n_{ertps} is the number of ertPS connections
- n_{be} is the number of BE connections.

The algorithm is described in the following.

In lines 15 to 19 of Algorithm 1, the BS allocates a unicast poll for C_i (an ertPS connection) in three conditions. First, when there is no bandwidth requested by that connection (BWR_i is 0). Second, if maximum latency requirement will be violated, or when ($T_c+t_f-T_i>=d_i$). Note that t_f is added since the actual data grant occurs in the next frame. If a data grant is allocated, then piggybacking is to be used by the corresponding SS, and thus the SS does not have to contend or wait for a unicast poll in the subsequent frames. Finally, a unicast poll is granted with a probability of π, i.e the probability the connection queue has some data to transmit.

On the other hand, the utilization of contention region may be reduced since more connections are expected to compete. Therefore, as shown by line 3 to 14 of Algorithm 1, we need to adjust contention when ($n_a>1$) where n_a refers to the number of ertPS connections who are expected to contend for the channel. In other words, n_a represents the expected number of active ertPS connections, and it is equivalent to $n_{ertps}\pi$. Starting from M and up to M_{max} ($M_{max} = W$), we find M_i that satisfies $((n_{be}+n_a)\frac{1}{M_i}(1-\frac{1}{M_i})^{n_{be}+n_a-1} >= n_{be}\frac{1}{M}(1-\frac{1}{M})^{n_{be}-1})$. In other words, we find the first M_i value that maintains the utilization of contention considering the new number of contenders $(n_{be}+n_a)$.

Algorithm 1 BS Side

Allocate data grants for ertPS
$n_a \Leftarrow \lfloor n_{ertps}\pi \rfloor$
if $n_a > 1$ **then**
    $p_s \Leftarrow n_{be}\frac{1}{M}(1-\frac{1}{M})^{n_{be}-1}$
    $i \Leftarrow M$
    while $i <= M_{max}$ **do**
        $p_i \Leftarrow (n_{be}+n_a)\frac{1}{M}(1-\frac{1}{M})^{n_{be}+n_a-1}$
        $i \Leftarrow i+1$
        if $p_i > p_s$ **then**
            break;
        end if
    end while
    $M \Leftarrow i$
end if
for $i = 1$ to n_{ertps} **do**
    if $BWR_i == 0$ AND $T_c + t_f - T_i >= d_{max}$ **then**
        Allocate a unicast polling for SS_i with probability π
    end if
end for
Allocate data grants for BE

6.6 Simulation

This section presents the simulation we used to evaluate the performance of the proposed scheme. We utilized Qualnet 4.5 (69) to implement the new scheme by modifying the IEEE 802.16 model.

As shown in Fig. 6.3, we consider a network of IEEE 802.16 with one BS and a number of SSs. In addition, each SS has one data traffic connection with a service type of BE or ertPS. The main network parameters are stated in Table 6.2. Moreover, it is worth mentioning

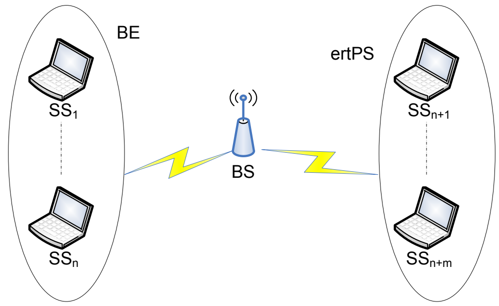

Figure 6.3 Network topology

that different main features of IEEE 802.16 are implemented including fragmentation, packing, admission control, ranging, burst profiles, Adaptive Modulation and Coding (AMC), and CRC. Each scheduler at the BS, or SS, follows a strict priority of different service types (*management* > UGS > ertPS > rtPS > nrtPS > BE). In addition, the BS uses a WFQ scheduling for fairness of each service type in the UL subframe. Finally, IP networking is used and nrtPS service type is used for routing and transport layers. These connections have higher priority as they are classified as *management* connections.

6.6.1 Traffic Characteristics

Each SS has one data connection which is a CBR or VoIP application which are provided by Qualnet modeler. A CBR source is used for every BE connection, and a VoIP source is associated with each ertPS connection. Each VoIP source is modeled by an NO-OFF model with the ON and OFF periods are both exponential with means of $0.352s$ and $0.648s$, respectively. In addition, the voice source uses silence encoder with a 160 bytes per voice packet, and each voice packet is generated every $20ms$ only during the ON period. Also, the maximum latency is set to $0.1s$ for each VoIP connection. On the other hand, a CBR application generates packets with a constant rate of $0.0007s$, and the data packet size is 1024 bytes. This allows an almost saturated BE connections providing a high effect on ertPS connections.

6.6.2 Performance Metrics

For performance analysis, we use the following metrics:

Table 6.2 IEEE 802.16 Network Parameters

Parameter	Value	Parameter	Value
PHY	IEEE 802.16 OFDMA	Channel frequency	2.4GHz
Channel bandwidth	20MHz	Transmission power	30.0 dBm
FFT size	2048	Cyclic prefix	8
MAC propagation	1μs	Duplexity	TDD
Frame duration	20ms	DL duration	10ms
Request backoff min	3	Request backoff max	15
Ranging backoff min	3	Ranging backoff max	15
T16 interval[1]	100ms	T3 interval[2]	200ms
Max request retries	16	Max ranging correction retries	16
TTG	10μs	RTG	10μs
SSTG	4μs	Service flow timeout interval	15s
DCD broadcast interval	5s	UCD broadcast interval	5s
SS wait DCD timeout interval	25s	SS wait UCD timeout interval	25s
Antenna connection loss	0.2	Antenna mismacth loss	0.3
Antenna efficiency	0.8	PHY noise factor	10.0
Symbols per DL PS	2	Symbols per UL PS	3
Preamble symbol length	1		

1- Bandwidth request timeout 2- Ranging timeout

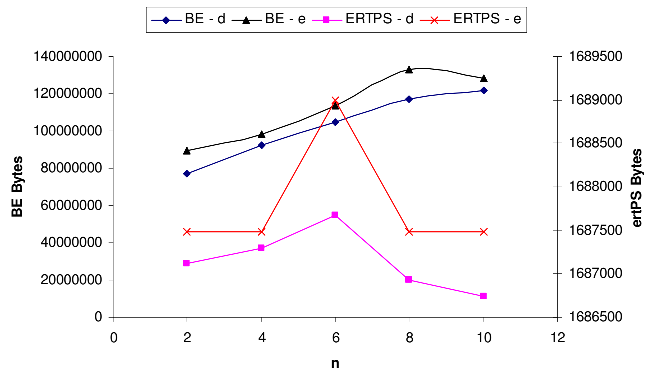

Figure 6.4 Throughput - 2 ertps SSs

1. Throughput: the throughput, in bytes, of both BE and ertPS connections.
2. Delay: the average end-to-end delay of ertPS connections in seconds.
3. Jitter: the average jitter, or delay variation, of ertPS connections in seconds.

6.6.3 Results

We start with a network where the number of ertPS connections is fixed, and the number of BE SSs is variable. Then, we fix the number of BE connections and vary the number of ertPS connections. In all figures presented, "e" refers to the proposed scheme being enabled, and "d" represents the case where it is disabled.

6.6.3.1 2 ertPS connections

Here, the number of ertPS connections is set to 2, and n refers to number of BE connections. Figure 6.4 shows the throughput of BE and ertPS connections. As illustrated in these figures, the throughput is enhanced for both types of services. While the gain is very small for ertPS, it ranges from 5% to 16.19% for BE. With the proposed scheme, bandwidth is allocated more efficiently. In other words, the new scheme reduces much of the wasted bandwidth, and allocates such bandwidth to other connections that actually require it.

Figure 6.5 shows that the new scheme enhances the delay performances except for when n = 4 (gain of −2%). In all other cases, the gain goes from 4% to 11% approximately. However,

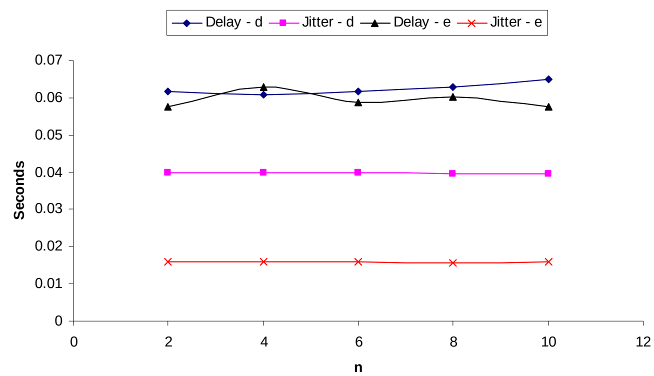

Figure 6.5 Delay and jitter - 2 ertps SSs

the maximum latency is not violated in all cases, and delay is kept very small. Moreover, the jitter performance is enhanced for all cases with gains up to around 60% as demonstrated in figure 6.5. Delay requirement is met and jitter is improved because of the use of unicat polling when the maximum latency is to be violated. In addition, an ertPS connection does not have to wait for polling once it requires bandwidth, i.e. it would send a bandwidth request using contention. Also, ertPS connections have a higher priority than that of BE connections in contention because of the introduced service differentiation in our proposed scheme.

6.6.3.2 4 ertPS connections

In this experiment, we increase the number of ertPS connections by 2. The value of n represents the number of BE connections. The throughput of BE and ertPS connections are given in figure 6.6. There is still a very small improvement in ertPS throughput. On the other hand, there is a higher gain for BE with a range of 6%−155%.

Figure 6.7 shows an improvement of delay for all values of n. The gain is from around 36% to 44%. Again, the maximum latency is not violated, and delays are kept at low values. In addition, the jitter gain is almost 63% as can be seen from figure 6.7.

Note that there is a higher gain than that of the pervious scenario of 2 ertPS in subsection 6.6.3.1. This is due to the increase of number of ertPS connections. In other words, more bandwidth is better utilized with the proposed scheme.

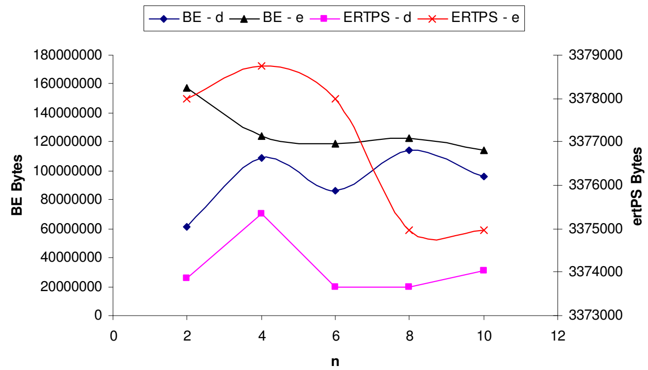

Figure 6.6 Throughput - 4 ertps SSs

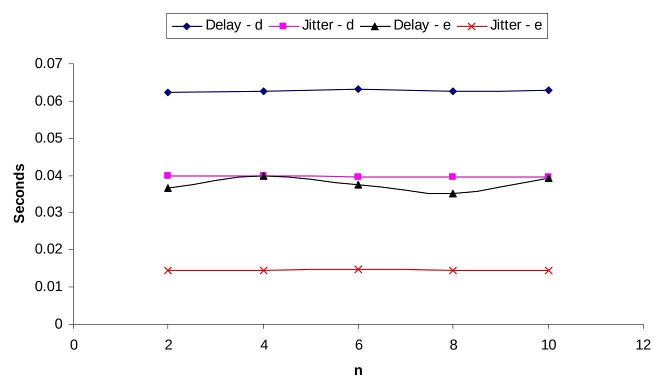

Figure 6.7 Delay ertps - 4 ertps SSs

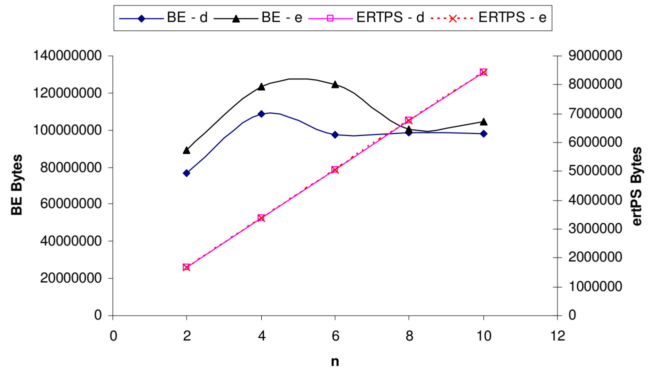

Figure 6.8 Throughput BE - 4 BE SSs

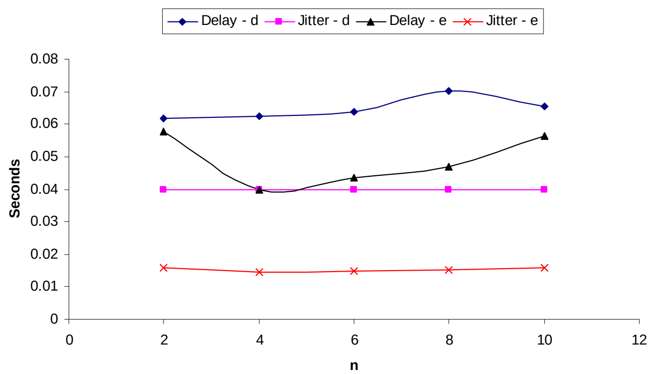

Figure 6.9 Delay ertps - 4 BE SSs

6.6.3.3 4 BE connections

Finally, we set the number of BE connections to 4 and vary the number of ertPS connections. Here, n refers to the number of ertPS connections. The throughputs of BE and ertPS connections are given in figure 6.8. While there is almost no difference in ertPS throughput, gains of BE throughput goes from around 2% to 27%.

As explained by figure 6.9, delay results show gains in all cases with values from around 6% to 36%. In addition, the jitter performance is increased as can be inferred from figure 6.9. Gains of jitter are about 60% to 63%.

As it can be seen from different results, the proposed scheme improves the throughput of the network, the throughput of ertPS and BE connections, and thus does improve the utilization of bandwidth allocation. The total throughput of the network is improved since both throughputs of ertPS and BE connections are increased. The improvement of different throughputs is due to the fact that the bandwidth is allocated only when requested via bandwidth requests for the ertPS connections, and thus the wasted bandwidth is much reduced and is given to BE connections that actually need it. In addition, the improvement occurs without violating the maximum latency required by voice applications. Moreover, the jitter is much reduced, and in almost all cases the delay is enhanced. Hence, the priority of ertPS is maintained due to the differentiation used in contention. The results illustrate the effectiveness of the proposed scheme as it adapts to different requirements and number of SSs.

6.7 Conclusions

The IEEE 802.16 provides a promising broadband wireless access technology, and is expected to replace or extend the already existing broadband communication technologies. Therefore, the IEEE 802.16 efficiency is of high importance for academia and industry. We consider voice applications using ertPS class that was introduced by the IEEE 802.16e-2005.

We investigated encouraging ertPS connections to benefit from contention to improve the network performance without violating any delay requirements of voice applications. Instead of always allocating bandwidth to ertPS connections, we proposed an algorithm that adaptively uses a mix of contention and unicast polling. However, as there is no differentiation between different classes in contention, a problem occurs when ertPS connections compete with many BE connections as the collision rate increases within the contention region. As a result, there would be more delays to get the required bandwidth of ertPS connections. Therefore, in our scheme we propose that a SS schedules the start time of a bandwidth request of a BE connection one contention slot after the beginning of the contention region. This would maintain the priority of the delay-sensitive ertPS connections in the contention region. We also applied the new scheme to voice applications using the well-known ON-OFF model. Finally, we used Qualnet Modeler for the performance evaluation. Results showed that the proposed scheme improves the jitter measures (with gains around 60%) and the throughput performance (about 2% to 155% of gain) without violating any latency requirements.

CHAPTER 7. Conclusions and Future Work

During the last few years, wireless networking has attracted much of the research and industry interest. In addition, almost all current wireless devices are based on the IEEE 802.11 and IEEE 802.16 standards for the local and metropolitan area networks (LAN/MAN) respectively. Both of these standards define the medium access control layer (MAC) and physical layer (PHY) parts of a wireless user.

In a wireless network, the MAC protocol plays a significant role in determining the performance of the whole network and individual users. Accordingly, many challenges are addressed by research to improve the performance of MAC operations in IEEE 802.11 and IEEE 802.16 standards.

We proposed and studied solutions to enhance the performance of an IEEE 802.11 WLAN and an IEEE 802.16 networks:

1. HDCF: The performance of 802.11 DCF degrades especially under larger network sizes, and higher loads due to higher contention level resulting in more idle slots and higher collision rates. We proposed HDCF to address the problem of wasted time in contention resolution of DCF via classifying stations into active and inactive ones. The objectives are to coordinate transmissions from different active stations with no collisions or idle slots, and limit the contention to newly transmitting stations. HDCF utilizes an interrupt scheme with active transmissions to enhance the fairness and eliminate, or reduce much of, the costs of contention in DCF (idle slots and collisions) without adding any assumptions or constraints to DCF.

   We provided a simple analytical description of HDCF compared to DCF. We used a simple but a well-known and an accurate model of the IEEE DCF which is presented in (2), and we started with assumptions like that used in (2). We explained how new arrivals affect the probability of collision, and how the collision level is reduced. We also showed that like DCF, HDCF operation consists of cycles such that each cycle includes on average a transmission by each user in the network. While DCF achieves this fairness property with the cost of idle slots and collisions, HDCF reduces much of such overheads, and thus is expected to enhance the throughput and fairness of the network.

In general, HDCF has the following advantages: 1) No idle slots wasted when there are no new stations trying to transmit, or no need to stop active transmissions. 2) Fairness to new stations as they can contend for the channel directly (like in DCF) without long delays as the contention cost is much smaller. 3) Stations transmit in random order without the need for a slotted channel, reserved periods, time synchronization, central control, or knowledge of number of active users.

Finally, we used Opnet to provide a simulation study for networks of two different PHYs (the IEEE 802.11b and 802.11g). In addition, the experiments considered different loads, network sizes (number of users in the network), noise levels, packet sizes, and traffic types. Results illustrated that HDCF outperforms DCF with gains up to 391.2% of throughput and 26.8% of fairness level.

2. Taking Advantage of the Existence of Hidden Nodes: When wireless users are out of range, they would not be able to hear frames transmitted by each other. This is referred to as the hidden terminal problem, and significantly degrades the performance of the IEEE 802.11 DCF because it results in higher collision rates.

   Although the problem is addressed by different works, it is not totally eliminated. Hence, we proposed a simple protocol that enhances the performance of DCF in the existence of hidden terminal problem. Opposite to other approaches, we proposed to take advantage of the hidden terminal problem whenever possible. We investigated if non-hidden stations could help each other retransmit faster to enhance the performance of 802.11 WLANs. Such cooperative retransmissions are expected to be faster since with DCF a non-collided station mostly transmits earlier than collided stations that double their backoff values. The proposed scheme modifies 802.11 DCF, is backward compatible, and works over the 802.11 PHY. We also presented an analysis model to calculate the saturation throughput of the new scheme and compare it to that of DCF.

Capture effect is an advancement in wireless networks that allows a station to correctly receive one of the collided frames under some conditions like a threshold of the received signal's SNR (signal-to-noise ratio). Thus, captures would enhance the throughout of the network while decreasing the fairness level. Consequently, we considered capture effect as it may reduce the gains of the proposed scheme, and would make it possible for the new scheme to be used even in a fully-connected WLAN where there are no hidden nodes.

Using Opnet simulation, we evaluated the new scheme with and without the capture effect for different topologies. Results showed gains of the number of retransmissions per packet, throughput, fairness, delay, and packet drops. However, there was small trade-off regarding fairness in some scenarios. Finally, simulation was used to validate the analytical model.

3. NZ-ACK: The 802.11e standard is designed to be backward compatible with the 802.11. Thus wireless networks are expected to have mix of EDCA (802.11e) and legacy DCF (802.11, 802.11b, 802.11g, and 802.11a) users. As a result, EDCA users' performance may be degraded because of the existence of legacy users, and therefore would get a lower priority of service. The main reason for such influence is due to the fact that EDCA users are controlled through the use of different contention parameters, which are distributed by a central controller. Nevertheless, there is no control over legacy users because their contention parameters are PHY dependent, i.e. they have constant values.

   We provided an insight on the effects of coexisting legacy DCF and EDCA devices, and presented general desirable features for any proposed mitigating solution. Based on these features, we then proposed a simple distributed scheme, called NZ-ACK (Non Zero-Acknowledgement), to alleviate the influence of legacy DCF on EDCA performance in networks consisting of both types of users.

NZ-ACK introduces a new ACK policy, and has the following features: 1) Simple and distributed. 2) Fully transparent to legacy DCF users, and thus backward compatibility is maintained. 3) No changes to the 802.11e standard frames formats. 4) Minimal overhead to EDCA users as all processing is at the QAP. 5) Adaptively provide control over legacy stations, and reserve more resources for the EDCA users as necessary.

Two variants of NZ-ACK were proposed. First, we used a simple intuition based on number of users of both types and expected traffic at EDCA users. This variant requires the AP to maintain virtual buffers for EDCA flows, and update these buffers depending on admission information. Second, we provided a model for solving the main challenges of NZ-ACK such that the priority of EDCA users is maintained. The model includes contention parameters, the number of users, and transmission activities of both types of users without the need for any virtual buffers or admission information.

Opnet simulation was used to evaluate both variants of NZ-ACK. Simulation results showed that NZ-ACK maintains the priority of service and delay bounds of EDCA users while providing acceptable throughput for legacy users.

4. Enhancing Bandwidth Utilization for the IEEE 802.16e: The IEEE 802.16 provides a promising broadband wireless access technology, and is expected to replace or extend the already existing broadband communication technologies. Therefore, the IEEE 802.16 efficiency is of high importance for academia and industry. We consider voice applications using ertPS class that was introduced by the IEEE 802.16e.

   We investigated encouraging ertPS connections to benefit from contention to improve the network performance without violating any delay requirements of voice applications. Instead of always allocating bandwidth to ertPS connections, we proposed that the BS follows an algorithm that adaptively allocates contention or unicast polling for an ertPS connection. Moreover, as there is no differentiation between different classes in contention, a problem occurs when ertPS connections compete with low priority connections

connections within a contention region. This would cause more delays for an ertPS connection to get its required bandwidth. Therefor, we proposed to implement a mechanism at the SS's UL scheduler of bandwidth requests to maintain the priority of the delay-sensitive ertPS connections in the contention region. We also applied the new scheme to VoIP applications using the well-known ON-OFF model. Finally, we used Qualnet Modeler for the performance evaluation. Results showed that the proposed scheme improves the jitter measures (with gains around 60%) for ertPS and the throughput performance (about 2% to 155% of gain) without violating any latency requirements.

Future Work

In the following, we identify some open issues for future work:

1. HDCF was designed with the assumption of equal weights of different users. Thus, future directions include extending HDCF to provide service differentiation among different flows.

2. NZ-ACK was designed to maintain priority for high priority flows in EDCA when some DCF users exist. Therefore, NZ-ACK can be redesigned to provide such priority among different access categories in EDCA.

3. Due to the rich features of IEEE 802.16, different problems may need investigation. First, allocation algorithms may consider a non-strict priority scheduling. Thus allowing higher throughput for the low priority classes. Second, when allocating bandwidth, the BS may consider the used data rate and energy level in addition to the class type of a connection. Third, there is a need to consider different traffic types like video. Then, guidelines may be provided to choose the right algorithm, or a generalized algorithm, for different types. Fourt, channel errors and the use of ARQ (Automatic Repeat Request) may be considered in the scheduling algorithm at BS.

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