FAIRNESS ISSUES IN MULTIHOP WIRELESS AD HOC NETWORKS

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NATIONAL UNIVERSITY OF SINGAPORE

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Abbreviation List

AAT	Average Attempts per packet
ACK	Acknowledgment
AT	Aggregate Throughput
ATDMA	Asynchronous Time Division Multiple Access
BEB	Binary Exponential Backoff
CBFAIR	Connection Based Fair backoff
CE	Channel Efficiency
CS	Carrier Sense
CTS	Clear To Send
DCF	Distributed Coordination Function
DCP	Double Contention Areas Problem
DWOP	Distributed Wireless Ordering Protocol
EDWOP	Extended DWOP
EHATDMA	Extended Hybrid ATDMA
EMLM	Enhanced Maximize-Local-Minimum fair queueing
FI	Fairness Index
FI_m	Max-min Fairness Index
HATDMA	Hybrid ATDMA
LCP	Lack of Coordination Problem
LSP	Lack of Synchronization Problem
MANET	Multihop wireless Ad Hoc Network
MBFAIR	Measurement Based Fair backoff
NAV	Network Allocation Vector
RI	Receiver Initiated
R_{min}	Minimum Throughput Rate
RTS	Request To Send
SI	Sender Initiated

Summary

A multihop mobile wireless ad hoc network (MANET) is a self-organizing and self-configuring network that can be instantly set up, and it operates without any pre-existing communication infrastructure except nodes, which themselves may move around in an arbitrary way. One of the important issues in the design of a MANET is how network bandwidth is to be shared among competing users. Fairness is one of the important properties desired in allocating bandwidth. Although much research has been done on in fairness of bandwidth allocation in the context of wireline networks, the algorithms developed in wireline networks for fair bandwidth provision cannot be easily extended to this new context. This is due to the unique characteristics of multihop wireless ad hoc networks. In this thesis, we investigate the fairness problem in multihop wireless ad hoc networks. We look into the fairness problem at two levels: the MAC/link layer and the network layer, with particular emphasis on MAC/link layer fairness, which is believed to be an important foundation for better network layer fairness.

By simulation, we demonstrate that when applied in multihop wireless ad hoc networks, the widely used MAC protocol IEEE 802.11 could suffer from a severe fairness problem: some link layer flows could seize the whole channel bandwidth (one) while others get virtually nothing (zero). Three causes leading to the one/zero fairness problem are identified: the lack of synchronization problem (LSP), the lack of coordination problem (LCP) and the double contention areas problem (DCP). Based on the analysis,

we propose a new MAC protocol named extended hybrid asynchronous time division multiple access (EHATDMA), which employs three mechanisms addressing the three problems mentioned above. Comprehensive simulations show that while various enhancements have been proposed to improve the fairness of MAC protocols in multihop wireless networks, most of them are still strongly biased towards throughput when a conflict between throughput and fairness arises. On the other hand, EHATDMA strikes a good balance between throughput and fairness. Our simulation results also reveal that the most important mechanism for improving fairness of wireless channel sharing is the non-work-conserving mechanism.

A three-dimensional Markov model is proposed to further analyze the throughput and fairness properties of IEEE 802.11 in multihop wireless ad hoc networks. Our analysis reveals that the RTS/CTS access method with the default parameters operates almost optimally in terms of saturation throughput. By extrapolating from the analytical model, we confirm the conclusion that non-work-conserving principles will improve MAC/link layer fairness.

Since end-to-end traffic in MANETs is expected to be mostly TCP-like, just as in Internet, we evaluate and compare the performance of TCP over IEEE 802.11 and three fair MAC protocols. The results show that fair MAC protocols do improve fairness of bandwidth allocation among TCP flows. In addition, they could also improve other performance aspects of TCP flows, such as stability and compatibility.

Publications

[1] **Jun He** and Hung Keng Pung, "One/Zero Fairness Problem of MAC Protocols in Multi-hop Ad Hoc Networks and Its Solution," *International Conference on Wireless Networks (ICWN)*, Las Vegas, Nevada, USA, 2003, pp. 479-485.

[2] **Jun He** and Hung Keng Pung, "A Fairer Multiple Access Protocol for Multi-hop Wireless Networks: Hybrid Asynchronous Time Division Multiple Access Protocol (HATDMA)," *IEEE Conference on Local Computer Networks (LCN)*, Bonn/Königswinter, Germany, 2003, pp. 356-365.

[3] **Jun He** and Hung Keng Pung, "Performance Modeling and Evaluation of IEEE 802.11 Distributed Coordination Function in Multihop Wireless Networks," *IEEE International Conference on Networks (ICON)*, Singapore, 2004, pp. 73-79. **(Best Student Paper)**

[4] **Jun He** and Hung Keng Pung, "Fairness of Medium Access Control Protocols for Multihop Ad Hoc Wireless Network," *accepted by Computer Networks Journal.*

CHAPTER 1

Introduction

A multihop mobile wireless ad hoc network (MANET[1]) is a self-organizing and self-configuring network that can be instantly set up, and it operates without any pre-existing communication infrastructure except nodes, which themselves may move around in an arbitrary way [1]. While these intriguing features (instant setup, infrastructure independence) enable a MANET to be deployed in many situations where traditional networks are either unavailable, infeasible or impossible — such as battle fields, disaster recovery, law enforcement, etc., they also impose huge challenges. Many fundamental issues need to be further investigated before MANETs can be applied in real life. Such issues include MAC layer protocols, routing protocols, security, etc. [2–4]. In this thesis, we are interested in the fairness of bandwidth allocation that a MANET delivers to end-users. Fairness is used to measure how entities in consideration share a resource. It is a desirable and important property for best effort service as well as for differentiated service (DiffServ) [5], in which flows belonging to the same class require bandwidth allocated to that class to be fairly share. Although much research has been done on fairness of bandwidth allocation in the context of wireline networks, the algorithms developed for wireline network fair bandwidth provision cannot be easily extended to

[1]In [1], MANET stands for Mobile Ad hoc NETwork. In this thesis, we use the acronym in a slightly different way. It stands for Multihop wireless Ad hoc NETwork (including both static and mobile networks) to emphasize the multihop property.

MANETs due to the unique characteristics of MANETs. It is hence our objectives in this thesis to: (i) investigate the factors affecting fairness of bandwidth provision, and (ii) propose solutions to improve fairness in the context of MANETs.

1.1 MAC/Link Layer Fairness and Network Layer Fairness

Fairness is a measure reflecting how a resource is shared among competing entities. In a network, the resource in consideration is usually bandwidth. It could be the bandwidth of a link or the bandwidth of the whole network. Therefore, the fairness problem in a network can be investigated at two different levels: *the media access control (MAC)/link layer* and *the network layer*. In wireline network research, much has been done on fair bandwidth provision at both levels.

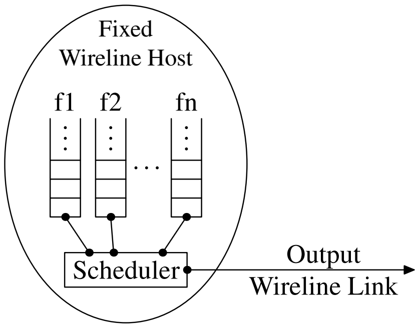
1.1.1 MAC/Link Layer (Hop-to-Hop) Fairness in Wireline Networks

At the link layer, a flow is defined as a packet stream between neighboring nodes. Several flows compete with one another to access an output link. In wireline networks, all flows competing for an output link are maintained in the same node, which has full control over the output link (Figure 1.1(a)). A scheduler in the node is used to determine the order of service so as to satisfy certain fairness criteria. The scheduler usually needs to decide:

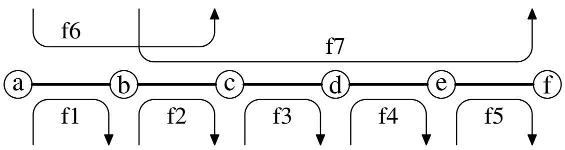
(a) which flow is to be served next;

(b) when to put a packet from the next flow into transmission.

Since in wireline networks, all competing flows reside in the same node, the scheduler has all the information needed to perform task (a). Furthermore, the scheduler is notified



(a) MAC/Link layer fairness in wireline network: local property — the scheduler in a host has full control over an output link and it also has precise and complete information of all flows competing for the output link



(b) Network layer fairness in wireline network: global property — schedulers in different hosts need to cooperate to achieve a desired fairness criterion

Figure 1.1: MAC/Link layer fairness and network layer fairness in wireline networks

immediately by the transmitter at the end of a transmission, which is the exact point in time to put the next packet into transmission (task (b)). MAC/Link fairness in wireline networks is therefore a *local property*, i.e., the scheduler does not need to consult the schedulers of other nodes to perform these two tasks. We will show later on that the locality of these two tasks no longer holds in multihop wireless ad hoc networks, which makes fair bandwidth provision far more difficult in that new context.

The classic fairness criterion in the allocation of link layer bandwidth among multiple flows is *max-min fairness.* It reflects the intuitive notion of fairness that any flow should be entitled to as much channel share as any other flow [6]. The *Generalized Processor Sharing (GPS)* policy ([7–9]) is an ideal fair scheduling discipline which exactly realizes the max-min fairness criterion. The GPS scheduler uses a fluid flow model, visiting backlogged flows in a round-robin fashion and serving each flow an infinitesimal amount of data that is proportional to the weight associated with the flow. In GPS, if N flows being served by the server (output link) have positive weights $\phi(1), \phi(2), \ldots, \phi(N)$, then the server (output link) serves $S(i, \tau, t)$ amount of data from the i^{th} flow in the interval $[\tau, t]$, such that for any flow i backlogged (the queue for this flow is not empty) in $[\tau, t]$, and for any other flow j, we have:

$$\frac{S(i, \tau, t)}{S(j, \tau, t)} \geq \frac{\phi(i)}{\phi(j)} \tag{1.1}$$

GPS ensures that non-backlogged flows get as much service as they requires, while backlogged flows share the remaining bandwidth in proportion to their weights, i.e., GPS achieves max-min (weighted) fair bandwidth allocation. GPS is not practical since it requires formula (1.1) be satisfied in any infinitesimal interval. Many packetized scheduling disciplines have been proposed to approximate GPS, e.g., Weighted Round Robin (WRR), Deficit Round Robin (DRR), Weighted Fair Queuing (WFQ), Self-Clocked Fair Queuing (SCFQ), Worst-case Fair Weighted Fair Queuing (WF^2Q) [10]. All these queuing disciplines are centralized and require precise information about the contending flows.

1.1.2 Network Layer (End-to-End) Fairness in Wireline Networks

At the network layer, a flow is defined as a stream of packets which traverse from a source to a destination along a predefined route (Figure 1.1(b)). The network as a resource is to be shared by all flows in the network. Since the length of the route may be longer than one hop, fairness at the network layer is no longer a local property, but a *global property*: nodes in a network must cooperate to achieve network layer fairness. Therefore, distributed algorithms are desired.

Network model of wireline networks: A wireline network can be modeled as a set of interconnected links $\mathcal{L}$ where each link $l \in \mathcal{L}$ has a *fixed* capacity C_l. A set of flows $\mathcal{F}$ compete for access to these links. Each flow f in $\mathcal{F}$ associates with a route r which is a subset of $\mathcal{L}$. $l \in f$ denotes that flow f goes through link l. $f \ni l$ denotes the set of flows which goes through l. λ_f denotes bandwidth allocation for flow f. A feasible bandwidth allocation scheme must satisfy the *capacity constraint*:

$$\sum_{f \ni l} \lambda_f \leq C_l, \qquad l \in \mathcal{L}. \tag{1.2}$$

Subject to the capacity constraint (1.2), various fairness criteria have been proposed in the literature. Among these, three fairness models have been of particular interest to the research community: max-min fairness, proportional fairness and minimum potential delay fairness [11]. We introduce these fairness criteria[2] in the next section. It should be noted that these fairness criteria are also applicable at the MAC/link layer.

[2]In this thesis, we use fairness criterion and fairness model interchangeably.

1.2 Fairness Criteria

1.2.1 Max-min Fairness

The objective of max-min fairness is to maximize the bandwidth allocated to each flow f, subject to the capacity constraint (1.2) that an incremental increase in f's allocation does not cause decrease in some other flow's allocation that is already as small as f's or smaller [6]. Formally, for every flow f, there is at least one link $l \in f$, such that

$$\sum_{f' \ni l} \lambda_{f'} = C_l \qquad \text{and} \qquad \frac{\lambda_f}{\phi_f} = max\{\frac{\lambda_{f'}}{\phi_{f'}}, f' \ni l\}, \tag{1.3}$$

where ϕ_f is the weight for flow f. In the above requirements, $\sum_{f' \ni l} \lambda_{f'} = C_l$ indicates link l is a bottleneck link, i.e., it has been fully utilized. And $\frac{\lambda_f}{\phi_f} = max\{\frac{\lambda_{f'}}{\phi_{f'}}, f' \ni l\}$ means that the weighted bandwidth $\frac{\lambda_f}{\phi_f}$ allocated to flow f is the largest among all flows passing through the bottleneck link l. When the number of resources and the number of flows are both finite, there is only one allocation satisfying max-min fairness.

1.2.2 Proportional Fairness

The objective of proportional fairness is to *maximize*:

$$\sum_{\mathcal{F}} \phi_f \log \lambda_f, \tag{1.4}$$

subject to the capacity constraint (1.2), where ϕ_f is the weight for flow f. Or equivalently, for any other feasible $\lambda_f^{'}$, the aggregate of the weighted proportional rate changes with respect to the optimum allocation λ_f is negative, i.e., $\sum_{\mathcal{F}} \phi_f(\lambda_f^{'} - \lambda_f)/\lambda_f \leq 0$.

Again, proportional fairness allocation is unique to networks with finite flows and finite resources.

1.2.3 Potential Delay Minimization Fairness

The objective of potential delay minimization fairness is to *minimize*

$$\sum_{\mathcal{F}} \phi_f/\lambda_f, \tag{1.5}$$

subject to the capacity constraint, where ϕ_f is the weight for flow f. This model tries to minimize aggregate transfer delay since transfer time is approximately proportional to the reciprocal of bandwidth.

1.2.4 General Fairness Model

Proportional fairness penalizes long routes more severely than max-min fairness in the interest of greater overall throughput; potential delay minimization fairness lies between them [11]. It should be noted that besides these three fairness models, other fairness models are also possible. [12] has shown that there is a general equivalence between maximizing utility functions and achieving some system-wide notion of fairness. A *utility function* $U(\lambda_f)$ is a function of bandwidth allocation λ_f, which is continuous, differentiable, increasing and strictly concave over the range $\lambda_f \geq 0$. Thus by giving utility function $U(\lambda_f)$, a new fairness model can be defined. In [13], the following class of utility functions is proposed:

$$U_i(\lambda_i, \alpha) = \begin{cases} (1-\alpha)^{-1}\lambda_i^{1-\alpha} & \text{if } \alpha \neq 1 \\ \log \lambda_i & \text{if } \alpha = 1 \end{cases}$$

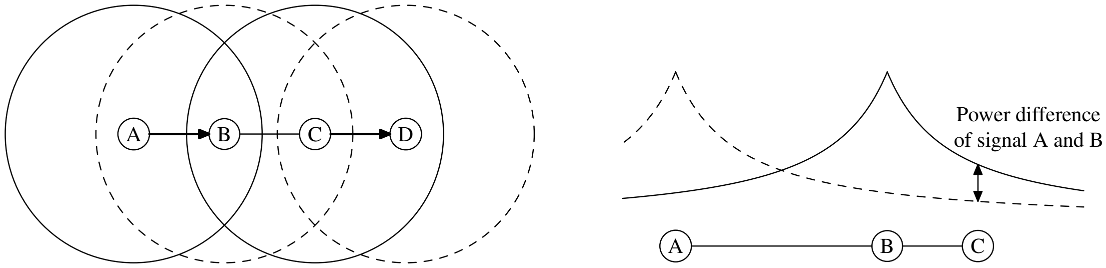
for $\alpha \geq 0$. This includes all the previously considered allocation policies — max-min proportional fairness ($\alpha = \infty$), proportional fairness ($\alpha = 1$) and minimum potential delay ($\alpha = 2$).

1.2.5 Fairness Models and Fairness Algorithms

We would like to point out that although the fairness criteria introduced in this section were originally developed for wireline networks, they do not depend on the characteristics of the underlying network. Therefore, they can be applied in wireless networks as well. However, the algorithms developed for wireline networks to achieve these fairness models usually depend on properties of wireline networks which are absent in wireless networks. For example, scheduling algorithms take advantage of the local property of a wireline output link as we have shown in Subsection 1.1.1. Hence these algorithms cannot be applied directly to wireless networks. In the following sections, we will discuss the characteristics of a multihop wireless channel and the fairness problems in a MANET in detail.

1.3 Characteristics of Multihop Wireless Channel

The wireless channel of a MANET is very different from a traditional wireline channel. The bandwidth available on a wireless channel is much narrower than that of a wireline



(a) Hidden terminal and exposed terminal

(b) Capture effect

Figure 1.2: Hidden terminal, exposed terminal and capture

channel. In addition, due to the effect of multi-path propagation and interference, the received signal strength at the receiver varies as a function of time; as a result the capacity of a wireless channel varies as a function of time. Furthermore, the wireless medium is inherently a shared medium: nodes within the joint neighborhood of the sender and the receiver of a flow contend for the limited bandwidth; thus the available bandwidth between a pair of neighbors is even lower and it is not fixed. Finally, the bit error ratio (BER) of a wireless channel is much higher than that of a wireline channel. Besides capacity limitation and high BER, another negative property of a wireless channel is location-dependent carrier sensing [14]. Since the transmission range of a radio signal is limited, only the nodes within a specific radius of the transmitter can detect the carrier on the channel. Location-dependent carrier sensing results in following three types of nodes that are problematic to the designers of multihop wireless networks, especially MAC protocol designers:

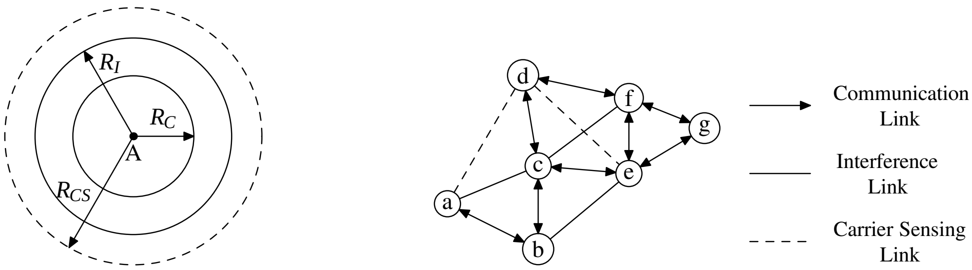
- *Hidden Nodes:* A hidden node is one that is within the range of the intended destination but out of the range of the sender. For example, in Figure 1.2(a), node C is in the transmission range of node B but out of the range of A. Thus A and C are hidden terminals to each other. If C starts transmission while there is an on-going transmission from A to B, a collision occurs at node B and the data packet from A to B becomes corrupted. Collisions caused by hidden terminals not only reduce the capacity of a wireless channel, but also contribute to the fairness problem.

- *Exposed Nodes:* An exposed node is one that is within the range of the sender but out of the range of the destination. For example, in Figure 1.2(a), when there is an on-going transmission from C to D, B is an exposed node. Since B is in the range of C, it cannot transmit even if its intended receiver (e.g., A) is out of the range of C. Similar to hidden terminals, exposed terminals reduce channel capacity and contribute to the fairness problem.

- *Capture:* Capture is said to occur when a receiver can correctly receive a transmission from one of two simultaneous transmissions, both within its range. In Figure 1.2(b), C is in the transmission ranges of both A and B. When A and B transmit to C at the same time, if the signal strength difference between the transmission of B and the transmission of A is larger than a threshold, C can still receive the transmission of B clearly. The capture effect reduces collision and can improve channel throughput, but it may result in unfair sharing of bandwidth.

1.3.1 Model of Multihop Wireless Ad Hoc Networks

We consider a MANET with a single physical channel with capacity C; transmissions are locally broadcast and only receivers within the *transmission range* of a sender can receive its packets. We further assume: (a) all links are symmetric; (b) nodes operate



(a) Communication range R_C, interference range R_I and carrier sensing range R_{CS}

(b) A MANET with three types of links

Figure 1.3: Model of a Multihop Wireless Ad Hoc Network

in half-duplex mode, i.e., a node cannot transmit and receive simultaneously. However, we do not exclude the capture effect.

In practice, carrier sensing wireless networks are engineered in such a way that the *carrier sensing (CS) range* is larger than the *interference range*, which is in turn larger than the *communication range* [15–17] (Figure 1.3(a)). To account for these differences, we model a MANET as a set of nodes N, interconnected by three types of links:

- *Communication link:* Nodes linked by a communication link can communicate directly. For example, in Figure 1.3(b), only node b could correctly receive node a's transmissions.
- *Interference link:* Transmission of a node at one end of an interference link prevents the node at the other end from receiving a packet correctly. For example, in Figure 1.3(b), transmission from node a to node b will be corrupted if node e starts to transmit during the transmission of node a.
- *Carrier sensing (CS) link:* Transmission of a node at one end of a carrier sensing link prevents the node at the other end from transmitting a packet. For example,

in Figure 1.3(b), node d is not allowed to transmit if node a is transmitting. Thus, at any instance, a MANET can be modeled as a combination of three graphs: a communication graph $G_C = (N, L_C)$, an interference graph $G_I = (N, L_I)$, and a carrier sensing graph $G_S = (N, L_S)$, where L_C, L_I and L_S are a communication link set, an interference link set and a CS link set respectively, and $L_C \subseteq L_I \subseteq L_S$ (Figure 1.3). In this thesis, we will discuss fairness issues which arise as a result of the different ranges.

One important observation is that at any instance only a subset of communication links L_C can be activated simultaneously. Ideally, all links in L_C should be activated at least once within a time interval which should be as small as possible. However, an unfair MAC protocol may prevent the activation of some links in L_C, leading to severe unfairness. Another observations is that the capacity C_l of a link l ($l \in L_C$) is not fixed. Links in L_C compete with one another to share channel capacity. A link may increase its capacity by "stealing" capacity from other competing links (i.e., to gain more chances to win the contention) or have its capacity decreased because of some capacity being "stolen" by others (i.e., getting fewer chances to win the contention).

1.4 Fairness Issues in Multihop Wireless Ad Hoc Networks

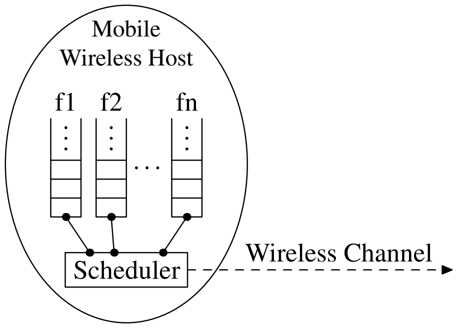
As in wireline networks, fairness in multihop wireless ad hoc networks can also be investigated at two levels: the MAC/link layer and the network layer.

1.4.1 MAC/Link Layer (Hop-to-Hop) Fairness in MANETs

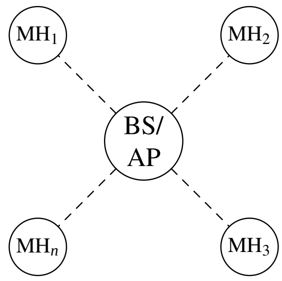
At the link layer, one-hop flows (which are defined as packet streams between neighboring nodes) compete with one another to share a wireless channel. In wireless networks, the link layer is tightly coupled with the MAC layer. Link layer fairness cannot be achieved without the support of MAC protocols. Hence, we use the term MAC/link layer fairness to highlight the importance of MAC protocols in providing hop-to-hop fairness. A wireless channel is a shared medium in which contending flows are distributed in different nodes. As a result, hop-to-hop fairness is no longer a local property — the scheduler in a host cannot correctly perform the two tasks introduced in subsection 1.1.1 without the cooperation of schedulers of other competing hosts. Hence, scheduling algorithms developed for the link layer of wireline networks can no longer be applied directly. The related issues are elaborated below.

1.4.1.1 Difficulties of Applying Fair Queueing-Scheduling over a Wireless Channel

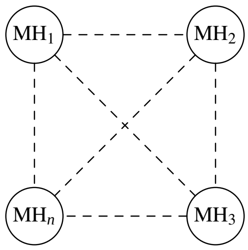
As we have discussed in Subsection 1.1.1, to achieve a certain fairness, the scheduler needs to perform two tasks: (a) determine the next flow to be served; (b) determine when a packet from the selected flow should be transmitted. In a wireline network, the information needed to perform both tasks can be obtained within the host itself; hence scheduling is a local operation. However, in a wireless network, nodes located in a region share a channel, and therefore flows residing in these nodes compete with one another to access the wireless channel. Figure 1.4(a) depicts a mobile wireless host and its scheduler. Compared with a fixed wireline host (Figure 1.1(a)), the wireless channel is not totally controlled by the mobile host; it is shared by all mobile hosts located in the contention region. Therefore, schedulers of competing nodes need to cooperate to perform the two tasks in order to achieve the desired fairness.



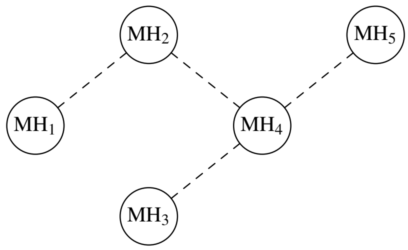
(a) A mobile wireless host and its scheduler



(b) Packet cellular network and wireless LAN



(c) Single-hop wireless ad hoc network



(d) Multihop wireless ad hoc network

Figure 1.4: MAC/Link layer fairness in cellular network/wireless LAN, single-hop wireless ad hoc network and multihop wireless ad hoc network

Several fair scheduling algorithms have been proposed for packet cellular networks and/or wireless LAN ([18,19]), and for single-hop wireless ad hoc networks ([20,21]). In packet cellular networks and wireless LAN (Figure 1.4(b)), all nodes communicate with a central control node (Base Station (BS) for cellular networks, and Access Point (AP) for wireless LAN). The logical structure of these networks is very similar to a wireline output link (Figure 1.1(a)). Thus the centralized fair scheduling algorithm developed for wireline networks can be conveniently extended for packet cellular networks and wireless LAN with the BS/AP acting as coordinator. The major concern of fairness in these networks is due to location-dependent error [19]. With location-dependent error, some nodes experience more transmission errors; thus, the actual throughputs of these nodes are much lower than those of others even when they are given the same chance to transmit. To achieve actual throughput fairness, mechanisms to compensate hosts whose packets are corrupted by transmission errors should be incorporated in the scheduling algorithms. [19] has provided a general fairness framework for this purpose.

In a single-hop wireless ad hoc network, nodes communicate with one another directly (Figure 1.4(c)); there is no central control node and all nodes have identical responsibilities. Information about flows is distributed in each node. Since in a single-hop wireless ad hoc network all nodes are within communication range of one another, a node can learn information of backlogged flows located in all other nodes in the network by overhearing. It is therefore possible to determine the service order of competing flows in a distributed manner (task a). For the same reason, all nodes know the exact time when an on-going transmission ends (synchronized), and hence, it can precisely determine when to transmit the next packet (task b). With some extra bookkeeping, the centralized scheduling can be easily adapted into a distributed one to achieve the desired fairness in a single-hop ad hoc network ([20, 21]).

Unfortunately, a wireless channel in a multihop mobile ad hoc network has characteristics that are totally different from those of packet cellular networks, wireless LANs and single-hop ad hoc networks. Hence it is costly or even impossible to apply fair queueing-scheduling algorithms that are meant for those others on a wireless channel in a MANET. In a MANET, there is no central control and not all nodes are within the communication range of one another Figure 1.4(d). A MANET wireless channel has no clear-cut boundary. Instead, it consists of a series of partially overlapped regions which compete with one another, and change their positions and shapes as the network evolves. It is costly or even impossible to collect, maintain and update all necessary information about competing flows to determine the order of service. Furthermore, competing nodes are no longer synchronized (synchronization is vital to perform task (b)) in the sense that a node does not necessarily know whether its competing nodes are transmitting or not (lack of synchronization). For example, in Figure 1.4(d), MH_1 and MH_4 compete with each other to transmit a packet to MH_2 and MH_5 respectively, but neither of them knows the activities of its competitor because $MH1$ is out of the carrier sensing range of $MH4$ and vice versa. In addition, the hidden terminal problem, the exposed terminal problem, the capture effect, and the different ranges of the communication link, interference link and carrier sensing link further complicate the hop-to-hop fairness problem in multihop wireless ad hoc networks. Although several fair MAC protocols have been proposed in the literature, none of them could provide high throughput and acceptable fairness regardless of topologies, traffic load and radio settings. New approaches are needed to achieve better hop-to-hop fairness, which is a fundamental element supporting end-to-end fairness.

1.4.2 Network Layer (End-to-End) Fairness in MANETs

An end-to-end network flow is defined as a stream of packets which traverse from a source to a destination along a predefined route. Though the fairness models and criteria developed for wireline networks may be applied in MANETs, the provision of end-to-end fairness in MANETs is a much more challenging task that has not been addressed adequately. The fairness of TCP flows in MANETs is particularly interesting to researchers because the traffic in MANETs is expected to be mostly TCP-like, just as it is in the Internet. However, even in wireline networks, providing fair bandwidth sharing among TCP flows is a challenging task. Many factors could affect the fairness of TCP flows: the MAC protocol [22–26], the routing protocol, the length of a route [27], buffer size [28], the active queue management algorithm [29], and congestion control algorithms [30–33], etc.. To achieve acceptable end-to-end fairness, further investigation into the interaction of all these factors is required. This is obviously a non-trivial task.

In this thesis, we focus on MAC/link layer fairness, which is a fundamental element in achieving ultimate end-to-end fairness. In addition, we also investigate the interaction between MAC protocols and TCP.

1.5 Contributions and Structure of Thesis

The key contributions of this thesis are:

- Through simulation, we demonstrate that the widely used MAC protocol IEEE 802.11 [34] could suffer from the one/zero fairness problem when operating in a multihop wireless ad hoc network, as some flows in the network may completely seize the channel capacity while others are virtually starved. We have identified three main causes for severe MAC/link layer unfairness (Chapter 2): the lack of synchronization problem (LSP), the double contention areas problem (DCP), and the lack of coordination problem (LCP).
- Based on the analysis of Chapter 2, we propose a new MAC protocol named extended hybrid asynchronous time division multiple access (EHATDMA) as a

solution. It employs three control schemes to address the three identified causes: a sender-initiated/receiver-initiated (SI-RI) hybrid scheme dealing with LSP, an asynchronous time division multiple access (ATDMA) scheme dealing with DCP, and a power control scheme dealing with LCP (Chapter 3).

- For better assessment of fairness, we design a new fairness index named max-min fairness index which is scenario-independent and reflects the difference between the fair sharing provided by a protocol and the ideal max-min fair sharing (Chapter 3).
- We carry out comprehensive simulations to compare the fairness of our protocol with the existing ones (Chapter 3). The results show that although the existing protocols employ various enhancements to improve their fairness property, most of them are still strongly biased towards optimizing throughput when there is a conflict between throughput and fairness. In addition, the fairness performance of these protocols varies widely from one scenario to another. On the other hand, EHATDMA strikes a good balance between throughput and fairness. It delivers a consistently high level of fairness regardless of network topology, traffic load and radio parameters, yet maintains high throughput whenever possible. Our simulation results also reveal that the most important mechanism affecting the fair sharing of radio channels among flows is the non-work-conserving mechanism.
- We propose a three-dimensional Markov model for analyzing and evaluating the throughput and fairness property of IEEE 802.11 (Chapter 4). Our mathematical model reveals several important results: (1) The model indicates that the RTS/CTS access method of IEEE 802.11 with default parameters operates almost optimally in terms of saturation throughput. (2) It shows that the instant

throughput[3] of a flow over a short distance may be tens or even hundreds times larger than that of a flow over a long distance. However, by introducing non-work-conserving principles, the variation in throughput can be reduced substantially. Simulation results reveal that, in addition to fairness, non-work-conserving principles can also improve the overall throughput of dense networks.

- We demonstrate by simulations that fair MAC protocols do improve fairness in TCP flows. In addition, they may also improve other aspects of TCP performance in multihop wireless ad hoc networks, for example, stability and compatibility (Chapter 5).

The rest of the thesis is organized as follows. Chapter 2 demonstrates an extreme fairness problem of IEEE 802.11 — the one/zero fairness problem, in which some MAC/link layer flows totally seize the channel bandwidth and others get nothing. The three main causes leading to the severe fairness problem are also identified. In Chapter 3, we present our fairness solutions for the MAC/link layer, which is known as extended hybrid asynchronous time division multiple access (EHATDMA). A max-min fairness index is proposed for comparative study of various fairness protocols. The index is scenario-independent. It reflects the difference between the fair sharing provided by a protocol and ideal max-min fair sharing. Comprehensive simulations show the fairness of our protocol against that of some existing protocols. In Chapter 4, we present a three-dimensional Markov model for the analysis and evaluation of the throughput and fairness property of the distributed coordination function (DCF) of IEEE 802.11 in mul-

[3]The instant throughput of a flow is the throughput of the flow observed over the period of time when a packet of that flow is scheduled for transmission. Chapter 4 will discuss the instant throughput in more details.

tihop wireless ad hoc networks. Chapter 5 demonstrates that fair MAC protocols do improve the performance of TCP over multihop wireless ad hoc networks. Chapter 6 summarizes the key results, identifies possible future work, and concludes the thesis.

CHAPTER 2

MAC Layer Fairness Problem Demonstration and Analysis†

One of the main design challenges imposed by multihop wireless ad hoc networks is the design of wireless medium access control (MAC) protocols. Since there is no central control node, it is difficult to have time synchronized across the network. Therefore, contention based asynchronous MAC protocols are preferred in such networks.

However, it is well known that contention based asynchronous protocols suffer from the hidden terminal problem and the exposed terminal problem as shown in Figure 1.2(a). Two nodes out of the transmission range of each other may interfere at a common node. These two nodes are referred to as hidden terminals of each other. Hidden terminals impair the throughput of a network. When a transmission is progressing, terminals hidden from the sender may initiate transmissions that would collide with the on-going one. As a result, throughput would be reduced. Exposed terminals are terminals that are within the transmission range of a transmitting terminal and thus are not allowed to initiate transmissions of their own. Similar to hidden terminals, exposed terminals can also reduce network throughput. Many protocols have been proposed to address

†The contents of this chapter has been presented in the paper for ICWN 2003. **Jun He** and Hung Keng Pung, "One/Zero Fairness Problem of MAC Protocols in Multi-hop Ad Hoc Networks and Its Solution," *International Conference on Wireless Networks (ICWN)*, Las Vegas, Nevada, USA, 2003, pp. 479-485.

the hidden terminal problem and the exposed terminal problem. Basically, these protocols can be classified into two categories: the multiple channel approach and the single channel with message exchange approach. In the multiple channel approach, besides the basic data channel, additional channels are employed for signaling between nodes. Protocols in this category include BTMA [35], DBTMA [36, 37], DCCA [38], BB [16]. Although these protocols are effective in dealing with the hidden terminal problem and the exposed terminal problem, they need multiple channels, which substantially complicates the design of the transmitter and receiver; therefore, the single channel approach is preferred in practice.

Single channel protocols include DFWMAC[1] [34], MACA [39], MACAW [40], MACABI [41], RIMA-SP, RIMA-DB and RIMA-BP [42], and FAMA [43, 44]. The basic idea of these protocols is the exchange of control frames between a sender and a receiver before the actual transmission of a data frame. The purpose of the control frames is to reserve the channel around the sender and the receiver for the forthcoming data frames. For example, in IEEE 802.11, a four-way handshake RTS-CTS-DATA-ACK is used. To transmit a packet, node A (Figure 1.2(a)) first sends a short control frame RTS to node B. Upon receiving the RTS, node B responds with a short control frame CTS, if it is not restrained from transmitting by other nodes; otherwise, it discards the RTS. Upon receiving the CTS, node A transmits the data frame (DATA) to node B, which responds with an ACK if the DATA frame has been received without error. Each control frame carries information about the remaining time of the current handshake. Any node over-

[1]The basic MAC protocol of IEEE 802.11 standard, hereafter, we use IEEE 802.11 and DFWMAC interchangeably

hearing any of the four frames is not allowed to transmit within the indicated period. After the initial RTS-CTS exchange, ideally, all neighbours of the sender and the receiver (node C in this case) should have been notified of the transmission intention and refrain from their own transmissions. Therefore, the data frame will be collision free. However, RTS-CTS exchange cannot eliminate all hidden terminals since RTS and CTS may collide with other frames and thus are not guaranteed to be received correctly by all neighbors.

The hidden/exposed problems can be even more prominent in real-life networks where the carrier sensing (CS) range is typically larger than the interference range, which in turn is larger than the communication range (Figure 1.3). In such a network, the RTS-CTS exchange cannot eliminate all hidden terminals even if it experiences no collisions. For example, in Figure 1.3(b), the CTS from node b can only prevent node c from transmitting, but not node e, which has no communication link but an interference link with node b.

It is also well known that contention based asynchronous protocols suffer from a fairness problem — some nodes (or flows) yield larger throughput than others [40]. Much research work has been done to address this issue [20,45–49]. However, most work is based on the assumption that the communication range, the interference range and carrier sensing are equal in their networks. Fairness in real-life networks as described in Chapter 1 remains an uncharted area. In this chapter, by using IEEE 802.11 as a study case, we demonstrate that in the context of real-life networks, single channel MAC protocols are vulnerable to a fundamental fairness problem which we refer to as the one/zero fairness problem, i.e., some flows seize the whole channel capacity while others

Simulation Parameters	**Value**
Communication Range	250m
Carrier Sense (CS) Range	550m
Capture Threshold	10db
Interference/Communication Ratio	1.78
Basic Rate, Data Rate	2Mbps
Packet Size	880 bytes
Packet Rate	250 packets/sec
Simulation Time	1000 seconds

Table 2.1: NS-2 simulation parameters for one/zero fairness problem

get virtually nothing. We identify three causes leading to the one/zero fairness problem, namely, the lack of synchronization problem (LSP), the lack of coordination problem (LCP) and the double contention areas problem (DCP). We choose IEEE 802.11 as a study case for two main considerations: (1) it is the most mature wireless LAN MAC protocol and has been standardized with products widely available, and (2) it is the *de facto* standard MAC protocol in the research of multihop wireless ad hoc networks.

2.1 Lack of Synchronization Problem (LSP) and Lack of Coordination Problem (LCP)

In this section, by using IEEE 802.11 as a case protocol, we demonstrate the one/zero fairness problem by running simulations for some typical scenarios. The simulation tool we have used is NS-2 version ns-2.1b9a [50]. Table 2.1 lists the parameters used in our simulations. We assumed an error-free channel, i.e., no transmission error, and all errors are due to collision. The radio model is based on existing commercial wireless network with a radio communication range of 250 meters and a channel capacity of 2Mbit/sec (more specifically, the radio model is based on WaveLan-II [15]). The CS range is 2.2

times the communication range (a default setting of NS-2; with this setting, all nodes in a cell of WLAN are within the CS range of each other [15]). Furthermore, we took the capture capability of a node into consideration. A node with capture capability may decode correctly one of two simultaneous transmissions if the difference in signal strength between the two transmissions is large enough. When the capture capability of a node is turned off, the interference range equals the CS range (550m). When capture is enabled, the capture threshold used was 10db (the default value of NS-2). We used the default capture behavior of NS-2, i.e., to enable the capturing effect, the stronger signal must come first. With capture enabled, once the stronger signal from sender S has been locked by a receiver R, transmissions from a third node T will not interfere with the reception of R if $d(T, R) > \alpha \times d(S, R)$, where $d(T, R)$ is the distance between node T and node R, $d(S, R)$ is the distance between node S and node R, and α is a fixed value determined by by capture threshold (for example, in NS-2, $\alpha \approx 1.78$ if the capture threshold is 10db). Following common practice in the literature, we used CBR-driven (constant bit rate) UDP traffic in our simulations. The packet rate was so chosen that each single flow alone could consume all channel capacity. Each simulation was run for 1000 seconds, which is long enough to show the fairness property of a MAC protocol. Throughput is computed as the number of successfully transmitted packets during the 1000 seconds simulation time.

Table 2.2 shows two scenarios used in our simulations. In both scenarios, there is an interference link between node b and c. The only difference between the two scenarios is that in scenario (A), $d(c, b) > \alpha \times d(a, b)$, therefore even if node c transmits at the same time, node b can still capture the transmission from node a as long as the transmission

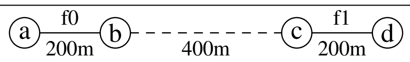
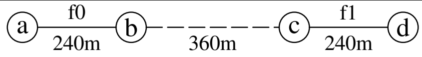
Scenario	Topologies and Flows	Descriptions
(A)		Capture is possible
(B)		Capture is impossible

Table 2.2: Scenarios used to show LSP and LCP

Configuration No.	Scenario	Capture	f0	f1	f1/f0
(a)	(A)	On	41245	173034	4.2
(b)	(A)	Off	0	194336	-
(c)	(B)	On	0	194290	-
(d)	(B)	Off	0	194290	-

Table 2.3: Simulation results for scenario (A) and (B) with IEEE 802.11

from node a arrives at node b first. However, in scenario (B), node b cannot capture any transmissions from node a because $d(c, b) < \alpha \times d(a, b)$. Table 2.3 reports the simulation results for scenario (A) and (B) with the capture capability on and off. It can be seen that except configuration (a), all configurations suffer from the one/zero fairness problem: flow 1 gets full channel capacity while flow 0 gets nothing. Even in configuration (a), the throughput of flow 0 is much smaller than that of flow 1, at approximately 25% of flow 1.

The extreme unfairness is caused by the hidden terminal problem and the exposed terminal problem. In scenario (A) and (B), node a and node c are hidden from each other. Whenever node c is transmitting, node a is in a disadvantageous position in contending for the channel because node a is out of the carrier sensing range of node c but its destination (node b) is exposed to node c. Unless arriving at node b during the gap between two consecutive transmissions of node c, transmissions from node a are doomed to failure. After each failure, node a doubles its contention window size (IEEE 802.11 employs the binary exponential backoff (BEB) algorithm) which puts node a in an even worse situation. Unfortunately, node a has no knowledge about the occurrence of the gap (this is known as the lack of synchronization problem (LSP)). It continues to send packets nevertheless. Occasionally, it may find a gap. And if capture takes effect, the packet is transmitted successfully. This is shown in configuration (a). However, if the receiver has no capture capability or if capture fails due to strong interference, node a can hardly have a successful transmission, as shown in configurations (b), (c) and (d). The result is that flow 0 gets nothing. We refer to this phenomenon as the one /zero fairness problem.

One possible solution to LSP is to shift the initiator role of flow 0 from sender a to receiver b because although sender a does not know when the gap between two consecutive transmissions of node c starts, receiver b knows. If we let receiver b take part in the contention to acquire the channel, LSP is resolved. This leads to a hybrid MAC protocol, in which both senders and receivers may take the initiator role. Traditional MAC protocols either designate the sender as initiator (sender-initiated, shortened as SI) [34,39,40,43,44], or receiver as initiator (receiver-initiated, shortened as RI) [41,42]. However, such approaches cannot resolve LSP. The underlying assumption of all these protocols is that if the initiator (sender in SI protocols, receiver in RI protocols) detects a free channel, there is a high probability that the channel around the responder (receiver in SI protocols, sender in RI protocols) is also free. The assumption would be valid if the responder is idle most of time. However, if the responder is located in a busy area while the initiator is usually idle, this assumption is wrong. It would be more efficient to switch the initiator role between the sender and the receiver. That is what the hybrid MAC protocol tries to achieve — choosing the initiator of a transmission based on the

Scenario	Topologies and Flows
(C)	a —f0 (200m)— b - - - 400m - - - c —f1 (200m)— d - - - 400m - - - e —f2 (200m)— f
(D)	a —f0 (240m)— b - - - 360m - - - c —f1 (240m)— d - - - 360m - - - e —f2 (240m)— f

Table 2.4: Scenarios used to show DCP

channel conditions around the sender and the receiver. A general rule is that the one within a busier area should assume the initiator role.

However, the hybrid scheme alone cannot resolve the fairness problem. To illustrate this, consider scenario (B): when receiver node b takes part in and wins the channel contention, a successful transmission does not follow. Once node b wins the channel contention, it will transmit a control frame (say POLL) to solicit a data frame transmission from node a. However, since node c is out of the communication range of node b, it cannot receive the control frame and therefore does not know there will be a forthcoming transmission from a to b. After a backoff time, node c starts a new transmission which leads to the corruption of transmission $a \rightarrow b$ because in scenario (B), $d(b,c) < \alpha \times d(a,b)$. We refer to this as the lack of coordination problem (LCP). Clearly new schemes are needed to deal with LCP. One possible solution is to boost the transmission power of the control frames if node b detects that it is suffering from LCP, so that the new communication range R_C covers node c. We will discuss the power control algorithm in the next chapter.

2.2 Double Contention Areas Problem

Table 2.4 shows two scenarios where the sender and the receiver of flow 1 are exposed to two different busy contention areas. Flow 1 competes with flow 0 and flow 2. To

No.	Scenario	Capture	f0	f1	f2
(e)	(C)	On	191396	2639	194675
(f)	(D)	On	194282	0	196066

Table 2.5: Simulation results for scenario (C) and (D) with IEEE 802.11

have a successful transmission, flow 1 must win in both contention areas. Table 2.5 reports the corresponding simulation results. In scenario (D), the IEEE 802.11 suffers from the one/zero fairness problem again. In scenario (C), even with capture enabled, the throughput of flow 1 is still extremely low: less than three packets per second. The extreme unfairness is expected since both the sender and the receiver of flow 1 are suffering from LSP. To have a successful transmission of flow 1, the channel around node c and d must be idle at the same time, which is highly unlikely due to the aggressiveness of node b and e in accessing the channel. To avoid severe unfairness in these scenarios, some kind of channel etiquette is required.

2.3 Further Analysis of the One/Zero Fairness Problem

In the previous sections, we have demonstrated that when operating in multihop wireless ad hoc networks, the widely used MAC protocol IEEE 802.11 suffers from a severe fairness problem: the one/zero fairness problem, i.e., some flows entirely seize the channel bandwidth while others are starved. We have identified three causes leading to the severe unfairness:

- *The lack of synchronization problem (LSP):* The sender of a flow has no information about when the receiver is/will be idle.
- *The lack of coordination problem (LCP):* In a real-life multihop network, not all

the interferers can be notified by the CTS/POLL control frame, which may lead to the one/zero fairness problem.

- *The double contention areas problem (DCP):* The sender and the receiver of a flow are exposed to two different contention areas. If both the areas are busy most of the time, the flow is likely to be starved.

Although the topologies used to demonstrate the one/zero fairness problem are simple linear patterns, we believe that these topologies can provide insight into the fairness property of a MAC protocol. Since every realistic topology can be decomposed into these linear patterns, the one/zero fairness problem can occur in any realistic multihop ad hoc network. Actually, as long as the three causes (LSP, LCP and DCP) exist, the one/zero fairness problem cannot be avoided. Unfortunately, in a realistic multihop ad hoc network, it is impossible to completely eliminate LSP, LCP and DCP for several reasons. First, in any kind of multihop ad hoc networks — no matter whether it has identical or different ranges of CS/interference and communication — as long as there are exposed terminals, LSP occurs. Since exposed terminals are inevitable in a multihop ad hoc network, LSP is also unavoidable. Second, in any multihop ad hoc networks where the channel is reused, DCP is also likely to occur because channel reuse inevitably generates different contention areas and the flows bridging these contention areas are vulnerable to DCP. Finally, unless the network is so engineered that all interferers of a receiver are covered by the carrier sensing range of the sender, LCP cannot be eliminated completely. In a network where LCP is not totally eliminated, we can estimate the occurrence probability of LCP. Suppose that the node density of the network is known and the distance between the sender and the receiver of a flow is uniformly distributed between 0 and the communication range R_C. We can calculate the average number of nodes located in the vulnerable area of a flow (Figure 2.1) by using the following formula:

$$vn\_num = density \times \int_0^{R_C} \frac{1}{R_C} va(x)dx.$$

Here, *vn_num* is the average number of nodes in the vulnerable area of a flow; *density* is the node density of the network; R_C is the communication range; x is the distance between the sender and the receiver of a flow; $va(x)$ is a function of x, denoting the extent of the vulnerable area. For example, in an area of $1000m \times 1000m$, 50 nodes are uniformly distributed (this configuration is typically used in performance analysis of routing protocols in the literature). By using the default setting of NS-2, we compute that the average number of nodes located in the vulnerable area of a flow is about 0.42. Supposing that only $\frac{1}{4}$ nodes in the network have traffic, then the probability that a flow will suffer from LCP is 0.105. Furthermore, supposing there are 18 flows in the network, then the probability that some flows suffer from LCP is $1-(1-0.105)^{18} \approx 0.87$. To verify our analysis, we randomly generate 10 scenarios with the parameters given above, and find that eight of them suffer from LCP, which indicates that our analysis is very accurate.

LCP is closely related with capture capability. As the capture capability of a node increases (i.e., the capture threshold decreases), LCP becomes less prominent. For the same $1000m \times 1000m$ scenario, we calculate the LCP vulnerable probability versus the capture threshold for two cases (Figure 2.2): (1) CS range is equal to the communication range ($R_{CS} = R_C$); (2) CS range is equal to the maximum interference range, which

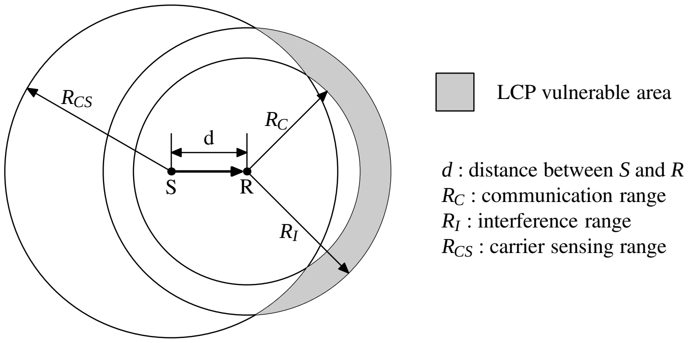


Figure 2.1: LCP vulnerability analysis

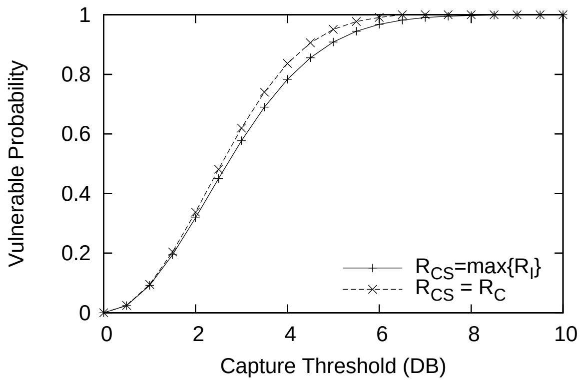


Figure 2.2: Vulnerable probability versus capture threshold

is determined by the communication range and the capture ratio ($R_{CS} = max\{R_I\} = \alpha \times R_C$). In both cases, the interference range for a transmission between x and y is $R_I(x, y) = \alpha \times d(x, y)$, where α is a value determined by the capture threshold and $d(x, y)$ is the distance between node x and y. The results show that even for the 2db capture threshold, the LCP vulnerable probabilities for the two cases are still larger than 0.3, which are significant.

2.4 Summary

A MAC protocol is a key element of multihop ad hoc networks. Throughput and fairness are two major concerns. Most MAC protocols used in multihop ad hoc networks are adapted from protocols initially designed for wireless LANs, where throughput is the major concern. When applied to multihop ad hoc networks, these protocols suffer from a severe fairness problem. The fairness problem becomes more acute in real-life networks where the interference and carrier sensing range are much larger than the communication range. In this chapter, we have considered the fairness problem in real-life networks and highlighted a fundamental fairness problem: the one/zero fairness problem. That is, some flows seize all channel capacity while others get virtually nothing. We have identified three causes leading to the one/zero fairness problem, namely, the lack of synchronization problem (LSP), the lack of coordination problem (LCP) and the double contention areas problem (DCP). We have further indicated that the one/zero fairness problem could occur in any realistic multihop ad hoc network. Based on the analysis in this chapter, we will propose in the next chapter a new MAC protocol named extended hybrid asynchronous time division multiple access (EHATDMA) as a solution; it employs three control schemes to address the three identified causes: a sender-initiated/receiver-initiated (SI-RI) hybrid scheme dealing with LSP, an asynchronous time division multiple access (ATDMA) scheme dealing with DCP, and a power control scheme dealing with LCP.

CHAPTER 3

Fairness of Medium Access Control Protocols for Multihop Wireless Ad Hoc Networks†

In the previous chapter, we have demonstrated that IEEE 802.11 could suffer from a severe fairness problem when applied in multihop ad hoc networks. In extreme cases, some flows would completely dominate network access while others are starved. Such a level of unfairness is obviously unacceptable. Several solutions have been proposed to address this problem [45, 47, 48, 51–56]. In [45], the authors proposed a systematic approach for translating a given fairness model into a corresponding contention resolution algorithm. In [56], the authors looked into the fairness problem from the game theory perspective and proposed a fairness model similar to that in [45]. [51] and [52] proposed a connection-based backoff algorithm and a measurement-based backoff algorithm respectively to replace the widely used binary exponential backoff (BEB) algorithm. In [47,48,53,54], the authors tried to apply fair queueing-scheduling algorithms in the MAC layer to achieve fairness. Although these studies have shed some light on tackling the

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Part of this chapter has been presented in a paper for LCN 2003: **Jun He** and Hung Keng Pung, "A Fairer Multiple Access Protocol for multihop Wireless Networks: Hybrid Asynchronous Time Division Multiple Access Protocol (HATDMA)", *IEEE Conference on Local Computer Networks (LCN)*, Bonn/Königswinter, Germany, 2003, pp. 356-365.

fairness problem, the effectiveness of their solutions in dealing with the fairness problem is confined to certain scenarios. There are also several limitations in the approaches:

a) Most of these methods attribute the fairness problem to the backoff algorithm used by the MAC protocol. Hence, they try either to provide a new and fair backoff algorithm or adapt some fair queueing-scheduling discipline originally developed for wireline to improve fairness in wireless channel sharing. As the backoff algorithm is only one of the many causes of the fairness problem, these solutions alone are inadequate.

b) All fairness metrics used for measuring the fairness properties of MAC protocols suffer a severe shortcoming: the fairness index of MAC is strongly dependent on the scenario under examination. While the approach may be acceptable for a comparative study of a given scenario, a metric that can overcome the shortcoming is clearly desirable.

c) Comprehensive and systematic simulation studies are indispensable in evaluating the fairness property of a MAC protocol. However, most simulation work reported has been carried out for some special network scenarios. Furthermore, there is no comprehensive comparative study of performance for different fairness mechanisms.

In this chapter, the limitations listed above will be addressed. The rest of this chapter is organized as follows. We review related work in Section 3.1. The strength and weakness of each study are discussed in detail. Based on the discussions and the analysis in the previous chapter, we propose a new protocol known as extended hybrid asynchronous time division multiple access (EHATDMA) in Section 3.2. Section 3.3 presents a comparative performance study of fairness for EHATDMA and several related protocols via simulations. In Section 3.5, we look into the effects of individual mechanisms and identify the mechanisms most critical to improving fairness. Section 3.6 concludes this chapter.

3.1 Fairness Problems in Multihop Wireless Networks and Related Work

Carrier sense multiple access (CSMA) is one of the most pervasive mechanisms in ad hoc wireless networks and many MAC protocols have adopted it to improve performance. The carrier sense (CS) mechanism requires that nodes listen to the channel prior to their transmissions. Nodes are not allowed to transmit when the channel is busy. In real life, carrier sense wireless networks are so engineered that the CS range is larger than the interference range which in turn is larger than the communication range ([15, 16]). As given in Chapter 1, a multihop wireless network can be modeled as a set of nodes N, interconnected by three sets of links: communication link set L_C, interference link set L_I and carrier sense link set L_{CS} (Figure 3.1). A node is not allowed to transmit if it has a CS link with another transmitting node; for instance in Figure 3.1, node a has a CS link with node d. In other words, a node cannot transmit if it is within the CS range of another node that is transmitting. Suppose node j has an interference link with node i. Then any packet transmission to node j overlapping in time at node j with another transmission from node i will be corrupted. For example, in Figure 3.1, the transmission of $a \rightarrow b$ will be corrupted if node e starts to transmit at the same time.

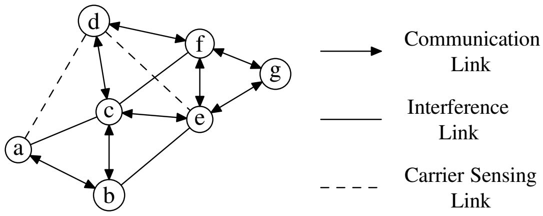


Figure 3.1: A carrier sense wireless network with three types of link

Only the nodes having a communication link with a transmitting node will accept that transmission.

In this model, only a subset of L_C can be activated simultaneously at any instance. Therefore, to be fair, a MAC protocol should successfully activate each communication link at least once within a reasonable time interval. If $L_A(t, t + \Delta t)$ denotes the link set that has been successfully activated during the time interval $[t, t + \Delta t]$, then for any t, a fair MAC protocol should keep the convergence time Δt (the minimal Δt that satisfies $L_A(t, t+\Delta t) = L_C$) as small as possible. However, it has been shown that most current MAC protocols cannot deliver a small enough convergence time Δt. In some extreme cases, the convergence time Δt is infinite (one/zero fairness problem), as we have shown in the previous chapter. In Chapter 2 we have identified three causes for the one/zero fairness problem: the lack of synchronization problem (LSP), the double contention areas problem (DCP) and the lack of coordination problem (LCP). To achieve an acceptable level of fairness, all three causes should be dealt with in the MAC layer. Although there are numerous studies on the fairness problem ([45, 47, 48, 51–56]), most do not explicitly deal with the three causes. They attribute the fairness problem to an improper backoff algorithm at the MAC protocol. Hence, they try to present new and fair backoff algorithms or adapt fair queueing and scheduling techniques meant for wireline networks to multiple access radio networks.

In [45], the authors proposed a systematic approach for translating a given fairness model into a corresponding rate adaptation algorithm. One of the key elements of the model is the *flow contention graph*. Each clique of this graph is viewed as a server and each flow as a client. However, as indicated by the authors themselves, for the derivation of the rate adaptation algorithm to hold, two conditions must be satisfied: (i) the contention loss probability for a clique is very small; (ii) all flows in a clique observe the same loss probability. Unfortunately, these two conditions can hardly be maintained in multihop networks, which are likely to operate in asynchronous mode. First, when operating in asynchronous mode, the derived algorithm of [45] acts very aggressively, which leads to non-negligible loss probability in multihop networks, and hence, violates condition (i). Second, condition (ii) can only hold if all contentions among flows are symmetrical, which can hardly be true in multihop networks. With these two conditions violated, the derived algorithm can no longer achieve its original fairness objective.

In [51], a connection-based balanced media access method (hereafter denoted as CBFAIR in this thesis) is proposed to achieve fairness. In CBFAIR, node i having packets for node j will access the channel after a backoff period with a pre-calculated probability of p_{ij}, or back off with probability $1 - p_{ij}$ using the same backoff window size. p_{ij} is calculated based on the number of connections possessed by node i and its neighbors. In addition, CBFAIR also employs a backoff-window-exchange scheme. The sender inserts the information of the last backoff window size into the control frame. Any node overhearing this information will use the received window size in place of its own if the received window size has a smaller value.

Another scheme that can replace BEB is the measurement-based algorithm (denoted as MBFAIR) proposed in [52]. The BEB algorithm used by IEEE 802.11 is replaced by the proposed algorithm which operates as follows: each node regards all of its neighbors as a single entity and has only the notion of "myself" and "others". It monitors channel usage arising from itself and others, and adjusts its backoff window size according to the ratio of the channel usage of itself to that of others. If the ratio is larger than C (C is a design parameter of MBFAIR, and is suggested to be close to 1), a node concludes that it has received a bigger share than it should and will double its contention window (CW) size, which is bounded by a minimal value CW_{min} and a maximal value CW_{max}. On the other hand, if the ratio is smaller than $1/C$, a node halves its backoff window size; otherwise, it keeps the window size unchanged.

Another common approach to the provision of fairer channel sharing is to incorporate fair queueing-scheduling into MAC protocols [47, 48, 53, 54]. These proposals invariably emulate fair queueing operations (i.e., assign start and finish tags to each packet) in a distributed manner by exploiting the broadcast nature of a wireless channel. In these schemes, each frame carries the tag of the head-of-line packet or the next-in-line packet. By overhearing, a node can establish a tag table, which ideally records precise and up-to-date tag information for each flow destined for or departing from its one-hop neighbors. With this table, a node can emulate fair queueing operations and schedule the transmissions of its outgoing packets. The differences between these queueing-scheduling proposals lie in which scheduling strategy is used and how to deal with an out-of-order event. Luo et al. [48] proposed a fair queueing scheme named enhanced maximize-local-minimum fair queueing (EMLM). In EMLM, all outgoing flows in a node are allowed to contend for the channel (we refer to this as *the multiple-scheduling strategy*). A flow is scheduled to transmit based on its rank in the sender as well as the rank in the receiver. The backoff time CW for each outgoing flow is calculated by the formula $CW = map(R_f^S + R_f^R) + random$, where R_f^S is the flow's rank in its sender, R_f^R is the estimated rank in its receiver, *map* is a linear function and *random* is a small random variable used to break a tie. When the receiver detects an out-of-order transmission intention (that is, the tag of the forthcoming packet is not the smallest in the receivers table), it simply ignores the transmission request. Kanodia et al. [54] proposed a protocol named the distributed wireless ordering protocol (DWOP). In DWOP, only the flow with the smallest tag in the sender's tag table is allowed to transmit (we refer to this as *the single-scheduling strategy*). When the receiver detects an out-of-order transmission intention, it will continue the transmission. However, at the end of the transmission, the receiver will request the sender (through ACK) to refrain from transmission for a time given by $T_{backoff} = R(eifs + difs + T_{success} + CW_{min})$, where $T_{success}$ is the longest possible time required to transmit a data packet, R is the rank of the received packet in the receiver's tag table, $eifs$, $difs$ and CW_{min} are parameters as specified in the IEEE 802.11 standard.

We have noted that neither fair backoff nor fair queueing-scheduling can efficiently resolve the fairness problem for several reasons. First, backoff algorithms are actually some kind of congestion control algorithms. Improper backoff policies may lead to unstable behavior. Although there are abundant studies on the stability of backoff policies ([57–59] and references therein), most studies have been carried out for *single-hop* networks[1] under simplifying assumptions (e.g., unbound backoff counter, no deletions) that render the results of these studies less relevant to the choice of backoff policy in a multihop wireless network. Without any good theoretical guideline, it is wiser to follow the conventional wisdom that exponential backoff is the best choice among acknowledgment-based policies for systems with a large overall arrival rate. This is because BEB has been proven to work well in both large, worldwide scale (TCP congestion control [60]) and small, local scale (Ethernet). Second, the backoff algorithm is not the preliminary cause of the fairness problem. It amplifies the unfairness initially caused by LSP, DCP or LCP. Therefore, any efficient fairness solution should address these three primary causes first before modifying the problematic but indispensable binary exponential backoff algorithm.

It should be pointed out that the effectiveness of fair queuing-scheduling in dealing with the fairness problem would be compromised in multihop wireless networks due to the constraints of these networks. To emulate fair queueing in a distributed manner, each node in a multihop wireless network must perform two basic tasks: (i) determine the service order of packets located in its contending region, and (ii) transmit its own packets according to order at the right time. As we have discussed in Chapter 1, it is very difficult to perform the tasks precisely in multihop wireless networks. To fulfill task (i), a node should at least obtain information of all the flows located in its three-hop neighbors. However, this would lead to prohibitive overhead due to excessive exchanges and maintenance of information. To reduce overhead, most proposed fair queueing schemes try to approximate ideal fair queueing by using only the start or finish tag

[1] In a single-hop wire/wireless network, the carrier of a node can be detected by all other nodes.

of flows located in a node's one-hop neighborhood. This approximation compromises the effectiveness of the fair queueing schemes. For example, consider typical scenario 3 in Figure 3.4(on page 59): each flow believes that there is only one flow (itself) in its neighborhood. In this case, the network operates as if there is no scheduling at all; consequently, the desired fairness cannot be ensured. Another difficulty of fair queueing arises from task (ii). Even if nodes in a multihop wireless network are able to establish an order of service for the flows in its neighborhood, the service order may not be enforced correctly due to LSP. For example, consider typical scenario 2 in Figure 3.4 (on page 59): supposing CS range equals to communication range), flow $0 \rightarrow 1$ backs off as it knows flow $2 \rightarrow 3$ (which is two hops away) has a smaller tag. However, it still does not know how long it should back off and when to start its own transmission because it has no way of knowing when the transmission of flow $2 \rightarrow 3$ ends. Note that task (ii) does not present a problem to wireline networks and single-hop wireless networks. In these networks, schedulers can immediately know the end of a transmission by notification from the transmitter (wireline networks) or by carrier sensing (single-hop wireless networks). This is a fundamental difference between multihop wireless networks and wireline/single-hop wireless networks.

Furthermore, in any real-life multihop wireless networks, it is very costly and sometimes impossible to exchange information between interfering and carrier sensing neighbors. This is because information needs to be relayed between interfering/CS neighbors by some intermediate nodes, and in some special cases, there is no such intermediate nodes at all (e.g., typical scenarios 10-21 of Figure 3.4 on page 59).

With these observations in mind, we propose a novel MAC protocol, known as the extended hybrid asynchronous time division multiple access protocol (EHATDMA), which has three schemes to deal with LSP, DCP, and LCP respectively. The schemes are described in Section 3.2.

3.2 Extended Hybrid Asynchronous Time Division Multiple Access Protocol (EHATDMA)

3.2.1 SI-RI Hybrid Scheme

The SI-RI hybrid scheme is designed to deal with LSP. In Figure 1.2(a), node A does not know the commencements of the gaps between consecutive transmissions of node C but node B does. So if we let node B instead of node A contend to acquire the channel when a gap is detected, LSP would be resolved. This observation leads to the SI-RI hybrid scheme for the MAC protocol, in which both sender and receiver take an initiator role. Based on the observed efficiency, the hybrid scheme chooses a proper collision avoidance (CA) mechanism to transmit a packet: either using the sender-initiated (SI) CA mechanism or using the receiver-initiated (RI) CA mechanism. Our previous work ([61, 62]) shows that the SI-RI hybrid scheme is very efficient in resolving LSP. We have refined our previous work and the resultant hybrid scheme is summarized as follows:

- *Multiple FIFO queues:* Unlike IEEE 802.11, we employ multiple FIFO queues in EHATDMA. Each queue buffers outgoing data packets for a receiver and may operate in one of three modes: sender initiated (SI) mode, receiver initiated (RI) mode and SI-RI mode. Packets in an SI-mode queue are sent by the sender actively using a four-way handshake procedure: RTS-CTS-DATA-ACK. Packets in an RI-

Octets: 2	2	6	6	2	2	4
Frame Control	Duration	RA	TA	Next PktLen	Period	FCS

Figure 3.2: Format of control frames

mode queue are polled by the receiver using a three-way handshake procedure POLL-DATA-ACK. If a flow is operating in SI-RI mode, both the sender and the receiver try to send or poll the head-of-line packet of the flow by using the four-way or three-way handshake procedure respectively. In addition, each queue also maintains the state of the queue: operating mode (SI, RI or SI-RI), contention window (CW) size, period of the queue $flow\_period$ (to be explained in the next subsection) and statistics used to choose the operating mode of the queue.

- *Frame format:* EHATDMA extends the frame structures of IEEE 802.11. It has four control frames (RTS, CTS, POLL and ACK) and a data frame (DATA). All control frames are of the same format, as shown in Figure 3.2. Two new fields are introduced: the *next_pktlen* field and the *period* field. These two fields are also added to the DATA frame. To poll a data packet, a receiver needs to know the length of the packet. The *next_pktlen* field of a frame carries this information. The source of a flow sets the *next_pktlen* field of every frame it transmits (i.e., RTS or DATA) to the length of the next packet in the queue. Upon receiving an RTS (or DATA), the destination copies the value of the *next_pktlen* of the received frame to the replying frame (i.e., CTS or ACK). The period field is used in ATDMA and will be explained in the next subsection.

- *neighbor table and flow table:* By overhearing, a node establishes a neighbor table

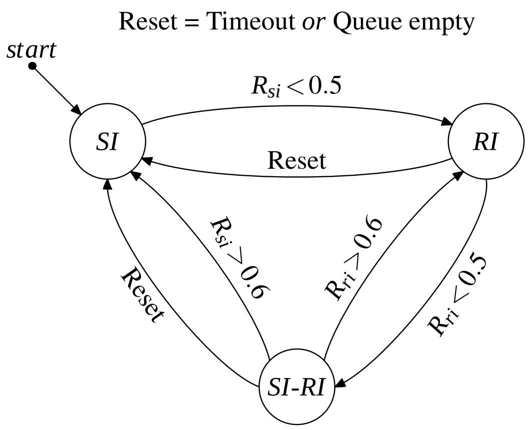


Figure 3.3: Selection of operating mode for a flow

— which records *period*s of all communicable neighbors, and a flow table — which records information regarding flows generated by or destined for its communicable neighbors. The information to be recorded include source, destination of the flow, length of the next packet (*next_pktlen*), etc. If the flow is destined for the node itself (an incoming flow), the flow table also maintains the mode of the flow, the contention window size (CW), *flow_period*, statistics, etc., to support RI operation. An incoming flow also maintains the state of *power_mode* to support power control (see Section 3.2.3). The *period* of a neighbor and the *next_pktlen* of a flow are reset to 0 if they have not been updated within a certain time (500ms in our implementation).

- *Selection of operating mode (SI/RI/SI-RI) for a flow:* The operating mode of a flow is managed by the initiator of a flow. For an SI flow, the sender is the initiator and the receiver is the responder; this is otherwise in an RI flow; for an SI-RI flow, both the sender and the receiver take the role of initiator as well as responder. Initially, all the flows operate in SI mode. The initiator of each flow monitors the transmission success rate of the flow periodically (in our implementation, the period is 200ms). At the end of a period, the initiator selects an operating mode for the next period. Figure 3.3 shows the mode selection diagram. For an SI flow, the sender is authorized to decide the flow's mode. The flow remains in SI mode if the success rate of the SI operation R_{si} is larger than 0.5;[2] otherwise, it chooses RI as the next operating mode. For an RI flow, it is the receiver that selects the next operating mode. The flow remains in RI mode if the success rate of the RI operation R_{ri} is larger than 0.5; otherwise, SI-RI becomes the next operating mode. For an SI-RI mode, both the sender and the receiver make a choice. At the sender side, the flow will get into SI mode if $R_{si} > 0.6$; otherwise, it remains in SI-RI mode. At the receiver side, the flow gets into RI mode if $R_{ri} > 0.6$, and remains in SI-RI otherwise. Note that in this case, the sender and receiver may choose different modes for the next period. The real operating mode depends on the outcome of the switching operation explained in the next item. If the sender of a flow operating in RI or SI-RI mode has not been polled for a period of time $\beta \cdot flow\_period$ ($flow\_period$ is the period of the flow, explained in the following subsection; $\beta = 4$ in our implementation) or the flow becomes empty, the flow is reset to SI mode. Similarly, the receiver of an RI or SI-RI gives up polling (the $next\_pktlen$ of the flow is also set to zero) and changes the flow to SI mode if it fails to poll for a certain number of times (seven in our implementation). Once the operating mode for the next period is decided, the initiator of the current period

[2]Note that all the thresholds used in our implementation are empirically assigned. One clue leads us to chose threshold around 0.5 is that the average number of transmission attempts for a packet should be less than 2.

starts the mode switching operation to tell the responder its decision, so that the state of flow can be consistent between sender and receiver.

- *Switching operating mode:* Two bits of the *Frame Control* field (Figure 3.2) are redefined as $flow\_mode$ to indicate the next operating mode[3]: 00=unchanged, 01=SI, 10=RI, 11=SI-RI. Once the mode has been decided, the initiator operates in the new mode immediately. To tell the responder its choice, it sets the $flow\_mode$ as the new mode for each frame transmitted to the responder. Upon receiving a frame with a non-zero $flow\_mode$, the responder changes the operating mode of the flow to the new mode and copies the mode to the $flow\_mode$ field of the replying frame to confirm that it has changed to the new mode. The initiator can tell whether the switching operation is successful or not by checking the $flow\_mode$ of the replying frame. If it is successful, it will not set the $flow\_mode$ for the following transmissions. Otherwise, the procedure continues until it succeeds. For an SI-RI flow, once the responder accepts a switching request from the initiator, it gives up its own switching operation. Hence, where the sender and the receiver choose different modes, the mode that has been transmitted successfully earlier will be used.

3.2.2 ATDMA — Asynchronous Time Division Multiple Access

While the SI-RI hybrid scheme is designed to resolve LSP, ATDMA is designed to address DCP. To improve the throughput of a flow $S \rightarrow R$ suffering from DCP, it is necessary to increase the probability that both S and R are idle at the same time. One way to achieve

[3]In our implementation, To DS bit (B8) and From DS bit (B9) are used. These two bits are not needed in multihop wireless networks.

this is to reduce the number of competing flows in both contention areas where the flow is exposed. With fewer competitors, the flow can get a higher chance to simultaneously win the channels in both areas. Following this cue, ATDMA requires a flow to freeze for a period of time (*flow_period*) after successful transmission. The value of *flow_period* should be so chosen that at any time, the number of active flows (i.e., flows trying to acquire the channel) is indeed reduced. Interestingly, IEEE 802.11 has already employed a similar idea. In IEEE 802.11, a node *must* enter into a backoff stage after a successful transmission; this is to prevent the node from monopolizing the channel. In ATDMA, this backoff stage is replaced with the frozen stage, which varies in length according to traffic load and is usually much longer than the backoff stage.

The core of ATDMA is the *flow_period* adaptation algorithm. For each flow, the *flow_period* should be wisely chosen so that it improves fairness without overly impairing overall throughput. In addition, it should promptly respond to any change in traffic load. In a fair scheduled access scheme, a flow which has just had a successful transmission is unlikely to be scheduled again before all other flows in the same contention area have transmitted one packet each. Hence, a reasonable *flow_period* is the total time needed by all other flows in the same contention area to transmit one packet each. Recall that each frame carries information about the length of the next packet meant to support the operation of the RI mode. The same information also gives the time needed by that flow to transmit the next packet, or we can view it as a reservation request for the channel. It is therefore very convenient to derive the total time needed by all flows located in communicable neighbors. Since a node cannot directly receive reservation requests made by its CS/interference neighbors, we have designed an algorithm to estimate the total requested time of these nodes.

With the introduction of a frozen stage, the scheme operates in a fashion similar to the conventional time division multiple access (TDMA) scheme in the following ways. In a contention area scale, a *flow_period* is similar to a time-frame of TDMA; each flow in the area has one slot (transmission) in the time frame. In a network scale, nodes are divided into several groups and the channel is shared among groups in a time division fashion. However, time frames in ATDMA are neither universally synchronized nor are they of the same length. Each flow maintains a local frame structure which varies as the scenario evolves. When the slot for a flow comes (i.e., at the end of a frozen state), the flow still needs to contend for the channel. For these reasons, we name the scheme asynchronous time division multiple access (ATDMA).

We describe the operation of ATDMA in more detail below:

- *Reservations of communicable neighbors $P_{i,cn}$:* As we have mentioned in the previous subsection, each node maintains a flow table that records all overheard reservations of flows due at its communicable neighbors (i.e., the *next_pktlen* of each overheard frame). For each flow, only the latest reservation is maintained. The sum of the *next_pktlen*s for all flows in the flow table at node i is denoted as $P_{i,cn}$ (communicable-neighbor-period), indicating the total time required to transmit the next packets of all flows of the communicable neighbors.
- *Estimating reservations of CS/interference neighbors $P_{i,csi}$:* It is very difficult to precisely estimate the reservations made by CS/interference neighbors. The estimation algorithm proposed here is only one of many possible solutions. To make an estimation, the algorithm requires a node to monitor channel activity periodically

(the period is $100ms$ in our implementation). At any time, the channel around a node that is not transmitting can only be in one of the following three states: *idle* — none of its neighbors (including all communicable and CS/interference neighbors) is transmitting; *clear-busy* — the channel is busy (i.e., one or more of its neighbors are transmitting), and at the end of this state, the node receives a frame without error; *noise-busy* — the channel is busy but at the end of this state, the node does not receive an error-free frame. As the node detects the *idle* state, the *noise-busy* state and the *clear-busy* state in the channel, it also records the number of occurrences and the duration of each channel state. At the end of a monitoring period, the node (say node i) estimates the reservations made by all its interfering/CS neighbors $P_{i,csi}$ by:

$$\begin{aligned} P_{i,csi} &= max(\text{average } \textit{busy} \text{ state duration, 1.5ms}) \\ &\quad \times \frac{\text{occurrences of } \textit{noise-busy} \text{ state}}{\text{occurrences of } \textit{clear-busy} \text{ state}}. \end{aligned} \tag{3.1}$$

Here, the *busy* state includes the *clear-busy* state and the *noise-busy* state. The first term of equation (3.1) can be viewed as the average length of reservations made by CS/interference nodes; the second term can be viewed as the number of CS/interference nodes. Note that both terms are estimated conservatively. We restrict each reservation of CS/interference nodes to be no larger than than $1.5ms$. A more reasonable estimation of the number of CS/interference neighbors should take the number of communicable neighbors into consideration. We choose a conservative policy because the presence of a CS/interference link does not necessarily mean two flows cannot access the channel simultaneously. Our simulation results show that this policy is feasible and can work well in most scenarios.

- *Node period:* the period of node i (denoted as P_i) is computed by:

$$P_i = P_{i,cn} + \min(P_{i,cn}, P_{i,csi}) + eifs.$$

  As specified in IEEE 802.11, $eifs$ is the period of time that a node has to wait before it could transmit if it detectes a *noise-busy* state . Thus, in a real-life network, a successful transmission of node i may put some nodes such as node j in the $eifs$ waiting state, thereby putting node j in a disadvantageous position in contending with node i. To put nodes which have transmitted successfully on an equal footing with others in channel contention, we add an $eifs$ time to their periods as shown in the above expression. For each frame transmitted by node i, the period field of the frame (Figure 3.2) is therefore filled with P_i. By overhearing, a node maintains a table that records periods of its communicable neighbors.

- *Flow period:* The period of a flow f is defined as $P_f = \max(P_S, P_R)$, where P_S and P_R are the node period of the sender and the node period of the receiver, respectively.

3.2.3 Power Control

The power control mechanism is designed to address LCP. The idea of power control, which is shown in Algorithm 1, is quite straightforward. As in IEEE 802.11, EHATDMA requires a node hearing a collision to keep silent for $eifs$ time, which is long enough for

Algorithm 1 Power Control Algorithm

```
// power_mode = [ normal | enhanced ]
// N_en: number of successful receptions in enhanced mode
if power_mode == normal then
  if DATA_fail_rate > 0.5 then
    power_mode = enhanced
    N_en = 0
  end if
else
  if DATA_fail_rate > 0.25 or N_en > 30 then
    power_mode = normal
  end if
end if
```

the reception of a CTS or an ACK. This requirement almost ensures that the sender will receive a replying CTS/ACK without collision. Hence, after sending a CTS for a flow, the receiver can safely attribute a failure to the corruption of DATA frame instead of the corruption of the CTS. Two causes may lead to DATA corruption: (i) the CTS collides at some hidden nodes of the sender; (ii) the flow is suffering from LCP. The power control algorithm is designed to reduce collisions due to the second reason. To support power control, a node monitors the DATA failure rate for each incoming flow. It calculates the DATA failure rate after every 20 transmissions of CTS for a flow. If the DATA failure rate is high (> 0.5), it assumes that the flow is suffering from LCP and switches to *enhanced* power mode. For an incoming flow in *enhanced* power mode, the receiver adjusts the transmission power upward for POLL/CTS to a level so that all potential interferers are within the new communication range. All other frames are still transmitted in *normal* power. The power mode is switched back to *normal* if the receiver has successfully received more than 30 packets Again, 30 packets is an empirical data for a flow (to limit the adverse effect of *enhanced* power), or if the DATA failure rate is higher than 0.25 (in this case, the *enhanced* power mode is deemed to be inefficient in improving the throughput of the flow).

Increasing transmission power certainly worsens the exposed terminal problem. However, since only the CTS/POLL frames of an *enhanced* mode flow are transmitted in increased power, and the flow reverts to *normal* mode soon after it gets a certain number of successful reception, the adverse effect of increasing power is minimized. In addition, due to the ATDMA scheme, the power control mechanism is unlikely to be triggered frequently[4]; consequently the adverse impact is further reduced.

3.3 Performance Evaluation and Comparison

Ideally, a MAC protocol for wireless networks should provide high throughput and maintain acceptable fairness with minimum transmission and computation overhead regardless of network topology, traffic load and radio parameters (i.e., CS range, interference range and communication range). We have carried out extensive simulations for IEEE 802.11, CBFAIR, MBFAIR, EMLM, DWOP, EDWOP and EHATDMA for a comparative performance study.

The implementation of EMLM in our simulation is slightly different from that proposed in [48] in the following ways: Instead of the proposed RTS-CTS-DS-DATA-ACK procedure, we use a four-way handshake procedure RTS-CTS-DATA-ACK because in a carrier sense network, DS is unnecessary. In addition, we do not use the R_f^R estimation algorithm since simulation results show that it does not improve performance. In EMLM, the backoff time of a flow f is calculated with the formula $CW_f = 16 \times R_f^S + random[0, 15]$,

[4]Simulation results have confirmed this. Results presented in Section 3.5.2 show that the power control scheme is seldom triggered in general scenarios.

where R_f^S is the flow f's rank in the sender(we refer to this as *the multiple-scheduling strategy*). If a flow experiences failure, normal BEB backoff is used.

DWOP emulates Self-Clocked Fair Queueing (SCFQ) in the same way as in EMLM. However, only the flow having the smallest tag is allowed to access the channel (we refer to this as *the single-scheduling strategy*). If an out-of-order transmission is detected, the sender is requested not to contend for the channel for $T_{backoff}$ time as specified in Section 3.1. In our implementation, we set $T_{success}$ to be the time needed to transmit a packet of size 1500B.

We combine EMLM and DWOP and derive a new scheme named EDWOP. In EDWOP, the *multiple-scheduling strategy* of EMLM is used. When an out-of-order event is detected, the $T_{backoff}$ strategy of DWOP is used.[5]

The simulation tool we used was NS-2 (version 2.19ba) [50]. Table 3.1 lists the parameters used for the simulations presented in this section. We assumed an error-free channel, and that all errors were due to collisions only. The communication range and CS range were 250m and 550m respectively (the default setting of NS-2). Furthermore, we took the capture capability of a node into consideration. A node with capture capability may decode correctly one of two simultaneous transmissions if the difference in signal strength between the two transmissions is large enough. The capture threshold used was 10db, which meant that nodes within the $1.78 \times d(S, R)$ range of node R would interfere with the transmissions of flow $S \rightarrow R$, where $d(S, R)$ was the distance between node S and node R. We used the default capture behavior of NS-2, i.e., for capture to take

[5] Since the source codes of CBFAIR, MBFAIR, EMLM and DWOP have not been made publicly available, our implementations of those protocols are based on the respective papers. Except the differences explicitly given here, we have tried our best to reproduce these protocols as accurately as possible.

Simulation Parameters	**Value**
Communication Range	250m
Carrier Sense (CS) Range	550m
Capture Threshold	10db
Interference/Communication ratio	1.78
Basic Rate	1Mbps
Data Rate	2Mbps
Packet Size	880 bytes
Packet Rate	250 packets/s
Simulation Time	100 seconds $\times$ 31

Table 3.1: NS-2 Simulation Parameters

effect, the stronger signal must come first. Following common practice in the literature, we used CBR-driven (constant bit rate-driven) UDP traffic in our simulations. The packet rate was so chosen that each single flow alone would consume all the channel capacity. We ran each scenario 31 times and the simulation time for each run was 100 seconds. We will present the average value of the 31 runs. Four performance metrics were used in the performance evaluation, as outlined in the following subsection.

3.3.1 Performance Metrics

- *Throughput:* Throughput can be measured in two metrics — *aggregate throughput (AT)* and *channel efficiency (CE)*. $AT = \sum_{f \in \mathcal{F}} T_f$, where $\mathcal{F}$ is the flow set in the scenario; T_f is the number of successfully transmitted packets for flow f in a simulation run. Channel efficiency is the portion of time used to successfully transmit the payload; it can be derived from AT by:

$$CE = \frac{AT \cdot E[P]/sim\_time}{channel\_rate}.$$

Here, $E[P]$ is the average size of the payload, sim_time is the simulation time. For our simulations, $E[P] = 880 \times 8 = 7040 bits$, and $sim\_time = 100s$, $channel\_rate = 2Mbps$. In a single hop wireless network, $CE \leq 1$. In a multihop wireless network, the channel may be spatially reused; hence, CE may be larger than 1.

- *Average attempts per packet (AAT):* A packet may collide with other packets several times before it is transmitted successfully. We use the average attempts per packet (AAT) to measure the collision avoidance efficiency of a MAC protocol. AAT is defined as:

$$AAT = \frac{Total\ Number\ of\ Transmission\ Attempt}{AT}$$

- *Max-min fairness index (FI_m):* Although much work has been done on the fairness problem in MAC protocols, there are no widely accepted fairness metrics which can be used to measure the fairness property of a MAC protocol. In [51], a fairness index is defined as the ratio of the maximal flow throughput to the minimal flow throughput in a network. However, a fairness index defined in such a way is scenario-dependent and the comparison of fairness indexes in different scenarios is meaningless. For example, suppose that the fairness index is 3 in scenario A and 2 in scenario B; we cannot say that the channel allocation in scenario B is fairer than that in A. A good fairness index should reflect the fairness property of a protocol without the need to refer to the scenario so that the fairness indexes of different scenarios can be compared. We have proposed such a fairness metric named the max-min fairness index (FI_m). The proposed metric is based on the widely used max-min fairness model. The max-min fairness model reflects the intuitive notion of fairness that any flow should be entitled to as much channel share as any other flow is. Or more formally, it maximizes the allocation of each flow i, subject to the constraint that an incremental increase in i's allocation does not cause a decrease in some other flow's allocation that is already as small as i's or smaller [6]. To calculate the max-min fairness index, we first calculate the max-min fair share ϕ_f for each flow f by using the algorithm proposed in [49]. FI_m is then calculated with the popular fairness formula [63]:

$$FI_m = \frac{\left(\sum_{f \in F} \frac{T_f}{\phi_f}\right)^2}{n \sum_{f \in F} (\frac{T_f}{\phi_f})^2} \tag{3.2}$$

By introducing max-min fair share, the influence of a scenario is eliminated from the fairness index. The max-min shares of a scenario calculated by the algorithm of [49] represent a perfect max-min fairness scheme. To assess fairness, all other channel sharing schemes for that scenario are measured against the perfect scheme by using equation (3.2). The result is a value within the range of $[0, 1]$, reflecting the difference between a channel sharing scheme and the ideal max-min sharing scheme. The larger the value is, the closer the channel allocation scheme will be to the ideal max-min scheme. A value of 1 indicates that a channel allocation scheme achieves perfect max-min sharing. Since fairness indexes of different scenarios are all measured against their respective perfect max-min sharing schemes and mapped into the same range $[0, 1]$, they can be compared with one another without any need to refer to scenarios. For example, if the fairness index is 0.97 in scenario

A and 0.8 in B, we can say that the channel allocation scheme in scenario A is fairer than that in B. We do not need to know the exact topologies of A and B. It is in this sense that we claim that the fairness index is scenario independent.

- *Minimum Flow Throughput Rate R_{min}:* A fairness index such as FI_m reflects the overall fairness of a protocol. Sometimes, the minimum flow throughput rate R_{min} is also important:

$$R_{min} = \frac{\min\{T_f | f \in F\}}{sim\_time},$$

where sim_time is the simulation time (100 seconds in our simulations).

3.3.2 Simulation Scenarios

As we have mentioned, a fair MAC protocol must provide acceptablely fair sharing to each flow in all wireless network configurations. To verify the fairness property of a MAC protocol, we have designed three sets of benchmark scenarios. Any MAC protocols claiming to be fair should provide acceptably fair sharing of the channel in all these benchmark scenarios.

3.3.2.1 Wireless LAN Scenarios

In wireless LAN, all nodes are within communication ranges of one another. One interesting question is whether a MAC protocol can still be fair as the number of active flows increases. We have designed 19 simulation scenarios with the number of active flows in each scenario corresponding to the scenario number (i.e., scenario 2 has two flows; scenario 3 has three flows, and so on.)

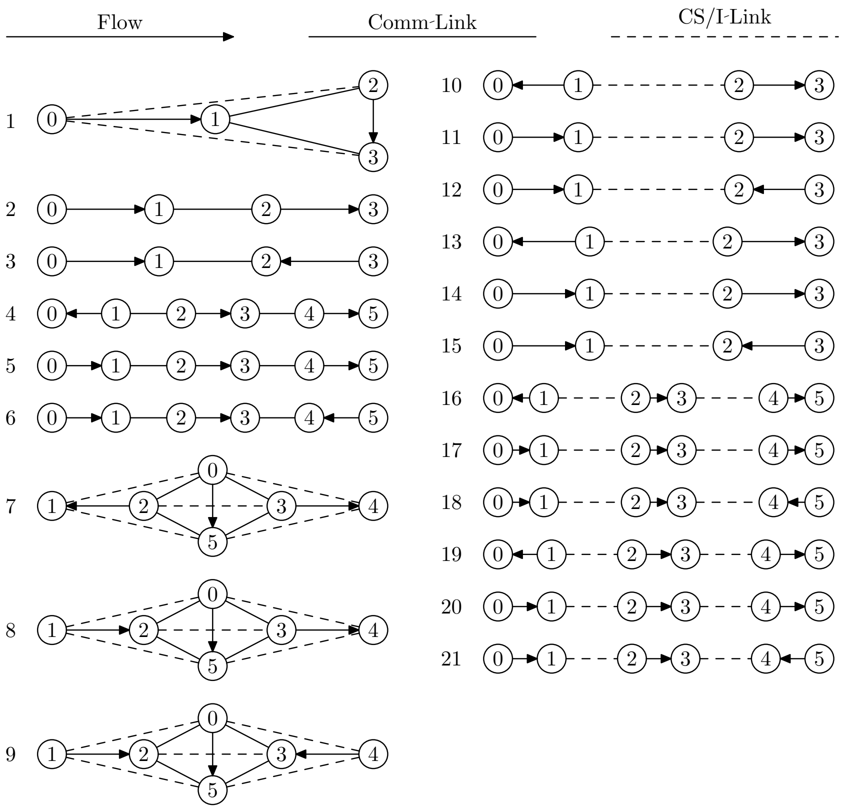


Figure 3.4: Typical scenarios

3.3.2.2 Typical Scenarios

For networks with different ranges in communication, interference and carrier sensing, we designed 21 typical scenarios as shown in Figure 3.4. Each scenario contains either two flows or three flows, and may suffer one or more of the three fairness problems (LSP, DCP and LCP). In scenario 1, all nodes are within CS range of each other[6]. In scenarios 2-6, the distance between any two adjacent nodes i and j is $d(i,j)=200m$. Thus, one can infer that scenario 2 suffers none of the three problems, scenario 3 suffers from LSP and scenarios 4, 5 and 6 suffer from DCP (readers are reminded that we have a CS range of 550m). In scenarios 7, 8 and 9, $d(1,2)=d(3,4)=200m$ and $d(2,3)=300m$. Therefore,

[6]To avoid cluttering, not all CS links are shown in Figure 3.4

scenarios 7 and 8 suffer none of the three problems and scenario 9 suffers from LSP between flow 1→2 and 4→3. In scenarios 10, 11 and 12, $d(0,1) = d(2,3) = 200m$ and $d(1,2) = 400m$. For these three scenarios, the link between node 1 and node 2 is a CS link and only scenario 11 suffers from LSP. In scenarios 13, 14 and 15, $d(0,1) = d(2,3) = 240m$ and $d(1,2) = 360m$. Hence, scenarios 14 and 15 suffer from LSP and LCP. In scenarios 16, 17 and 18, $d(0,1) = d(2,3) = d(4,5) = 200m$ and $d(1,2) = d(3,4) = 400m$; all of them suffer from DCP. In scenarios 19, 20 and 21, $d(0,1) = d(2,3) = d(4,5) = 240m$ and $d(1,2) = d(3,4) = 360m$; all of them suffer from DCP and LCP.

3.3.2.3 General Scenarios

Figure 3.5 shows the scenarios in the general category. General scenarios 1, 2, 4 and 7 were used in [48]. General scenarios 5, 6 and 8 are randomly generated scenarios. To avoid cluttering, we do not show CS links and interference links in the figure.

3.3.3 Simulation Results

We have carried out extensive simulations for IEEE 802.11, CBFAIR, MBFAIR, EMLM, DWOP, EDWOP and EHATDMA. However, to avoid cluttering the figure, we only show the simulation results for MBFAIR, EMLM, DWOP, EDWOP and EHATDMA. This is because we are more interested in the fairness property of a MAC protocol, and IEEE802.11 and CBFAIR are far less competitive in terms of fairness, as we have shown in [62].

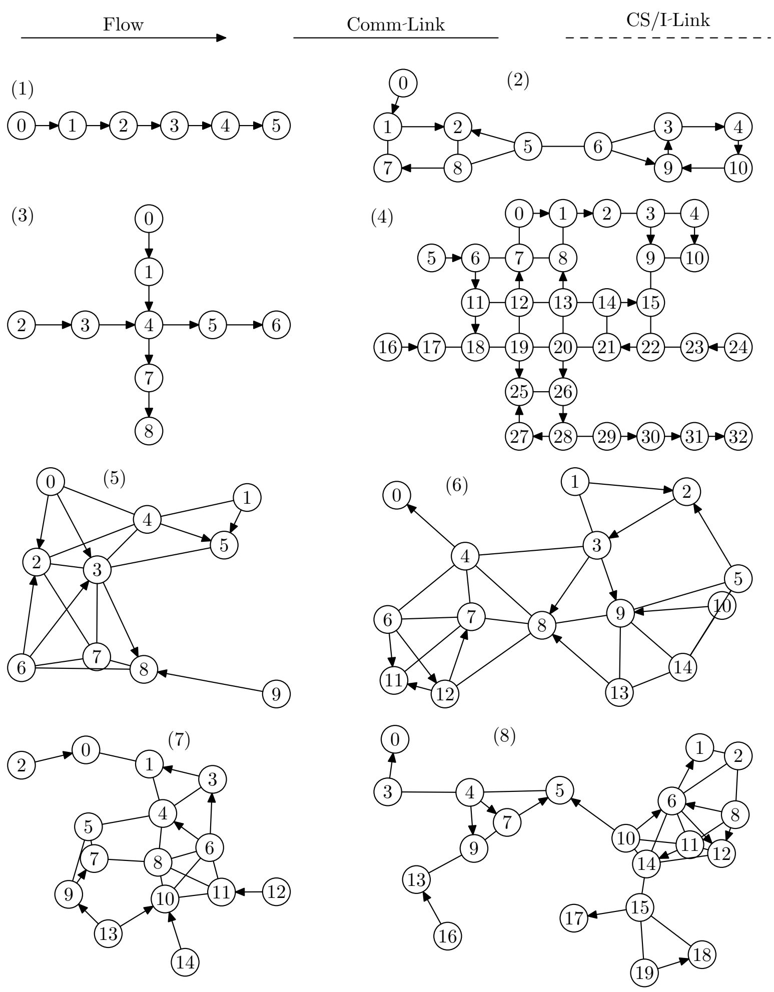
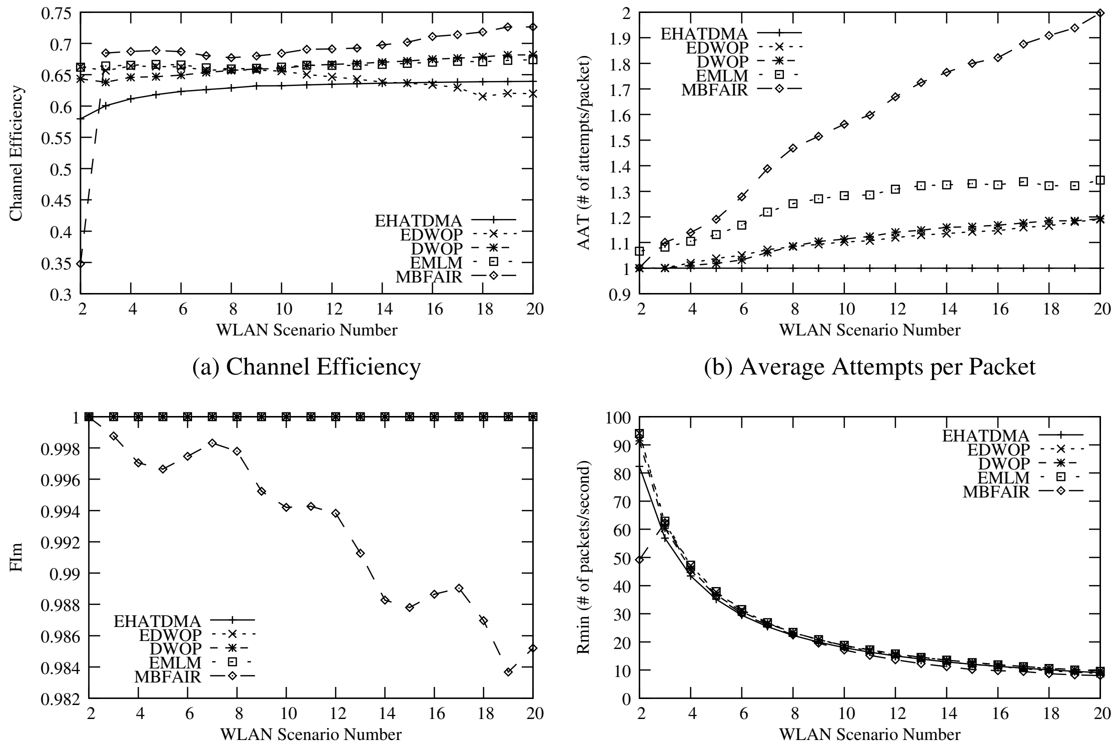


Figure 3.5: General scenarios

3.3.3.1 Simulation Results of WLAN Scenarios

Figure 3.6 shows the simulation results of the WLAN scenarios (the scenario number equals to the number of flows in a scenario). From Figure 3.6(c), we can see that the FI_ms of EMLM, DWOP, EDWOP and EHATDMA remain nearly unchanged at around 1 in all WLAN scenarios. The results indicate that these four protocols achieve perfect max-min fair channel sharing in all WLAN scenarios. Although the FI_m of MBFAIR decreases as the number of flows N increases, it still performs very well — $FI_m > 0.984$



(a) Channel Efficiency (b) Average Attempts per Packet

(c) Max−min Fairness Index (d) Minimal Packet Throughput Rate

Figure 3.6: Simulation results of WLAN scenarios

in all scenarios.

Figure 3.6(a) shows that when there are only two active flows ($N = 2$), MBFAIR has extremely low channel efficiency (CE). However, in all other scenarios, it achieves the highest CE among all protocols. This can be explained as follows: In a WLAN scenario, all nodes can communicate with one another. As a result, the channel usage ratio R of "itself" to "other" observed by each node is around $1/(N-1)$. When there are only two flows in the network ($N = 2$), $R \approx 1$. After successful transmission, a flow is likely to find that $R > C$ (in our implementation, $C = 1.1$). As a result, the contention window size is likely to double, which leads to throughput degradation. When $N > 2$, R is almost definitely less than 1; accordingly, nodes are inclined to choose small contention window sizes, which causes CE to increase. However, a smaller contention window size incurs more collisions, which is clearly reflected in Figure 3.6(b) — the AAT of MBFAIR is much larger than those of others. Without capture capability, the CE of MBFAIR would decrease.

Figure 3.6(a) also shows that EHATDMA has the lowest CE in most scenarios. Its throughput is about 10% less than that of MBFAIR. The low CE of EHATDMA is due to its conservativeness in channel access. After a successful transmission, the ATDMA scheme restrains a flow from accessing the channel for a $flow\_period$ time, which is equal to the time needed for all other flows to transmit a packet. When the network reaches a stable state, the ATDMA scheme successfully reduces the number of active flows (i.e., flows trying to transmit) at any instance to 1. Hence, the backoff time of a flow does not overlap with those of others. As a result, throughput is reduced. The gain is that collision is completely avoided. As shown in Figure 3.6(b), the AAT of EHATDMA is 1 in all scenarios.

The CEs of DWOP and EMLM lie in between those of MBFAIR and EHATDMA (Figure 3.6(a)). As the number of flows N increases, so do the collisions (Figure 3.6(b)). EMLM acts more aggressively than DWOP. As a result, it experiences more collisions. However, collisions do not affect the CE too much because one of the colliding transmissions will survive with high probability due to capture. Although the number of collisions increases as N increases (Figure 3.6(b)), the CEs of DWOP and EMLM remain almost unchanged (Figure 3.6(a)). However, the CE of EDWOP does decrease as N increases. The decrease results from an out-of-order penalty. EDWOP employs a multiple-scheduling strategy as in EMLM, and is therefore very aggressive in channel access. Consequently, more out-of-order events occur and more flows are penalized to take a $T_{backoff}$ time backoff, which leads to throughput degradation.

There are no significant differences in minimum flow throughput R_{min} among all protocols examined except the special case of MBFAIR (when $N = 2$).

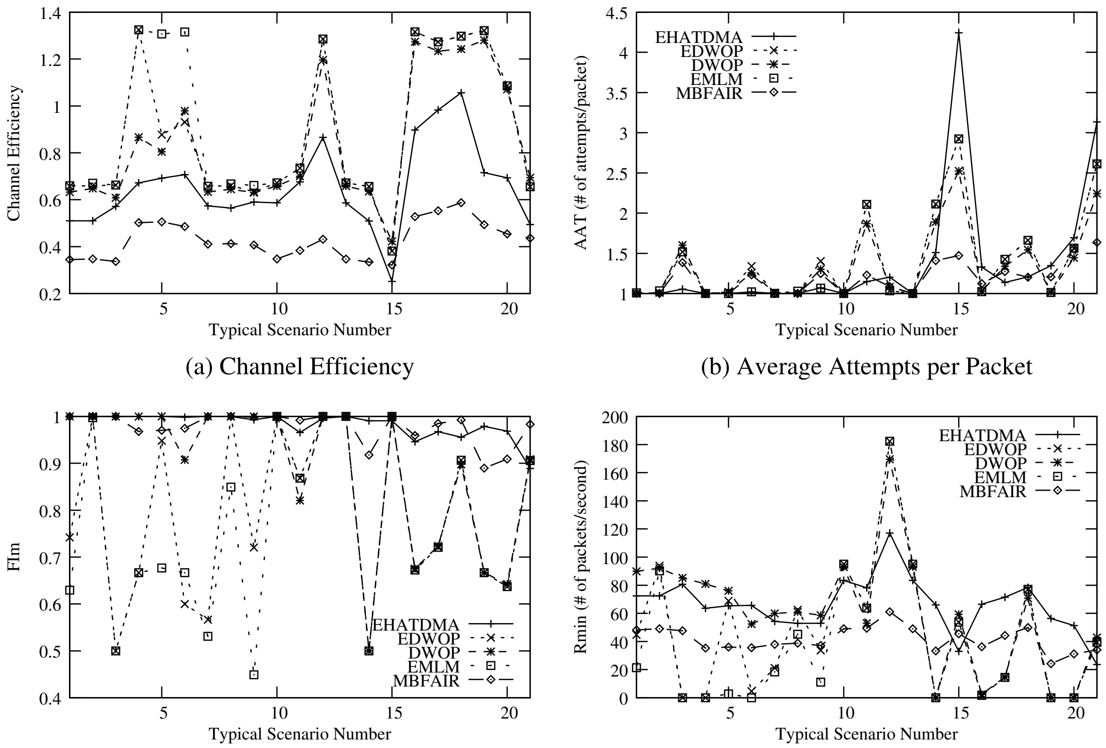
From the results shown above, we conclude that in WLAN scenarios, all the schemes perform very well in terms of fairness. MBFAIR achieves the highest throughput ($N > 2$) while EHATDMA achieves perfect max-min fairness sharing without incurring any collisions.

The CE of a protocol in WLAN scenarios can serve as a benchmark to determine the degree of channel reuse in multihop wireless networks. For example, EHATDMA achieves a channel efficiency of about 0.63 in WLAN. In a multihop scenario, if the CE of EHATDMA is 1.2, we can say that it achieves a channel reuse degree of 2. This knowledge allows us to have a better understanding of the relationship between channel reuse and fairness.

3.3.3.2 Simulation Results of Typical Scenarios

The typical scenarios are extreme cases specially designed to evaluate the effectiveness of a MAC protocol in dealing with the three problems — LSP, DCP and LCP. Figure 3.7 shows the simulation results of the typical scenarios.

From Figure 3.7(a) and Figure 3.7(c), we observe a fundamental conflict between throughput and fairness. Figure 3.7(a) shows that fair queueing-scheduling based protocols (EMLM, DWOP and EDWOP) have higher throughput than EHATDMA and MBFAIR. However, the higher throughput is achieved at the expense of fairness (Fig-



(a) Channel Efficiency

(b) Average Attempts per Packet

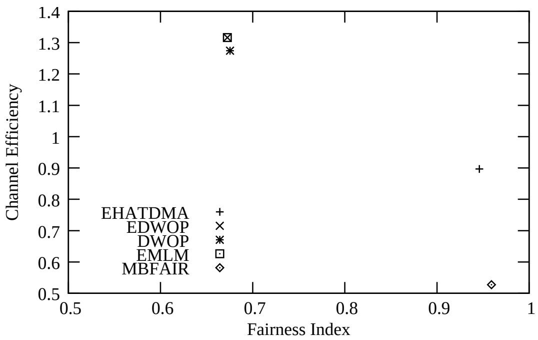
(c) Max−min Fairness Index

(d) Minimal Packet Throughput Rate

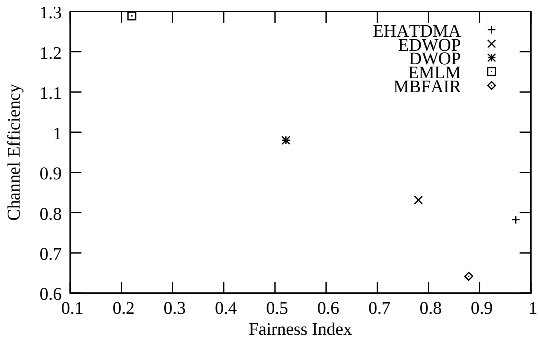
Figure 3.7: Simulation results of typical scenarios

ure 3.7(c)). In typical scenarios 4, 5 and 6, EMLM almost achieves a reuse degree of 2, much higher than others. However the high degree of reuse severely impairs fairness — the fairness of EMLM in these scenarios is the worst among all protocols. A similar phenomenon occurs in scenarios 16-19. In these scenarios, DWOP, EMLM and EDWOP achieve much higher overall throughput by sacrificing fairness. On the other hand, EHATDMA and MBFAIR attain much better fairness in these scenarios. However, their throughputs are also much lower.

Figure 3.7(a) and Figure 3.7(c) also reveal a fundamental difference in the philosophies adopted by different protocols in dealing with the fundamental conflict between throughput and fairness. Although EMLM, DWOP and EDWOP employ various mech-



(a) FI vs CE (typical scenario 16)



(a) FI vs CE (general scenario 7)

Figure 3.8: Fundamental conflict between fairness and throughput

anisms to improve fairness, they are still strongly in favor of overall throughput when a conflict between throughput and fairness arises. On the other hand, EHATDMA and MBFAIR prefer fairness to throughput. The difference is clearly shown in Figure 3.8(a), which plots the throughput and fairness achieved by different protocols in typical scenario 16.

Figure 3.7(c) shows that the FI_ms of EMLM, DWOP and EDWOP vary widely, which indicates that the fairness of these protocols are very sensitive to scenarios. For example, DWOP, which is the best of the three queueing-scheduling based protocols in terms of fairness, has $FI_m \approx 1$ in scenarios 1-5 and 7-10 while it has $FI_m < 0.7$ in scenarios 16, 17, 18 and 19. Figure 3.7(d) shows that in some scenarios, some protocols suffer the one/zero fairness problem ([61]) — some flows dominate the channel while others get nothing, such as with protocol EMLM and EDWOP in scenarios 3-6.

In contrast, the fairness performance of EHATDMA and that of MBFAIR are consistent. In most cases, the FI_ms of MBFAIR are very close to 1, and even in the worst case, its FI_m has a value as large as 0.9. However, Figure 3.7(a) indicates that the fairness of MBFAIR is achieved at a much higher cost — in most scenarios, the CEs of MBFAIR are less than half of those of EHATDMA. As we have mentioned, when the number of contending flows is 2, MBFAIR is inclined to choose a larger window size, which improves fairness but degrades aggregate throughput. EHATDMA has the best trade-off between throughput and fairness: it achieves very impressive fairness ($FI_m > 0.95$ for most cases) and maintains high throughput whenever possible. (Figure 3.7(a)).

The only scenario in which EHATDMA does not perform well is scenario 15, which suffers from LSP and LCP. It appears that the hybrid scheme, the ATDMA scheme and the power control scheme cannot figure out a way to work around the situation. In this scenario, algorithms with more intelligence are needed.

From the results presented above, we conclude that the fairness performance of the various schemes in typical scenarios are as follows: queueing-scheduling protocols prefer throughput to fairness and the fairness of these protocols is scenario-dependent whereas EHATDMA and MBFAIR favor fairness over throughput and their fairness performance is almost scenario independent; EHATDMA has the best overall performance. By employing explicit mechanisms dealing with LSP, DCP and LCP, EHATDMA achieves a high level of fairness while maintaining high throughput whenever possible in most typical scenarios. Since general scenarios can be decomposed into a combination of these typical scenarios, we can expect the same performance patterns from EHATDMA and queueing-scheduling protocols in the general scenarios[7].

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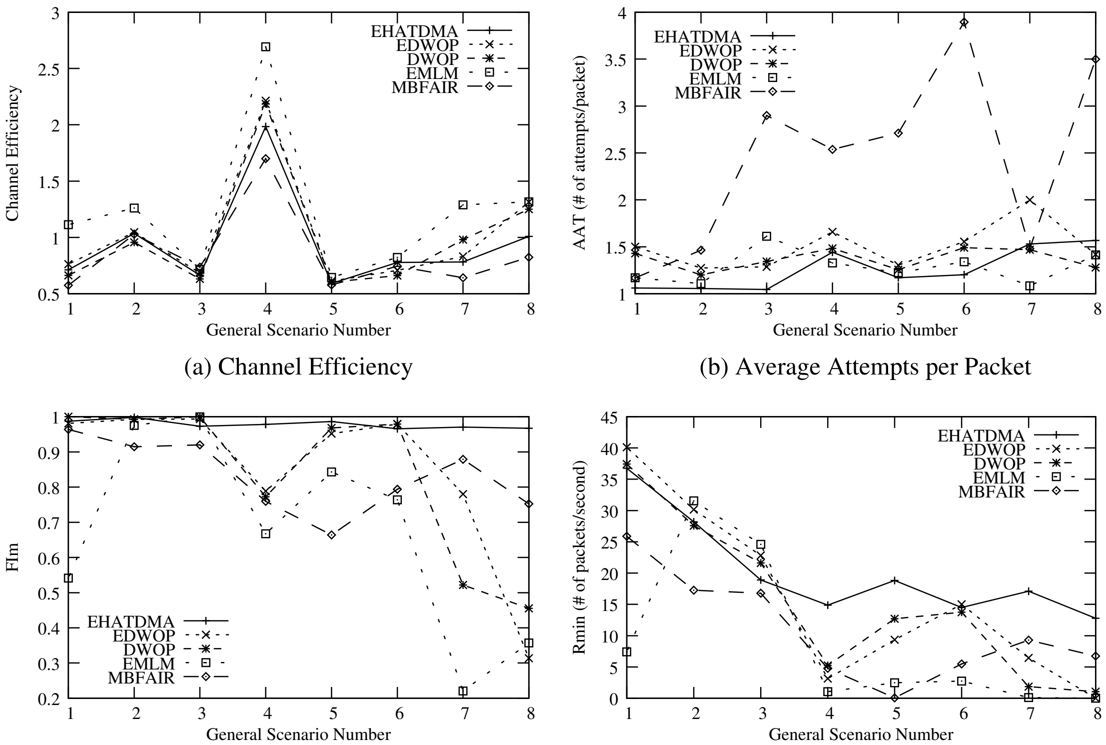
[7]We do not include MBFAIR in the statement because MBFAIR displays very different behaviors for cases where $N > 2$ and cases where $N = 2$



(a) Channel Efficiency (b) Average Attempts per Packet
(c) Max−min Fairness Index (d) Minimal Packet Throughput Rate

Figure 3.9: Simulation results of general scenarios

3.3.3.3 Simulation Results of General Scenarios

As expected, the FI_ms of the queueing-scheduling protocols vary widely from one scenario to another (Figure 3.9(c)): the FI_m ranges for EMLM, EDWOP and DWOP are $[0.2, 1]$, $[0.3, 1]$ and $[0.45, 1]$ respectively; MBFAIR is less sensitive to scenarios ($[0.65, 0.96]$); EHATDMA has the best fairness performance: in all the general scenarios, its $FI_m > 0.96$. The minimum flow throughput rate of EHATDMA is among the largest in all general scenarios (Figure 3.9(d)) while the AAT of EHATDMA is the smallest in most scenarios (Figure 3.9(b)).

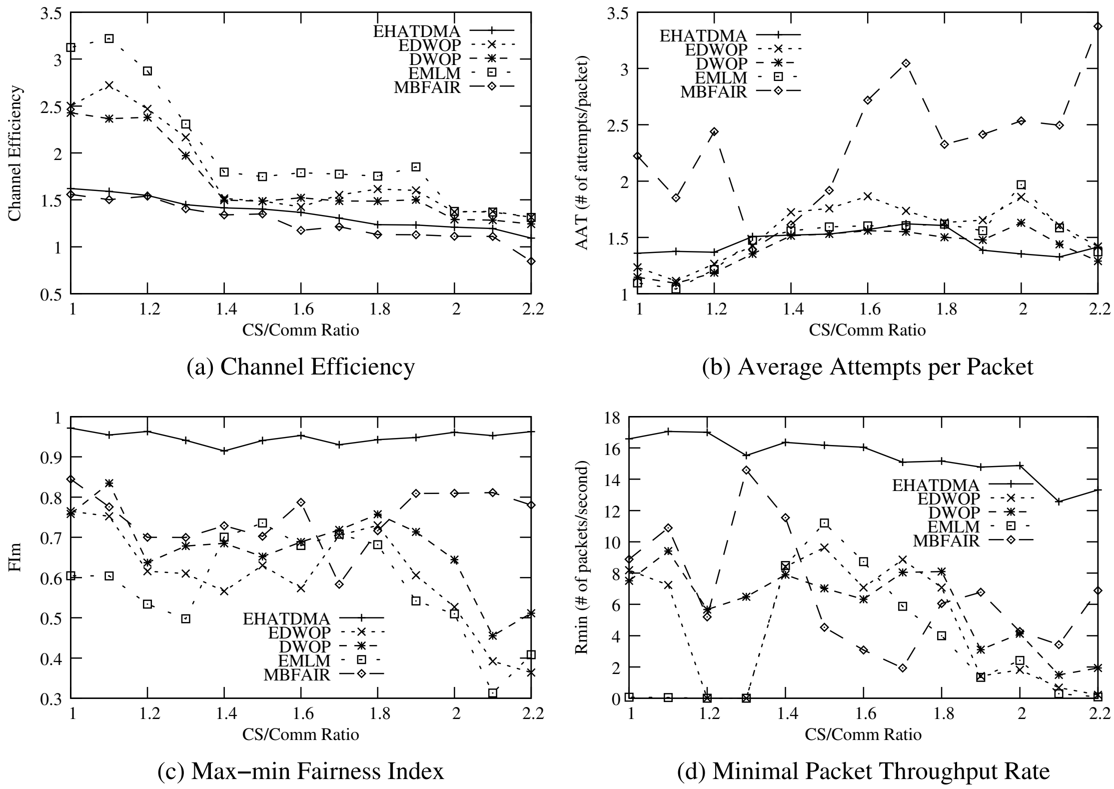
The fundamental conflict arises again. In general scenarios 1, 4, 7 and 8, EMLM achieves much higher CE than others. However, its fairness is also the worst. In these scenarios, EMLM acts very aggressively; the multihop network is virtually divided into several disconnected subnetworks. The flows bridging these subnetworks are starved (Figure 3.9(d)). To highlight the conflict, Figure 3.8(b) plots the channel efficiency and fairness achieved by the various protocols in general scenario 7.

MBFAIR performs very differently in general scenarios and in typical scenarios. In general scenarios, a node usually has more than one neighbor; hence, MBFAIR is inclined to choose a small contention window size, which increases throughput (compared with the one neighbor case) at the expense of AAT — the AAT of MBFAIR is much larger than those of others (Figure 3.9(c)), and fairness — the FI_ms of MBFAIR are much worse in general scenarios than in typical scenarios. On the other hand, the desirable properties of EHATDMA shown in the typical scenarios are preserved in the general scenarios.

3.3.4 The Impact of the Ratio of the Carrier Sensing Range to the Communication Range

In this subsection, we investigate the impact of the ratio of the CS range to the communication range (CS/Comm ratio) on the performance of MAC protocols. We have run a series of simulations in general scenarios 5-8 (Figure 3.5) with the CS/Comm ratio varying from 1.0 to 2.2. The capture threshold used in the simulation was 4db; all other parameters and assumptions remained unchanged. We only present the simulation results of scenario 8, as it is more complex than other scenarios, and statistically speaking, its result (Figure 3.10) is more representative.

Figure 3.10(c) shows that the fairness indexes of EMLM, DWOP, EDWOP and



(a) Channel Efficiency

(b) Average Attempts per Packet

(c) Max−min Fairness Index

(d) Minimal Packet Throughput Rate

Figure 3.10: The effects of carrier sensing range

MBFAIR vary widely when the CS/Comm ratio changes. For example, as the ratio increases, the fairness index of DWOP fluctuates in the range $[0.45, 0.8]$. However, the FI_m of EHATDMA remains stable as the range ratio changes, and is by far larger than those of others. Similarly, the R_{min} of EHATDMA is larger than those of others and more stable (Figure 3.10(d)). In addition, the AAT of EHATDMA is also very small (Figure 3.10(b)).

These behaviors are actually expected. The effect of the changing CS/Comm ratio is equal to the changing topology. As we have shown in previous subsections, EHATDMA exhibits weaker dependency on topology than the others do.

Note that when CS/Comm ratio < 1.4, EHATDMA and MBFAIR achieve much lower throughput than queueing-scheduling protocols do. This is again due to the fundamental conflict between throughput and fairness. For example, when the CS/Comm ratio is small, suppressing flow 4→9 may enable three flows (3→0, 16→13 and 7→5) to transmit simultaneously, which obviously increases throughput substantially.

3.3.5 The Impact of Mobility and the Convergence Time of EHAT-DMA

Until now, we have only considered static scenarios where topology and traffic remain unchanged. For a mobile network, the scenario is indeed a dynamic one since topology and traffic load vary as nodes move around. However, we can view the evolution of a dynamic scenario as shifting through a chain of static scenarios: $S_1 \rightarrow S_2 \rightarrow \ldots$, where S_i is a static scenario and $S_i \neq S_{i+1} (i = 1, 2, \ldots)$. Hence, the performance of a MAC protocol in dynamic scenarios can be investigated from two aspects: (i) performance in static scenarios, which we have addressed in previous subsections; (ii) ability in dealing with scenario shift.

A cogent indicator for the ability of a MAC protocol in coping with scenario shift is convergence time — the time needed to reach a stable state after a scenario shift. If a MAC protocol converges rapidly after any scenario shift and maintains high performance thereafter, we can conclude that the protocol works well in dynamic scenarios.

The definition for the fairness index FI is extended to explore convergence time. Suppose that a scenario shift $S_{i-1} \rightarrow S_i$ occurs at time 0, the fairness index at time t

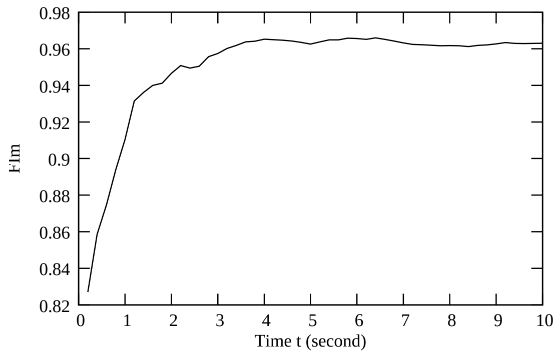


Figure 3.11: An example of the $FI_m(t)$ of EHATDMA in general scenario 8

$(FI(t))$ is calculated by:

$$FI_m(t) = \frac{\left(\sum_{f \in F} \frac{T_f(t)}{\phi_f}\right)^2}{n \sum_{f \in F} (\frac{T_f(t)}{\phi_f})^2}$$

where $T_f(t)$ is the throughput of flow f during the interval $(0, t)$ and ϕ_f the fair share for flow f in the new static scenario S_i. The convergence time can be easily found in the plot of $FI(t)$. Figure 3.11 is an example of the $FI_m(t)$ of EHATDMA in general scenario 8. During the time interval $(-1, 0)$, the 14 flows in general scenario 8 randomly start one after another, hence triggering a series of swift scenario shifts. After all flows have started, $FI_m(t)$ is calculated in a step of 0.2 seconds.

Figure 3.11 shows that although the scenario shifts swiftly — within one second, 14 flows randomly start one after another, EHATDMA can still accommodate the shift in a timely manner — within just 0.2 second, the fairness index $FI_m(t)$ reaches beyond 0.82, which is already much better than the stable FI_ms of other protocols. Within less than three seconds, the $FI_m(t)$ reaches 0.96 and remains stable hence. In other scenarios, EHATDMA achieves even shorter convergence times. The rapid convergence of EHATDMA clearly indicates that it can deal with mobile scenarios elegantly.

3.4 Overhead and Implementation Complexity of EHATDMA

Our simulation results show that EHATDMA performs very well in all the benchmark scenarios. Another attractive feature is its low communication and computation overhead. In EHATDMA, for each packet transmitted by SI mode, the communication overhead is (RTS + CTS + ACK) + DATA header = 24 + 24 + 24 + 32 =104 bytes. For each packet transmitted by RI mode, the corresponding communication overhead is (POLL +ACK) + DATA Header = 24 + 24 + 32 = 80 bytes. For a data packet of size 512 bytes, the efficiencies of the SI and RI modes are thus 83% and 86% respectively. This is comparable to the efficiency of IEEE 802.11, for which the overhead is 72 bytes/packet, and the channel efficiency is 88% for a packet with the size of 512 bytes. Furthermore, the computation overhead of EHATDMA is also very low. EHATDMA does not require service tags to be computed for packets nor for the flow table to be sorted. Usually, these two operations are necessary in the fair queueing and scheduling schemes, thus incurring heavy computation loads.

EHATDMA is also simpler to implement. Most operations of EHATDMA are already available in IEEE 802.11 and it only requires two additional simple tables to be maintained — the flow table and the neighbor table. All the information needed in these two tables can easily be obtained by overhearing the packets. The channel monitor is also easy to implement (the IEEE 802.11 standard requires the channel status to be provided by the physical layer to the MAC layer, which facilitates the implementation of channel monitoring). EHATDMA is also robust; incomplete and imprecise information will not generate deadlock problems (deadlock is possible in fair queueing and scheduling schemes for multihop wireless networks [48]).

3.5 The Analysis of Individual Mechanisms

In this section, we analyze the effects of individual control mechanisms on the performance of the various fairness schemes under investigation. First, we focus our analysis on the single/multiple scheduling strategy and the $T_{backoff}$ out-of-order treatment of fair queueing-scheduling schemes, and then on the individual mechanisms employed in EHATDMA.

3.5.1 The Effects of Single/Multiple Scheduling Strategy and Out-of-order Backoff

The key differences of all queueing-scheduling schemes lie in their scheduling policy and treatment of out-of-order events. In these respects, there are three basic elements in the core of the three queueing-scheduling protocols (EMLM, DWOP and EDWOP):

- *Multiple-scheduling strategy:* All the outgoing flows of a node are allowed to contend for the channel, and the contention window for each flow is derived from its rank in the tag table.
- *Single-scheduling strategy:* Only the flow with the lowest tag in a node is allowed to contend for the channel.
- *Out-of-order backoff:* When an out-of-order event is detected, a flow is penalized to backoff for $T_{backoff}$ time.

EMLM employing only the multiple-scheduling strategy acts very aggressively in channel access. The resulting effect favours throughput over fairness. This is clearly reflected in the results of the general scenarios (Figure 3.9), where EMLM always produces the highest throughput and yields the worst fairness.

To improve the fairness property of EMLM, we introduce the out-of-order backoff scheme proposed in DWOP into EMLM; the resultant protocol is named EDWOP. The out-of-order backoff scheme improves fairness dramatically in most cases (Figure 3.9(c)).

Comparing the performance of EDWOP and DWOP, it seems that neither of them has a clear advantage over the other. Although the multiple-scheduling strategy is likely to improve throughput, it is also likely to compromise fairness. Simulation results show that the throughput gained from the multiple-scheduling strategy is indeed marginal.

From the above analysis, we conclude that the most important mechanism improving the fairness property of a MAC protocol employing queueing-scheduling is the out-of-order backoff strategy; the benefits of multiple scheduling strategy is quite marginal.

3.5.2 The Effects of Hybrid, ATDMA and Power Control

In this section, we analyze how the individual control mechanisms (namely the hybrid scheme, the ATDMA scheme and the power control scheme) employed in EHATDMA affect fairness performance.

Figure 3.12 shows the simulation results of the general scenarios in the presence of these control mechanisms, which are denoted as H, A and P respectively, for the SI-RI hybrid scheme, the ATDMA scheme and the power control scheme. The results indicate that the hybrid scheme alone can improve fairness substantially in most cases

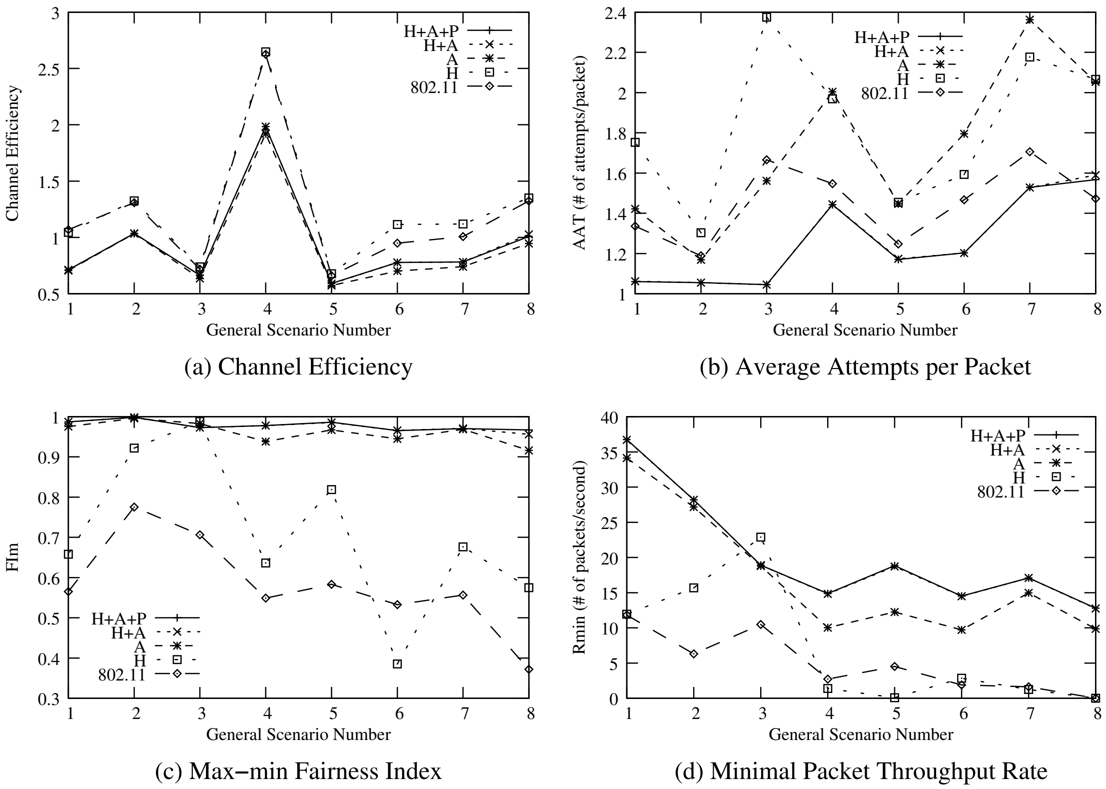


Figure 3.12: The effects of the hybrid scheme, the ATDMA scheme and the power-control scheme

(Figure 3.12(c)). In some cases, the hybrid scheme also improves throughput (Figure 3.12(a), scenarios 6 and 7). However, it cannot improve the minimal throughput rate and it increases AAT. This is mainly due to the inability of the hybrid scheme to deal with the double contention areas problem (DCP).

The most influential fairness control mechanism of EHATDMA (i.e., H+A+P) is ATDMA. As shown in (Figure 3.12(c)), ATDMA is the dominating contributor to better fairness with respect to the other two schemes (i.e., the hybrid scheme and the power control scheme). The CE and R_{min} of ATDMA are also very close to those of EHATDMA (Figure 3.12(a,d)). The only drawback of ATDMA is its higher AAT, which can be overcome by combining the hybrid scheme and ATDMA. As shown in Figure 3.12, the performance curve for H+A almost overlaps with that of EHATDMA. This implies the irrelevancy of power control in the presence of both the hybrid scheme and ATDMA in these scenarios. However, the power control scheme may still be useful in other rare scenarios.

The results presented in this section reveal that the non-work-conserving channel access mechanism plays a vital role in providing fair sharing to contending flows. In fair queueing-scheduling schemes, the out-of-order backoff scheme serves as the non-work-conserving mechanism. A flow is scheduled not to work for $T_{backoff}$ time if it violates the channel-access order determined by the queueing algorithm. In EHATDMA, the ATDMA scheme is the non-work-conserving mechanism. After a successful transmission, a flow is paused for a time of $flow\_period$, which is derived from the traffic load observed by the flow. One major difference between these two mechanisms is that ATDMA is a self-adaptive mechanism (as the traffic load around a flow changes, the $flow\_period$ also changes) while out-of-order backoff is not. This property allows ATDMA to work well with different network connections and different traffic loads. $T_{backoff}$ is somehow fixed. It is defined as: $T_{backoff} = R(eifs + difs + T_{success} + CW_{min})$, where R is the number of packets that should have been transmitted before the violating packet, and $T_{success}$ is a fixed value representing the longest possible time required to transmit a data packet. Fixing the value of $T_{success}$ may present a performance problem. If the actual time needed to transmit an average-length packet is much smaller than the value of $T_{success}$, the throughput of flows may be penalized severely. On the other hand, if the average packet length approaches the maximal packet length, $T_{success}$ may not be large enough to maintain the desired level of fairness.

The relationship between non-work-conserving mechanisms and fairness is quite straightforward. As we have identified, one fundamental cause of the fairness problem is LSP (DCP can be reduced into LSP): the receiver of a flow f is exposed to a hidden terminal of the sender (say, node i) that is transmitting most of the time. If node i works in a work-conserving manner; it contends for the channel persistently, and once it sends a packet successfully, it builds up its advantage in channel acquisition and eventually starve flow f. However, if node i works in a non-work-conserving manner, it relinquishes the channel for a time after each successful transmission, which gives the flow f a higher success probability in channel acquisition. As a result, the fairness problem is relieved. The key issue is the design of a non-work-conserving mechanism that can achieve acceptable fairness while maintaining high throughput. ATDMA makes a very good trade-off in this aspect.

3.6 Summary

In this chapter, we have investigated the fairness properties of MAC protocols in multihop ad hoc carrier sense networks in which the carrier sense range is larger than the interference, and the latter is in turn larger than the communication range. We have proposed a new MAC protocol known as extended hybrid asynchronous time division multiple access (EHATDMA) to deal with the severe unfairness caused by the lack of synchronization problem (LSP), the double contention areas problem (DCP) and the lack of coordination problem (LCP). The protocol has three control schemes. The first is the SI-RI hybrid scheme for dealing with LSP. It employs both SI and RI collision avoidance mechanisms. The second is ATDMA, which addresses DCP. It requires a flow that has just successfully transmitted a packet to refrain from accessing the channel for a $flow\_period$ estimated based on the traffic load around the flow. The third is a power control algorithm, which deals with LCP. A node adjusts its transmission power for CTS/POLL when experiencing LCP. For better assessment of fairness, we have also designed an index named the max-min fairness index, which is scenario-independent and reflects the difference between the fair sharing provided by a protocol and the ideal max-min fair sharing.

We have carried out comprehensive simulation experiments with EHATDMA and other related protocols (IEEE 802.11, CBFAIR, MBFAIR, EMLM, DWOP and EDWOP) in a series of comparative performance studies. Our simulation results show that while various enhancements have been proposed to improve the fairness of MAC protocols of multihop wireless networks, most of them are still strongly biased towards throughput when a conflict between throughput and fairness arises. In addition, the fairness performance of these proposals vary widely from one scenario to another. On the other hand, EHATDMA strikes a good balance between throughput and fairness. It delivers a consistently high level of fairness regardless of network topology, traffic load and radio parameters, yet maintains high throughput whenever possible. Furthermore, EHATDMA is also able to deal with mobility elegantly; it can rapidly reach a new stable state after a scenario shift. Our simulation results also reveal that the most important mechanism affecting the fair sharing of radio channels among flows is the non-work-conserving mechanism.

In the next chapter, we will establish a mathematical model under some simplified assumptions to evaluate the throughput and fairness properties of IEEE 802.11 in multihop wireless ad hoc networks. The model will confirm that non-work-conserving mechanisms do improve the fairness property of IEEE 802.11.

CHAPTER 4

Fairness and Throughput Analysis of IEEE 802.11 in Multihop Wireless Ad Hoc Networks†

There are three techniques commonly used for performance evaluation: analytical modeling, simulation and measurement [63]. The viability of performance measurement is often hampered by the the limited access of the firmware of commercial wireless cards. Consequently, performance evaluation of MAC protocols is often carried out either by means of simulation as we have done in the previous chapter [64, 65], or by means of analytical modeling. In this chapter, we opt for the analytical approach in studying the fairness and throughput of IEEE 802.11 in the context of multiple wireless ad hoc networks.

In the literature, analytical modeling of wireless MAC protocols has been confined to either single-hop networks [66–71], or multihop ad hoc networks with the assumption of simple backoff rules [35, 72–77]. Due to the hidden terminal problem, analytical models developed for singe-hop networks cannot be applied in multihop networks. The

†Part of this chapter (throughput analysis) has been presented in the paper for ICON 2004: **Jun He** and Hung Keng Pung, "Performance Modeling and Evaluation of IEEE 802.11 Distributed Coordination Function in Multihop Wireless Networks", *IEEE International Conference on Networks (ICON)*, Singapore, 2004, pp. 73-79. **(Best Student Paper)**

analytical modelings for multihop networks have restricted the study of these networks in relation to throughput performance ([35, 72–75, 77]) to the simple backoff schemes such as constant or geometrically distributed backoff window with parameter p (i.e., for every slot, a node transmits a packet with probability p, and otherwise with probability $1 - p$, where $0 < p \leq 1$). In these studies, theoretical throughput is usually given as a function of p. Although these theoretical results are useful for comparative study, they cannot predict throughput of a MAC protocol with a given exponential backoff scheme. In addition, these studies have not addressed fairness problems in their analysis.

We have shown in the previous chapters that MAC protocols in multihop wireless networks are vulnerable to the fairness problem, as some nodes (or flows) yield larger throughput than others ([40]). Several enhancements to wireless MAC protocols have been proposed to alleviate this problem ([20, 45, 51, 61, 62, 78]). Although these studies have shed some light on the fairness problem, their investigations have mainly been confined to simulation. To our knowledge, there is no analytical model for fairness analysis in multihop wireless networks. Though Chhaya and Gupta analyzed the fairness problem in [67], their proposed model is only valid for cases where the number of hidden nodes is small. Hence their analysis cannot be applied to multihop wireless networks, where hidden nodes prevail.

In this chapter, we propose an analytical model to investigate the throughput and fairness properties of IEEE 802.11 [34] in multihop wireless networks, with nodes randomly placed according to a two-dimensional Poisson distribution. We are interested in the saturation throughput of a node in such a network and the fairness of channel sharing among one-hop flows of various source-destination[1] distances. IEEE 802.11 is chosen because of its pervasive use in wireless LANs and its being the *de facto* standard MAC protocol used in research on multihop wireless networks. In IEEE 802.11, the fundamental mechanism to access the medium is called distributed coordination function (DCF), which is based on the carrier sense multiple access with collision avoidance (CSMA/CA) scheme. It also employs a binary exponential backoff (BEB) algorithm to manage retransmission of collided packets. DCF describes two channel access methods. The default method is a two-way handshaking technique called the basic access method, in which a positive ACK is transmitted by the destination to confirm a successful packet transmission. The second method is a four way handshaking mechanism which uses a request-to-send/clear-to-send (RTS/CTS) technique to reserve the channel prior to data transmission. The purpose of the RTS/CTS exchange is to reduce performance degradation due to hidden terminals. In this chapter, we analyze the throughput and fairness properties of both access methods.

The rest of the chapter is organized as follows. In Section 4.1, we briefly review both access methods of DCF. In Section 4.2, we present our analytical model of DCF for multihop wireless networks. In Section 4.4 and Section 4.5, we apply the model to derive throughputs for the basic access method and the RTS/CTS access method, respectively. Throughput and fairness evaluation of DCF are carried out in Section 4.6 and Section 4.7, respectively. Section 4.8 summarizes this chapter.

[1]Throughout this chapter, "source" and "destination" refer to MAC layer source and destination, respectively

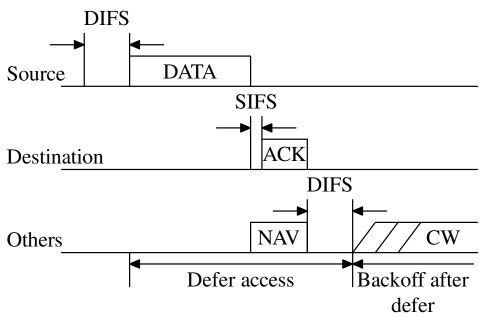


Figure 4.1: The basic access method in DCF

4.1 Distributed Coordination Function of IEEE 802.11

This section briefly describes the operations of the basic method and the RTS/CTS method of DCF. Readers should refer to the IEEE 802.11 standard [34] for a more detailed presentation.

4.1.1 The Basic Access Method

A node with a new packet may proceed with its transmission if the channel is sensed to be idle for an interval larger than the distributed inter frame space (DIFS). Otherwise, the node persists to sense the channel until the channel is idle for a DIFS and then generates a random backoff interval, which is uniformly drawn in the range $(0, w-1)$. The value of w is known as the contention window (CW), which is an integer within the range determined by PHY characteristics CW_{min} and CW_{max}. Initially, w is set to be CW_{min}. After each failure, w is doubled, up to a maximum CW_{max}. This procedure is known as binary exponential backoff (BEB). The backoff timer is decreased as long as the channel is idle, frozen when the channel is busy, and resumed when the channel is idle again for a DIFS. The node transmits when the backoff timer reaches zero.

Upon receiving a DATA frame correctly, the destination waits for an SIFS and then replies with an ACK to confirm the reception (Figure 4.1). In case the source does not receive an ACK, the data frame is assumed lost, and the source either reschedules the transmission according to the given backoff rules, or discards the packet and resets w to CW_{min} if the number of retries reaches an upper limit.

DCF has a virtual carrier sense mechanism at the MAC layer. Within each frame (RTS, CTS, DATA and ACK), there is a field conveying the remaining time of the current transmission. All other nodes overhearing a frame adjust their network allocation vector (NAV) and will not try to transmit until NAV expires.

4.1.2 The RTS/CTS Access Method

DCF also provides an optional channel access mechanism known as the RTS/CTS access method. Two short control frames RTS and CTS are introduced to reduce collision time and to deal with the hidden terminal problem. Prior to the transmission of a data frame, the source transmits an RTS to the destination. Upon receiving the RTS, the destination will reply with a CTS after an SIFS. The source is allowed to transmit its packet only if the CTS frame has been correctly received. Ideally, after the exchange of RTS and CTS, all neighbors of the source and the destination will be notified of the transmission attempt (via actual or virtual carrier sense mechanism), and will refrain from transmission; consequently, the data frame can be transmitted without collision (Figure 4.2).

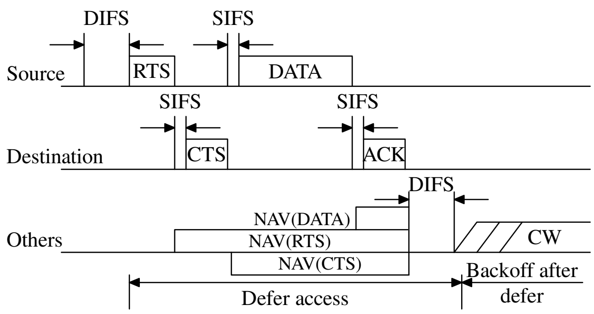


Figure 4.2: The RTS/CTS access method in DCF

4.2 Analytical Model of DCF in Multihop Wireless Networks

Our analytical model is based on the work of Bianchi in [70], where a two-dimensional Markov model is proposed for *single-hop* networks to derive the saturation throughput of DCF. We have extended Bianchi's model significantly and succeeded in providing an analytical model of DCF for *multihop* wireless networks.

4.2.1 Assumptions, Throughput and Fairness Definitions

The following is a list of assumptions of our analytical model for DCF.

1. Nodes in a network are Poisson distributed over a plane with density λ, i.e., the probability $p(i, A)$ of finding i nodes in an area of A is given by:

$$p(i, A) = \frac{(\lambda A)^i}{i!} e^{-\lambda A}.$$

2. We assume a heavy traffic load condition, i.e., a node always has a packet to send and the destination is chosen randomly from one of its neighbors. Furthermore, all packets are of the same size.

3. All nodes use the same fixed communication range of R. Therefore, the hearing region of any node is πR^2, and the average number of neighbors of a node, denoted as N, is $N = \lambda \pi R^2$. For simplicity of notation, we normalize all distance with respect to R, and set $R = 1$. Hence, we have $N = \lambda \pi$. We assume identical ranges of communication, interference and carrier sensing.
4. The channel is ideal and all errors are due to collision.
5. For simplicity, we do not take the capture effect into consideration; that means any overlap of two transmissions arriving at a node will lead to a collision at that node.
6. The collision probability of a packet is independent of the number of retransmissions (if any) and the location of the source, *but* is dependent on r, which is the distance between the source and the destination. We denote $p(r)$ as the collision probability of a transmission with a distance of r.
7. All-time related parameters are normalized with respect to the slot time θ, which is specified by the PHY in IEEE 802.11. For simplicity of notation, we set $\theta = 1$. Let l_{sifs}, l_{difs}, l_{rts}, l_{cts}, l_{data} and l_{ack} denote the length of the SIFS, DIFS, RTS, CTS, DATA and ACK frames, respectively.

Remarks: Two-dimensional Poisson node distribution is commonly assumed in studies on the performance analysis of radio networks ([74, 75, 77]). It should be noted that this assumption is introduced not just for mathematical convenience. Two other important reasons have led us to the choice. First, we are interested in the throughput and fairness properties of a MAC protocol in "general" scenarios. Without *a priori* knowledge about the disposition of nodes, two-dimensional Poisson is a reasonable approximation of node distribution. This is particularly true if nodes can move about randomly. Second, the throughput of a channel and the fairness of channel shares allocated by a MAC protocol could vary widely from one scenario to another. With the two-dimensional Poisson distribution assumption, nodes are randomly and uniformly placed in a plane. Throughput and fairness achieved in such an arrangement can be viewed as the "average" throughput and fairness, which we believe is a better measure to reflect the throughput and fairness properties of a MAC protocol. A heavy traffic load condition (assumption (2)) is assumed because the fairness problem is not prominent when traffic load is light and we are more interested in the fairness property of a MAC protocol operating in a saturation state. The heavy traffic load condition models the scenario in which every node is busy in forwarding packets to others. Assumption (6) extends the *conditional collision probability* assumption made in [70]. It enables us to establish a three-dimensional Markov model for a node in multihop wireless networks. Assumptions (3) and (5) are made to avoid unnecessary complexity of presentation. With some minor modifications, the analytical model can be applied to networks where the carrier sensing range is different from the communication range. The capture effect can also be easily subsumed under the model by incorporating it into the computation of $p(r)$.

To investigate the throughput and fairness properties of DCF, we define three kinds of throughput as follows:

- TH — *node saturation throughput*: It is defined as the fraction of time used by a node to successfully transmit payload bits.
- $TH(r)$ — *long-run throughput of a flow with a distance of* r: It is defined as

the fraction of time used by a flow with a source-destination distance of r to successfully transmit payload bits. Later on, we will show their relationship as $TH = \int_0^1 TH(r) \cdot N \cdot 2r \cdot dr$. In our model, flows of different distances have the same chance to be scheduled to transmit a packet because a node randomly chooses one of its neighbors as its destination (assumption (2)). Hence, ideally we should have:

$$TH(r_i) = TH(r_j), r_i, r_j \in [0, 1].$$

- $TH_I(r)$ — *instant throughput of a flow with a distance of* r: It is defined as the ratio of the time used by a flow to successfully transmit payload bits to the total time used by that flow. The difference between $TH(r)$ and $TH_I(r)$ is that $TH(r)$ is the throughput of a flow observed over long-run time while $TH_I(r)$ is the throughput of a flow observed over only the period of time when a packet of that flow is scheduled for transmission. $TH_I(r)$ is closely related to the average service time of a flow, i.e., when a packet of a flow is scheduled, how much time is needed to transmit it successfully. For an ideally fair MAC protocol, we should have:

$$TH_I(r_i) = TH_I(r_j), r_i, r_j \in [0, 1].$$

For the fairness property, we are interested in the problem of whether the relationships:

$$\begin{cases} TH(r_i) & = & TH(r_j) & r_i, r_j \in [0, 1] \\ TH_I(r_i) & = & TH_I(r_j) & r_i, r_j \in [0, 1] \end{cases}$$

hold for the two access methods of DCF. If not, we want to find out to what extent the deviation is and what kinds of mechanism can be employed to deal with the fairness problem. In the following sections, we will establish an analytical model to calculate TH, $TH(r)$ and $TH_I(r)$.

4.3 Three-Dimensional Markov Chain

With the assumptions (1), (2) and (6), the activities of a node can be modeled by a three-dimensional Markov chain $\{s(t), b(t), r(t)\}$. Here, $s(t)$ is the backoff stage of a node at time t and it records the number of retransmissions a packet has suffered; $b(t)$ is the backoff counter of a node at time t; $r(t)$ is the distance between a node and its destination at time t.

In IEEE 802.11, a packet is discarded after m retries. The value of m is specified in the standard and is 4 and 7 for the data frame and RTS frame respectively. Hence, the maximum value of $s(t)$ is m.

A node's contention window (CW) size is either doubled or kept at a maximum value CW_{max} after each failure, and is reset to CW_{min} after a successful transmission or when a packet is discarded. CW_{min} and CW_{max} are specified by PHY in IEEE 802.11. In this chapter, we use parameters assigned for direct sequence spread spectrum (DSSS) PHY. For DSSS, CW_{min} and CW_{max} are 31 and 1023 respectively. Therefore, for a node in backoff stage i, the backoff window size W_i is:

$$\begin{cases} W_i = 2^i W & i \leq m' \\ W_i = 2^{m'} W & i > m', \end{cases}$$

where $W = (CW_{min} + 1)$ and $2^{m'}W = (CW_{max} + 1)$; hence for DSSS, we have $m' = 5$.

$r(t)$ is the distance between the node and its destination at time t. Obviously, it is a continuous process in the range $[0, 1]$. For simplicity of analysis, we discretize the range $[0, 1]$ into n equal intervals: $I_1 = [0, 1/n), I_2 = [1/n, 2/n), \ldots, I_n = [(n-1)/n, 1]$ (correspondingly, the hearing region of a node is divided into n rings), and any value lying in the k^{th} interval I_k is replaced with the middle point of I_k. With this conversion, we get a discrete process $r(t)$, $r(t) \in \{r_k | k \in [1, n]\}$, where:

$$r_k = (2k-1)/(2n).$$

Given that each source uniformly chooses one of its neighbors as destination (assumption (2)), and that the average number of nodes within a region of radius r is proportional to r^2, the probability density function of the distance r between a source and its destination is:

$$f(r) = 2r,\ 0 \le r \le 1.$$

And the corresponding probability mass function is:

$$P\{r = r_k\} = 2r_k \cdot 1/n = 2r_k \cdot \Delta r, \tag{4.1}$$

where $\Delta r = 1/n$.

Since we assume that collision probability is independent of $s(t)$ and destination is randomly chosen, $\{s(t), b(t), r(t)\}$ is a discrete-time Markov chain, which is shown in Figure 4.3 and Figure 4.4.

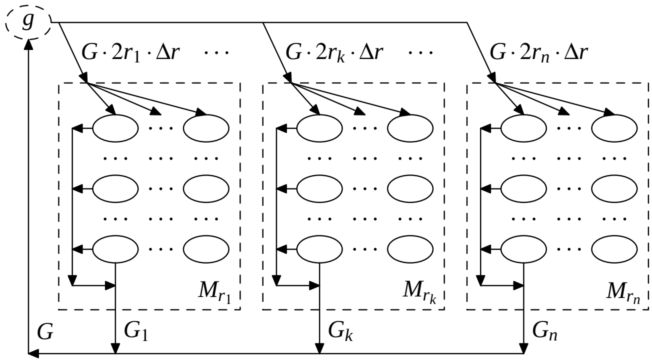


Figure 4.3: Overall Markov chain model for a node

Figure 4.3 shows the overall Markov chain of a node. After a successful transmission or after discarding a packet, a node randomly chooses one of its neighbors as the destination of the next packet. The probability that the destination lies in k^{th} ring I_k is $2r_k \cdot \Delta r$ (equation (4.1)). Note that the state g in Figure 4.3 is a pseudo-state; it is introduced for the convenience of presentation. Figure 4.4 shows the detailed Markov chain for transmissions with a distance of r_k.

The states of this Markov chain can be divided into two groups, a *wait* group:

$$wait = \{(i, j, k) | i \in [0, m], j \in [1, W_i - 1], k \in [1, n]\}$$

and a *trans* group:

$$trans = \{(i, 0, k) | i \in [0, m], k \in [1, n]\}.$$

A node in *trans* state is transmitting. The outcome of a transmission is either success or collision. A node in *wait* state defers to other nodes if it detects a busy channel; it makes a state transition $(i, j+1, k) \to (i, j, k)$ by decreasing its backoff counter by 1 if it detects an idle slot. We define the time interval between two consecutive backoff

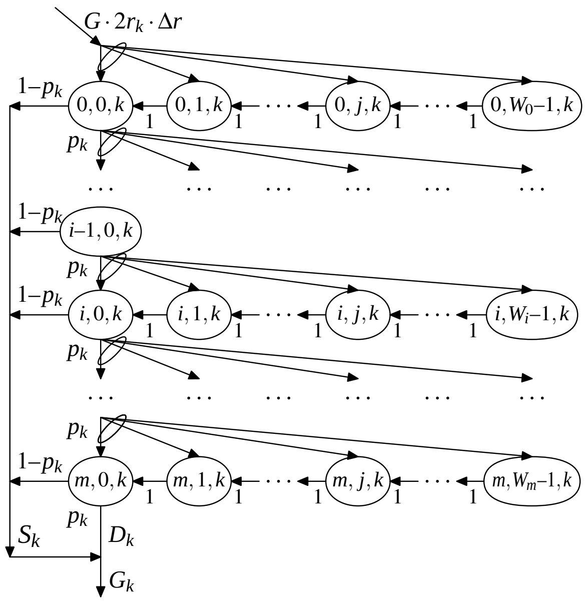


Figure 4.4: M_{r_k} : Markov chain model of the binary exponential backoff (BEB) window scheme for transmissions with a distance r_k

time counter decrements as a *macro-slot* (denoted as Θ), i.e., the time a node dwells in *wait* state. Obviously, a *macro-slot* is a variable interval consisting of a busy period (the length of the busy period is variable and might be zero) and an idle slot (real slot θ, as specified by PHY).

The only non-null one-step transition probabilities in this Markov chain are[2]:

$$\begin{cases} P\{i,j,k \mid i,j{+}1,k\} = 1 & i \in [0,m], j \in [0,W_i{-}2], k \in [1,n] \\ P\{i,j,k \mid i{-}1,0,k\} = \dfrac{p_k}{W_i} & i \in [1,m], j \in [0,W_i{-}1], k \in [1,n] \\ P\{g \mid i,0,k\} = 1{-}p_k & i \in [0,m{-}1], k \in [1,n] \\ P\{g \mid m,0,k\} = 1 & k \in [1,n] \\ P\{0,j,k \mid g\} = \dfrac{2r_k \cdot \Delta r}{W_0} & j \in [0,W_0{-}1], k \in [1,n], \end{cases} \tag{4.2}$$

[2]We adopt the short notations: $P\{i_1, j_1, k_1 | i_0, j_0, k_0\} = P\{s(t+1) = i_1, b(t+1) = j_1, r(t+1) = k_1 | s(t) = i_0, b(t) = j_0, r(t) = k_0\}$. And [a,b] $= \{x | a \leq x \leq b\}$

where $p_k = p(r_k)$. The first equation in (4.2) accounts for the fact that the backoff time is decremented. The second equation accounts for the fact that the source reschedules a collided packet. After a successful transmission or the discarding of a packet (the third and the fourth equations), the source will randomly choose a destination for a new packet (the fifth equation).

Let G denote the sum of all transition rates joining pseudo state g

$$G = \sum_{k=1}^{n} G_k.$$

From the local balance of M_{r_k}, we get:

$$G_k = G \cdot 2r_k \cdot \Delta r.$$

Let $b_{i,j,k}$ be the stationary distribution of the Markov chain. By using the local balance equation for each stage in M_{r_k}, we get:

$$b_{0,0,k} = G \cdot 2r_k \cdot \Delta r \tag{4.3}$$

and

$$b_{i,0,k} = p_k \cdot b_{i-1,0,k},\ 0 < i \leq m. \tag{4.4}$$

Therefore, we have:

$$b_{i,0,k} = p_k^i \cdot b_{0,0,k},\ 0 < i \leq m. \tag{4.5}$$

Owing to the regularities of the chain, for each $j \in [1, W_i - 1]$, we have:

$$b_{i,j,k} = \frac{W_i - j}{W_i} \begin{cases} G \cdot 2r_k \cdot \Delta r & i = 0 \\ p_k \cdot b_{i-1,0,k} & 0 < i \leq m. \end{cases} \tag{4.6}$$

With equations (4.3) and (4.4), equation (4.6) can be simplified as:

$$b_{i,j,k} = \frac{W_i - j}{W_i} b_{i,0,k} \quad 0 \leq i \leq m. \tag{4.7}$$

Finally, by imposing the normalization condition, we have:

$$\begin{aligned} 1 &= \sum_{k=1}^{n} \sum_{i=0}^{m} \sum_{j=0}^{W_i - 1} b_{i,j,k} \\ &= \sum_{k=1}^{n} G \cdot 2r_k \cdot \Delta r \sum_{i=0}^{m} p_k^i \frac{W_i + 1}{2} \\ &= G \sum_{k=1}^{n} 2r_k \cdot \Delta r \cdot A(r_k), \end{aligned} \tag{4.8}$$

where

$$A(r_k) = \sum_{i=0}^{m} p_k^i (W_i + 1)/2.$$

Note that p_k is a function of r_k (yet to be known). For DSSS, $A(r_k)$ is given by equation (4.9):

$$A(r_k) = \begin{cases} \frac{2(1-2p_k))(1-p_k)}{W(1-2p_k^{m+1})(1-p_k)+(1-2p_k)(1-p_k^{m+1})} & m \leq m' \\ \frac{2(1-2p_k)(1-p_k)}{W(1-2p_k^{m'+1})(1-p_k)+(1-2p_k)(1-p_k^{m+1})+W2^{m'}p_k^{m'+1}(1-p_k)(1-p_k^{m-m'})} & m > m' \end{cases} \tag{4.9}$$

The continuous version of equation (4.8) is:

$$1 = G \int_0^1 A(r) \cdot 2r \cdot dr.$$

Hence, we have:

$$G = \frac{1}{\int_0^1 A(r) \cdot 2r \cdot dr}.$$

Once we get G, the stationary probability of any state (i, j, k) $b_{i,j,k}$ can be calculated by equations (4.7), (4.5) and (4.3).

Some quantities are of particular interest to us. For example, $\bar{G}$ — the probability that a node transmits in a *macro-slot*:

$$\begin{aligned}
\bar{G} &= \sum_{k=1}^{n}\sum_{i=0}^{m} b_{i,0,k} = G\sum_{k=1}^{n} \frac{1-p_k^{m+1}}{1-p_k} \cdot 2r_k \cdot \Delta r \\
&= \sum_{k=1}^{n}\sum_{i=0}^{m} p_k^i \cdot G \cdot 2r_k \cdot \Delta r \\
&= G\sum_{k=1}^{n} \frac{1-p_k^{m+1}}{1-p_k} \cdot 2r_k \cdot \Delta r \\
&= G\int_0^1 \frac{1-[p(r)]^{m+1}}{1-p(r)} \cdot 2r \cdot dr,
\end{aligned}$$

S_k — the successful rate of transmissions from a node to its neighbors located in the ring of r_k:

$$S_k = \sum_{i=0}^{m} (1-p_k) \cdot b_{i,0,k} = G(1-p_k^{m+1}) \cdot 2r_k \cdot \Delta r,$$

and S — the overall successful rate of a node:

$$S = \sum_{k=1}^{n} S_k = G\int_0^1 (1-[p(r)]^{m+1}) \cdot 2r \cdot dr.$$

To calculate throughput, we need to know:

1. T_w: the average length of *wait* state, which is equal to the average length of a *macro-slot* Θ;
2. T_s: the average length of *trans* state resulting in a successful transmission;
3. T_c: the average length of *trans* state resulting in a collision;
4. $p(r)$: the collision probability of a transmission of distance r.

T_s and T_c are fixed values and can be easily calculated. T_w is the average length of *macro-slot* which is still an unknown. In the following two sections, we will present formulas for T_w and $p(r)$ of the basic access method and of the RTS/CTS access method. With these quantities, TH, $TH(r)$ and $TH_I(r)$ defined in Section 4.2.1 can be derived.

4.4 Throughputs of the Basic Access Method

4.4.1 Transmission Probability τ and Idle Probability Π_I

As we have mentioned, a *macro-slot* Θ consists of a busy channel period and an idle slot θ. The ratio θ/Θ is actually the probability (Π_I) that a channel is detected to be idle by a node when the node is not transmitting: $\Pi_I = \theta/\Theta$. Accordingly, we have $T_w = \Theta = 1/\Pi_I$.

Let τ denote the transmission probability of a node in an arbitrary slot when the node is not transmitting. If Π_I is known, we have:

$$\tau = \frac{\bar{G}}{\bar{G} + (1-\bar{G})T_w} = \frac{\bar{G}}{\bar{G} + (1-\bar{G})/\Pi_I}. \tag{4.10}$$

On the other hand, given τ, the probability $P_{x,I}$ that a node detects that one of its neighbors x is not transmitting in an arbitrary slot is:

$$P_{x,I} = \frac{1-\tau}{T_t \cdot \tau + (1-\tau)}. \tag{4.11}$$

Here, T_t is the channel occupy time of a transmission (normalized with respect to slot time θ). For the basic access method, $T_t = l_{data} + l_{sifs} + l_{ack} + l_{difs}$.

For the basic access method, a node will detect an idle channel if none of its neighbors is transmitting. To simplify the analysis, we assume that the transmissions of a node's neighbors are independent. This assumption has also been used in [35, 74, 75, 77], and our simulation results later will confirm its validity. According to Poisson distribution, the probability of having i nodes within the transmission range of a node is $e^{-N}N^i/i!$, where $N = \lambda\pi$ is the average number of neighbors of a node. With the assumptions of independent transmission and Poisson distribution, we have:

$$\begin{aligned} \Pi_I &= \sum_{i=0}^{\infty} \frac{N^i}{i!} e^{-N} \cdot P_{x,I}^i \\ &= e^{-N} \sum_{i=0}^{\infty} \frac{(N \cdot P_{x,I})^i}{i!} = e^{-N(1-P_{x,I})} \end{aligned} \tag{4.12}$$

Equations (4.10) and (4.12) represent a nonlinear system in the two unknowns τ and Π_I, which can be easily solved using numerical techniques.

4.4.2 Transmission Collision Probability $p(r)$

Now we determine the collision probability $p(r)$ for a transmission with a distance of r. In DCF, nodes hearing a collision must wait for an EIFS time that is large enough for

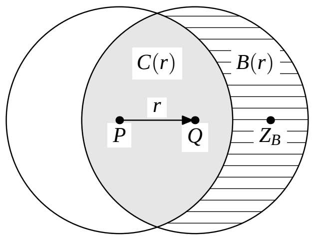


Figure 4.5: Illustration of hidden area for RTS and DATA frames

the reception of an ACK frame. Hence, it is safe to assume that ACK is always received successfully by the source. The collision probability of a transmission is therefore equal to the collision probability of the data frame of the transmission. Figure 4.5 illustrates a transmission from node P to node Q. Obviously, we have:

$$p(r) = 1 - P_s(r),$$

where $P_s(r)$ is the probability that a transmission is successful. According to the conditions of a successful transmission, we have:

$$P_s(r) = P_1(r) \cdot P_2(r) \cdot P_3(r),$$

where

$$\begin{aligned} P_1(r) &= \text{Prob.}\{\text{Q does not transmit in the same slot} \\ &\qquad \mid \text{P transmits in a slot}\}, \\ P_2(r) &= \text{Prob.}\{\text{nodes in } C(r) \text{ does not transmit in the} \\ &\qquad \text{same slot} \mid \text{P transmits in a slot}\}, \\ P_3(r) &= \text{Prob.}\{\text{nodes in } B(r) \text{ does not transmit for} \\ &\qquad 2l_{data} + 1 \text{ slots} \mid \text{P transmits in a slot}\}. \end{aligned}$$

The condition "P transmits in a slot" means that P has just detected an idle slot and will start to transmit at the end of this idle slot (or at the beginning of the next slot). The reason for $P_3(r)$ is that the vulnerable period for a data frame is $2l_{data}+1$. $C(r)$ is the common neighborhood of the sender and the receiver; $B(r)$ is the hidden area. This has been shown in [74]:

$$\begin{aligned} C(r) &= 2\left(arccos(\frac{r}{2}) - \frac{r}{2}\sqrt{1-(\frac{r}{2})^2}\right), \\ B(r) &= \pi - C(r). \end{aligned}$$

Given the transmission probability of a node τ, we can get $\bar{G}$ by equations (4.10), (4.11) and (4.12). $\bar{G}$ is the transmission probability of a node in a *macro-slot*. Since each *macro-slot* consists of a busy period and an idle slot, $\bar{G}$ is actually the transmission probability when a node detects an *idle* slot. Hence:

$$\begin{aligned} P_1(r) &= (1-\bar{G}) \cdot \text{Prob.}\{\text{Q detects an idle channel} \\ &\quad \mid \text{P detects an idle idle channel}\} \\ &= (1-\bar{G}) \cdot \text{Prob.}\{\text{none nodes in } B(r) \text{ transmits}\}. \end{aligned}$$

Following the same line of reasoning as equation (4.12), we have:

$$\text{Prob.}\{\text{none nodes in } B(r) \text{ transmits}\} = e^{-\lambda B(r)(1-P_{x,I})}.$$

Therefore:

$$P_1(r) = (1-\bar{G}) \cdot e^{-\lambda B(r)(1-P_{x,I})}.$$

When node P detects an idle channel, the chance that nodes in $C(r)$ also detect an idle channel is higher than usual due to spatial dependency; hence, the transmission probability increases accordingly. Let τ_c denote the average transmission probability of nodes in $C(r)$, given the condition that node P detects an idle slot. We have:

$$P_2(r) = \sum_{i=0}^{\infty}(1-\tau_c)^i \frac{(\lambda C(r))^i}{i!} e^{-\lambda C(r)} = e^{-\tau_c \lambda C(r)}.$$

To simplify the calculation, τ_c can be replaced with τ. Our experience shows that this approximation has little impact on the results.

To calculate $P_3(r)$, we first find out the probability $P_{idle}(y, \tau)$ that a node does not transmit for y slots with a given transmission probability τ:

$$P_{idle}(y, \tau) = 1 - \sum_{i=0}^{y-1}(1-\tau)^i \tau = (1-\tau)^y.$$

If we assume that the transmission probability of nodes in $B(r)$ remains constant and be τ during the whole vulnerable period, we have:

$$\begin{aligned} P_3(r) &= \sum_{i=0}^{\infty}(P_{idle}(2l_{data}+1, \tau))^i \frac{(\lambda B(r))^i}{i!} e^{-\lambda B(r)} \\ &= e^{-\lambda B(r)(1-(1-\tau)^{2l_{data}+1})}. \end{aligned}$$

This assumption works well when the node density is low ($N \leq 5$). For a higher density network, we need to consider the temporal dependency of transmission probability. After node P starts to transmit, neighbors of P that have not been transmitting will detect the carrier, and thus will not transmit. The silence of these nodes increases the transmission probability of nodes in $B(r)$. Let τ_b denote the average transmission probability of nodes in $B(r)$ when node P is transmitting. For dense networks, $P_3(r)$ is given by:

$$P_3(r) = e^{-\lambda B(r)[1-(1-\tau)^{l_{data}+1}(1-\tau_b)^{l_{data}}]}.$$

We use the transmission probability of the node at Z_B (Figure 4.5) to approximate τ_b. Z_B is at a distance of $1 + r/2$ to node P, which is the average of the nearest (=1) and the farthest distance (=1+r) between node P and its hidden nodes. Following the same line of reasoning as $P_1(r)$, we get τ_b by:

$$\tau_b = \bar{G}e^{-\lambda B(1+r/2)(1-P_{x,I})}. \tag{4.13}$$

4.4.3 Throughputs of the Basic Access Method

By solving the nonlinear systems formed by equations (4.10) and (4.12), we can compute all the quantities (τ, Π_I, G, $\bar{G}$, p_k, S_k, S, etc.) needed to calculate the three throughputs defined in Section 4.2.1.

For the basic access method, the average length of the successful *trans* state T_s, the average length of the collision *trans* state T_c and the average length of the *wait* state T_w are:

$$\begin{aligned} T_s &= T_c = l_{data} + l_{sifs} + l_{ack} \\ T_w &= 1/\Pi_I. \end{aligned}$$

The three throughputs are obtained as follows:

$$\begin{aligned} TH &= \frac{S \cdot E[P]}{\bar{G}T_s + (1-\bar{G})T_w} \\ TH(r_k) &= \frac{S_k \cdot E[P]/(N \cdot 2r_k \cdot \Delta r)}{\bar{G}T_s + (1-\bar{G})T_w} \\ TH_I(r_k) &= \frac{S_k \cdot E[P]}{\bar{G}_k T_s + (H_k - \bar{G}_k)T_w} \end{aligned} \tag{4.14}$$

where $E[P]$ is the average length of payloads. Since we assume fixed packet size, $E[P]$ can be calculated directly from packet size. $\bar{G}_k$ is the probability that a node is transmitting to a destination located in the ring of r_k. H_k is the probability that a node is in *trans* state or in *wait* state for a destination located in the ring of r_k (i.e., the probability a node in state of M_{r_k} (Figure 4.4)). $\bar{G}_k$ and H_k are given by:

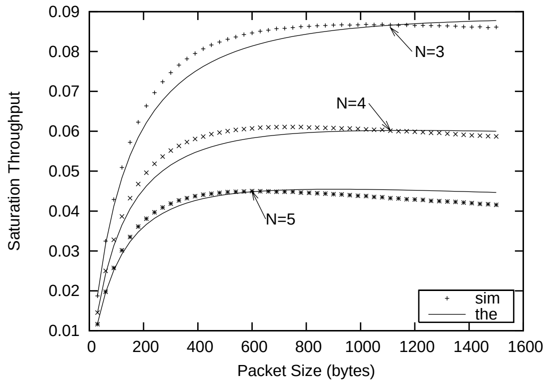
$$\bar{G}_k = \sum_{i=0}^{m} b_{i,0,k}, \quad \text{and} \quad H_k = \sum_{i=0}^{m} \sum_{j=0}^{W_i - 1} b_{i,j,k}.$$

Since S_k is the total success rate of transmissions for all neighbors located in the k^{th} ring, it is divided by the term $N \cdot 2r_k \cdot \Delta r$ (average number of nodes in the k^{th} ring) to get the long-run throughput of a flow. Given $S = \sum_{k=1}^{n} S_k$, we have $TH = \int_0^1 TH(r) \cdot N \cdot 2r \cdot dr$.

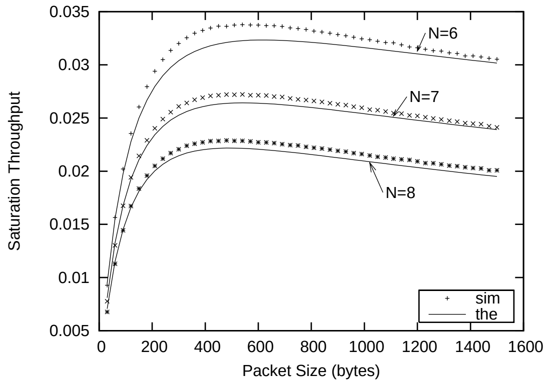
4.4.4 Model Validation for the Basic Access Method

To validate our model, we have compared the theoretical results with simulation results. The simulator we have used is NS-2 [50]. Ten scenarios have been randomly generated. In each scenario, about 5000 nodes are uniformly distributed in a square area[3]. The Ns of these scenarios are $3, 4, \ldots, 12$ respectively. N is the average number of neighbors of

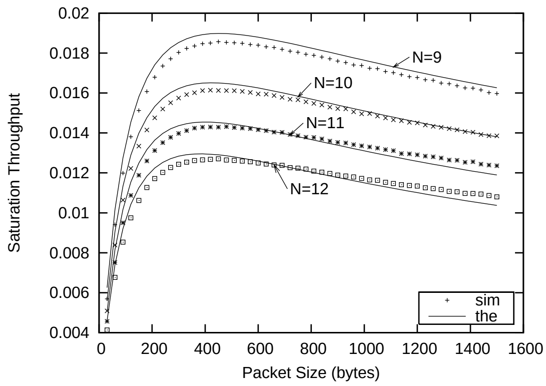
[3] The large number is a statistical necessity because we need enough flows of various distances.



(a) N=3,4,5



(b) N=6,7,8



(c) N=9,10,11,12

Figure 4.6: Saturation throughput of the basic access method: analysis versus simulation

a node in a scenario. All nodes have a communication range of 250m and the transmission rate is 2Mbps. Packet size varies from 30 bytes to 1500 bytes, with a step of 30 bytes. Node saturation throughput is calculated as the average throughput of nodes in a scenario. Since nodes along the edges of the square area do not have a Poisson distribution neighborhood, only the nodes having a larger than $6 \times 250m$ distance to the edges are taken into consideration in the calculation of the throughput. Figure 4.6 shows numerical results and simulation results of saturation throughputs versus packet size for different Ns. In all the figures, solid lines represent theoretical results and dotted lines represent simulation results. The results show that our analytical model is very accurate for the basic access method. For most N and packet sizes, the difference between theoretical results and simulation results is below 5%, and the maximum difference is less than 10%.

4.5 Throughputs of the RTS/CTS Access Method

The analysis for the RTS/CTS access method is much more difficult. It is unrealistic to capture all details of the RTS/CTS access method in the analysis. In this section, we provide an approximation to the analysis.

4.5.1 Transmission Probability τ and Idle Probability Π_I

The transmission probability τ and idle probability Π_I of the RTS/CTS access method are the same as those for the basic access method; they are given by equation (4.10) and equation (4.12), respectively.

One of the difficulties in analyzing the RTS/CTS access method is the estimation of $P_{x,I}$ — the probability a node finding its neighbor x not using the channel. In the RTS/CTS access method, the channel occupancy time of a transmission is not fixed but dependent on the transmission outcome of an RTS frame. If the RTS is transmitted successfully, the actual channel occupancy time is $T_{succ} = l_{rts} + l_{cts} + l_{data} + l_{ack} + 3 \cdot l_{sifs} + l_{difs}$. If the RTS fails, the actual channel occupancy time is $T_{fail} = l_{rts} + l_{cts} + l_{sifs} + l_{difs}$. However, the occupancy time of the transmission observed by other nodes may be different. Even when the intended receiver fails to receive the RTS, other neighbors of the source may still observe a full occupy time T_{succ} because these nodes may have received the RTS correctly and therefore defer their transmissions. Furthermore, in the RTS/CTS access method, not only one-hop neighbors, but also two-hop neighbors will influence the observation.

To simplify analysis, we assume that any transmission of a node will be observed by its neighbors as a full transmission. Furthermore, to account for two-hop neighbors' influences, a transmission received successfully by a node is also assumed to be observed by its neighbors as a full transmission. For the RTS/CTS access method, we use $\tilde{P}_{x,I}$ to denote the idle probability of x observed by x's neighbors. Due to the symmetry of traffic, the successful reception probability of a node is equal to the successful transmission probability. With these assumptions, $\tilde{P}_{x,I}$ is given by:

$$\tilde{P}_{x,I} = \frac{1-\tau}{\tau(1+S/\bar{G})T_{succ} + (1-\tau)}.$$

4.5.2 RTS Frame Collision Probability $p_{rts}(r)$ and Data Frame Collision Probability $p_{data}(r)$

Similar to the basic access method, we assume that the CTS and ACK sent by the destination are always received correctly by the source. The collision probability of a transmission is equal to the sum of RTS collision probability $p_{rts}(r)$ and data frame collision probability $p_{data}(r)$. $p_{rts}(r)$ can be derived by following the steps given in Section 4.4.2. In this section, we concentrate on $p_{data}(r)$ — the probability that a transmission fails due to collision of the data frame. Ideally, after an RTS/CTS exchange, all neighboring nodes of source P and destination Q should have been notified of the

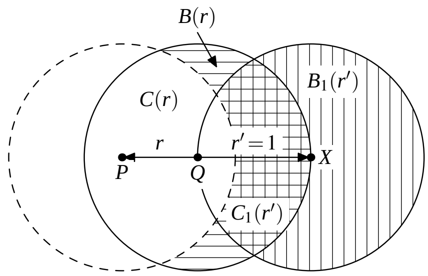


Figure 4.7: Illustration of hidden area for CTS frame

forthcoming data frame transmission, and the data frame should be collision-free, i.e., $p_{data}(r) = 0$. However, due to the collision of CTS, some nodes in $B(r)$ may not know the on-going transmission from P to Q, and may start to transmit and therefore lead to the collision of the data frame at node Q.

To calculate $p_{data}(r)$, we need to know $p_{cts}(r)$ — the average CTS collision probability in area $B(r)$. An accurate calculation of $p_{cts}(r)$ is very difficult. We use the collision probability at point X to replace the average probability (Figure 4.7). Similar to Section 4.4.2, we have:

$$p_{cts}(r) = 1 - P_1(r) \cdot P_2(r) \cdot P_3(r),$$

where

$$\begin{aligned}
P_1(r) &= \text{Prob.}\{\text{X does not transmit in the same slot} \\
&\qquad \mid \text{RTS successes }\}, \\
P_2(r) &= \text{Prob.}\{\text{nodes in } C_1(r') \text{ does not transmit in the} \\
&\qquad \text{same slot} \mid \text{RTS successes}\}, \\
P_3(r) &= \text{Prob.}\{\text{nodes in } B_1(r') \text{ does not transmit for} \\
&\qquad 2l_{cts} + 1 \text{ slots} \mid \text{RTS successes}\}.
\end{aligned}$$

$p_{cts}(r)$ can be calculated by following the same steps in Section 4.4.2.

Let $\tilde{p}_{data}(r)$ = Prob.{data frame fails | RTS succeeds}. Given $p_{cts}(r)$, the average number of hidden terminals that fail to receive CTS is $M = \lambda B(r) p_{cts}(r)$. We have:

$$\begin{aligned}\tilde{p}_{data}(r) &= 1 - \sum_{i=0}^{\infty} (P_{idle}(l_{data}+1, \tau))^i \frac{M^i}{i!} e^{-M} \\ &= 1 - e^{-\lambda B(r) p_{cts}(r)(1-(1-\tau)^{l_{data}+1})}.\end{aligned} \tag{4.15}$$

Note that here we use τ because we over-estimate $p_{cts}(r)$. $p_{data}(r)$ is given by:

$$p_{data}(r) = (1 - p_{rts}(r)) \cdot \tilde{p}_{data}(r)$$

and the total transmission collision probability $p(r)$ is given by:

$$p(r) = p_{rts}(r) + p_{data}(r).$$

Given $p_{rts}(r)$ and $p_{data}(r)$, we can calculate the average RTS frame fail rate $\bar{G}_{rf}$ and DATA frame fail rate $\bar{G}_{df}$ by:

$$\begin{aligned}\bar{G}_{k,rf} &= \sum_{i=0}^{m} b_{i,0,k} \cdot p_{rts}(r_k) \\ \bar{G}_{rf} &= \sum_{k=1}^{n} \bar{G}_{k,rf} \\ &= G \int_0^1 \frac{1 - [p(r)]^{m+1}}{1 - p(r)} \cdot p_{rts}(r) \cdot 2r \cdot dr\end{aligned}$$

and

$$
\begin{aligned}
\bar{G}_{k,df} &= \sum_{i=0}^{m} b_{i,0,k} \cdot p_{data}(r_k) \\
\bar{G}_{df} &= \sum_{k=1}^{n} \bar{G}_{k,df} \\
&= G \int_0^1 \frac{1-[p(r)]^{m+1}}{1-p(r)} \cdot p_{data}(r) \cdot 2r \cdot dr.
\end{aligned}
$$

4.5.3 Throughputs of the RTS/CTS Access Method

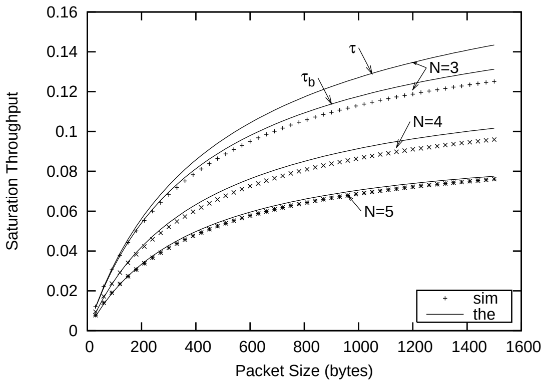
Throughputs of the RTS/CTS access method are calculated in a similar way as the basic access method, except that for the RTS/CTS access method, we need to further consider two types of collision: RTS collision and DATA frame collision. The average time for the successful *trans* state T_s, the average time for the RTS collision *trans* state T_{c1}, the average time for the DATA collision *trans* state T_{c2}, and the average time for the *wait* state T_w are given as follows:

$$
\begin{aligned}
T_{c1} &= l_{rts} + l_{sifs} + l_{cts} \\
T_{c2} &= T_s = l_{rts} + l_{data} + l_{data} + l_{ack} + 3 \cdot l_{sifs} \\
T_w &= 1/\Pi_I.
\end{aligned}
$$

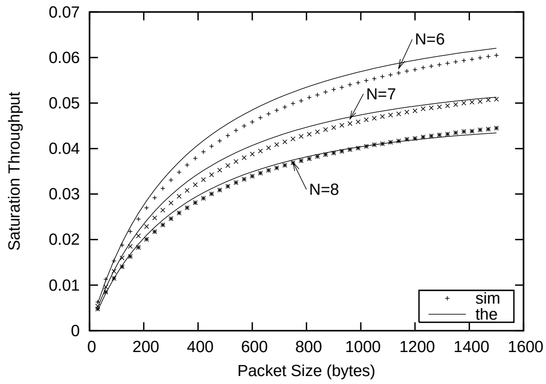
The three throughputs are calculated by:

$$
\begin{aligned}
TH &= \frac{S \cdot E[P]}{\bar{G}_{rf}T_{c1} + (S + \bar{G}_{df})T_s + (1-\bar{G})T_w} \\
TH(r_k) &= \frac{S_k \cdot E[P]/(N \cdot 2r_k \cdot \Delta r)}{\bar{G}_{rf}T_{c1} + (S + \bar{G}_{df})T_s + (1-\bar{G})T_w} \\
TH_I(r_k) &= \frac{S_k \cdot E[P]}{\bar{G}_{k,rf}T_{c1} + (S_k + \bar{G}_{k,df})T_s + (H_k - \bar{G}_k)T_w}
\end{aligned}
\tag{4.16}
$$

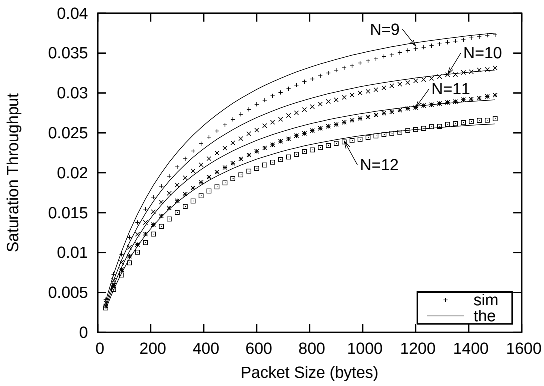
4.5.4 Model Validation for the RTS/CTS Access Method



(a) N=3,4,5



(b) N=6,7,8



(c) N=9,10,11,12

Figure 4.8: Saturation throughput of the RTS/CTS access method: analysis versus simulation

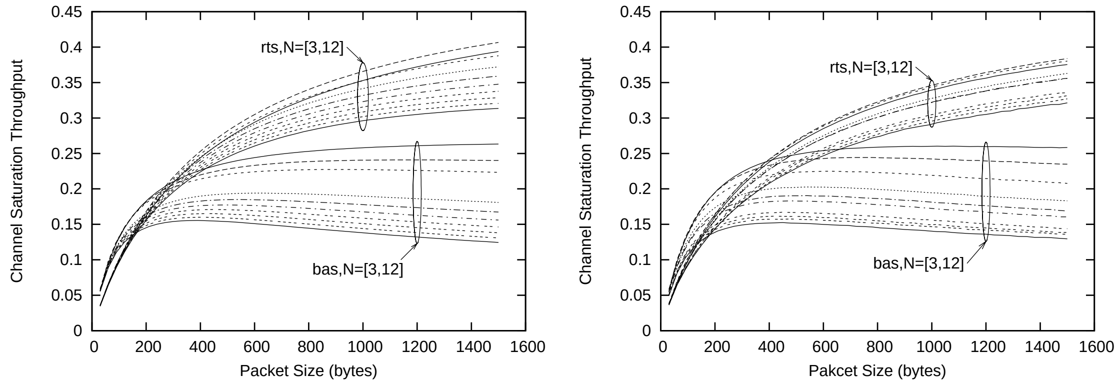
Figure 4.8 shows numerical results and simulation results for the RTS/CTS access method. The simulation scenarios and settings are the same as those for the basic access method. In all the figures, solid lines represent theoretical results and dotted lines represent simulation results. The results show that our analytical model is very accurate. In cases of $N > 3$, the difference between theoretical results and simulation results is below 5% for most packet sizes, and the maximal difference is less than 10%. For the case $N = 3$, our model overestimates the saturation throughput (the solid line indicated by τ). However, the theoretical results are still quite close to the simulation results — the maximal difference is about 13.6%, which occurs at packet size of 1500 bytes. For $N = 3$, if we replace the τ in equation (4.15) with τ_b (equation (4.13)), we can further improve the accuracy of our model. The line indicated by τ_b in Figure 4.8(a) shows this improvement. It can been seen that with this change, accuracy is improved to smaller than 5%.

Compare with the results of the basic access method, RTS/CTS improves saturation throughput substantially when the packet size is large (packet size > 400 bytes). In the extreme case ($N = 12$, packet size=1500), RTS/CTS improves saturation throughput by 140%.

4.6 Throughput Performance Evaluation

4.6.1 Channel Saturation Throughput

The saturation throughput used in the analysis is per-node based. Sometimes, the saturation throughput of a channel is also of interest. However, unlike wireline channels or wireless channels in single-hop wireless networks, where a channel is a single entity shared by all nodes attached to it, the channel in a multihop wireless network actually consists of multiple spatially overlapping channels that compete with one another and have neither absolute nor readily observable boundary. To measure throughput of a multihop channel, we define channel throughput as the throughput per unit of area. One



(a) Theoretical channel saturation throughput

(b) Simulation channel saturation throughput

Figure 4.9: Channel saturation throughput (PHY=DSSS)

natural "unit of area" is the hearing region of a node. Hence, the channel saturation throughput of multihop wireless networks is:

$$C = TH \cdot N,$$

where TH is per-node saturation throughput, and N is the average number of neighbors of a node. With this definition, the channel saturation throughput of DCF is ready to be calculated.

Figure 4.9 shows the theoretical and simulative channel saturation throughput with default parameters for DSSS. It clearly shows that the saturation throughput of channel under DCF is dependent not only on node density and packet size, but also on access methods.

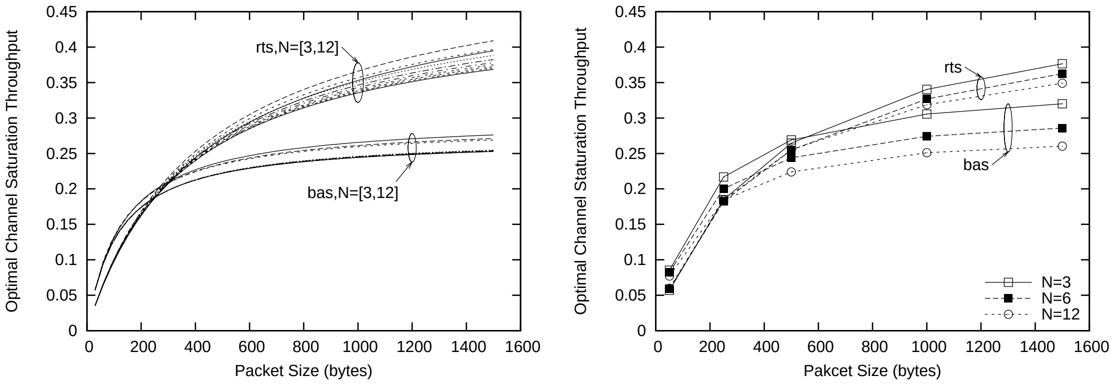
When the packet size exceeds 300 bytes, the RTS/CTS access method outperforms the basic access method regardless of node density. As packet size increases further, the advantage of the RTS/CTS access method also increases. When packet size is less than

300 bytes, the basic access method outperforms the RTS/CTS access method marginally. This suggests that the RTS/CTS access method should be used in most practical cases.

For the basic access method, channel saturation throughput is very sensitive to node density. When the packet size is large, the throughput of network with $N = 12$ is less than half of that with $N = 3$. This is expected because the basic access method has no mechanisms to deal with the hidden terminal problem. In a network with high node density, hidden terminals prevail; hence, transmissions are more vulnerable to collisions, which leads to extremely low throughput. In contrast, the RTS/CTS access method is much less sensitive to node density because the RTS/CTS exchange prior to the transmission of a data frame can eliminate most hidden terminals.

Another interesting finding is that for a given N, the channel throughput of the basic access method almost remains constant when packet size is larger than 300 bytes. This finding is counter-intuitive. One expects throughput to decrease because the vulnerable period of a transmission increases as packet size increases. This can be explained as follows: Although the vulnerable period of a transmission increases as packet size increases, hidden nodes are also more likely to be exposed to other transmitting nodes (due to longer transmission time). Hence, the transmission probability τ decreases correspondingly. More importantly, BEB will cause τ to become smaller if collision probability increases. The overall effect of increasing vulnerable period and decreasing τ is an almost constant throughput.

The trend of constant throughput does not occur in the RTS/CTS access method. For a given N, the channel saturation throughput keeps increasing as packet size increases. Due to the RTS/CTS exchange, the adverse impact of hidden terminals is



(a) Theoretical maximum channel saturation throughput

(b) Simulation maximum channel saturation throughput

Figure 4.10: Maximum channel saturation throughput (PHY=DSSS)

reduced. As packet size increases, the overhead (due to RTS/CTS exchange, backoff time) per transmission is also reduced. Hence, the throughput increases.

4.6.2 Maximum Channel Saturation Throughput

The analytical model is very convenient for determining the maximally achievable channel saturation throughput. The analysis clearly shows that the saturation throughput depends on traffic load (N and packet size) and system parameters (m and W). Since traffic load is not directly controllable, the only way to achieve optimal performance is to employ adaptive techniques to tune the values m and W.

The analytical maximum channel saturation throughput can be easily calculated by searching all reasonable combinations of m and W. Since adjusting m has the similar effect as adjusting W, a simpler way of finding maximum throughput is to change W only while keeping m constant. Figure 4.10(a) shows the analytical maximum channel saturation throughput.

One surprising finding is that the maximum channel saturation throughput is almost independent of N for both methods. This independence has also been found in [70], where the performance of DCF is analyzed in the context of single-hop networks. In [70], it has also been found that the maximum saturation throughput is also independent of access methods, i.e., the basic access method and the RTS/CTS method can achieve similar maximum saturation throughputs. However, access method independency is not observed in multihop wireless networks. The theoretical results clearly show that maximum channel saturation throughput is very much dependent on access method. When packet size exceeds 300 bytes, the RTS/CTS access method has an advantage over the basic access method. When packet size exceeds 600 bytes, the advantage of the RTS/CTS access method is clear-cut.

Another interesting finding is that the throughput improvement of the RTS/CTS access method by optimizing W is only marginal — in most cases, theoretical throughput improvements are less than 5%, and the maximum improvement is about 15%. However, the improvement for the basic method is substantial when N is large. In the extreme case ($N = 12, \text{packet size} = 1500$), the improvement is as large as 110% (Figure 4.9(a) and Figure 4.10(a)). Since the basic access method has no mechanisms for dealing with the hidden terminal problem, it operates in a region far from optimal when the traffic load is heavy. An increased W reduces the actual offered traffic load and hence alleviates the adverse effect of the hidden terminal problem. As a result, throughput is improved. On the other hand, the RTS/CTS access method can deal with the hidden terminal problem more efficiently; an increased W has only little impact on performance. The implication of this finding is that the RTS/CTS access method with the default parameters is already operating in a region of almost optimal performance; any effort to optimize it by adjusting parameters may only lead to marginal improvement.

We have carried out simulations to reaffirm these findings. For each combination of $N = 3, 6, 12$ and packet size $= 50, 250, 500, 1000, 1500$, we find the maximum throughput by varying the initial backoff window size W from 32 to 1024 with a step of 32. The results are shown in Figure 4.10(b). Except the case of $N = 3$ for the basic access method, the theoretical results fit quite well with the simulation results. It appears that when N is small, our model is likely to underestimate the maximum channel throughput of the basic method. Nevertheless, the properties revealed by the theoretical analysis can still be found in simulations.

4.7 Fairness Evaluation

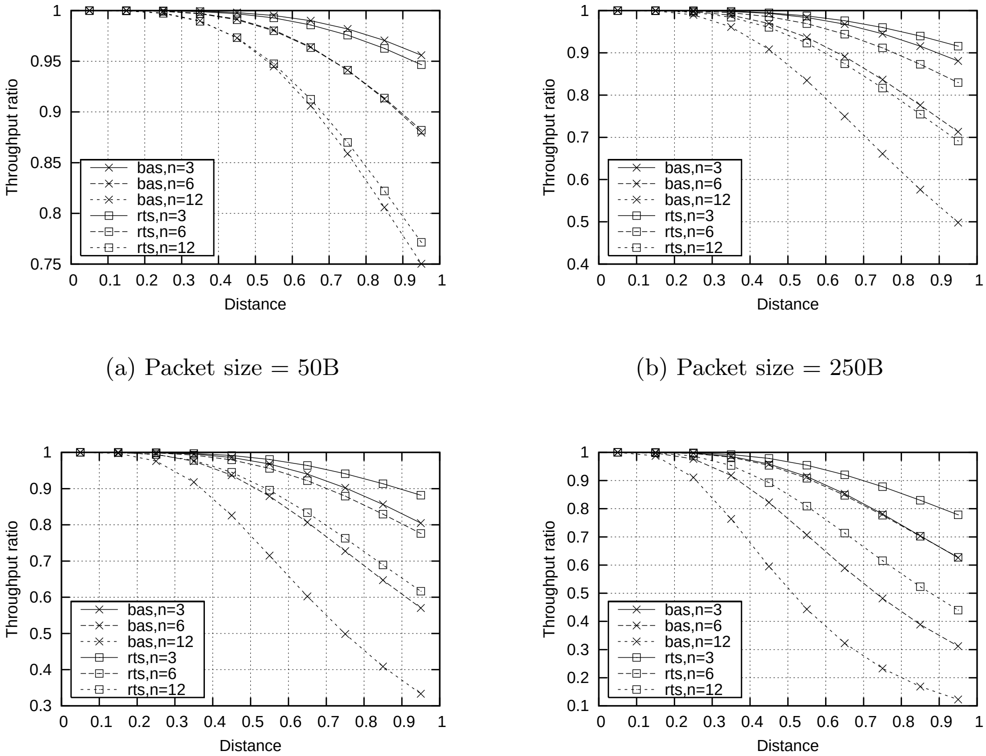
In this section, we evaluate the fairness property of DCF.

4.7.1 Fairness of Long-run Flow Throughput

Since a node randomly chooses a neighbor as destination for a packet, each neighbor (i.e., each flow) should have the same chance to be scheduled. Ideally, we should have:

$$TH(r_i) = TH(r_j), r_i, r_j \in [0, 1].$$

We calculate $TH(r), r = 0.05, 0.15, 0.25, \ldots, 0.95$ for different Ns and packet sizes. For ease of comparison, the calculated throughputs are normalized with respect to $TH(0.05)$. Figure 4.11 shows the normalized theoretical throughput versus distance. It is clearly



(a) Packet size = 50B

(b) Packet size = 250B

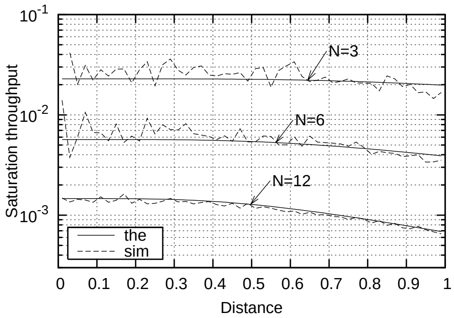
(c) Packet size = 500B

(d) Packet size = 1500B

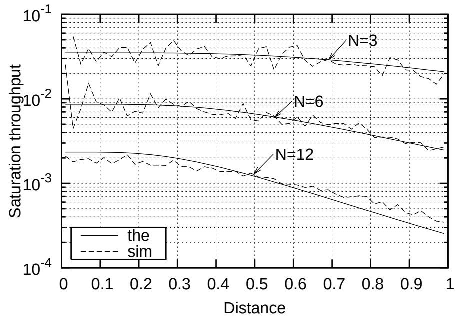
Figure 4.11: Long-run flow throughput fairness: theoretical results

shown in Figure 4.11 that as the distance r increases, the long-run throughput $TH(r)$ decreases inevitably even for scenarios with very sparse density and very small packet size (Figure 4.11(a), N=3). The decrement of long-run flow throughput indicates that some packets of long distance flows are discarded, and as the distance r of a flow increases, the number of discarded packets increases as well.

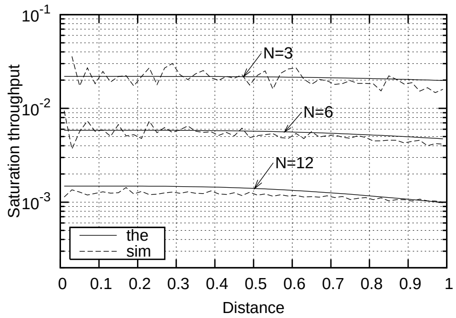
Figure 4.11 also shows that as packet size or node density increases, the long-run throughput gap between short flows and long flows widens. Another finding is that the RTS/CTS access method performs much better than the basic method when packet size is large. These results are expected. As packet size or node density increases, the



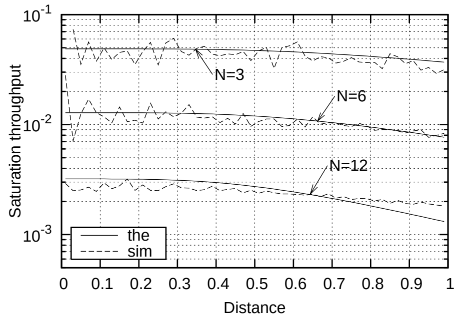
(a) Packet size=250B, basic access method



(b) Packet size=1500B, basic access method



(c) Packet size=250B, RTS/CTS access method

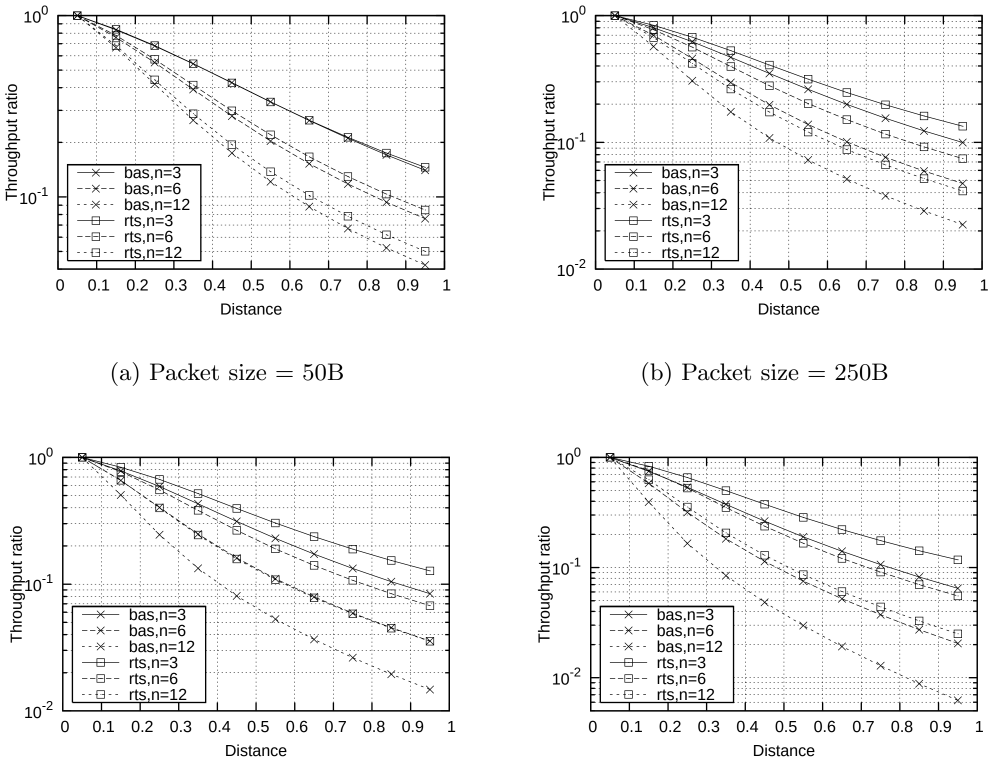


(d) Packet size=1500B, RTS/CTS access method

Figure 4.12: Long-run flow throughput validation: theoretical and simulation results

hidden terminal problem becomes more prominent, which worsens the fairness problem. The RTS/CTS access method has a mechanism to reduce the adverse impact of hidden terminals and hence performs better than the basic method.

We have carried out simulations to reaffirm the theoretical results of long-run throughput. The same parameters and topologies for node throughput validation are used. Theoretical and simulation long-run flow throughputs are calculated in a step of distance 0.02 (i.e., total 50 points). Figure 4.12 shows both theoretical and simulation results. It can be seen that the theoretical results match the simulation results closely.



(a) Packet size = 50B

(b) Packet size = 250B

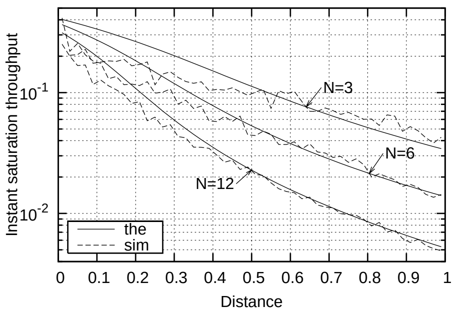
(c) Packet size = 500B

(d) Packet size = 1500B

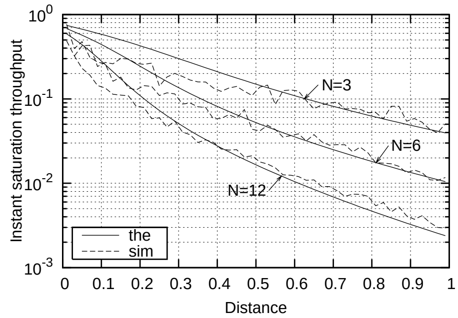
Figure 4.13: Instant flow throughput fairness: theoretical results

4.7.2 Fairness of Instant Flow Throughput

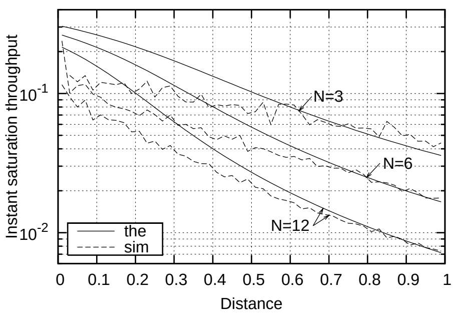
Long-run throughput $TH(r)$ is the throughput of a flow observed over a long time. It does not reflect the actual time a node engages in transmitting for that flow whereas instant throughput $TH_I(r)$ does. $TH_I(r)$ is the efficiency of a node in using the channel when it is trying to transmit a packet for a flow of distance r. A flow having a large $TH_I(r)$ means that when scheduled, it can successfully transmit a packet quickly (i.e., the service time of a packet is small) while a flow having a smaller $TH_I(r)$ takes a longer time to have a successful transmission. Figure 4.13 shows the theoretical results for the instant throughput ratio. The findings presented in the previous subsection still hold:



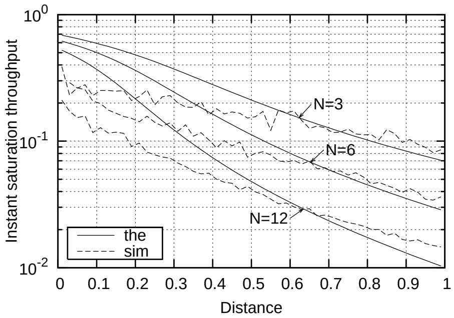
(a) Packet size=250B, basic access method



(b) Packet size=1500B, basic access method



(c) Packet size=250B, RTS/CTS access method



(d) Packet size=1500B, RTS/CTS access method

Figure 4.14: Instant flow throughput validation: theoretical and simulation results

(1) as packet size or node density increases, the instant throughput gap between long flows and short flows widens as well; (2) the RTS/CTS method performs better than the basic method in terms of instant throughput fairness. The difference is that in this case, the fairness problem is much more prominent. The results in Figure 4.13 indicate that the average service time of a long flow may be tens, even hundreds times larger than that of a short flow. This means that a node is locked in transmitting for long flows with very low efficiency most of the time. The implication of this phenomenon is that improving fairness might improve node throughput as well. If we even out instant

throughput by reducing the service time of long flows and increasing the service time of short flows, we might get improved node throughput.

We have carried out simulations to validate the theoretical results of instant flow throughput. The theoretical and simulation results are reported in Figure 4.14. Figure 4.14(a) and Figure 4.14(b) show that for the basic method, the theoretical results fit quite well with the simulation results. For the RTS/CTS access method, the theoretical results only fit well with the simulation results when r is large. Nevertheless, the two findings from the theoretical model still hold in simulations. Furthermore, the fairness problem is also very prominent in simulations, although not as bad as predicted by the analytical model. Overall, the model is very useful in providing insights into the fairness property of DCF.

4.7.3 Non-work-conserving Principles

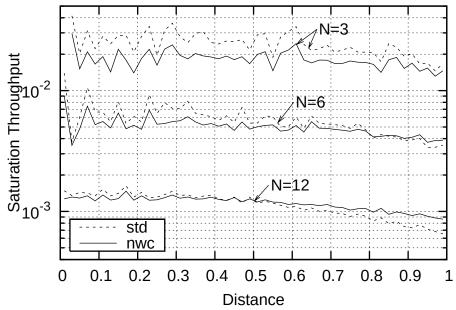
We have found that as packet size or node density increases, the differences between the long-run/instant throughputs of long flows and those of short flows increase. To even out throughput among flows, we can either: (1) reduce packet size, or (2) reduce node density in a network. Reducing packet size is a feasible choice for the basic method. Figure 4.6 indicates that node saturation throughput remains almost constant or only slightly decreases when packet size exceeds 400 bytes for a given N. This suggests that by segmenting a large packet into several packets of size around 400 bytes, we can improve the fairness of the basic method while maintaining a similar (or slightly higher) node saturation throughput achievable by original packet size. However, for the RTS/CTS access method, reducing packet size would also reduce node saturation throughput, as shown in Figure 4.8.

The alternative is to reduce node density. Node density is not a directly controllable parameter. However, a close check of the analytical model reveals that only active nodes influence performance. An active node is a node trying to transmit a packet of its own. Hence, if we could reduce the number of active nodes at any instance, we can reduce node density. A convenient way to achieve this is to make an active node enters into non-active state (i.e., not try to transmit) for a period of time after a successful transmission. The duration of the non-active state should be large enough to ensure that the number of active nodes at any instance has indeed decreased. By employing non-active state, we actually introduce the concept of non-work-conserving disciplines [79]. It will be interesting to precisely analyze the impact of non-working-conserving disciplines on throughput and fairness. However, to do so, we need to take dependency into consideration, which complicates the analysis considerably. We do not try do a precise analysis of non-work-conserving disciplines here[4]. Instead, we employ simulation to confirm the general conclusion drawn from the current analytical model that non-working-conserving disciplines will improve fairness.

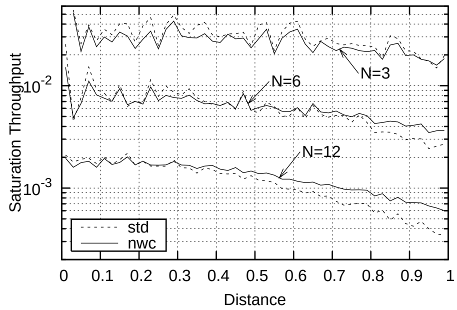
Figure 4.15 and Figure 4.16 report the simulation results for standard IEEE 802.11 and non-work-conserving IEEE 802.11 for long-run throughput and instant throughput, respectively. For non-work-conserving IEEE 802.11, the duration of the non-active state of a node i is:

$$dur(i) = T_t \times max\{D_j | j \in N_i \cup \{i\}\},$$

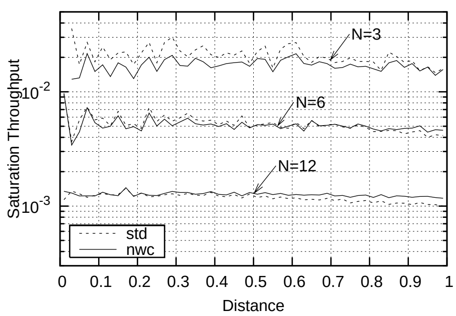
[4]We believe that our analysis is still valid as long as the traffic generated by all active nodes keeps the whole network in saturation state.



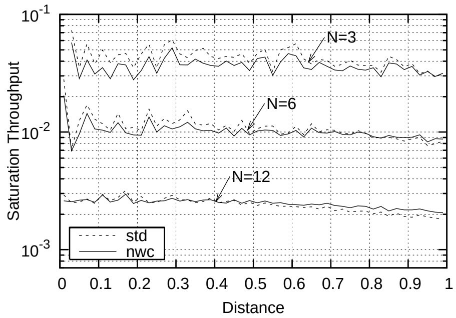
(a) Packet size=250B,bas



(b) Packet size=1500B,bas



(c) Packet size=250B,RTS



(d) Packet size=1500B,RTS

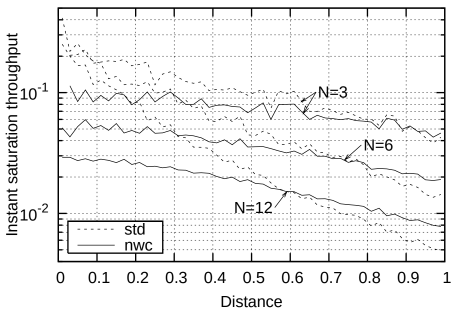
Figure 4.15: Long-run flow throughput versus distance: simulation results for standard IEEE 802.11 and non-work-conserving IEEE 802.11

where T_t is the time needed to transmit a packet; D_j is the degree of node j (i.e., the number of neighbors); N_i is the set of one-hop neighbors of node i. Here, we are not interested in how a node gets its non-active state duration $dur(i)$. In our simulation, $dur(i)$ is pre-calculated and preset for each node. In practice, algorithms like the ATDMA scheme introduced in the previous chapter can be employed to estimate and adjust $dur(i)$ on-the-fly.

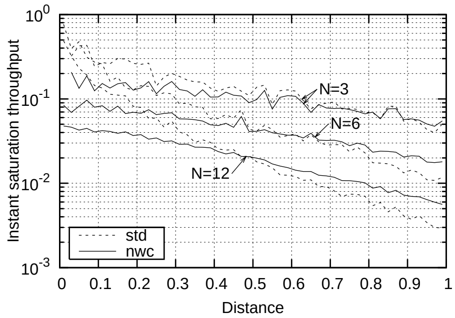
Figure 4.15 shows that the fairness of long-run flow throughput is improved by the non-work-conserving mechanism, especially for the basic access method. For the RTS/CTS access method, although the improvement is not as large as that for the basic method, it is still observable. The drastic effect of non-work-conserving principles occurs in instant flow throughput. Figure 4.16 indicates that the fairness of instant throughput is improved dramatically by the non-work-conserving mechanism. Note that a fairer instant flow throughput can also benefit the operations of upper layer protocols. With a fairer instant throughput distribution, the average service time (i.e., the instant flow throughput) for a flow is less sensitive to the flow's distance. Hence, as the the source and the destination move closer or further apart, the MAC protocol maintains a consistent and predictable service, which is vital for the operations of some upper layer protocols (e.g., TCP). Shortest path routing protocols can also benefit from a fairer MAC protocol because these routing protocols prefer long one-hop flows which are otherwise penalized by an unfair MAC protocol.

Non-work-conserving disciplines also reduce the average number of retransmissions experienced by a packet. In the analytical model, the number of retransmissions is calculated by $(\bar{G}/S-1)$. Theoretical results show that $(\bar{G}/S-1)$ also increases as node density increases. Hence, we expect the average number of retransmissions of non-work-conserving IEEE 802.11 to be smaller than that of standard IEEE 802.11. Simulations results confirm this predication. For example, the average number of retransmissions reduces from 8.9 to 5.9 for the basic access method, and from 5.2 to 3.3 for the RTS/CTS access method for the case $N = 12$ and packet size = 1500B. This is a significant advantage for nodes powered by battery.

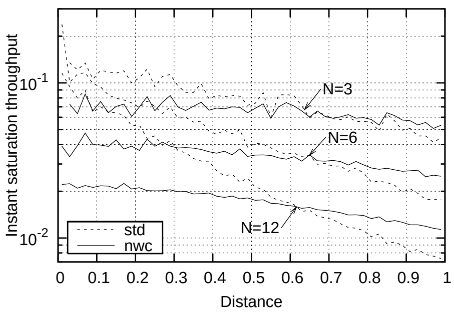
As we have conjectured, a fairer MAC protocol might have a higher throughput. Figure 4.17 shows the node saturation throughputs of standard IEEE 802.11 and non-work conserving IEEE 802.11. The results confirm the conjecture: for $N > 6$, the



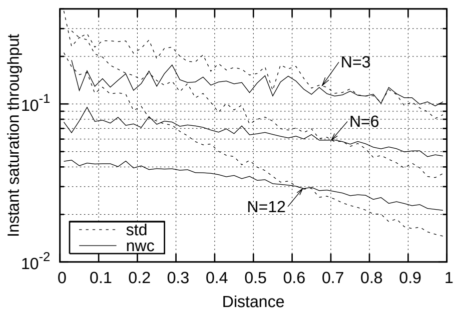
(a) Packet size=250B,bas



(b) Packet size=1500B,bas



(c) Packet size=250B,RTS



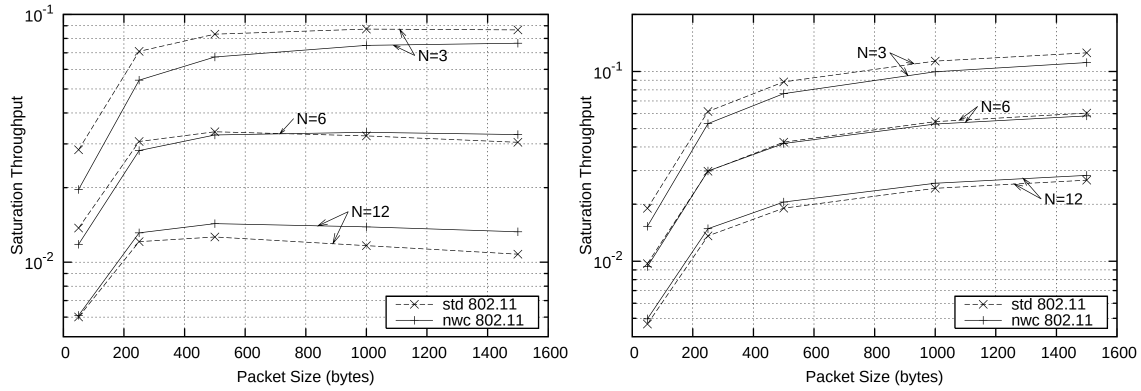
(d) Packet size=1500B,RTS

Figure 4.16: Instant flow throughput versus distance: simulation results for standard IEEE 802.11 and non-work-conserving IEEE 802.11

node saturation throughput of non-work conserving IEEE 802.11 is larger than that of standard IEEE 802.11.

4.8 Summary

In this chapter, we have proposed an analytical model to analyze the throughput and fairness properties of the distributed coordination function (DCF) of IEEE 802.11 in multihop wireless networks with nodes randomly placed according to a two-dimensional Poisson distribution. The model is applicable to both access methods of DCF. Simula-



(a) Basic access method

(b) RTS/CTS access method

Figure 4.17: Node throughput comparison, standard IEEE 802.11 versus non-work conserving IEEE 802.11 (PHY=DSSS)

tion results show that our model is very accurate in predicting saturation throughput. Using the analytical model, we have evaluated the saturation throughput of DCF. The results reveal that the RTS/CTS access method is much more superior to the basic access method in most cases. More importantly, our model indicates that the RTS/CTS access method with the default parameters operates in a region almost optimal in terms of saturation throughput.

Our model also enables us to analyze the fairness property of IEEE 802.11 operating in a multihop wireless ad hoc network. We are interested in the fairness of channel shares allocated by IEEE 802.11 among one-hop flows of various source-destination distances. We have defined two throughputs to explore the fairness property of DCF: long-run flow throughput and instant flow throughput. We have found that both long-run throughput and instant throughput of a flow decrease as the flow's distance increases. As node density or packet size increases, short flows get a larger share of throughput than long flows do. Particularly, the difference of instant throughputs between short flows and long flows may be in one or even two order of magnitude, which means that the average service time of a long flow may be tens or even hundreds times larger than that of a short flow. Such huge gaps are harmful to the operations of the upper layer protocols. Non-work-conserving principles are proposed to reduce the gap. By extrapolating from the analytical model, we establish the conclusion that non-work-conserving principles will improve the fairness of both throughputs we have defined. We have substantiated the conclusion with simulation results. In addition to fairness, non-work-conserving principles can also reduce the average number of retransmissions experienced by packets. It may even improve the overall throughput of dense networks.

In this and the previous chapter, we have investigated the fairness problem from the perspective of the MAC/link layer. As we discussed in Chapter 1, MAC/link layer fairness is a fundamental supporting element in achieving network layer (end-to-end) fairness. A fairer MAC protocol could deliver a more consistent and predictable service to and thereby produce positive impacts on upper layer protocols. There are studies showing that a fair MAC protocol indeed improves the performance of upper layer protocols ([78]). In the next chapter, we will investigate the impact of the MAC layer protocol on the performance of TCP flows in some typical scenarios, with the purpose of verifying that a fair MAC protocol does produce positive impact on TCP flows.

CHAPTER 5

Evaluation and Comparison of TCP Performance over Four MAC Protocols for Multihop Wireless Ad Hoc Networks

In the previous chapters, we have investigated the MAC/link layer fairness in multihop wireless ad hoc networks and have shown that the widely used IEEE 802.11 cannot provide satisfactory MAC/link layer fairness. We have also proposed a new MAC protocol which can deliver better MAC/link layer fairness. In this chapter, we investigate the impact of MAC fairness on the performance of the network layer. Since end-to-end traffic in MANETs is expected to be mostly TCP-like, just as in the Internet, we focus our interest on the performance of TCP over MANETs. As we have pointed out in Chapter 1, many factors could affect the performance of TCP, e.g., MAC protocol, routing protocol, the length of a route, buffer size, active queueing management algorithms, congestion control algorithms, etc. The impacts of these factors and their interactions on TCP performance are one of the most active research topics in multihop wireless ad hoc networks [22–33]. In this chapter, we look into the problem from the perspective of the interaction between the MAC/link protocols and TCP. We are particularly interested in the impact of fair MAC protocols on the performance of TCP flows.

5.1 TCP Performance Problems in Multihop Wireless Ad Hoc Networks

It is widely acknowledged that TCP does not perform well in the presence of wireless links [80]. Originally designed for wireline networks, TCP interprets a packet loss as an indication of congestion and enters into a congestion control state whenever a packet loss event is detected. This is likely the case in a wireline context in which transmission error is rare. However, this assumption is most likely invalid in a wireless context due to much higher channel error rates and hence more packet losses. Node mobility in MANETs may lead to frequent re-routing and further loss of packets in the event of failure in re-routing. The indiscrimination of TCP in dealing with lost packets keeps TCP in the congestion control state unnecessarily. As a result, the performance of TCP degrades dramatically. In view of this, several enhancements to TCP have been proposed by researchers ([30, 32, 33, 81]) to improve the performance of TCP in MANETs. The basic ideas of these proposals are similar: they try to employ explicit or heuristic mechanisms to identify the reasons of packet losses (i.e., congestion, transmission and route failure) and guide the actions of TCP accordingly.

Recently, researchers have found that medium access control (MAC) protocols have a profound impact on the performance of TCP. Tang et al. [25] investigated the interaction between three MAC protocols (CSMA, FAMA and IEEE 802.11) and TCP, and found that the interaction between TCP and MAC layer backoff timer can cause severe unfairness and capture conditions. Xu et al. [82] pointed out that even in a static multihop ad hoc network (no route change) with a perfect channel (i.e., no transmission error), TCP could still suffer severe performance problems, namely, instability, serious unfairness, and incompatibility. [82] revealed that these problems are rooted in the MAC layer (IEEE 802.11 in this case), thus, enhancements to the MAC layer are necessary to improve the performance of TCP.

Several enhancements to the fairness of IEEE 802.11 have been proposed. In the previous chapters, we have discussed some of them: EMLM [48], MBFAIR [52], DWOP [54] and our proposed EHATDMA. As we have shown in Chapter 3, all these protocols can provide fairer bandwidth allocation among one-hop flows than IEEE 802.11 can (though their effects in improving MAC/link layer fairness are different). Although these protocols are promising in providing MAC/link layer fairness, their impacts on upper layer protocols (especially TCP) remain unknown. Since fairer MAC protocols could deliver a more consistent and predictable service to and thereby bring positive impacts on upper layer protocols, we expect that the fairness as well as other performance aspects of TCP would be improved accordingly by these fair MAC protocols. In this chapter, we look into the interaction between TCP and MAC by evaluating and comparing the performance of TCP over fair MAC protocols and IEEE 802.11. Our aim is to find out whether the fairness at the MAC layer is also observed at the session layer (TCP), and whether a fair MAC protocol can have some positive impacts on the performance of TCP.

It should be noted that even confined within the scope of the interaction between TCP and MAC protocol, the task of performance evaluation for TCP over MANETs remains a complex one. Comprehensive simulations including various topologies and traffic loads are necessary to draw a convincing conclusion, which almost certainly brings

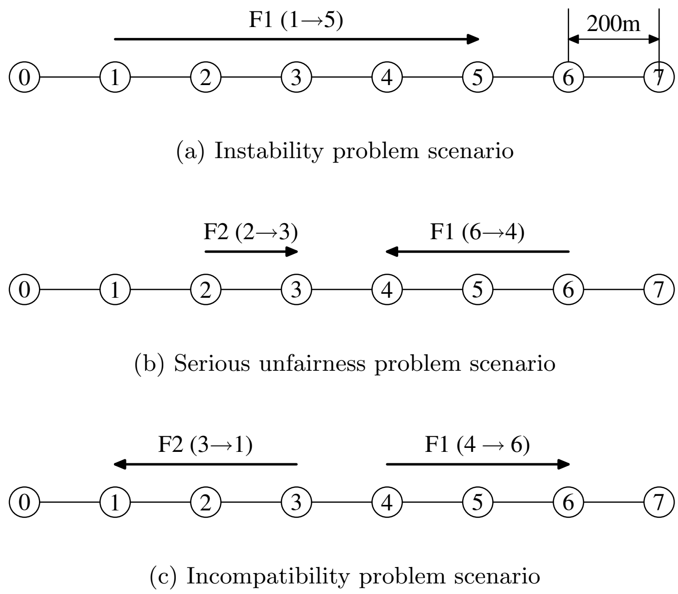


Figure 5.1: Simulation scenarios

other factors into action. For example, when two TCP flows pass through a common node, the queue management in the common node will obviously affect the fairness of bandwidth allocation between them. To avoid the complexity of this kind, in this chapter, we only consider several well-known typical scenarios in which TCP over IEEE 802.11 performs badly. We select these typical scenarios as benchmark to verify whether various enhancements to IEEE 802.11 can improve the performance of TCP in MANETs. The scenarios we use are shown Figure 5.1; they were first used in [82] to manifest the TCP performance problems incurred by TCP/IEEE 802.11 interaction. In [82], it has been shown that when operating over IEEE 802.11, TCP flows could suffer from the following performance problems:

- *Instability:* Due to the contention between forward *data* packets and backward *ack*, the throughput of a TCP flow varies widely from one instant to another.

(e.g., the four-hop TCP flow in Figure 5.1(a)).

- *Serious unfairness:* Two simultaneous TCP traffic flows may suffer from unacceptable unfairness. For example, in Figure 5.1(b)), once F2 starts, F1 is completely forced down.
- *Incompatibility problem:* Two simultaneous TCP traffic flows cannot coexist in the network at the same time. Once one flow develops, the other one will shut down. For example, in Figure 5.1(c), F1 and F2 cannot coexist. Most of the time, only one flow has non-zero throughput.

In the next section, we demonstrate all three performance problems by simulations and investigate whether they can be avoided with fair MAC protocols.

We would like to point out that although these scenarios are very simple, they represent typical problems in more general scenarios. It is therefore natural to select them as benchmarking scenarios to test whether a new protocol or a new combination of protocols can deliver better TCP performance and deserve to be investigated in more complex scenarios.

5.2 TCP Performance Evaluation and Comparison

In this section, we evaluate and compare the performance of TCP over four MAC protocols: IEEE 802.11, MBFAIR, EDWOP and EHATDMA. We try to find out whether the three performance problems of TCP revealed in [82] can be avoided by using fair MAC protocols. We have carried out extensive simulations. The simulation tool used is NS2 [50] and the parameters used in simulations are shown in Table 5.1. Reno-TCP

Simulation Parameters	**Value**
Communication Range	250m
Carrier Sense (CS) Range	550m
Capture Threshold	10db
Interference/Communication ratio	1.78
Basic Rate	1Mbps
Data Rate	2Mbps
Packet Size	880 bytes
Packet Rate	250 packets/s
Simulation Time	1000 seconds

Table 5.1: NS-2 simulation parameters for TCP performance evaluation

was used and the maximal window size of TCP was 32. AODV [83] was used as the routing protocol. To focus on the interaction between TCP and MAC, we assume an error-free channel, i.e., no transmission errors; all errors are due to collisions. Figure 5.1 shows the scenarios used in this study (the same scenarios were used in [82]). In each scenario, a simple string topology consisting of eight evenly distributed nodes is used. The distance between neighboring nodes is 200 meters. For each scenario, we vary the MAC protocol and data packet size (32B, 64B, 128B, 256B, 512B, 1024B, 1460B). For each MAC/packet-size combination, we run the simulation 32 times. The average results of the 32 runs are presented.

Three performance metrics are used in the study:

- *Goodput rate:* the number of bits a TCP flow transmits successfully per unit time;
- *Stability Index (SI):* The stability index (SI) is used to measure the stability of a TCP flow. It is defined by:

$$SI = \frac{\left(\sum_{i=1}^{n} r_i\right)^2}{n \sum_{i=1}^{n} r_i^2},$$

where r_i is the goodput rate of a flow in the i^{th} interval (we use 1.0-second interval). Actually, SI is the widely used fairness index (FI) calculated in the time domain. The closer SI is to 1, the more stable a TCP flow is.

- *Fair index (FI):* defined by the following formula:

$$FI = \frac{(at_1 + at_2)^2}{2(at_1^2 + at_2^2)},$$

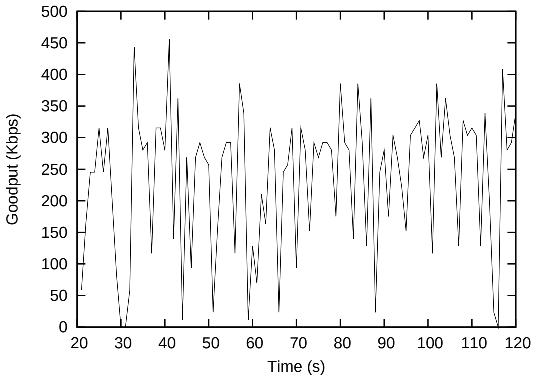
where at_1 and at_2 are the average throughputs of flow 1 and flow 2 respectively.

5.2.1 Instability Problem

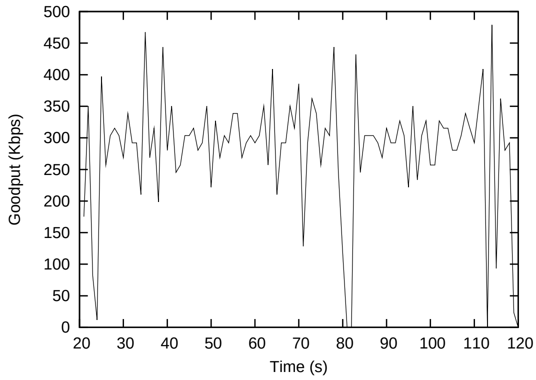
Due to the contention between forward *data* packets and backward *ack* packets, the goodput rate of a TCP flow could vary widely from one instant to another. Figure 5.1(a) shows a scenario in which TCP displays the instability problem. In this scenario, a four-hop TCP flow is established between node 1 and node 5. The TCP flow starts at the 20^{th} second and ends at the 120^{th} second.

Figure 5.2(a) shows the result of one simulation for the scenario of Figure 5.1(a). It shows the goodput rate measured in each 1.0-second interval against time for the IEEE 802.11/1460B combination. It can be seen that the goodput rate varies widely during the 100 seconds lifetime of the flow. As shown in the figure, the SI for this instance is 0.82. The result is consistent with that presented in [82].

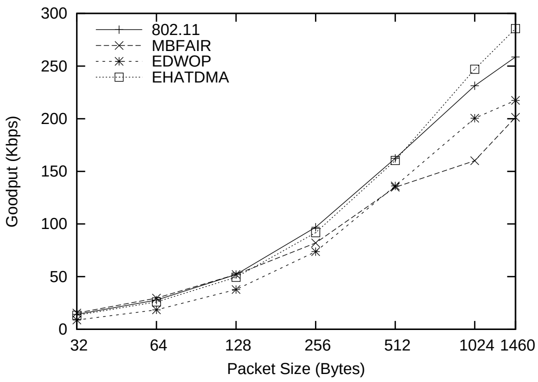
Figure 5.2(b) and Figure 5.2(c) report the average goodput rate and SI respectively for the various MAC protocols. All the results presented in these two figures are the average of the 32 runs. The results show that EHATDMA performs quite well in both



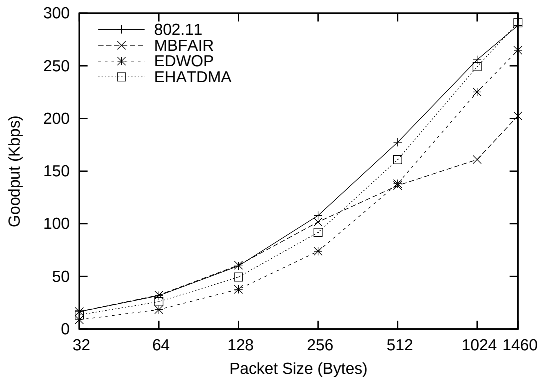
(a) TCP/802.11, packet size=1460B, FLBP=0ms (SI=0.82)



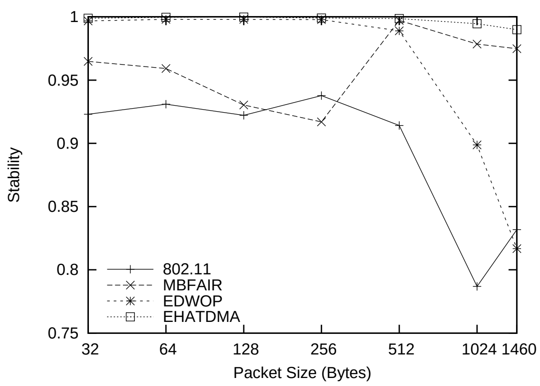
(d) TCP/802.11, packet size=1460B, FLBP=100ms (SI=0.9)



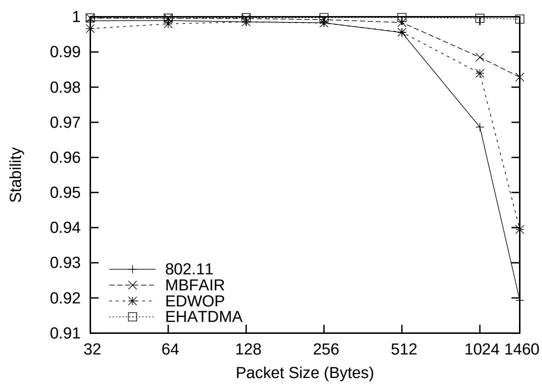
(b) Goodput versus packet size (FLBP=0ms)



(e) Goodput versus packet size (FLBP=100ms)



(c) Stability versus packet size (FLBP=0ms)



(f) Stability versus packet size (FLBP=100ms)

Figure 5.2: Simulation results for instability problem

aspects. It not only has the largest goodput rate in most cases (Figure 5.2(b)), but also performs very stably across all data packet sizes. The SI of EHATDMA remains nearly constant around 1 (Figure 5.2(c)). On the other hand, although IEEE 802.11 performs quite well in terms of goodput rate, its stability is the worst among all the four MAC protocols. EDWOP performs stably with some sacrifices in goodput when the data packet size is small ($\leq 512B$). However, it breaks down when packet size exceeds 512B. This behaviour is expected. In EDWOP, when a receiver detects an out-of-order transmission, it will notify the sender to backoff for a period of time that is a function of $T_{success}$ (the longest time to transmit a data packet). This backoff stage turns EDWOP into a non-work-conserving scheme to improve the fairness and stability properties. In our implementation, we set the largest packet size as 1500B. As the data packet size approaches 1500B, the difference between the backoff time and the time actually needed to transmit a data packet reduces. As a result, the contention increases, which in turn leads to instability. Increasing the value of $T_{success}$ will improve the stability property of EDWOP. However, this will certainly further reduce the goodput of a TCP flow. Contrary to expectations, MBFAIR does not deliver better performance to TCP in this scenario. It performs unstably when data packet size is small and achieves lower goodput rate when data packet size is large. This can be explained as follows: In this scenario, all nodes participating in transmission have at least two "other" nodes; thus, the channel usage ratio is smaller than C most of the time. As a result, all nodes tend to choose a small CW, which in turn leads to more collisions. Consequently, the link breakage event is triggered frequently, which causes the lower goodput rate and instability.

As pointed out in [82], the reason for the instability problem is the MAC protocol.

After several attempts, the MAC protocol discards a data packet and reports to the route protocol (in our case, AODV) that the link is broken. Upon receiving a link breakage event, the route protocol discards all packets waiting on that link and tries to find a new route. Before a new route is found, it is highly probable that the TCP flow will time out, which is well known to be very costly in terms of throughput loss. From the end users point of view, a frequent timeout TCP flow is very unstable and usually unacceptable.

However, the link breakage event reported by the MAC protocol in this situation is false and unnecessary. Actually, the link is still there. It is only because the MAC protocol cannot fairly schedule channel access, which prevents a link from being activated successfully for a long time, leading the nodes of the link to draw the false conclusion that the link is broken. A fair MAC protocol should avoid most of these false events, as demonstrated by EHATDMA. However, different fair MAC protocols alleviate this problem to different extents. EHATDMA is much better than MBFAIR in this case.

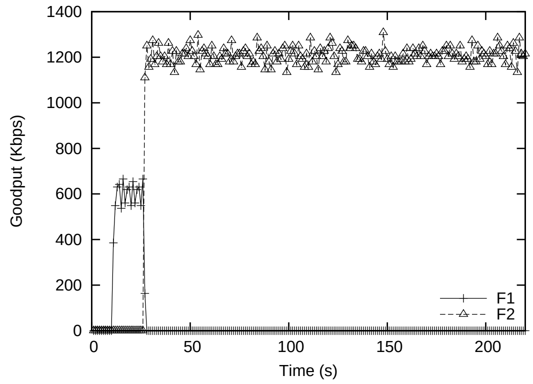
To further reduce the false link breakage event, we have designed a simple mechanism named false link breakage prevent (FLBP) which can be incorporated into all MAC protocols. The idea of FLBP is very simple. After exceeding the maximal number of channel access attempts, an accessing node shall not trigger a link breakage event if it has successfully sent (received) a packet to (from) the other end of the link within the last α time units. The parameter α should be so chosen that it does not introduce substantial delay in the detection of a true link breakage (say due to mobility of nodes) while preventing most false link breakages. Without FLBP, it takes at least 25ms for a node to generate a link breakage event (the time needed to make the maximal number of attempts). In our simulation, we therefore choose α to be $4 \times 25ms = 100ms$, which is large enough to avoid most false link breakage events and but small enough to detect a true one. Note that $\alpha = 100ms$ is far smaller than the TCP coarse timeout interval which is usually several times of 500ms; therefore, FLBP will not trigger TCP timeout events unnecessarily.

Figures 5.2(d)(e)(f) show the simulation results for the same configurations with FLBP enabled. It can be seen that the performance of TCP over all the MAC protocols improves, especially for IEEE 802.11 and MBFAIR. The stability of TCP over IEEE 802.11 and MBFAIR improves substantially with data packet size $\leq 512B$. With the data packets $> 512B$, although the instability problem is eased further by FLBP, it is still prominent. Increasing α will increase stability. However, a larger value of α will certainly delay the detection of a true link breakage event, which may be undesirable in mobile scenarios.

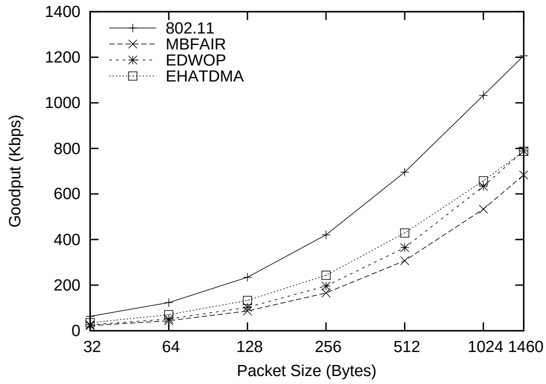
From the above analysis, we conclude that a fair MAC protocol does not necessarily lead to stable TCP behaviour. Neither MBFAIR nor EDWOP can provide stable goodput rate to the four-hop TCP flow for all data packet sizes; only EHATDMA can. FLBP is a useful mechanism that benefits all the four MAC protocols.

5.2.2 Serious Unfairness Problem

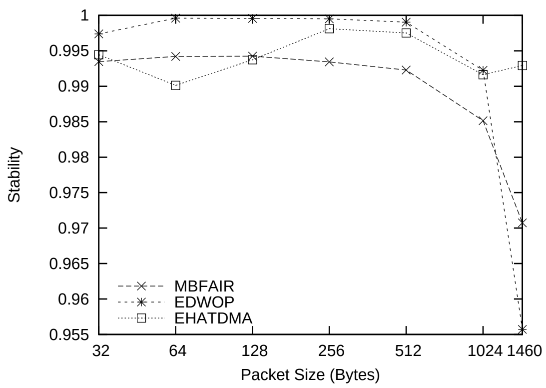
In this scenario (Figure 5.1(b)), two TCP flows (F1 and F2) are set up and start at 10s and 20s respectively. Each simulation last 220 seconds. Figure 5.3(a) plots the simulation results for TCP over IEEE 802.11 with a packet size of 1460B. It clearly shows that once F2 develops, F1 is completely forced out: the goodput rate of F1



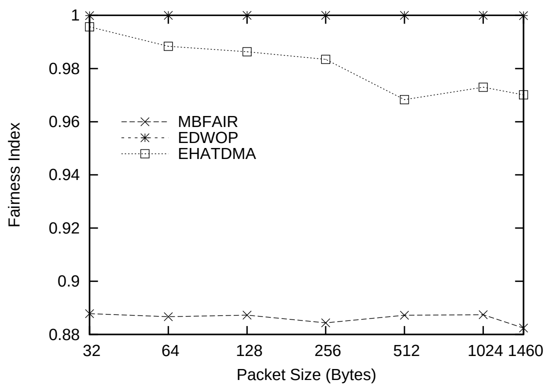
(a) Serious unfairness problem



(b) average Goodput versus Packet Size (FLBP=100ms)



(c) Stability versus Packet Size (FLBP=100ms)



(d) Fairness versus Packet Size (FLBP=100ms)

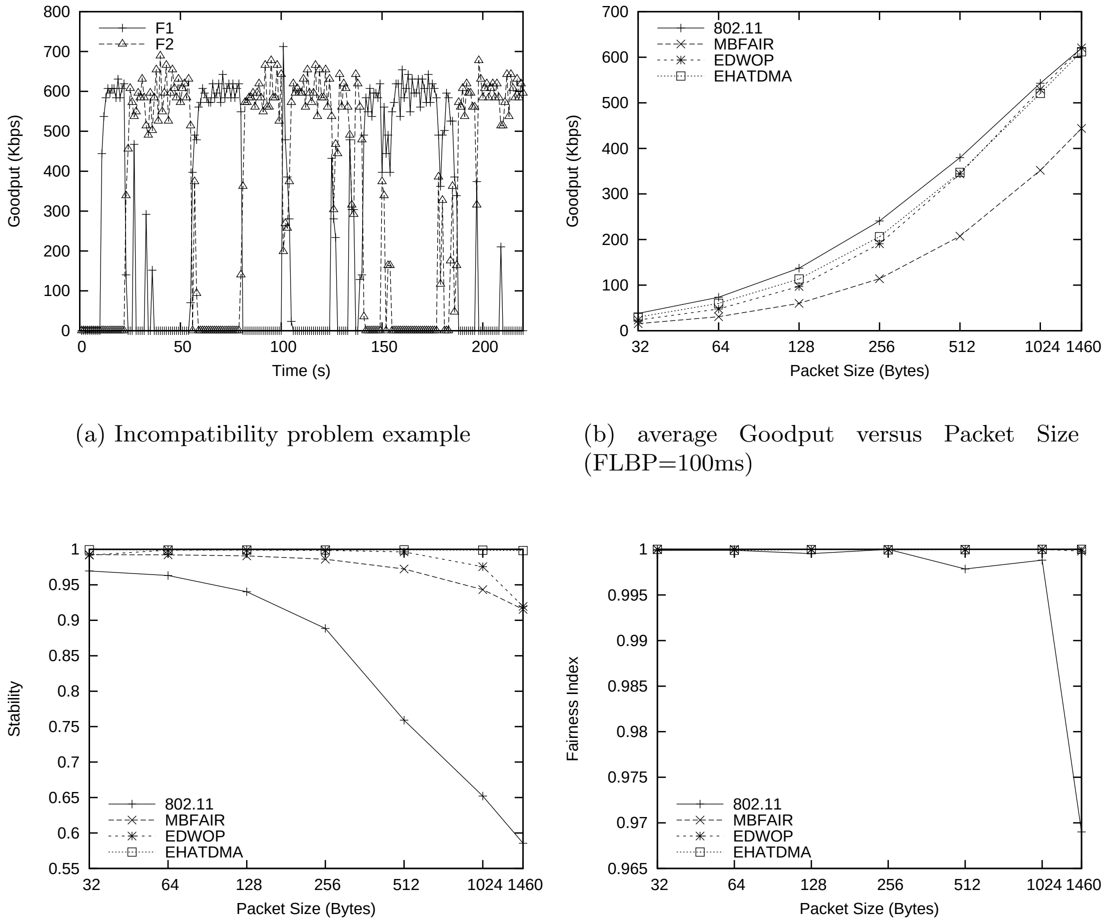
Figure 5.3: Simulation results for serious unfairness problem

remains 0 for the rest of the time. For other data packet sizes, TCP over IEEE 802.11 behaves similarly. Clearly, such kind of extreme unfairness is not desirable.

Figures 5.3(b)(c)(d) report the simulation results for all combinations with FLBP set at 100ms. To isolate the effect of the routing protocol, we only consider the simulation results between time T and $T + 100$, where T is the first time when a route for flow F2 is found. Figure 5.3(b) shows the average goodput rate of the network (i.e., the sum of goodput rates of F1 and F2). It can be seen that the average goodput rate of TCP/IEEE 802.11 combination is by far larger than other combinations. However, the high goodput rate of TCP/IEEE 802.11 combination comes at the cost of fairness. Actually, once a route for F2 is found, F1 is almost starved (for this reason, we do not include the results of IEEE 802.11 in Figure 5.3(c) and Figure 5.3(d)). Since F2 has only one hop, the high goodput rate is expected. Figure 5.3(c) reports the stability index of the network, which is the average of the SIs of F1 and F2. The figure shows the same trend as Figure 5.2(f). When the size of the data packet is smaller than 512B, all the fair MAC protocols deliver a stable channel to the TCP flows ($SI > 0.99$). When the size of the data packet is larger than 512B, the stability of TCP over MBFAIR and EDWOP reduces significantly. However, the stability of EHATDMA is rather insensitive to data packet size. The SI of EHATDMA is larger than 0.99 for all data packet sizes. Figure 5.3(d) reports the fairness index. It appears that different fair protocols allocate channel differently. In this scenario, EDWOP treats the two TCP flows absolutely fairly — in spite of the number of hops, both of the flows are allocated the same goodput rate (FI is 1). However, MBFAIR favors short flows; the rate of flow F2 is twice that of F1 (FI is around 0.9). The fairness of EHATDMA lies between them.

5.2.3 Incompatibility Problem

Figure 5.1(c) shows the incompatibility problem scenario. Two TCP flows are set up. F1 starts at 10s and F2 starts at 20s. In [82], it has been revealed that F1 and F2 cannot coexist: once one flow develops, the other flow will shut down. This phenomenon is clearly manifested in Figure 5.4(a), which plots one of the simulation results with TCP



(a) Incompatibility problem example

(b) average Goodput versus Packet Size (FLBP=100ms)

(c) Stability versus Packet Size (FLBP=100ms)

(d) Fairness versus Packet Size (FLBP=100ms)

Figure 5.4: Simulation results for incompatibility problem

over IEEE 802.11 and a packet size 1460B. Figure 5.4(a) shows that most of the time, only one flow, either F1 or F2, has a non-zero goodput rate.

Figure 5.4(b)(c)(d) report the simulation results for all combinations. All the measures are the same as given in Section 5.2.2. We still use SI to measure the compatibility of two flows because from a single flow point of view, the incompatibility problem is actually the instability problem.

The results in Figure 5.4(c) and Figure 5.4(d) show that EHATDMA performs very well for all data packet sizes in terms of compatibility (or stability) and fairness. For all cases, both indexes of these two protocols are almost equal to 1 — in other words, the two TCP flows run very stably and equally share the network bandwidth. MBFAIR and EDWOP can also allocate bandwidth fairly (Figure 5.4(d)). However, when operating over MBFAIR and EDWOP, the compatibility of F1 and F2 degrades as packet size increases. Furthermore, as shown in Figure 5.4(b), the average goodput of TCP/MBFAIR is much lower than that of TCP/EHATDMA and TCP/EDWOP. For example, when packet size is 512B, the average throughput of TCP/EHATDMA is 1.67 times that of TCP/MBFAIR. The reason is that MBFAIR is inclined to select a large contention window in this scenario because the TCP flows only have two hops. TCP/IEEE 802.11 has a slightly higher goodput rate than others. However, the compatibility of the two TCP flows over IEEE 802.11 is the worst. Interestingly, in this scenario, TCP/IEEE 802.11 achieves long-run fairness (Figure 5.4(d)) when packet size is not very large: $FI > 0.995$ for packet size ≤ 1024. The fairness in this case is mainly due to the symmetry of topology and traffic.

5.3 Summary

In this chapter, we have evaluated and compared the performance of TCP over IEEE 802.11 and three fair MAC protocols: MBFAIR, EDWOP and EHATDMA. Simulation results for representative scenarios indicate that a fair MAC protocol does not necessary lead to satisfactory TCP performance (e.g., MBFAIR). However, compared with IEEE

802.11, the fair MAC protocols do improve the stability of TCP flows and allocate bandwidth among contending TCP flows more fairly. With the help of THE FLBP mechanism, the fairness and stability of TCP flows are further improved. The amounts of improvement made by different fair MAC protocols are different. Overall, EHATDMA performs the best. It achieves fairness and stability for all three scenarios and the full range of data packet sizes without sacrificing too much goodput. On the other hand, MBFAIR and EDWOP do not work well for all configurations. For large data packet sizes, they make a trade-off between goodput rate and stability (or fairness).

The study in this chapter provides positive evidence that fair MAC protocols do improve the fairness of bandwidth allocation as well as other performance aspects of TCP flows. Previous studies (e.g., [80]) have shown that a reliable link-layer protocol that is TCP-aware is the most effective scheme in improving the performance of TCP over wireline-cum-wireless connections. Since an unfair MAC protocol is tantamount to presenting unreliable links (which is up and down in an unpredictable way) to upper layer protocols, to make the MAC protocol fairer is therefore equivalent to making it more reliable. In this sense, our results for multihop wireless ad hoc network echo the discoveries in [80] for wireline-cum-wireless connections; both sets of results strongly support investing in the fairness of the MAC/link layer to improve the performance of TCP.

CHAPTER 6

Conclusion and Future Research

6.1 Summary

Multihop wireless ad hoc networks are an ideal technology to establish an on-demand communication system for civilian and military applications. With this technology, users can set up a network instantly as the need arises. How network bandwidth is shared among users is an important issue that needs to be considered from the very beginning of the design of the network. Fairness is a desirable property in bandwidth allocation for best effort service as well as for differentiated service (DiffServ) [5], where flows belonging to the same class need to fairly share bandwidth allocated for that class. Although much research has been done on fairness of bandwidth allocation in the context of wireline networks, the algorithms developed for their fair bandwidth provision cannot be easily extended to multihop wireless ad hoc networks. In this thesis, we have investigated the fairness issues in multihop wireless ad hoc networks. We have looked into the fairness problem at two levels: MAC/link layer fairness and network layer fairness, with emphasis on MAC/link layer fairness, which we believe is the fundamental supporting element for network layer fairness. The key contributions of this thesis are as follows:

1. Through simulations, we have demonstrated that the widely used MAC protocol IEEE 802.11 could suffer from the one/zero fairness problem when operating in a

multihop wireless ad hoc network: some flows in the network may completely seize the channel capacity while others are virtually starved. Three causes leading to the one/zero fairness problem have been identified: the lack of synchronization problem (LSP), the double contention areas problem (DCP), and the lack of coordination problem (LCP).

- *The lack of synchronization problem (LSP):* The sender of a flow has no information about when the receiver is/will be idle.
- *The lack of coordination problem (LCP):* In a real-life multihop network, not all the interferers can be notified by the CTS/POLL control frame, which may lead to the one/zero fairness problem.
- *The double contention areas problem (DCP):* The sender and the receiver of a flow are exposed to two different contention areas. If both the areas are busy most of the time, the flow is likely to be starved.

2. We have proposed a new MAC protocol known as extended hybrid asynchronous time division multiple access (EHATDMA) to deal with the severe unfairness caused by the lack of synchronization problem (LSP), the double contention areas problem (DCP) and the lack of coordination problem (LCP). The protocol has three control schemes. The first is the SI-RI hybrid scheme for dealing with LSP; it employs both SI and RI collision avoidance mechanisms. The second is the ATDMA, which deals with DCP. It requires a flow that has just completed a successful data transmission to restrained from accessing the channel for a $flow\_period$, which is estimated based on the traffic load around the flow. The third is a power control algorithm, which deals with the LCP. A node adjusts its

transmission power for CTS/POLL when experiencing LCP.

3. For better assessment of fairness, we have designed an index named the max-min fairness index, which is scenario-independent and reflects the difference between the fair sharing provided by a protocol and the ideal max-min fair sharing.

4. We have carried out comprehensive simulation experiments for EHATDMA and other related protocols (IEEE 802.11, CBFAIR, MBFAIR, EMLM, DWOP and EDWOP) in a series of comparative performance studies. Simulation results show that while various enhancements have been proposed to improve the fairness of MAC protocols of multihop wireless networks, most of them are still strongly biased towards throughput when a conflict between throughput and fairness arises. In addition, the fairness performance of these proposals varies widely from one scenario to another. On the other hand, EHATDMA strikes a good balance between throughput and fairness. It delivers a consistently high level of fairness regardless of network topology, traffic load and radio parameters, yet maintains high throughput whenever possible. Furthermore, EHATDMA is able to deal with mobility swiftly; it can rapidly reach a new stable state after a scenario shift. Our simulation results also reveal that the most important mechanism affecting the fair sharing of radio channels among flows is the non-work-conserving mechanism.

5. We have proposed an analytical model to analyze the throughput and fairness property of the distributed coordination function (DCF) of IEEE 802.11 in multihop wireless networks, with nodes randomly placed according to a two-dimensional Poisson distribution. The model is applicable to both access methods

of DCF, i.e., the basic access method and the RTS/CTS access method. Simulation results show that our model is very accurate in predicting saturation throughput.

6. Using the analytical model, we have evaluated the saturation throughput of DCF. The results reveal that the RTS/CTS method is much more superior to the basic access method in most cases. More importantly, our model indicates that the RTS/CTS access method with the default parameters operates in a region almost optimal in terms of saturation throughput.

7. Our analytical model has also enabled us to analyze the fairness property of IEEE 802.11 operating in multihop wireless ad hoc networks. We are interested in the fairness of channel shares allocated by IEEE 802.11 among one-hop flows of various source-destination distances. We have defined two throughputs to explore the fairness property of DCF: long-run flow throughput and instant flow throughput. We have found that both long-run throughput and instant throughput of a flow decrease as the flow's distance increases. As node density or packet size increases, short flows get a larger share of throughput than long flows do. Particularly, the difference of instant throughput between short flows and long flows may be in one or even two orders of magnitude, which means that the average service time of a long flow may be tens or even hundreds of times larger than that of a short flow. Such a huge gap is harmful to the operations of upper layer protocol. We have proposed non-work-conserving principles to reduce the gap. By extrapolating from the analytical model, we have established the conclusion that non-work-conserving principles will improve the fairness of both

the throughputs that we have defined. We have substantiated the conclusion with simulation results. In addition to fairness, non-work-conserving principles can also reduce the average number of retransmissions experienced by packets; it may even improve the overall throughput of dense networks.

8. We have evaluated and compared the performance of TCP over IEEE 802.11 and three fair MAC protocols: MBFAIR, EDWOP and EHATDMA. Simulation results for representative scenarios indicate that a fair MAC protocol does not necessarily lead to a satisfactory performance of TCP (e.g., MBFAIR). However, compared with IEEE 802.11, the fair MAC protocols do improve the stability of TCP flows and allocate bandwidth among contending TCP flows more fairly. With the help of the FLBP mechanism, the fairness and stability of TCP flows are further improved. The amounts of improvement made by different fair MAC protocols are different. Overall, EHATDMA performans the best. It achieves fairness and stability for all three scenarios and the entire range of data packet sizes without sacrificing too much goodput. On the other hand, MBFAIR and EDWOP do not work well for all configurations. For large data packet sizes, they make a trade-off between goodput rate and stability (or fairness).

6.2 Directions for Future Work

In this thesis, we have mainly focused on MAC/link layer fairness, which is a fundamental element supporting end-to-end fairness. Our preliminary work in Chapter 5 indicates that fair MAC protocols not only improve the fairness of TCP flows, but also has the potential to improve other performance aspects of TCP flows (e.g., stability and compatibility). Clearly, detailed further study and investigation are needed to fully understand and support fair sharing among end-to-end flows in multihop wireless ad hoc networks. We present here some directions for future research on end-to-end fairness:

- *Routing protocols and fairness:* Routing protocols have a profound impact on the fairness of end-to-end flows. For example, if routes generated by a routing protocol for two end-to-end flows share some common nodes or pass though a common contention region, the fairness problem may arise because the two flows compete with each other in the common nodes or common region. However, if the routes are disjointed, i.e., the two routes have no common nodes and do not pass through any common contention regions, there will be no fairness problems arising from contentions for resources. Another fairness problem related with routing protocols is the fairness of route discovery delays. If the routing protocol is an on-demand based protocol [2], different flows may experience very different route discovery delay. For example, in Figure 5.1(b), if F1 starts before F2, the AODV can find a route for F2 very quickly. However, if F2 starts before F1, the route discovery delay for F1 will be significantly increased. In extreme cases, no route for F1 may be found during the whole simulation time. From the end users point of view, this is another kind of fairness problem: the admission fairness problem. With fairness in consideration, the routing problem in multihop wireless ad hoc networks becomes even more challenging.
- *Multiple factors interaction:* In the literature, in addition to the MAC protocol, researchers have also identified other factors that impact the fairness of TCP

flows, e.g., routing protocol, length of a route [27], buffer size [28], active queueing management algorithm [29], and congestion control algorithms [30–33], etc. The impacts of these individual factors on the fairness and throughput of TCP have been studied, and corresponding enhancements have also been proposed. However, the performance observed by users is the results of interactions of all these factors. Therefore, it is important to characterize the interactions and investigate the combined effects of these individual enhancements. The method of "simulation based experiments coupled with rigorous statistical analysis" used in [84] can be employed to empirically study the effects of various possible interactions.

- *Cross layer design and optimization:* Recently, researchers of multihop wireless ad hoc networks have achieved promising progress in cross layer design [85–87] and optimization with various objectives: e.g., to increase end-to-end (TCP) throughput and energy efficiency of the network [88], to increase single-hop throughput and reduce power consumption [89], or joint routing, scheduling and power control to minimize power consumption of the whole network while meeting other requirements [90–93]. It would be interesting to incorporate fairness as a requirement into these models, from which distributed algorithms can be derived so that a certain level of fairness can be maintained.

- *End-to-end Fairness modeling in multihop wireless ad hoc networks:* In wireline networks, it has been widely accepted that the fair bandwidth allocation problem can be modeled as a general utility based constrained maximization problem ([13, 94, 95]). It will be valuable to extend the model to multihop wireless ad hoc networks. Attempts have been made in this direction ([55]). However, caution must be exercised in extending the model since a wireless link in a multihop wireless ad hoc network channel has no fixed bandwidth and links competing with one another may not have information of each other (lack of synchronization problem). To be relevant, the extended model *must* explicitly take into consideration the essential characteristics of multihop wireless ad hoc networks, more specifically, the incomplete information problem and the lack of synchronization problem.

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