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Distributed Scheduling and Multi-channel
Opportunistic Media Access for Ad Hoc Networks

by

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Distributed Scheduling and Multi-channel Opportunistic Media Access for Ad Hoc Networks

Vikram Kanodia

Abstract

Current wireless ad hoc networks suffer from two main performance limitations. First, current ad hoc networks do not efficiently utilize the scarce and dynamic wireless spectrum. As a result, the goodput of current ad hoc networks is often lower than the maximum radio transmission rate. Second, current ad hoc networks provide only best effort service, and there is little related work to provide general mechanisms to enable more powerful services (such as guaranteed services, differentiated services and flow protection). Consequently, current ad hoc networks are unable to provide quality-of-service (throughput or delay targets, QoS differentiation and fairness) and can incur severe unfairness even in simple topologies. In this thesis I design and evaluate mechanisms that together address the above mentioned two main performance limitations of current ad hoc networks. In particular this thesis has two main contributions.

First, I propose and evaluate Distributed Wireless Ordering Protocol (DWOP), which provides a framework for design of join scheduling and MAC in ad hoc networks. The goal of DWOP is to ensure that to the closest extent possible, packets are serviced in the order as defined by a centralized reference scheduler. By ensuring that packets access the medium in an exact reference order, DWOP serves as a
framework to apply the wealth of packet scheduling service disciplines developed for wireline networks to wireless ad hoc networks thereby making it possible to achieve the desired goals of fairness, throughput and delay targets and service differentiation in such networks.

Second, I propose MAC mechanisms to *opportunistically* exploit the scarce and variable wireless channel to maximize net system throughput of ad hoc networks. In particular I devise *Multi-channel Opportunistic Auto Rate (MOAR)*, a distributed MAC protocol which exploits the presence of *frequency diversity* in ad hoc networks to maximize the net throughput of such networks. MOAR is opportunistic across both users and channels and exploits temporal variations across multiple frequency channels present at the physical (PHY) layer to opportunistically transmit data at a higher rate on high quality channels.

The two contributions, namely *joint design of distributed MAC and scheduling* and *design of multi-channel opportunistic MAC protocol*, together form a framework for high performance ad hoc networks which not only support QoS but also achieve high throughput by efficiently exploiting the scarce and dynamic wireless channel.
To my parents and brother.
Acknowledgments

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Chapter 1

Introduction

An ad hoc network is a collection of wireless nodes that can communicate with each other without any dependence on a fixed infrastructure (such as base station or access points) or centralized administration [1]. Nodes within transmission range can communicate directly with each other while those out of radio range of each other must rely on other nodes to forward along packets to their final destination. Since ad hoc networks do not require any fixed infrastructure, they can be set up inexpensively as needed. The potential for low cost deployment and high availability makes ad hoc networks attractive for a number of applications. While most current non-military ad hoc network test-beds are experimental in nature, future deployment scenarios for ad hoc networks include military operations and disaster relief, collaborative computing and communications in smaller areas (buildings, classrooms, conferences etc.) and rapidly reconfigurable metropolitan wireless networks among others [2, 3]. However, in order to make the successful transition from experimental environments to commercial environments ad hoc network designers will need to address two critical challenges as described below.

The first challenge faced by network designers while designing high performance ad hoc networks is to ensure that future ad hoc networks are able to efficiently utilize the scarce and dynamic wireless spectrum at the physical (PHY) layer in order to maximize throughput. The goodput of current ad hoc networks is often lower than the maximum radio transmission rate after encountering the effects of multiple
access, fading, noise, and interference. Thus it is critical for future high performance ad hoc networks to efficiently utilize the scarce wireless channel in order to maximize throughput. Recent advances in wireless communications [4, 5, 6] have shown that spectral efficiency can be dramatically increased if physical layer rate adaptation is employed to serve users with better channel conditions at a higher data rate while maintaining acceptable error rate. However, since rate adaptation is primarily a physical layer protocol, Medium Access Control (MAC) mechanisms are needed to exploit this capability. Traditionally, design of MAC protocols has only been coarsely coupled with the design of physical layer protocols. However, the uniquely variable nature of the wireless channel coupled with scarcity of spectrum at the PHY layer motivates for designing novel MAC protocols which not only fully exploits the entire range of capabilities provided by the PHY layer but are also able to translate the unique characteristics of the wireless channel into additional throughput gain at the MAC layer. While the most popular MAC standard for ad hoc networks (namely the IEEE 802.11 [7, 8, 9] MAC standard) is not equipped to ensure efficient utilization of the scarce and limited wireless spectrum, data rate adaptation at the MAC layer based on physical channel conditions [10, 11] has been shown to result in a moderate network throughput gain. Moreover, significant throughput gains can be achieved by designing opportunistic MAC mechanisms which efficiently exploit the inherent diversity of the wireless channel in time (across users). For example Opportunistic Auto Rate (OAR) [12, 13] is a protocol that exploits the fact that the time scale over which received signal strength is correlated is on the order of multiple packet transmission times to opportunistically send back to back data packets when the channel is of good quality. In this way OAR is able to achieve a higher throughput then current state-of-art rate adaptation protocols. While OAR is opportunistic across users
to the maximal extent, it is still unable to exploit diversity in frequency domain pro-
vided by the presence of multiple frequency channels within many popular wireless
systems (for example the IEEE 802.11 standard supports multiple frequency chan-
nels at the physical layer). In particular, if channel quality on a particular frequency
channel is poor, significant throughput gains are available via opportunistically skip-
ning frequency channels in search of better quality channels. However, designing an
efficient opportunistic channel skipping MAC protocol for wireless ad hoc networks
is quite challenging because of the following three reasons. First, the absence of a
central controller in wireless ad hoc networks makes it necessary for the transmis-
ter and the receiver of a flow to coordinate a skip decision in a totally distributed
manner. Thus there is a need for mechanisms to enable the sender and the receiver
to negotiate and skip decision between themselves without relying on a centralized
entity. Second, channel state information regarding other frequency channels is not
available a priori in most practical systems. Thus an efficient channel skipping MAC
protocol needs to measure channel state information on the current frequency chan-
nel prior to making a decision to skip which introduces the need to incorporate the
overhead of channel measurement into protocol design. Finally, although in theory
nodes can skip indefinitely in search of a better channel until the frequency channel
with the highest possible transmission rate is found, due to the overhead of channel
measurement and skipping, the throughput gains available via opportunistic skip-
ning diminish with each skip. Moreover, when the average channel conditions are
poor, the probability of finding the highest quality channel (and the highest possible
data rate) is very low. Consequently, in order to maximize the gain in throughput
it is critical to devise an optimal skipping rule which limits the number of times a
node skips in search of a better quality frequency channel.
The second challenge faced by wireless ad hoc network designers while designing high performance ad hoc networks is to address the critical need for ad hoc networks to support more powerful services (such as guaranteed services, differentiated services and flow protection) than the current best effort service provided by the most popular MAC protocol for ad hoc networks, namely IEEE 802.11 standard. In particular, ad hoc networks need to provide mechanisms to meet the broad goals of throughput targets, delay targets, and some notion of "weighted fairness", wherein flows with larger weights receive correspondingly better service in accordance with a system-wide fairness model. In wireline networks, link layer fairness mechanisms serve as the foundation for achieving quality of service (QoS) at the network layer (for example Weighted Fair Queuing for the IntServ QoS service model [14]) and fluid fair queuing [14, 15] has been a popular paradigm for providing bounded delay channel access and fairness for packet flows over a shared unidirectional wired link. Popular wireline packet scheduling service disciplines\footnote{See [16] for a review of various packet scheduling service disciplines.} implement fair queuing by mapping the targeted QoS constraint to a priority index associated with each packet and then dynamically scheduling the transmission of each packet in a reference service order of smallest priority index first. At first glance it would seem that the wireline scheduling model of transmitting packets in accordance with a reference scheduling order is equally applicable or can be readily extended to wireless ad hoc networks. However, the following three unique characteristics of ad hoc networks makes the problem of adapting fair queuing to such networks quite challenging. First, due the absence of a centralized controller in wireless ad hoc networks, nodes are unable to assess their per-packet relative priorities prior to medium-access and as such have no means to defer access to a node with a higher priority packet. Thus in the absence of coor-
dination or cooperation nodes are unable to access the medium in accordance with an order as defined by a centralized reference scheduler. Second, the random nature of the most popular medium access control (MAC) protocols\(^2\) makes the problem of enforcing dynamic priority-based medium access in accordance with a reference scheduler very difficult. As opposed to centralized systems where scheduling and medium access are typically addressed independently, the distributed nature of ad hoc networks leads to a coupling between scheduling and medium access adding to the challenge of ensuring that packets are transmitted in accordance with a centralized reference schedule. Finally, due to spatial (location dependent) contention for the shared wireless channel some nodes may not have accurate knowledge of the contention even in their own neighborhood. In other words, transmission of a packet involves contention over the joint neighborhood of the sender and the receiver and asymmetry in topology can prevent some nodes from having a true knowledge of the level of contention at the other end (sender or receiver) of a flow which can lead to potential starvation of some flows. For example, in wireless ad hoc networks employing a CSMA/CA media access algorithm such as IEEE 802.11, even a simple topology with two flows and four nodes can result in starvation of one of the flows. As illustrated in Figure 1.1 and Reference [17], a topology in which the sender of one flow is out of radio range of the sender of another flow\(^3\) results in severe throughput degradation, and hence unfairness, for one of the flows. Thus in summary, due to the absence of a centralized controller, random nature of most popular MAC protocols and spatial dependent contention for the shared wireless channel, current ad hoc networks are unable to implement popular wireline fair queuing algorithms to

\(^2\)The IEEE 802.11 Distributed Control Function (DCF) is based on the CSMA/CA protocol.

\(^3\)Referred to as the Asymmetric Topology.
achieve QoS targets.

![Diagram of an asymmetric topology](image)

Figure 1.1: Example of an asymmetric topology where flow A gets 5% of the throughput while Flow B obtains 95% of the throughput.

The problem of providing fairness in ad hoc networks has received a lot of attention in recent years and nearly all the previous approaches have focused on providing new protocols targeted at providing MAC-layer fairness [18, 19, 20, 21, 22, 23, 24]. However, none of the proposed approaches target an ordering mechanism to approximate any given centralized reference scheduler in order to achieve fairness, QoS differentiation or to meet throughput and delay targets. A class of scheduling policies based on eliminating contention and achieving scheduled medium access in ad hoc networks by exchanging node information within a two hop neighborhood has been proposed in [25]. However, the proposed approach requires knowledge of contention in any node's two hop neighborhood and is not based on the popular IEEE 802.11 MAC scheme. Further the Enhanced Distributed Coordination Function (EDCF) [26] (an extension to the IEEE 802.11 DCF) and similar approaches [27, 28] have been proposed to provide class based service differentiation within the IEEE 802.11 MAC framework. However, such approaches only provide statistical priority
and does not guarantee that low priority packets will always wait until all higher priority packets have been transmitted. Thus, all the current approaches to achieve QoS in ad hoc networks fail to ensure that packets are serviced in the exact order as defined by a centralized reference scheduler and as a consequence are unable to apply the wealth of packet scheduling service disciplines developed for wireline networks to wireless ad hoc networks.

*The contribution of this thesis is the design of MAC mechanisms which together address the two critical challenges faced by network designers while designing high performance wireless ad hoc networks - namely inefficient utilization of the scarce and variable wireless spectrum and lack of mechanisms to provide stronger services than the best-effort service offered by current ad hoc networks.* The next section discusses the contributions of this thesis in detail.

### 1.1 Contribution of This Research

The contributions of this thesis can be broadly divided in two parts, namely *joint design of scheduling and medium access control* and *design of opportunistic multi-channel medium access control*, as described below.

- **Joint Design of Scheduling and MAC**

  In this part of the thesis I develop the *Distributed Wireless Ordering Protocol (DWOP)*, a joint media access and distributed scheduling scheme designed to achieve a reference scheduling service order in wireless ad hoc networks. *The goal of DWOP is to ensure that to the closest extent possible, packets are serviced in the order as defined by a centralized reference scheduler.* In this way DWOP serves as a framework to apply the wealth of packet scheduling
service disciplines developed for wireline networks to wireless ad hoc networks thereby making it possible to achieve the desired goals of fairness, throughput and delay targets and service differentiation in such networks.

I first, show that the IEEE 802.11 MAC protocol diverges significantly from an exact service order and can even starve some nodes in many cases. Further, I identify two topology related issues, asymmetric information and perceived collisions as one of the primary reasons behind the poor ordering property of IEEE 802.11. Using a graph theoretic framework I relate protocol behavior to flow of information across mobile nodes and demonstrate how topologies suffering from asymmetric information and perceived collisions can be readily identified and incorporated in the design of an ordering protocol within IEEE 802.11 channel access mechanism.

Next I exploit the broadcast nature of the wireless medium to share information regarding the priority indexes of the queued packets among mobile nodes. Within this context I design DWOP and introduce several MAC layer mechanisms that enable DWOP to use the shared information within the IEEE 802.11 medium access framework with the goal of achieving a perfect service order.

In practical systems where all nodes are not within radio range of each other enforcing exact system-wide perfect packet ordering is a conflicting objective with achieving high utilization and maximal spatial reuse. To address this issue, I introduce a simple mechanism as part of DWOP to increase spatial reuse by trading off controlled deviations from the system-wide perfect schedule. Thus network designers have the freedom to tune DWOP in order to achieve the desired level of tradeoff between spatial reuse and reference service ordering.
Realistic systems will have dynamic properties due to factors such as mobile nodes, sleeping nodes waking up, etc. With such behavior, nodes will not always have complete information about other nodes within their radio range. Hence, I develop a simple analytical model to study the transient characteristics and convergence properties of DWOP. The model shows that DWOP's convergence is sufficiently fast to allow (for example) high mobility speeds.

Finally I illustrate via extensive ns-2 simulations and theoretical analysis that DWOP is able to achieve a nearly perfect service order even in complex topologies with incomplete information and in dynamic scenarios beginning with no or limited information.

- **Opportunistic Multi-channel Media Access**

In this part of the thesis I develop *Multi-channel Opportunistic Auto Rate (MOAR)*, a distributed MAC protocol which exploits the presence of multiple frequency channels at the PHY layer to opportunistically skip frequency channels in search of a better quality channel. The key insight that serves as the motivating factor for designing MOAR is that in most wireless systems (for example IEEE 802.11 enabled ad hoc networks) two different frequency channels are spaced sufficiently apart to experience uncorrelated fading. Thus, if the measured channel conditions on the current frequency channel are poor, MOAR enables mobile nodes to skip frequency channels in search of a better quality channels. Since MOAR nodes transmit packets at a higher rate (on better quality channels), MOAR is able to achieve a *net* higher throughput as compared to state-of-art MAC protocols which exploit only the multi-rate capabilities of the PHY layer. Further MOAR allows the sender and the re-
ceiver of a flow to mutually negotiate a skip decision via transmission of control packets and thus MOAR is able to realize the opportunistic gains available via channel skipping in a distributed manner.

In realistic systems channel state information for all frequency channels may not be available \textit{a priori}. Thus, prior to each channel skip MOAR nodes need to measure the channel quality on the current frequency channel and base the decision to skip to another frequency channel on the measured channel quality. This introduces an additional overhead of channel skipping and thus the throughput gain available via continued opportunistic channel skipping diminish with each channel skip. Moreover, even though in theory nodes can skip indefinitely in search of the highest quality channel, in cases where the \textit{average} channel conditions are poor, the probability of finding the highest quality channel (and the highest possible data rate) is very low. Thus it is important to balance the time and resource costs of channel measurement/skipping with the potentially additional throughput gains available via continued skipping. Towards this end I design an \textit{optimal skipping rule for MOAR} to extract maximal throughput gains via opportunistic channel skipping. The optimal skipping rule for MOAR maps the channel conditions at the PHY layer to a MAC rule which limits the number of times nodes skip in search of a better quality channel.

In order to correctly implement the optimal skipping rule in practical systems, MOAR nodes need to be able to estimate the distribution of achievable data rates. Towards this end I introduce a measurement based estimation scheme to enable MOAR nodes to accurately estimate their channel conditions online.
While MOAR provides a general framework to enable nodes to skip channel opportunistically for systems which support multiple channels at the PHY layer, I focus on IEEE 802.11 standards as an example system to demonstrate the throughput gains available via skipping frequencies in search of better quality channels. I investigate the various factors affecting the performance of MOAR for IEEE 802.11 enabled ad hoc networks via extensive ns-2 simulations. In particular I show that MOAR is able to achieve 20% to 25% average gain in throughput over current state-of-art multi-rate protocols for IEEE 802.11.

1.2 Organization of Thesis

The remainder of this thesis is organized as follows. In Chapter 2 I develop Distributed Wireless Ordering Protocol (DWOP), a media access and scheduling scheme designed to ensure the packets access the wireless medium in an exact order as specified by a scheduling policy. Next, I present the Multi-channel Opportunistic Auto Rate (MOAR) protocol in Chapter 3. MOAR is a distributed MAC protocol which exploits the presence of multiple frequency channels in wireless networks to opportunistically skip channels in search of higher quality channels. Finally, I summarize our contributions in Chapter 4 and also present some directions for future research.
Chapter 2

Joint Scheduling and Medium Access Control

2.1 Introduction

In *ad hoc* networks employing a CSMA/CA media access algorithm such as IEEE 802.11 [7, 9, 8], even a simple topology with two flows and four nodes can result in near starvation. For example, as illustrated in Figure 2.1(a) and Reference [17], a topology in which the sender of one flow is out of radio range of the sender of another flow results in severe throughput degradation, and hence unfairness, for one of the flows. To address such performance problems and lack of fairness in IEEE 802.11 *ad hoc* networks, previous approaches have focused on introducing new protocols targeted at providing MAC-layer fairness [18, 19, 20, 21, 22, 23, 24].

Our approach in this thesis is quite different. Rather than providing node- or flow-level fairness at the MAC layer, I target an *ordering mechanism* that can be used to approximate a set of reference schedulers in order to achieve either QoS differentiation or fairness. That is, our *objective is to design a distributed MAC protocol in which, to the closest extent possible, packets are serviced in the order defined by a reference scheduler*. Our technique applies to a broad class of schedulers in which packets are serviced in increasing order of a priority index that can be computed locally.\(^1\) This class includes Earliest Deadline First, Virtual Clock [29], and FIFO. Both Earliest

\(^1\)That is, the index must be computable using only flow and node state and not the state of remote flows.
Deadline First and Virtual Clock schedulers target QoS differentiation, whereas FIFO combined with TCP provides proportional-fair bandwidth allocation.

In the remainder of this chapter, I consider FIFO as the reference scheduler where priority indexes are set to packets’ arrival times.

In this chapter I present the Distributed Wireless Ordering Protocol (DWOP) [30], a media access and scheduling scheme designed to achieve a reference scheduling service order in wireless ad hoc networks via information sharing. Our contributions are as follows.

First, I study the performance of IEEE 802.11 from the perspective of information sharing. Specifically, I present several scenarios in which IEEE 802.11 diverges significantly from the reference service schedule (e.g., FIFO), resulting in severe performance degradations for a subset of flows. I describe how the root of these problems is incomplete information sharing which can be classified into scenarios of asymmetric information and perceived collisions.

I next introduce a graph-theoretic formalism to explore the role of information sharing in ad hoc networks by generalizing the development of [21]. With a general flow-contention graph, scenarios of asymmetric information and perceived collisions are readily identified so that they can be incorporated into protocol design and analysis.

Within this context, I describe DWOP. The protocol exploits the broadcast nature of the wireless medium to piggyback the priority indexes (arrival times) of queued head-of-line packets on existing hand-shake messages. As the targeted (global) FIFO schedule would transmit packets in order of these arrival times, each node builds a scheduling table based on overheard information of other packet’s arrival times. With this information sharing, I devise a simple MAC rule such that a
node contends for the medium only if it’s locally queued packet has a smaller arrival
time than all packets in its table, i.e., if the node has inferred that it possesses the
next region-wide packet in the hypothetical reference FIFO schedule. Otherwise, if
there are higher priority (lower arrival time) packets in the table, the node defers
access. Contending nodes access the medium according to the IEEE 802.11 protocol,
and deferring nodes can be viewed as setting an extended NAV (Network Allocation
Vector) to wait for their turn. I show that DWOP attains a perfect FIFO schedule
for networks with continuously backlogged flows and all nodes within (symmetric)
radio range of each other. Moreover, we show that two additional table management
techniques, receiver participation and stale entry elimination, limit DWOP’s devi-
tions from the reference FIFO schedule in more complex topologies characterized by
flow graphs with asymmetric information or perceived collisions.

Topologies in which all nodes are not within radio range of each other may present
opportunities for multiple flows to transmit simultaneously without interference, i.e.,
spatial reuse. However, enforcing exact system-wide perfect packet ordering in such
scenarios is a conflicting objective with achieving high utilization and maximal spatial
reuse.\(^2\) To address this issue, I introduce a simple mechanism to increase spatial
reuse by trading off controlled deviations from the system-wide perfect schedule. In
particular, nodes assign a Time-to-Live (TTL) parameter to each table entry based
on a scaled version of their local contention level measured using the size of the
scheduling table. I show that the TTL scaling parameter can be tuned to obtain a
range of behaviors: at one extreme is IEEE 802.11 with high spatial reuse but poor
ordering properties (and hence poor fairness properties). At the other extreme is
high ordering accuracy possibly at the expense of lower spatial reuse.

\(^2\)See [18] for discussion of this tradeoff in the context of fairness.
Realistic systems will have dynamic properties due to factors such as mobile nodes, sleeping nodes waking up, etc. With such behavior, nodes will not always have complete information about other nodes within their radio range. Hence, I develop a simple analytical model to study the transient characteristics and convergence properties of DWOP. In particular, a new node must hear from each other node within its radio range in order to have a complete scheduling table and be assured not to transmit a packet in non-FIFO order. I show that DWOP's convergence is sufficiently fast to allow (for example) high mobility speeds, and that DWOP's determinism results in significantly faster convergence than the time required to hear from each user in an analogous scenario for IEEE 802.11.

Finally, I perform a set of ns-2 simulations to evaluate the ability of DWOP to schedule packets in order of their priority indexes. I revisit the adverse scenarios in which IEEE 802.11 performs poorly and show that near-perfect FIFO is achieved by DWOP, with deviations limited to four packets for DWOP, as compared to practically unbounded deviations for IEEE 802.11. Moreover, I show that even in simple topologies in which all nodes are within radio range of each other, DWOP dramatically improves the packet transmission order and hence fairness as compared to IEEE 802.11. I also consider a set of randomly generated topologies and the role of spatial reuse.

The remainder of this chapter is organized as follows. In Section 2.2, I describe the role of information sharing in the poor and unfair performance obtained by IEEE 802.11 in ad hoc networks. In Section 2.3, I present the DWOP protocol in the context of a graph-theoretic view of information sharing. Next, in Section 2.3.3, I present an analytical model used to explore the transient behavior of DWOP. Finally, in Section 2.6 I review related work and in Section 2.7 I conclude.
2.2 Information Sharing in IEEE 802.11

In this section, after briefly reviewing the IEEE 802.11 Distributed Coordination Function (DCF) protocol (see [7, 9, 8] for further details), I study the role of information sharing in the protocol’s ability to provide a FIFO-like service. In particular, I present two topologies in which IEEE 802.11 obtains severe performance degradations, in one case due to asymmetry of information, and in the latter, due to “perceived collisions”. While the former problem is previously documented in the context of fair bandwidth allocation [17, 21], our perspective of information sharing, presented in Section 2.3, addresses both problems and provides the context for distributed FIFO scheduling.

2.2.1 Review of IEEE 802.11 DCF

In this thesis, I consider the IEEE 802.11 four-way handshake protocol depicted in Figure 2.4. (The boxes marked CURRENT PACKET INFO and tables at the bottom are our proposed modifications to 802.11 and will be discussed in Section 2.3.) A node that intends to transmit a packet waits until the channel is sensed idle for a time period equal to Distributed InterFrame Spacing (DIFS). If the channel is sensed idle for DIFS seconds, the node generates a random backoff timer chosen uniformly from the range $[0, w - 1]$, where $w$ is referred to as the contention window. At the first transmission attempt, $w$ is set to $CW_{\text{min}}$ (minimum contention window). The backoff timer is decremented as long as the channel is sensed idle, stopped when a transmission is detected on the channel, and reactivated when the channel is sensed idle again for more than a duration DIFS.

After the backoff timer reaches 0, the node transmits a short request to send (RTS) message. When the receiving node detects an RTS, it responds after a time period
equal to the Short InterFrame Spacing (SIFS) with a clear to send (CTS) packet. The sending node is allowed to transmit its actual data packet only if the CTS packet is correctly received. The RTS and CTS packets have information regarding the destination node and the length of the data packet to be transmitted. Any other node which hears either the RTS or CTS packet can use the data packet length information to update its network allocation vector (NAV) containing the information of the period for which the channel will remain busy. Thus, any hidden node\textsuperscript{3} can defer transmission suitably to avoid collision. Finally, a binary exponential backoff scheme is used in IEEE 802.11 DCF: after each unsuccessful transmission, the value of $w$ is doubled, up to the maximum value $CW_{\text{max}} = 2^m CW_{\text{min}}$, where $m$ is the number of unsuccessful transmission attempts.

### 2.2.2 Full Information

In topologies such that all nodes are within radio range of each other, nodes have equal probability of capturing the channel since they have the same information regarding the system’s state.\textsuperscript{4} In particular, after transmission of an acknowledgment packet, each node sets its backoff timer according to the same distribution as described above. Thus, since nodes have equal probability of capturing the channel, nodes obtain equal shares of service in the long-term. However, over short timescales, the binary exponential backoff mechanism can result in significantly unequal service shares to backlogged flows.

In any case, I will show that even under such simple topologies, the service order

\textsuperscript{3}Readers are referred to [17] for more discussion on hidden terminal problem.

\textsuperscript{4}For clarity, I ignore the effects of transmission errors and propagation delay in the discussions below.
of IEEE 802.11 diverges significantly from a reference scheduling order due simply to the random access nature of the protocol.

2.2.3 Asymmetric Information

In topologies where all nodes are not within radio range of each other, nodes can have different probability of channel capture due to one node hearing an RTS or CTS that another node does not hear. This unequal channel access probability can result in large differences in the net throughput achieved by individual nodes even over long time scales, thereby resulting in further deviations from the centralized reference schedule. I refer to such scenarios as deriving from asymmetric information among nodes and provide an illustrative example as follows.

(a) Asymmetric Information  
(b) Perceived Collision

Figure 2.1: Illustrative example topologies

Consider the topology depicted in Figure 2.1(a) in which the receiver of Flow A (node 2) is in direct radio range of Flow B, whereas the sender (node 1) has no knowledge of Flow B. In the scenario of Figure 2.1(a), Flow B obtains a significantly higher throughput share as compared to Flow A, namely 95% vs. 5%.

5 All simulations were done using ns-2. Details about the simulation setup are presented in Section 2.5.
in total share can be attributed to the fact that Flow B can hear packets from the receiver of Flow A, and hence knows exactly when to contend for the channel. Thus Flow A has equal probability of capturing the channel after each successful packet transmission by either of the flows. On the other hand, the transmitter of Flow A does not hear any packets from Flow B, and continually attempts to gain access to the channel via repeated RTS requests. The receiver node of Flow A cannot reply as it is either deferring access to Flow B or detecting a collision between a packet from Flow B and RTS from the transmitter of Flow A. In either case, the transmitter of Flow A times-out and doubles its contention window after each failed attempt. Thus the transmitter of Flow A has to discover an available time-slot randomly. Since the DATA packet size is much larger than the control packet size and the contention window can become quite large, the probability of Flow A capturing the channel is significantly less compared to Flow B. After Flow B has finished its packet transmission, it picks a backoff timer from a smaller sized initial congestion window (for the next packet) and thus is more likely to obtain channel access again. Therefore, whenever Flow B obtains the channel it tends to keep it for an extended period of time.

The unequal bandwidth shares obtained in the topology of Figure 2.1(a) were also observed in References [17] and [21]. In [17], the authors propose an additional control packet termed RRTS (Request-for-RTS) as a mechanism to address similar unfairness issues. Although successful in some topologies, the RRTS mechanism leads to unfair throughput allocations in other topologies (an example is given in [17]), and is not a part of the IEEE 802.11 standard.

In [21], the authors attribute the above behavior to the asymmetry in information
available to each flow. Flow B has exact information through the receiver of Flow A, whereas the transmitter of Flow A has no information.

2.2.4 Perceived Collision

While in the above example, more information helps Flow B, this does not imply that more information always increases a node’s throughput share. Consider the topology shown in Figure 2.1(b) where Flow B has information about Flow A and Flow C, while Flow A and Flow C have no information about any other flow in the system. In this case, Flow B attains 28% of the total bandwidth share, whereas Flows A and C get 36% each. Even though Flow B has information about the other two flows in the system, it obtains a smaller share of the total throughput.

The reason that Flow B obtains a smaller bandwidth share is described as follows. Whenever either Flow A or Flow C captures the channel, Flow B sets the NAV accordingly upon hearing the CTS. Due to spatial reuse, Flows A and C can capture the channel simultaneously, thus causing Flow B to set consecutive NAVs. In this case, more information at Flow B about contending flows requires it to defer access to more flows. By extension, as the number of contending flows around Flow B increases, Flow B’s share can decrease. In this scenario, Flow B gains access whenever both Flow A and C are simultaneously in backoff or control packets of Flow A and C collide at Flow B. After acquiring the channel, Flow B can retain access to the channel for multiple packet transmissions for the reasons discussed for the topology in Figure 2.1(b). Note that as the number of contending flows increases, the probability of simultaneous backoffs reduces, but the probability of control packet

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6The information is the time when a flow should contend for the channel. In IEEE 802.11, virtual carrier sense (NAV time) contains that information.
collision increases. I term this phenomenon of control packet collision as *perceived collision*, as it reduces the amount of information at Flow B about other flows.

In summary, the node with more information gains from its knowledge when it has already acquired the channel but loses when it is deferring to other flows. The phenomenon of perceived collisions assists the nodes with more information to transition from no access to an access state by temporarily removing the information.

### 2.3 Information Sharing and the DWOP Protocol

In this section, I first present a graph-theoretic framework to describe the main mechanisms leading to unfair allocation in 802.11, asymmetric information and perceived collisions. I propose a two-step procedure to convert any topology into a flow-contention graph, using the concept of shared information among contending flows as the central idea. I then present the Distributed Wireless Ordering Protocol (DWOP) to closely approximate a centralized reference service order in ad hoc networks, even with complex topologies involving asymmetric graphs. I demonstrate that in simple topologies that have all nodes within radio range of each other, the protocol achieves perfect service order when all nodes are continuously backlogged. For more complex topologies I characterize the discrepancy between the DWOP schedule and the true global reference schedule in several special cases.

#### 2.3.1 Graph-theoretic Formalism

Here, I formalize the notion of shared information via a simple graph theoretic framework. In particular, the spatially distributed nature of ad hoc networks naturally leads to incomplete information about the other nodes of the network. The problems arising from the spatial separation of nodes can be captured in the following frame-
work, which follows a development similar to [21], but with an important difference to highlight asymmetric node information.

Framework

As the first step in the proposed framework, the geographical map of the nodes is converted into a connectivity graph based on the radio range of different nodes. In the second step, the connectivity graph is converted into a flow-contention graph with the knowledge of transmitter-receiver pair of every flow.\(^7\) For the subsequent development, assume a network of \(N\) nodes, denoted by the set \(\mathcal{N} = \{n_i : i = 1, \ldots, N\}\). Each flow constitutes a transmitter-receiver pair represented by a tuple \((n_i, n_j)\) with \(i \neq j\) and \(n_i, n_j \in \mathcal{N}\). The set of \(K\) flows is represented by \(\mathcal{F} = \{f_k : k = 1, \ldots, K\}\). The two steps in the procedure are formally defined as follows:

![Connectivity graphs for example topologies](image)

(a) Asymmetric information  (b) Perceived-collision

Figure 2.2 : Connectivity graphs for example topologies

1. Connectivity graph \(G = (V, E)\): The set of vertices, \(V\), in the connectivity graph \(G\) represent the nodes in the network, i.e., \(v \in V = \mathcal{N}\). An edge, \(e \in E\),

\(^7\)In this conversion, for simplicity I assume that the carrier sense range is the same as the packet detect range, ignoring the “double ring” effect explored in detail in [31].
exists between vertices \( v_i \) and \( v_j \) if the nodes \( n_i \) and \( n_j \) are within the radio range of each other.\(^8\)

2. **Flow-contention graph** \( G' = (V', E'_s, E'_w) \): The vertices of \( G' \) represent the flows in the network, \( V' = \mathcal{F} \). There is a directed **strong edge** \( e'_s \in E'_s \) from vertex \( v'_i \) to \( v'_j \) if the transmitter of flow \( j \) is in the radio range of at least one of the constituent nodes (transmitter or receiver or both) of flow \( i \). A directed **weak edge** from vertex \( v'_m \) to \( v'_n \) exists if only the receiver of flow \( n \) is in the radio range of at least one of the nodes of flow \( m \). If there is a strong edge between two flows, it immediately implies that there is a strong or weak opposite edge between the same two flows. Strong edges are denoted by solid lines and weak edges by dashed lines.

![Diagram](image)

(a) Asymmetric information  \hspace{2cm}  (b) Perceived collision

**Figure 2.3**: Flow-contention graphs for example topologies

The flow-contention graphs for the asymmetric-information topology (Figure 2.1(a)) and perceived-collision topology (Figure 2.1(b)) are shown in Figures 2.3(a) and 2.3(b), respectively. The connectivity graphs of the example topologies are shown in Figures 2.2(a) and (b) respectively. The flow-contention graph is independent of the

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\(^8\)Two nodes are considered to be within radio range of each other if they can decode each other’s packets reliably.
particular scheduling or media access protocols and is rather a flow-specific representation of a connectivity graph. However, the form of the information and specific use of it is dependent on a particular choice of a protocol. The flow-contention graph can thus be used to classify the flows which are directly affected by asymmetric information and perceived collisions.

Examples

Due to the passive nature of the receiver in IEEE 802.11 (the receiver only sends CTS or ACK, but does not influence the transmitter’s decision regarding when to contend), receiver information regarding timing of contending nodes is not used in medium access protocols. Thus, in 802.11, receiver information is “weaker” than the same information at the transmitter of a flow, which is why transmitter information is labeled as strong. Thus, two flows which have a directed strong edge in one direction and a weak edge in the other direction have asymmetric information about each other. The flow-contention graph of Figure 2.3(a) clearly shows the asymmetric information between Flow A and Flow B, where the shared information is the exact completion of a successful packet transmission.

The flows that have an incoming strong degree of at least two, and an outgoing weak degree of at least one are affected by perceived collisions. Recall that perceived collisions may not affect the flows which are transmitting the packets, but the nodes which are deferring to the active flows. Perceived collisions become more probable as the incoming strong information at a flow increases relative to the node’s outgoing weak information (proportional to number of strong incoming edges), and hence can hurt the nodes with more information about other nodes.

The flow contention graph only captures the information exchange between the
flows. Combined with a specific protocol, which defines the actions taken at each information exchange, the graph-theoretic framework can also be used to study the role of information sharing in other medium-access protocols.

2.3.2 DWOP: Distributed Wireless Ordering Protocol

In this section, I present DWOP, a protocol that approximates a reference scheduler in wireless ad hoc networks by exploiting overheard information from other nodes to estimate when to contend for the channel. I first describe how priority indexes (packet arrival times) can be communicated via piggybacking so that nodes can build local scheduling tables based on overheard information. I then describe how to exploit the piggybacked information to obtain a reference (FIFO) schedule within the framework of IEEE 802.11 for a topology where all nodes are within radio range of each other. I then show that a receiver's scheduling table information can be effectively used in more complex topologies, via receiver out-of-order notification which reduces the information disparity among nodes. Finally, since the local scheduling table can potentially have stale entries due to perceived collisions, mobility and channel errors, I propose a distributed stale entry detection method enabling a quick recovery to steady state.

Distributing Arrival Times

A FIFO schedule is realized by servicing packets in order of their arrival times to the network. To achieve a distributed FIFO schedule among multiple nodes in an ad hoc network, I communicate the arrival times of queued packets to other nodes via piggy-backing. Analogous to [32], all nodes will maintain a scheduling table based on overheard information to assess whether or not they possess the next packet for
Node 7 (destination neighbor) Scheduling Table Updates

Figure 2.4: Piggybacking on IEEE 802.11 four-way handshake, and the updating of scheduling tables.
service in the distributed FIFO scheduler. In particular, as illustrated in Figure 2.4, each packet has an associated arrival time. When a node issues an RTS message, it piggybacks the arrival time of its current packet. Nodes that overhear this RTS insert an entry into a local scheduling table. When the receiving node grants a CTS, it also appends the arrival time in the CTS frame to allow the hidden nodes (node 7 in Figure 2.4) which are unable to hear the RTS to add an entry in their scheduling tables upon hearing the CTS. Next, when the node transmits the DATA packet, it piggybacks the arrival time of its head-of-line (highest priority) packet, not including the one in transmission, which is also inserted in the local table by overhearing nodes.

With this information, each node can assess the priority of its own head-of-line packet in relation to its (necessarily partial) list of other head-of-line packets. In this way, nodes have the potential to approximate a "global" FIFO reference schedule in a distributed way.

Using Shared Information

The key idea for DWOP is that each node should contend for the channel only when it has the packet with the lowest arrival time (highest priority) among all the nodes within its radio range. The proposed design philosophy is in contrast with that of IEEE 802.11, in which nodes contend for the medium without consideration or knowledge of the arrival times of queued packets at other nodes. The operation of the DWOP protocol within the framework of IEEE 802.11's backoff policy is depicted in Figure 2.5. When a node has a packet to transmit, it checks its local scheduling to make a decision regarding if it should contend for the channel. If the node determines (perhaps incorrectly) that its locally queued packet is the highest priority packet in the region (i.e., the node's packet has an arrival time less than all entries in the
scheduling table), then the node will contend for the channel as it would in IEEE 802.11. On the other hand, if the node’s HOL packet is lower in priority than an entry in its local scheduling table, the node will back off, thereby deferring access to the higher priority packet.

For more complex topologies in which all nodes are not within radio range of each other, asymmetry of information between nodes (Section 2.2) will prevent DWOP from achieving a perfect FIFO transmission order since not all transmitting nodes are aware of all other nodes’ packet arrival times. Consider again the topology in Figure 2.1(a). In Section 2.2, it was shown how information asymmetry in IEEE 802.11 causes Flow B to have a larger share of throughput than Flow A. For DWOP, the effect of information asymmetry on the throughput share for each flow is reversed;
Flow A achieves a higher share compared to Flow B. The transmitter of Flow A has no knowledge about arrival times of packets queued at the transmitter of Flow B and thus always infers that it has the highest priority packet in the system. Therefore the channel access mechanism for Flow A defaults to IEEE 802.11 and it attempts to gain channel access continuously.

On the other hand, the transmitter of Flow B is aware of Flow A packet arrival times through the receiver of Flow A and thus defers access whenever there is a higher priority packet queued at the transmitter of Flow A. Thus, Flow B is less aggressive in channel access, and in case of the same arrival pattern as at Flow A, always defers channel access after one successful packet transmission. In this case, information asymmetry may cause Flow A to obtain a higher share of bandwidth than Flow B. Note that the receiver of Flow A is aware of packet priorities of both the flows in the system, and can thus “forward” the information about Flow B to the transmitter of Flow A. I use the above observation to exploit receiver scheduling table information with an aim to ameliorate information asymmetry.

**Receiver Participation**

To address the problems introduced by information asymmetry inherent to ad hoc networks, I propose an *out-of-order notification* piggybacked on a control packet sent by the receiver to the transmitter, on every FIFO violation according to the receiver’s scheduling table. In [25], it was shown that two-hop information, if available, is sufficient to achieve perfect FIFO and eliminate all contention. In essence, the proposed receiver participation technique passes information that is two-hops

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9This is in contrast to IEEE 802.11 where the receiver plays more of a passive role, replying to only packets sent by the transmitter.
away from its transmitter, but does so only when needed to avoid the overhead of propagating topology information. Since the notification is sent only when needed, the proposed technique cannot ensure perfect global FIFO rather provides a close approximation to FIFO.

On receiving an RTS for a packet that is out-of-order with respect to the receiver's local scheduling table, the receiver can send back a feedback message to the transmitter informing the transmitter of its actual rank with respect to the receiver's local scheduling table. The transmitter, on reception of such an out-of-order notification from the receiver, goes into a backoff after completing the current packet transmission for a time, given by

\[ T_{\text{backoff}} = R(EIFS + DIFS + T_{\text{success}} + CW_{\text{min}}) \]

where \( R \) is the rank of transmitter in the receiver scheduling table, and \( T_{\text{success}} \) is the longest possible time required to transmit a data packet successfully including handshake (RTS+CTS+DATA+ACK).

The purpose of \( T_{\text{backoff}} \) is to allow the higher priority packets in the radio range of the receiver to complete transmission. To ensure perfect FIFO, an alternate mechanism to achieve this would be to have the receiver not reply to any RTS that carries a priority tag larger than the smallest entry in the receiver's scheduling table. This would effectively force the transmitter to timeout and backoff, thus avoiding any out of order packet transmission. However, since the transmitter has already expended system resource while transmitting the RTS successfully for the out-of-order packet, the present transmission is allowed to complete and the receiver piggybacks the out-

\[ ^{10} \text{Since the transmitter chooses to send an RTS, this implies that the packet has a higher priority than the highest priority packet in the transmitter's scheduling table.} \]
of-order notification on the CTS/ACK. Hence, the transmitter reacts to the out of
order notification after the completion of the out-of-order packet. This is a tradeoff
between achieving perfect packet ordering and high system utilization. As I show in
Section 2.5 the deviation from the ideal schedule caused by allowing an out-of-order
packet to proceed is small.

In DWOP, nodes access the channel solely based on their rank in their own
scheduling table or in the receiver’s scheduling table. Thus, the performance of the
protocol depends critically on maintaining the consistency of the scheduling tables. I
next show how perceived collisions can cause stale entries in the scheduling table
and present a novel stale entry detection method.

Perceived Collisions and Stale Entry Detection

Recall that an overhearing node adds entries to its table upon hearing RTS/CTS
and removes entries when it hears the successful completion of a packet through
DATA/ACK. Thus, the only reason a table can have stale entries is if after hearing
an RTS/CTS a node fails to hear the succeeding ACK. In the absence of channel
errors and mobility, the reason a node would not hear an ACK after hearing an RTS
is because of a collision at that node. This can happen in case the other colliding
node was not aware of the previous RTS and thus was not deferring access. This
scenario can happen in graphs characterized by perceived collisions as described in
Sections II and III.

The effect of inconsistent tables is a possible large deviation from the ideal FIFO
schedule if the inconsistency is not corrected. In the worst case, stale entries could
lead a node to completely stop transmitting, as it defers access to the stale entry in
its table.
Figure 2.1(b) depicts a topology where stale entries can occur, where both transmitter and receiver of Flow B can have stale entries. Observe that in this case, Flows A and C have no stale entries and continue transmitting. This causes the local scheduling tables at Flow B to continuously update its table by adding and deleting new entries although the position of its HOL packet remains fixed. This is because the new additions/deletions for Flows A and C occur below the position occupied by the HOL packet of Flow B.

Thus, I observe that an indicator of stale table entries is when a node's own packet position remains fixed with entries below the HOL packet entry changing.\textsuperscript{11} I use the above observation as a stale entry detection method to estimate when a given node may have stale entries in its local scheduling table. Thus, when a node observes that its position remains fixed although packets with priority below its HOL packet are being transmitted, it immediately concludes that it has one or more stale entries in its table. In this way each node can independently identify the existence of one or more stale entries in its local scheduling table.

To remove the stale entry from the table after detection, I propose a simple heuristic that a node simply delete the oldest (smallest arrival time) entry assuming that this was the table's stale entry. As I later confirm through experiments in Section 2.5, the oldest entry is actually the stale entry in most cases. If it is not, it will be detected and removed in subsequent transmissions via the same mechanism, thus ensuring eventual removal of all stale entries so that the flow will eventually be ranked one in its table and resume contention for the channel.

Thus in summary, DWOP is characterized by the following. (1) Sharing informa-

\textsuperscript{11}A node's position could also be fixed with entries above the position of the HOL packet changing in the normal (no stale entry) case.
tion using a piggyback mechanism. (2) Introduction of determinism in IEEE 802.11 so that a node contends for the channel only when the arrival time of its HOL packet is the smallest among all the local scheduling table entries. (3) Combating asymmetry in information sharing through the use of active receivers which notify the transmitter of out-of-order packets. (4) Detection of stale entries (via observing the way scheduling table entries change) and acting upon stale entries by removing the oldest entries first. With these four mechanisms, DWOP closely approximates a reference FIFO schedule in wireless ad hoc networks, even with information asymmetry.

Spatial Reuse

The correctness of a perfect schedule in DWOP relies on the local scheduling table at each node and how aggressively the node uses this table for medium access. In the limiting case when a node is maximally aggressive in medium access and chooses to ignore its scheduling table altogether, DWOP reduces to IEEE 802.11. On the other end of the spectrum, if a node makes full use of its scheduling table, a close approximation to a perfect schedule is achieved at the cost of spatial reuse which can lead to a lower system-wide throughput in certain topologies.

I provide a mechanism to control the above tradeoff via introducing a Time-to-Live (TTL) parameter for each table entry. The TTL parameter is set at the time a node inserts an entry in its table and upon the expiration of this timer, the particular entry is deleted from the node's scheduling table. Thus while making the medium access decision, a node relies upon its table only for a limited time. Moreover, observe that by adaptively setting the TTL parameter to the contention level in a flow's neighborhood a node's aggressiveness can be tuned which in turn controls the deviation from a perfect schedule to allow spatial reuse. Note that the
size of each node's local scheduling table is an indicator of the number of contending flows. Hence, it is possible to achieve a controlled use of the scheduling information about other nodes by setting the TTL value for each entry to be $T_{\text{backoff}}$ (as given by Equation (2.1)) scaled by the size of the local scheduling table at the time the entry is inserted.

This is explored through simulations (presented in Section 2.5), where by varying the scaling factor (based on the table size) it is possible to a limit the deviation from an exact FIFO schedule and increase system throughput.$^{12}$

In Sections 2.3.3 and Section 2.5, I confirm that the four mechanisms introduced in Sections 2.3.2 allow DWOP to closely approximate a FIFO schedule.

### 2.3.3 Protocol Analysis

In practical systems, nodes typically have no initial knowledge of the system. In Section 2.4, it is shown that DWOP converges to the information steady state where all nodes have heard from their neighboring nodes.

In this section, I will study the protocol FIFO behavior in information steady state. To proceed, the following two definitions are required.

**Definition 1 (Information steady state)** The system is in steady state when it has information about all contending flows in its scheduling table, where the number of entries is equal to incoming strong degree for the transmitters and incoming weak degree for the receivers.

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$^{12}$An additional degree of spatial reuse can be achieved via use of modified backoff policies such as those proposed in [18].
Definition 2 (Tandem packet capture) Assume that all nodes transmit equal size packets of duration $T_{\text{length}}$ seconds. Further assume that all nodes are continuously backlogged. Let $\{t_1, t_2, \ldots, t_k\}$ represent the time of channel capture by an arbitrary subset of $k$ nodes in the system, such that $t_1 \leq t_2 \leq \cdots \leq t_k$. The channel capture by the $k$ nodes is said to be in tandem if $t_1 + 2T_{\text{length}} \leq t_k - CW_{\text{min}}$.

The following proposition shows that in a simple fully connected topology, DWOP achieves a perfect FIFO schedule in steady state.

Proposition 1 (Perfect FIFO case) If all nodes are in within radio range of each other and are continuously backlogged, then the proposed protocol achieves perfect FIFO in the steady state.

Proof In steady state with constant backlogged flows, each node is aware of all HOL packets of every other flow. This complete information implies that every node is aware of their global position, and hence never contends if it is not rank one. Since there are no collisions, the top ranked node always acquires the channel, implying perfect FIFO.

The above proposition is the motivation for preserving determinism in the protocol. In the presence of correct information, the protocol minimizes out of order transmissions. When the nodes are not within radio range of each other, perfect FIFO is not achievable in all cases.

The following result partially characterizes the performance of DWOP for general topologies.

Theorem 1 (FIFO Bound) Consider an arbitrary network of nodes with each
node transmitting equal size packets\textsuperscript{13} under constant backlog. Further assume that none of the nodes have any stale entries. In the information steady state, the transmission delay for a flow $f$ with the highest priority packet is characterized as follows:

1. If flow $f$ has no outgoing weak edges, it will transmit without any out-of-order transmissions before it.

2. If flow $f$ has at least one outgoing weak edge and the tandem packet capture condition is not satisfied for the neighboring nodes with asymmetric connection, then $f$ will have to wait for no more than one out-of-order packet transmission.

\textbf{Proof} From the hypothesis, stale entry detection is never triggered to delay the top ranked packet.

1. For flow $f$ with no weak outgoing edge, all its outgoing edges have to be strong. Thus, none of the contending flows will attempt to capture the channel out of order, which will allow $n$ to transmit immediately.

2. Let $\mathcal{A}$ denote the set of flows with weak incoming edge from flow $f$ with the highest priority packet. Due to receiver out-of-order notification, all competing flows in set $\mathcal{A}$ will backoff for at least one packet duration. Thus, if the tandem packet condition is not satisfied for flows in $\mathcal{A}$, then after successful transmissions by them, flow $f$ will transmit. \textsuperscript{14}

If the tandem packet capture condition is satisfied for the neighboring nodes with asymmetric connectivity, then the node may have to wait more than two packet

\textsuperscript{13}The analysis easily extends to unequal size packets with a modification to tandem packet capture condition

\textsuperscript{14}In the worst case, $t_k - t_1$ is approximately equal to one packet transmission.
transmissions but will eventually send the packet with probability increasing (due to random choice of backoff counters by each node) with time.

In the presence of stale entries, the delay in transmission of the highest priority packet depends on the time it takes to detect the stale entry. Furthermore, in the absence of the backlogging condition for some of the contending nodes, the time for successful transmission can deviate from the bounds in Theorem 1. The impact of traffic variations will be analyzed in the next section.

2.4 System Dynamics and Transient Behavior of Information Sharing

In realistic scenarios, each node will not have perfect information about other nodes’ head-of-line packets. For example, when a mobile node moves to within radio range of a new set of users, the new node does not have information about the others’ HOL packets and vice versa. Similar conditions occur when a new user enters the system and when a sleeping node wakes up. In such cases, there is a transient time until the new node overhears sufficient information for the system to reach the information steady state, in which all nodes have information about all other backlogged nodes within their radio range.

In this section, I study this transition time for two cases. For the transient case, I compute the time to reach the steady state for a scenario in which no node has information about any other node. For the perturbation case, I compute the time to return to the steady state when a new node enters a system consisting of a number of nodes that had previously reached the steady state. In both cases, our analytical model indicates that the system converges significantly faster than the time required
for an analogous IEEE 802.11 system to allow each user to transmit.\footnote{Each user must transmit at least one packet to reach the steady state.}

2.4.1 Model Description

In the following analysis, I consider a region of $m$ mobile nodes, all within radio range of each other, such that each node has at least two packets in its queue. Hence each transmitted packet always carries information of the next HOL packet in the node. Also in order to simplify the analysis, I assume that the values of random timer backoffs of all nodes are independent and uniformly distributed in the range of $[0,C_w]$ and ignore collisions.

2.4.2 Relationship Between Backoff Timers

Here, I compute two distributions regarding the ordering of backoff timers, which play an important role in system dynamics. Let $\xi_i, i = 1, \cdots, m$ denote the backoff timers chosen by the $m$ nodes. Rearranging $\xi_i, i = 1, \cdots, m$ in increasing order
\( \xi_{i_1} \leq \xi_{i_2} \leq \cdots \leq \xi_{i_m} \) (see Figure 2.6). According to [33], the probability distribution of the random variable \( R_{k_1, k_2}^m = \xi_{i_{k_2}} - \xi_{i_{k_1}} \) \( (k_2 > k_1) \) is given by

\[
P[R_{k_1, k_2}^m = r] = \frac{m!r^{k_2-k_1-1}C_{w-r}^{m-k_2+k_1}}{(k_2-k_1-1)!(m-k_2+k_1)!C_w^m}.
\]

(2.2)

Denote \( Z_h^m \) as the probability that node \( i_1 \) consecutively transmits \( h \) packets before node \( i_2 \) transmits a packet. This probability is equivalent to the probability that node \( i_1 \), after sending out its first packet, consecutively picks up \( h \) backoff timers \( \chi_{i_1}^1, \cdots, \chi_{i_1}^h \) such that \( \sum_{k=1}^{h-1} \chi_{i_1}^k \leq R_{1,2}^m \) and \( \sum_{k=1}^{h} \chi_{i_1}^k > R_{1,2}^m \) (see Figure 2.7). Thus,

\[
Z_h^m = \sum_{r=0}^{C_w} P[\sum_{k=1}^{h-1} \chi_{i_1}^k \leq R_{1,2}^m, \sum_{k=1}^{h} \chi_{i_1}^k > R_{1,2}^m | R_{1,2}^m = r] P[R_{1,2}^m = r].
\]

(2.3)

Let \( A_h \) denote a random variable given by the sum of node \( i_1 \)'s first \( h - 1 \) backoff timers, i.e., \( A_h = \{ \sum_{k=1}^{h-1} \chi_{i_1}^k \leq r \} \) and \( \bar{A}_h = \{ \sum_{k=1}^{h-1} \chi_{i_1}^k > r \} \), then

\[
P[\sum_{k=1}^{h-1} \chi_{i_1}^k \leq r, \sum_{k=1}^{h} \chi_{i_1}^k > r] = P[A_h \bar{A}_{h+1}].
\]
Since $\chi_{i_1}^k, k = 1, 2, \cdots, h$ are independent and identically distributed uniform random variables in the range $[0, C_w]$, I have

$$P[A_h] = \frac{1}{C_w^{h-1}} \int_0^r \int_0^{r-x_1} \cdots \int_0^{r-\sum_{k=1}^{h-2} x_k} dx_{h-1} \cdots dx_2 dx_1$$

$$= \frac{1}{C_w^{h-1} (h-1)!} \cdot r^{h-1}$$

(2.4)

Furthermore, since $\sum_{k=1}^{h-1} \chi_{i_1}^k \leq \sum_{k=1}^{h} \chi_{i_1}^k$, therefore $A_{h+1} \subseteq A_h$, so that

$$P[A_h \bar{A}_{h+1}] = P[A_h - A_{h+1}] = \frac{r^{h-1}}{C_w^h (r-1)!} \left[1 - \frac{r}{C_w h}\right]$$

(2.5)

Substituting Equations (2.5) and (2.2) into Equation (2.3)

$$Z_m^h = \sum_{r=0}^{C_w} \frac{1}{C_w^{h-1} (h-1)!} \cdot \frac{1}{C_w h!} \frac{r^h}{m(C_w - r)^{m-1}}$$

(2.6)

### 2.4.3 Transient Behavior from the Initial State

Here, I use the above distributions to study the system’s transient behavior from an initial state in which all nodes have no information about the HOL packets of any other nodes. With information sharing, the system will evolve from this initial state into the steady state after each node transmits at least one packet.

To address this issue, I compute the probability that the system enters the steady state after $n$ packet transmissions. Let $S(n, m)$ denote the event that a system with $m$ nodes is in the steady state after $n$ packet transmissions. Note that $P[S(n, m)] = 0$ if $n < m$, and $P[S(n, m)] = 1$ if $m = 1$. Furthermore, after sending out their first packet, nodes $i_1$ and node $i_2$ will coordinate with each other for channel access since each knows the priority index of the other’s next packet. From the performance analysis point of view, after node $i_2$ sends out its first packet, nodes $i_1$ and $i_2$ can be treated as a single “virtual node” and the system can be treated as consisting of $m - 1$ nodes still in the initial state. Therefore, Figure 2.8 charts the evolution of
the system from the initial state to the steady state. Thus, using the state diagram of Figure 2.8

\[ P[S(n, m)] = \sum_{h=1}^{n-m+1} Z^h_m P[S(n-h, m-1)]. \]  \hspace{1cm} (2.7)

where \( Z^h_m \) is given in Equation (2.6). The same procedure yields

\[ P[S(n-i, m-k)] = \sum_{h=1}^{n-i-m+k+1} Z^h_{m-k} P[S(n-i-h, m-k-1)], \]  \hspace{1cm} (2.8)

for \( k = 1, 2, \cdots, m - 1 \), where

\[ Z^h_{m-k} = \sum_{r=0}^{C_w} \frac{r^{h-1}}{C_w^{h-1}(h-1)!} - \frac{r^h}{C_w^h h!} \frac{(m-k)(C_w-r)^{m-k-1}}{C_m^{m-k}}. \]

Therefore, \( P[S(n, m)] \) can be computed using Equations (2.6)-(2.9). I now present numerical and \( ns-2 \) simulation investigations of the system’s time to reach the steady state. Figure 2.9 depicts the transition period’s probability distributions (in units of packets) for \( m = 4 \) and \( 8 \) nodes. I consider \( C_w = 32 \), the same as the minimum
Figure 2.9: Probability Distribution of Transition Duration
contention window size given in [7]. In order to highlight the impact of information sharing on the transient process, I also present the distributions of the duration for each node to send out at least one packet in the standard IEEE 802.11 MAC protocol.

I make the following observations regarding the figure. First, after $2m$ packet transmissions, the probability for the system with information sharing to enter the steady state is more than 0.95. For example, in Figure 2.9(a), after 8 packets are transmitted in a system with 4 flows (4 source nodes and 4 destination nodes), the probability for the system to enter the steady state is 0.97 from the model's prediction and 0.98 from simulation. Second, observe that this duration is significantly less than the time required for the IEEE 802.11 MAC protocol to reach a state in which every node has transmitted at least one packet. For example, for this to occur in the system with 4 flows with probability 0.85, 5 packets must be transmitted with information sharing, whereas 9 packets must be transmitted in IEEE 802.11. Thus, information sharing accelerates the system into an steady state in which each node has transmitted at least one packet.

2.4.4 Transient Behavior from the Perturbation State

Mobility, sleeping nodes awakening, etc., will perturb the system from its steady state and cause multiple nodes to contend for the channel simultaneously. To evaluate the distribution of the duration for the system to return to the steady state, I treat the existing $m$ nodes as a virtual node denoted by $A$, since there exists an order among these $m$ nodes such that only one contends for the channel with with new nodes. To simplify analysis, I consider a single new node indexed by $m + 1$. Without loss of generality, I assume that the priority of the $k^{th}$ packet of virtual node $A$ is higher
than the priority of the first packet of node $m + 1$ and the priority of the $(k + 1)^{th}$ packet of node $A$ is lower than the priority of the packet of node $m + 1$. I also assume that node $m + 1$ will know if node $A$ has another packet with higher priority than itself packets after node $A$ transmits a packet. This situation will occur if each original node has at least two packets with higher priorities than node $m + 1$ packets or every node piggybacks the information about the packet with the second highest priority in its scheduling table as well as the information of its next packet. Under

![Graph](image)

**Figure 2.10**: Probability Distribution of Perturbation Period Length

this assumption, the probability that the virtual node $A$ transmits $n - 1$ packets before node $m + 1$ transmits its first packet is given by $0.5Z_{2,1}^{n-k-1}$. With probability 0.5, virtual node $A$ obtains a backoff timer smaller than that of node $m + 1$ and sends out $k$ consecutive packets. This will occur because the new node $m + 1$ learns that virtual node $A$ has packets with higher priorities and waits until virtual node
A sends out the $k^{th}$ packet.

With probability $Z_2^{n-k-1}$, as defined in Equation (2.3), virtual node $A$ will consecutively send out $n - k - 1$ packets with lower priorities than the first packet of node $m + 1$ before the perturbed system returns to the steady state. Similarly, the probability that node $m + 1$ continuously transmits $n - 1$ packets before the perturbed system returns to the steady state is given by $0.5Z_2^{n-1}$. Specifically, if $n \leq k$ and virtual node $A$ first captures channel, then it is impossible for the system to return to the steady state after $n$ packets having been transmitted. This is because virtual node $A$ will continuously transmit $k$ packets before node $m + 1$ transmits its first packet.

Let $P(n, k)$ denote the probability that the system returns to the steady state after $n$ packets having been transmitted. According to above analysis

$$P(n, k) = \begin{cases} 0.5Z_2^{n-1}, & n \leq k, \\ 0.5Z_2^{n-k-1} + 0.5Z_2^{n-1}, & n > k. \end{cases}$$ (2.10)

Finally, in Figure 2.10 I present numerical and simulation investigations on the duration required for the perturbed system to return to the steady state. I consider a system with 8 nodes (4 source nodes and 4 destination nodes) in the steady state and 2 nodes (one source node and one destination node) joining the system. I further consider that there are initially 4 packets in the original nodes with higher priority than the first packet of the new node.

I make the following observations about the figure. First, note that information sharing allows the system to rapidly return to the steady state, as compared to waiting for each node to transmit in IEEE 802.11. Second, note that there is an inflection point in the distribution for information sharing occurring at 4 packet durations. This is corresponds to the number of packets in the original nodes with
higher priority than the first packet of new nodes. The reason for this inflection is that when one of the original nodes captures channel, there will be 4 packets transmitted before the new source node transmits its first packet and the perturbed system cannot return to the steady state because the original nodes do not have information about this node's HOL packet. Therefore, the probability for the system to return to the steady state after 4 packet transmissions depends only on the case in which node $m + 1$ captures the channel first and continuously transmits 3 packets before one of the original nodes sends out its packet. Thus, the distribution duration for the system to return to the steady state increases slowly before reaching 4 packet transmission times quickly thereafter.

### 2.5 Simulation Experiments

In this section I present simulation results to compare the proposed DWOP protocol with IEEE 802.11 with FIFO as the reference scheduler. The simulations were performed using the CMU Monarch wireless extensions to the ns-2 simulator.

I consider three topologies first without channel errors and then with a realistic channel model. The data packet size is set to 1000 bytes, while the data capacity of the wireless channel is 2 Mb/sec. For input traffic, I use constant-rate flows for each node with jittered inter-arrival times. To simulate heavy loads, I set the input rate for each flow to be high enough to individually saturate the channel. I plot the average of 5 random runs of 50 seconds each for each test case. All other physical layer parameters were set to the default parameter values in ns-2. All flows are single hop, but all topologies consist of nodes which are out of radio range of at least one node in the network. Further, two sets of simulations are performed for each topology, with carrier sense threshold the same as the data threshold, and
with the carrier sense threshold smaller than the data threshold (default values from CMU extensions were used in this case). The results for both values of carrier sense thresholds were found to be similar, and hence I primarily present them for the case where carrier sense threshold is equal to data threshold, to highlight the role of the graph-theoretic representation.

2.5.1 Asymmetric Information Topology

Here I present results for the topology shown in Figure 2.1(a). The simulation results are shown in Figure 2.11(a), which compares the throughput of IEEE 802.11 with DWOP. For IEEE 802.11, asymmetry of information helps Flow B to obtain 95% of total throughput whereas Flow A obtains only about 5% (the share is 70-30% with a smaller carrier-sense threshold). For DWOP, both flows have an equal share of throughput. For experiments with the carrier sense smaller than the data threshold, the total throughput is nearly identical to that of IEEE 802.11, whereas for the depicted case with identical thresholds, the total throughput of DWOP is approximately two-thirds that of IEEE 802.11.

Figure 2.11(b) shows the distribution of the number of consecutive packets sent by any flow before it relinquishes the channel to the other flow. For FIFO with constant-rate arrival patterns and identical rates, flows access the channel in a round-robin manner. Thus, in an ideal FIFO system no flow should keep the channel for more than one packet. In Figure 2.11(b), for IEEE 802.11, the distribution is spread out with a single flow (Flow B) keeping the channel for a large number of consecutive packets (the maximum number of consecutive packets transmitted by Flow B is 129). For DWOP, a flow never transmits more than 3 consecutive packets. Thus I see that the DWOP approximates the ideal FIFO schedule significantly more closely than
(a) Throughput Comparison

(b) Packet Order Distribution

Figure 2.11: Comparison for DWOP and IEEE 802.11 for Asymmetric Topology
IEEE 802.11.

Figure 2.12 depicts the number of packets sent by each flow sampled at 1 second intervals. For DWOP, both flows have equal throughput at this time scale. On the other hand, with IEEE 802.11 Flow B starves out Flow A consistently over the entire duration of the simulation.

![Figure 2.12: Bandwidth Share for Asymmetric Topology](image)

2.5.2 Perceived Collision Topology

In this section, I consider the topology considered in Figure 2.1(b), to study the impact of perceived collisions on the throughput share of Flow B and the extent of resulting deviation from reference FIFO.
Figure 2.13(a) compares the throughput of IEEE 802.11 with DWOP. For IEEE 802.11, Flow B obtains a smaller throughput share whereas Flows A and C approximately divide the rest of the share equally. The reasons for unequal share were discussed in Section 2.2.4.

However DWOP allows nearly equal share of the net throughput for all three flows, although with a net throughput of three-fourths (86% with a smaller carrier sense threshold) that of IEEE 802.11. Figure 2.13(b) shows the distribution of consecutive packets sent by a flow. As shown, IEEE 802.11 violates FIFO with a single flow keeping the channel for a maximum of 77 packets, while DWOP closely approximates FIFO with no flow keeping the channel for more than 4 consecutive packets, thereby confirming the efficacy of proposed stale entry detection mechanism.

2.5.3 10-Node Topology

In this section, I present the results for a more complex 10-node topology shown in Figure 2.14; in this topology, both Flows B and C can have stale entries. Figure 2.15(a) compares the throughput obtained by IEEE 802.11 and DWOP. Observe that Flow C which has the maximum strong information, obtains the lowest throughput in IEEE 802.11. Flows B, D and E practically divide the bandwidth among themselves due to asymmetric information and spatial reuse. However, DWOP results in a near equal share of the net throughput being allocated to all the flows, albeit at a total throughput loss. The total throughput of DWOP is three-fifths (two-third with lower carrier sense threshold) that of IEEE 802.11.

Figure 2.15(b) depicts the distribution of consecutive packets sent by a flow. Observe that IEEE 802.11 violates FIFO ordering with a single flow keeping the channel for a maximum of 8 packets. However, this plot coupled with 2.14(a) shows
Figure 2.13: Comparison of DWOP and IEEE 802.11 for Perceived Collision Topology
that it is in fact Flow B, D and E that keep relinquishing the channel to each other.
In contrast, DWOP closely approximates FIFO with no flow keeping the channel for more than 2 consecutive packets.

2.5.4 Spatial Reuse

Here I present results showing the effect of the scaled Time-To-Live (TTL) parameter on net throughput. Recall from Section 2.3.2 that by setting the TTL parameter adaptively (based upon the local scheduling table size) it is possible to obtain a throughput gain via spatial reuse at the cost of deviation from a perfect FIFO schedule. Figure 2.16 illustrates this tradeoff. To quantify the loss in a perfect FIFO schedule I introduce the FIFO-index which is the difference between the maximum and the minimum throughput obtained by any flow for a given topology, normalized with respect to the maximum value. In particular the FIFO-index, $\rho$, is given by:

$$\rho = \frac{\text{Throughput}_{\text{max}} - \text{Throughput}_{\text{min}}}{\text{Throughput}_{\text{max}}}$$  \hspace{1cm} (2.11)$$

where $\text{Throughput}_{\text{max}}$ and $\text{Throughput}_{\text{min}}$ denote the maximum and the minimum
(a) Throughput Comparison

(b) Packet Order Distribution

Figure 2.15 : Comparison of DWOP and IEEE 802.11 for the 10-node Topology
throughput respectively. For a perfect FIFO schedule and identical flow input rates, all flows will have the same throughput so that the FIFO-index is zero. Thus, the larger the FIFO-index value, the greater the deviation from a perfect schedule.

Figure 2.16 depicts the throughput loss and corresponding fairness index value for different values of the factor by which I multiply the scheduling table size to obtain the TTL value (as discussed in Section 2.3.2). The value of FIFO-index is scaled by 100 and the loss in throughput (in percentage) compared to throughput of IEEE 802.11. The values represent the 90 percentile value for 50 randomly generated topologies. The number of flows in each topology is a random number uniformly distributed between 5 and 10. The sender node of each flow is placed with independent uniformly random horizontal and vertical coordinates on a 2-D grid 2500 meters by
2500 meters. The destination node is placed randomly in a circle with the sender node as the center and radius equal to half the radio range of each node.\footnote{The radio range with default parameters in ns-2 is 250 meters.} In all topologies it is ensured that either the sender node or the destination node of each flow is within the radio range of at least one other flow, thus resulting in connected topologies.

The FIFO-index for IEEE 802.11 is around .98 showing extreme deviation from a perfect FIFO schedule. As I increase the value of TTL, the FIFO-index decreases indicating a reduced deviation from global FIFO. However this is accompanied by a loss in net system throughput as compared to IEEE 802.11.

## 2.6 Related Work

Distributed scheduling and media access to achieve fair bandwidth allocation in ad hoc wireless networks has been an intensive research topic in recent years, e.g., [18, 19, 20, 21, 22, 23, 24]. By exploiting the broadcast nature of the wireless medium, all of these schemes use some form of information sharing to allow distributed nodes to cooperate with each other to achieve a desired global behavior. For example, with passive information sharing (i.e., using measured information about channel idle times, collisions, etc.), the authors of [21] devise a distributed dynamic p-persistent MAC protocol designed to achieve proportional fairness. Using active information sharing (i.e., piggybacking), the authors of [24] devise a scheme to emulate Self-Clocked Fair Queueing (see also [34]) by piggybacking local virtual times and adjusting IEEE 802.11 backoff policies accordingly. Finally, the authors in [18] introduce three localized fair queueing models within the framework of the CSMA/CA paradigm to let distributed nodes to emulate Start-Time Fair Queueing
(see also [35]) and achieve global weighted fairness in ad hoc networks. However, none of the proposed approaches target an ordering mechanism to approximate any given centralized reference scheduler in order to achieve fairness, QoS differentiation or to meet throughput and delay targets. Further, the Enhanced Distributed Coordination Function (EDCF) [26] (an extension to the IEEE 802.11 DCF) and similar approaches [27, 28] have been proposed to provide class based service differentiation within the IEEE 802.11 MAC framework. However, such approaches only provide statistical priority and does not guarantee that low priority packets will always wait until all higher priority packets have been transmitted.

In contrast, our objective is to provide the reference scheduler service order at the MAC layer rather than per-flow or per-node fairness. This objective is shared with [36] as well as a degenerate case of the distributed priority scheduler in [32, 37]. In contrast to [36], I consider complex topologies in which complete information is not available, and provide a graph-theoretic framework and protocol to address these topologies. In contrast to [32], I target a deterministic behavior so that near-exact desired service ordering is achieved in most cases, whereas [32] focuses on meeting delay and rate targets. Thus, in [32], the precise service ordering is not a focus, so long as the quality-of-service targets are satisfied.

2.7 Summary

The goal of the work presented in this chapter is to design a distributed media access and scheduling algorithm to achieve desired service order in wireless ad hoc networks. Choosing FIFO as an example target scheduler, I showed that the IEEE 802.11 protocol diverges significantly from FIFO order, even starving nodes in many cases, due to asymmetric information sharing and "perceived collisions". Within this context I
presented Distributed Wireless Ordering Protocol (DWOP), a distributed media access and scheduling protocol to achieve perfect ordering in wireless ad hoc networks. I illustrated via simulations and theoretical analysis that DWOP exploits information sharing to achieve nearly perfect service order, even in complex topologies with incomplete information, and in dynamic scenarios beginning with no information.
Chapter 3

Multi-channel Opportunistic Medium Access Control

3.1 Introduction

The transmitted signal in a wireless network usually reaches the receiver via multiple propagation paths. These paths change with time due to reflectors in the environment and/or mobility. The changing strength of each path and the changing interference between these paths induces channel fading which is a fundamental trait of the wireless channel. Traditionally, channel fading has been viewed as a source of unreliability which has to be mitigated. However recent advances in wireless communications theory suggests an alternate view. Channel fluctuations can be exploited by transmitting information opportunistically when and where the channel is strong [38, 39, 40, 41, 42, 43, 44].

Nearly all the literature on opportunistic wireless communication has focused on exploiting multi-user diversity which has its roots in the work of Knopp and Humblet [45]. When many users are present, different users will experience peaks in their channels quality at different times. This effect is called multi-user diversity and can be exploited by scheduling transmissions when a user has favorable channel conditions. However, the presence of multiple frequency channels in such systems like IEEE 802.11 enabled wireless networks is a source of a different form of diversity which too can be exploited opportunistically to enhance the throughput of wireless
networks. In particular, if the channel conditions on a current frequency channel are not favorable, mobile nodes can skip to a better quality frequency channel enabling data transmission at a higher rate. In this way it is possible to increase the throughput of wireless networks by skipping frequency channels opportunistically. There is little previous work on opportunistically exploiting frequency diversity to enhance the throughput of wireless networks (the related work is discussed in detail in Section 3.3). Moreover, for a wireless ad hoc network with no central controlling entity, realizing the throughput gains available via opportunistic skipping of frequency channels introduces severe design challenges not incurred in centralized cellular systems.

The contributions of this chapter is the design and evaluation of an efficient opportunistic channel skipping protocol for wireless ad hoc networks which coordinates the channel skip decision among the mobile nodes in a decentralized manner. In particular, I develop the *Multi-channel Opportunistic Auto Rate (MOAR)* protocol, an enhanced MAC protocol for multi-channel and multi-rate IEEE 802.11 enabled wireless ad hoc networks. The key idea of MOAR is to exploit the variable nature of the wireless channel in a *distributed* way via opportunistically skipping frequency channels in search of a better quality channel. When measurements indicate low channel quality on the current frequency channel, MOAR allows the receiver and transmitter to negotiate a decision to skip frequencies in search of a better quality channel. Since different IEEE 802.11 frequency channels are spaced at a distance greater than the *coherence bandwidth*, the channel conditions on different channels are independent and hence there is a high probability that the skipping node will find better channel conditions on one of the other frequency channels. Consequently MOAR nodes exploit the presence of diversity in frequency domain in a distributed
manner to transmit packets at a higher rate (on higher quality channels) resulting in an enhanced net system throughput for MOAR. Moreover, MOAR is compatible with the state-of-art rate adaptation protocols (e.g., Auto Rate Fallback [10], Receiver Based Auto Rate [11] and Opportunistic Auto Rate (OAR) [13, 12]) and hence is able to fully exploit the diversity present at the physical (PHY) layer in frequency domain (across multiple frequency channels) and in time domain (across users).

The design of MOAR is constrained to only one active flow (among all the flows within radio-range of the active flow) to use the entire spectrum at any given time.

In theory nodes can skip indefinitely in search of a better channel until the frequency channel with the highest possible transmission rate is found. However, in realistic systems where channel state information is not available a priori, each skip decision incurs an additional overhead due to channel measurement. As a result the throughput gains available via opportunistic channel skipping can diminish with each skip. Moreover, when the average channel conditions are poor, the probability of finding the highest quality channel (and the highest possible data rate) is very low. Thus, in order to maximize the gain in throughput it is critical to balance the tradeoff between additional throughput gain via channel skipping and the time and resource costs of channel measurement and skipping. Consequently, I devise an optimal skipping rule for MOAR to limit the number of times a node skips in search of a better channel. The optimal skipping rule for MOAR is designed to balance the tradeoff between the throughput gain available via opportunistic channel skipping and the cost of channel quality measurement in order to maximize the system throughput. In particular, the optimal skipping rule for MOAR maps the channel conditions at the PHY layer to a MAC rule which allows nodes to limit the
number of times they skip in search of a better channel.

I explore the performance of MOAR via extensive ns-2 simulations and also study the various factor impacting the performance of MOAR. Our experiments show that MOAR outperforms the state-of-art multi-rate protocols by 20% to 25%.

The remainder of this chapter is organized as follows. First in Section 3.2 I describe the wireless channel model and review the multi-rate and multi-channel capabilities of the IEEE 802.11 standards. Next, in Section 3.3 I discuss the related work on exploiting multi-rate and multi-channel capabilities of the IEEE 802.11 and also related work on exploiting frequency diversity in wireless networks. I present the Multi-channel Opportunistic Auto Rate (MOAR) protocol in Section 3.4 and also discuss the various challenges encountered while designing an efficient channel skipping protocol within the IEEE 802.11 channel access framework. In Section 3.5 I devise the optimal skipping rule for MOAR and discuss issues relating to implementation of the optimal skipping rule in practical systems in Section 3.6. The results of simulation experiments are presented in Section 3.7. In Section 3.8 I investigate the joint performance of DWOP (Chapter 2) and MOAR. Finally, I summarize this chapter in Section 3.10.

3.2 Background and Motivation

3.2.1 Channel Model

The transmitted radio frequency signal is reflected by both natural and man-made objects. Based on the relative phases of different reflections at the receiver, the different copies of the same signal may add coherently or tend to cancel out. Coherent addition of the copies can result in large received signal power and cancellation
eventually leads to zero received signal power. Thus, the signal at the receiver is a superposition of different reflections of the same signal, received with varying delays and attenuations. This phenomena of interference between two or more versions of the transmitted signal is called multipath fading. There are three main physical factors in the radio propagation channel which influence fading, as described below:

- **Multipath propagation**

  The presence of reflecting objects and scatterers in the channel path creates a constantly changing environment resulting in multiple versions of the transmitted signal arriving at the receiver displaced with respect to one another in amplitude, time and phase as shown in Figure 3.1. The signal power strength of the various multipath components is heavily dependent on the spatial location of the transmitter, receiver, the reflecting objects and the material of the reflecting objects.

![Figure 3.1: Illustration of multipath fading.](image)

- **Speed of the mobile and surrounding objects**

  Any motion, be that of the surrounding environment, the sender, or the re-
ceiver, results in random frequency modulation due to different *Doppler shifts* on each of the multipath components. This causes the strength of the received signal to vary with time. The speed of variation is directly governed by the speed of change in the communication medium (which consists of all intermediate objects in the direct and the reflected space).

- **Transmission bandwidth of the signal**

  If the transmitted radio signal bandwidth is greater than the "bandwidth" of the multipath channel, the received signal will be distorted, but the received signal strength will not fade much over a local area. The bandwidth of the wireless channel can be quantified by *coherence bandwidth* which is a measure of the maximum frequency separation for which the received signals are strongly correlated in signal amplitude. In other words, two sinusoids with frequency separation greater than the coherence bandwidth are affected quite differently by the channel. The coherence bandwidth of a channel is strongly related to the multipath structure of the channel. If the transmitted signal has a narrow bandwidth compared to the coherence bandwidth the amplitude of the signal changes rapidly but the signal is not distorted in time.

Physical layer design and analysis typically consider detailed propagation models that characterize all reflections and their time-variations [6, 4]. An accurate and widely utilized model which considers time-varying multi-path propagation is [4]

\[
y(t) = \sum_{i=1}^{p(t)} A_i(t) \cdot x(t - \tau_i(t)) + z(t), \quad (3.1)
\]

\[1\text{This is a baseband model which assumes perfect carrier demodulation at the receiver radio frequency front-end.}
where $x(t)$ is the transmitted signal and $y(t)$ is the received signal. The time-varying multi-path propagation is captured by the attenuation of each path $A_i(t)$, the time delays $\tau_i(t)$ and the number of paths $p(t)$. The additive term $z(t)$ is generally labeled as the background noise and represents the thermal noise of the receiver. The loss suffered by the signal during its propagation along different paths is captured in $A_i(t)$, and depends on the distance between the sender and the receiver.

Typically, physical layer algorithms (error correcting codes, channel modulation, demodulation and decoding) use the elaborate models in Equation (3.1). The performance of any physical layer implementation is well captured by observing its packet loss rate as a function of the received signal to noise ratio (SNR). Received signal to noise ratio measures the extent of the received signal power over the channel background noise. Generally, the larger the SNR, the better the chance of any packet being received error free. Actual performance (packet loss rate as a function of SNR) is dependent on a particular implementation.\textsuperscript{2}

Recognizing that the received signal-to-noise ratio, SNR can be used to capture the packet level performance of any physical layer implementation, I use the following model for the received $SNR$, given the transmission power $P$ at packet transmission time $t_p$,

$$SNR(t_p) = P_r(d) \cdot \frac{\rho(t_p)}{\sigma^2},$$

(3.2)

where $\rho(t_p)$ is the average channel gain for the packet at time $t_p$, and $\sigma^2$ is the variance of the background noise $z(t)$ and $P_r(d)$ denotes the received power when

\textsuperscript{2}For example, I have found in controlled laboratory tests with hardware-emulated channel conditions, that 802.11b compliant cards from different manufacturers perform differently under identical channel conditions [46].
the distance between the sender and the receiver is $d$ and is given by

$$P_r(d) = \frac{P_t G_t G_r \lambda^2}{(4\pi)^2 d^\beta},$$

(3.3)

where $P_t$ is the transmit power (in Watts), $G_t$ and $G_r$ are transmit and receive antenna gains, $\lambda$ is the wavelength (in meters) and $\beta$ is the path loss exponent.

The short time-scale variation in the received SNR is captured by the time-varying parameter $\rho(t_p)$, known as the fast fading component of the fading process. The time-variation of $\rho(t_p)$ is typically modeled by a probability distribution and its rate of change [4]. An accurate and commonly used distribution for $\rho(\cdot)$ is the Ricean distribution,

$$p(\rho) = \frac{\rho}{\sigma^2} e^{-\frac{\rho^2}{2\sigma^2}} I_0(2K\rho),$$

(3.4)

where $K$ is the distribution parameter representing the strength of the line of the sight component of the received signal and $I_0(\cdot)$ is the modified Bessel function of the first kind and zero-order [4]. The Ricean distribution models the case where there is a dominant stationary (nonfading) signal component present (such as the line-of-sight component) and the random multipath components are superimposed on a this stationary dominant signal. For $K = 0$, the Ricean distribution reduces to the Rayleigh distribution, in which there is no-line-of-sight component.

The phenomenon of multipath on a mobile radio channel is characterized by two parameters the multipath delay spread, which is related to frequency selectivity, and the Doppler shift which is related to time selectivity. Multipath delay spread is a direct result of different propagation delay of the various multipath components. On the other hand Doppler spread describes the time varying nature of the channel caused by motion of the mobile node and its surrounding objects. Next I discuss these two parameters in detail and also describe how the effects of these two parameters
is modeled.

**Doppler Shift and Coherence Time**

The rate of change of $\rho(t_p)$ in Equation (3.4) depends on a mobile host’s relative speed with respect to its surroundings. Among the several models available in the literature I use the Clarke and Gans model [4]. The motion of nodes causes a Doppler shift in the frequency of the received signal, and the extent of the Doppler shift depends on the relative velocity of the sender and the receiver. Let $f_m$ denote the maximum Doppler frequency during communication between two nodes. Then according to the Clarke-Gans model, the received signal is modulated in the frequency domain by the following spectrum

$$S(f) = \frac{1.5}{\pi f_m \sqrt{1 - \left( \frac{f - f_c}{f_m} \right)^2}}.$$  \hspace{1cm} (3.5)

In Equation (3.5), $f_c$ represents the carrier frequency of the transmitted signal and $f_m$ denotes the maximum Doppler shift given by

$$f_m = \frac{v}{\lambda},$$

where $v$ denotes the speed of the mobile node and $\lambda$ denotes the wavelength of the transmitted signal.

The spectral shape of the Doppler spectrum in Equation (3.5) determines the time domain fading waveform and hence the temporal correlation. The coherence interval, $T_c$, represents the average time of decorrelation and is given by

$$T_c = \frac{1}{f_m}.$$  \hspace{1cm} (3.6)

\footnote{Also see [4] for a survey.}

\footnote{For 802.11b, carrier frequency is in 2.4-2.481 GHz range.}
Coherence interval is a statistical measure of the time duration over which the channel impulse response is essentially invariant. In essence, the channel SNR values ρ(·) separated by more than $T_c$, are approximately independent. At mobile speeds of 1 m/s (3.6 km/hr), the coherence interval is approximately 122.88 ms for a center frequency of 2.4 GHz. The coherence interval reduces to 24.57 ms, 12.28 ms and 6.14 ms for mobile speeds of 5 m/s (18 km/hr), 10 m/s (36 km/hr) and 20 m/s (72 km/hr). In engineering design [4], a more conservative estimate of the coherence interval is used which is around 43% of the above numbers: 51.98 ms, 10.39 ms, 5.20 ms and 2.59 ms for speeds of 1, 5, 10 and 20 m/s. At 2, 5.5 and 11 Mb/sec, the transmission time for a 1000 byte packet is 4 ms, 1.45 ms and 0.73 ms. The observation that the coherence interval is of the order of multiple packet transmission times motivated the design of Opportunistic Auto Rate (OAR) protocol [12, 13].

Multipath Delay Spread and Coherence Bandwidth

Doppler spread and coherence interval are parameters which describe the time varying nature of the channel caused by the motion of the mobile node and of the objects surrounding it. However they do not describe the time dispersive nature of the channel due to multipath propagation delays. Due to reflection off surrounding objects, the various multipath components arrive at the receiver displaced with respect to each other in time and amplitude. This time dispersion of the channel is called multipath delay spread. A common measure of multipath delay spread is the root mean square (rms) delay spread. Typical values of the rms delay spread are on the order of microseconds in outdoor mobile radio channels and on the order of nanoseconds in indoor radio channels [4]. In particular, measurement studies [47, 48, 49, 46] have shown that for the IEEE 802.11b standard rms delay spread for an indoor
environment ranges from 10-35 ns.

A dual representation of multipath delay spread in frequency domain is given by the coherence bandwidth, $B_c$. Coherence bandwidth can be defined as a statistical measure over the range of frequencies over which the channel passes all spectral components with approximately equal gain and linear phase [4]. In other words coherence bandwidth is the range of frequencies over which two frequency components have a strong potential for amplitude correlation and thus two sinusoids with frequency separation greater than $B_c$ are affected quite differently by the channel.

The rms delay spread and coherence bandwidth are inversely proportional to one another. Assuming frequency correlation between amplitudes of frequency components being above .9, the coherence bandwidth can be approximated by [4, 50]

$$B_c \approx \frac{1}{50 \cdot \sigma_r},$$  \hspace{1cm} (3.7)

where $\sigma_r$ represents the rms delay spread. For frequency correlation between amplitudes of frequency components above .5, the coherence bandwidth can be approximated by [4, 50]

$$B_c \approx \frac{1}{5 \cdot \sigma_r},$$ \hspace{1cm} (3.8)

Using the values of rms delay spread from measurement studies [47, 48, 49, 46], typical value of coherence bandwidth for IEEE 802.11 standards can be computed to be in the range 1-3 MHz in an indoor environment.

3.2.2 Review of IEEE 802.11

In this section I review the multi-rate and multi-channel properties of the IEEE 802.11 standard [7, 9]. Table 3.1 summarizes the multi-rate and multi-channel features of

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5Direct Sequence Spread Spectrum.

6Orthogonal Frequency Division Multiplexing.
<table>
<thead>
<tr>
<th>Physical Layer</th>
<th>802.11b</th>
<th>802.11a</th>
</tr>
</thead>
<tbody>
<tr>
<td>DSSS[^5]</td>
<td></td>
<td></td>
</tr>
<tr>
<td>OFDM[^6]</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Maximum Achievable Data Rate</td>
<td>11 Mb/sec</td>
<td>54 Mb/sec</td>
</tr>
<tr>
<td>Frequency Band</td>
<td>2.4 GHz</td>
<td>5 GHz</td>
</tr>
<tr>
<td>Number of Channels</td>
<td>11</td>
<td>12</td>
</tr>
<tr>
<td>Number of Orthogonal Channels</td>
<td>3</td>
<td>8</td>
</tr>
<tr>
<td>Channel Separation</td>
<td>5 MHz</td>
<td>20 MHz</td>
</tr>
<tr>
<td>Coherence Bandwidth [47, 48, 49, 46]</td>
<td>1-3 MHz</td>
<td>1-3 MHz</td>
</tr>
</tbody>
</table>

Table 3.1: Multi-rate and Multi-channel Features of IEEE 802.11 Standards

IEEE 802.11a and IEEE 802.11b standards.

**Multi-rate IEEE 802.11**

The IEEE 802.11a and IEEE 802.11b protocols are *multi-rate* in that they provide physical-layer mechanisms to transmit at higher rates than the base rate if channel conditions so permit. Figure 3.2 shows a sample channel variation with time for a mobile speed of 2.5 m/s and a Ricean channel fading model with the value of the Ricean parameter, $K$, set to 3. The received power shows wide fluctuations such that the supported data rate varies between 5.5 and 2 Mb/sec with almost equal probability. In practice, depending on the line-of-sight factor $K$ in Equation (3.4) and the distance between the transmitter and the receiver $d$ in Equation (3.2), the channel rates can vary within the entire range of the lowest to highest possible data
rate. As shown in Table 3.1, the highest available rate in IEEE 802.11a is 54 Mb/sec and 11 Mb/sec for IEEE 802.11b.

![Graph showing channel variation with time](image)

Figure 3.2: Illustration of channel variation with time

In particular for IEEE 802.11b systems I model the achievable data rates as a function of received signal to noise ratio, SNR, as

\[
\text{Data Rate} = \begin{cases} 
2 \text{ Mb/sec} & \text{if } SNR_2 \leq \text{SNR} < SNR_{5.5} \\
5.5 \text{ Mb/sec} & \text{if } SNR_{5.5} \leq \text{SNR} < SNR_{11} \\
11 \text{ Mb/sec} & \text{if } SNR_{11} \leq \text{SNR}
\end{cases} \quad (3.9)
\]

---

7Achievable data rates as a function of distance for 802.11a are available in a white paper from [http://www.atheros.com](http://www.atheros.com). For 802.11b, I use the specifications for the Orinoco™ wireless NIC which can be found at [http://www.orinocowireless.com](http://www.orinocowireless.com)
where $SNR_2$, $SNR_{5.5}$ and $SNR_{11}$ denote the minimum threshold signal to noise ratio to support data rates of 2, 5.5 and 11 Mb/sec respectively.

**Multi-channel IEEE 802.11**

Besides multi-rate capabilities, the IEEE 802.11 standard also provides for multiple frequency channels as summarized in Table 3.1. In case of IEEE 802.11b the allocated spectrum in the 2.4 GHz band is from 2400 MHz to 2483 MHz. For North America, there are 11 channels starting at 2412 MHz and spaced at an interval of 5 MHz each [7, 9, 8]. Each channel has an approximate bandwidth of 22 MHz and channels 1, 6 and 11 (which are 25 MHz apart) are completely orthogonal. Similarly, in case of IEEE 802.11a there are a total of 12 physical layer channels with 8 completely orthogonal channels.

Recall from Section 3.2.1 that the coherence bandwidth for IEEE 802.11 standards ranges from 1-3 MHz which is much less than the channel separation of 5 MHz for IEEE 802.11b and 20 MHz for IEEE 802.11a. Thus from the definition of coherence bandwidth it follows that two different IEEE 802.11 frequency channels experience uncorrelated fading. *The fact that coherence bandwidth is smaller than the channel separation for IEEE 802.11 provides a key motivating factor for designing a multi-channel opportunistic MAC protocol.* I exploit this observation to motivate the design of Multi-channel Opportunistic Auto Rate (MOAR) protocol in Section 3.4.

### 3.3 Related Work

In this section I discuss the related work in literature on multi-rate and multi-channel IEEE 802.11 and also discuss the related work on exploiting diversity for higher
throughput. I divide the related work into three main categories. In Section 3.3.1 I present the related work on exploiting multi-rate capabilities of IEEE 802.11. Next I discuss the literature on MAC protocol design to exploit the presence of multiple physical layer channels in Section 3.3.2. Finally, in Section 3.3.3 I present the related work in wireless communications literature on opportunistically exploiting diversity to enhance the throughput of wireless networks.

3.3.1 Multi-rate IEEE 802.11

Few rate-adaptation techniques have been designed for multi-rate wireless ad hoc networks. The first commercial implementation that exploits the multi-rate capability of IEEE 802.11 networks is termed Auto Rate Fallback (ARF) [10]. Another protocol to exploit the multi-rate capabilities of IEEE 802.11 termed Receiver Based Auto Rate (RBAR) was proposed in [11]. The key idea of RBAR is for receivers to control the sender’s transmission rate. In IEEE 802.11, all RTS/CTS messages must be sent at the base rate to ensure that all stations are able to receive these messages error free. RBAR uses physical-layer analysis of the received RTS message to determine the maximum possible transmission rate for a particular bit error rate. The receiver inserts this rate into a special field of the CTS message to inform the sender and other overhearing nodes of the potentially modified rate. Overhearing nodes modify their NAV values to the new potentially decreased transmission time. In this way, RBAR quickly adapts to channel variations and extracts significant throughput gains.

Opportunistic Auto Rate (OAR) protocol proposed in [12, 13] exploits the unique characteristics of the variable wireless channel to enhance the throughput of IEEE 802.11 enabled wireless ad hoc networks. In particular OAR exploits the fact that at mod-
erate velocities, channel coherence time\textsuperscript{8} is on the order of multiple packet times. Thus, when the channel is of good quality, significant throughput improvement can be obtained by opportunistically sending multiple back-to-back packets at a higher rate. Consequently, when a mobile user is granted channel access while encountering a high-quality channel, OAR grants the user a channel access time that allows multiple packet transmissions. As the subsequent packet transmissions are also highly likely to be successful at the higher data rate, OAR obtains a throughput gain as compared to state-of-art rate-adaptation protocols, e.g., RBAR and ARF. Moreover, OAR also limits the extent to which it is opportunistic in order to ensure that users with perpetually bad channels obtain their fair share of time accessing the channel. Hence, OAR is opportunistic while still maintaining the temporal fairness properties of the base-rate IEEE 802.11 protocol.

3.3.2 Multi-channel Medium Access Control

MAC protocol designs that exploit multiple physical layer frequency channels have received significant attention in the recent literature \cite{51, 52, 53, 54, 55, 56, 57}. For example, the protocols in \cite{52, 54} divide a common channel into multiple (two in \cite{52}, one data and one control) sub-channels to decrease contention in CSMA type networks and increase throughput. These protocols require each station to monitor all sub-channels at all times which may not be feasible in practice as it requires more than one transceiver per node. Hop-Reservation Multiple Access (HRMA) protocol \cite{57} is a multi-channel protocol for slow frequency hop ad hoc networks where all stations hop according to a predefined hopping pattern and exchange RTS/CTS. After a successful exchange of RTS/CTS the transmitter-receiver remain in a hop

\textsuperscript{8}Duration beyond which samples of received signal are uncorrelated(Section 3.2.1)
for further data exchange while other nodes keep hopping according to the prede-
fined hop pattern. The MAC protocol in [56] provides a means to load balance
users among the three orthogonal channels in IEEE 802.11b enabled wireless ad hoc
networks.

All of the above approaches exploit multiple frequency channels to reduce con-
tention or to increase throughput by ensuring that multiple communication can take
place in the same region simultaneously, each in a different non-interfering channel.
Although these approaches result in significant performance gains especially in a
targeted scenario of ad hoc networks, they do not address opportunistic scheduling
 gains available from a multi-rate medium access protocol and do not exploit the
unique properties of multiple frequency channels (namely independent fading) to
enhance the throughput of ad hoc networks. Our objective in this thesis is to isolate
the throughput gains available from opportunistically skipping channels in search of
better quality channels and address the various MAC mechanisms needed to capture
this potential gain in ad hoc networks.

Likewise, while [43] does address multi-channel opportunistic scheduling, it fo-
cuses on a cellular time slotted system with perfect channel information and is not
applicable to distributed systems such as ad hoc networks.

3.3.3 Exploiting Diversity via Opportunistic Communication

The existence of multiple channels is a source of diversity which can be exploited
to enhance the throughput of wireless ad hoc networks. The concept of enhanc-
ing throughput by exploiting diversity (be it multi-channel, spatial or multi-user
diversity) has been well studied in the wireless communications literature. One
such formulation is known as the problem of parallel Gaussian channels [58, 41],
where multiple simultaneous and orthogonal channels are available to the transmitter, and the transmitter appropriately allocates its power and/or time resources. Also, there is a growing literature on opportunistic and multi-rate scheduling, e.g., [59, 42, 39, 38]. Such schemes exploit channel variations to select high-quality-channel users and provably optimize system throughput while also satisfying user fairness constraints. However, the above cited works assume that the channel quality of each of the users is known a priori, which allows the transmitter to choose the user and/or the channel optimally. Moreover, such results address scheduling in centralized time-slotted systems more applicable to cellular networks and do not address the distributed MAC protocols required to extract the available performance gains.

Multi-user diversity has also been studied within the context of wireless ad hoc networks in [60] where the authors exploit mobility to increase the capacity of delay-insensitive wireless ad hoc networks. In [44] the authors jointly address both physical layer and medium access control issues to exploit multiuser diversity gains in a distributed fashion in CSMA networks. However, none of the above cited work exploits the presence of multiple frequency channels within the IEEE 802.11 protocol to enhance the throughput of wireless ad hoc networks.

In the next section, I describe the challenges involved in designing a realistic distributed MAC protocol which seeks to achieve significant throughput gains by skipping channels in search of higher-quality channels and present a detailed description of the MOAR protocol.
3.4 Multi-channel Opportunistic Auto Rate (MOAR)

3.4.1 Background

OAR [12, 13] can be characterized as opportunistic across users, better exploiting periods of high quality channel to achieve a significant throughput gain. However, OAR does not exploit the presence of diversity in frequency domain (in the form of multiple channels). In particular, short time-scale channel variations for different IEEE 802.11 channels have a low degree of correlation among themselves. Figure 3.3 depicts a typical sample path of the received SNR for two channels between the same two devices (at a fixed distance). The figure shows two horizontal lines which indicate the threshold SNR for receiving at 2 Mb/sec and 5.5 Mb/sec. The key point is that the two channels have a strong independent component despite being from the same pair of devices. This is due to the fact that the channel separation in the frequency domain (5 MHz, for IEEE 802.11b) is much larger than the coherence bandwidth\(^{10}\) which at 2.4 GHz ranges from 1-3 MHz [47, 48, 49, 46]. Namely, while different channels may have the same average conditions, measurement studies [4] and the example in the figure indicate that SNRs on different channels can be quite different at the same time such that there are significant potential throughput gains to be obtained by selection of a better quality channel.

3.4.2 Objectives

In this thesis I devise Multi-channel Opportunistic Auto Rate (MOAR), a distributed MAC protocol to exploit the fading diversity among different IEEE 802.11 frequency

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\(^9\)These channel conditions are obtained with the Ricean fading model with parameter \(K = 4\).

\(^{10}\)Recall from Section 3.2.1 that coherence bandwidth is the bandwidth over which the channel fading is correlated.
channels and opportunistically select a better quality frequency channel for transmission. The fundamental idea is that both the transmitter and the receiver of a flow opportunistically skip channels in search of a better quality channel, if the current channel is of low quality. Ideally, channel qualities on all the frequency channels would be known so that nodes could simply skip to the best channel to transmit on at all times. However for realistic systems, design of an efficient channel skipping protocol introduces the following challenges:

- **Measuring channel conditions before and after each skip.**

For realistic systems channel conditions on all the frequency channels are not known a priori. Moreover, since channel conditions are continually changing, past channel measurements (beyond several packet transmission times, i.e., co-
herence time interval) are not a useful predictor of current channel conditions. Hence, there is a need to introduce a mechanism to measure the current conditions on the present channel before making the decision whether to skip to another channel or not.

- **Coordinating a channel skip decision between the transmitter and the receiver.** Prior to skipping, the transmitter and the receiver of a flow need to mutually decide the frequency channel to skip to. Since a wireless ad hoc network does not have a central entity to coordinate skip decisions, there is a need for a distributed mechanism to coordinate the skip decision between the transmitter and the receiver.

- **Maintaining carrier sense for all overhearing nodes.** A potential problem with channel skipping in wireless ad hoc networks is the need to maintain carrier sense for all overhearing nodes to avoid the hidden terminal problem [17]. This involves making sure that all overhearing nodes are able to correctly set their defer timers so as to allow the transmitter-receiver pair sufficient time to skip to better quality channels.

- **Limiting the number of times nodes skip in search of a better quality channel.** Potentially, a transmitter-receiver pair can continue skipping multiple times in search of the highest quality channel. However, due to the overhead of channel measurement and estimation incurred at every skip, throughput gains of sending data on a better quality frequency channel are diminishing with each skip. Moreover, when the average channel conditions are poor, the probability of finding the highest quality channel is very low. Therefore it is important to balance the tradeoff between throughput gain and the time and resource cost
of opportunistic channel skipping. In particular, there is a need to devise a mechanism to optimally limit the number of times a transmitter-receiver pair skip in search of a better quality channel.

Next I present a detailed description of the MOAR protocol and also describe how I overcome the first three challenges mentioned above. In Section 3.5 I devise an optimal skipping rule for MOAR and show how a MOAR node can limit the number of times it skips in search of better quality channels to optimally balance the tradeoff between the throughput gain available via opportunistic skipping and the overhead of channel skipping/measurement.

3.4.3 MOAR Protocol Description

In this section I present the MOAR protocol in detail. Although our discussion of MOAR is within the context of the RTS/CTS mechanism within the DCF mode of IEEE 802.11 standard, the concepts are equally applicable to other RTS/CTS based protocols such as SRMA [61], MACAW [17] and FAMA [62].

Ideally, all channel qualities would be known so that a user could simply select the best channel to transmit on at all times. However, in practice, channel qualities are continually changing and not known until explicitly measured via control (RTS/CTS) or data packets. Thus, MOAR employs a channel skipping technique within the IEEE 802.11 framework as described below.

All nodes initially reside on a single common frequency channel, known as the home channel. The DATA transmission is preceded by the sender transmitting an RTS packet to the receiver on the home channel. On reception of the RTS frame, the receiver makes the decision to skip by comparing the measured signal to noise
ratio (SNR)$^{11}$ to a channel skip threshold. I describe how to set the value of the channel skip threshold in Section 3.5 where an optimal skipping algorithm for MOAR is devised. If the measured SNR is low, the sender and the receiver skip to a new channel in search of a better quality channel, whereas if the measured SNR is high, data is transferred on the current frequency channel as in the OAR [12, 13] protocol, in which nodes transfer a multiple number of packets in proportion to their channel quality. In this way by making opportunistic channel skipping compatible with OAR I seek to fully exploit the diversity present at the PHY layer in frequency domain (across multiple frequency channels) and in time domain (across users).

On making the decision to skip, the receiver selects a channel to skip to and piggy-backs this channel on the CTS packet. After transmitting the CTS frame, the receiver immediately skips to the new frequency channel and waits for another RTS from the receiver for a time equal to the CTS timeout value as mandated by the IEEE 802.11 standard [7, 8]. Since I assume that in a realistic setting channel conditions on other frequency channels are unknown, the channel to which the receiver decides to skip is selected randomly among the available frequency channels. Yet, if information regarding channel conditions or interference on some other frequency channel is known (e.g., in a wireless LAN scenario where the Access Point (AP) may have information regarding interference on other frequency channels), the receiver can take that into account to make a better decision about which channel to skip to. However, for the purpose of this discussion the existence of such information is not required.

If after skipping to a new frequency channel the receiver does not receive another

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$^{11}$A reasonably accurate estimate of the received SNR can be made from physical-layer analysis of PHY layer preamble to each packet.
RTS from the sender within a CTS timeout period, the receiver node switches back to the home channel and starts contending for channel access as mandated by the IEEE 802.11 standard.

Once the sender receives confirmation of the choice of frequency channel to skip to from the receiver (via a CTS frame), it immediately skips to that channel. The time elapsed for switching channels is 1\(\mu s\) [51] and of negligible overhead. After skipping to the selected channel, the transmitter and receiver renegotiate the data rate via another RTS/CTS exchange which also serves the dual purpose of measuring the channel conditions on the new frequency channel. Note that the transmission time of an RTS/CTS exchange represents approximately 5% of the DATA/ACK transmission time for a 1000 byte data packet at the base rate.\(^{12}\) As transmission above the base rate represents a 5.5-fold maximum increase for IEEE 802.11b and 27-fold maximum increase for IEEE 802.11a, significant throughput gains are available to MOAR even after accounting for the overhead of channel skipping and additional RTS/CTS messages. In case channel quality on the new frequency channel is measured to be below the skip threshold, the sender-receiver pair can choose to skip again in search of a better quality channel.

While channel skipping could potentially proceed multiple times, to derive maximal throughput gain it is necessary to balance the cost of channel measurement per skip against the expectation of obtaining a better quality channel via continued skipping. In Section 3.5 I devise an optimal skipping algorithm to decide the optimal

\(^{12}\)Recall that for the OAR [12] protocol nodes transmit multiple back-to-back data packets at higher data rates so that the net DATA/ACK transmission time for multiple packets transmitted at a higher data rate is the same as the transmission time for a single DATA/ACK exchange at the base rate.
time to stop in order to maximize the expected throughput gain of skipping.

Since RTS/CTS exchange prior to any channel skip is done at the base rate on the home channel, all nodes within radio range of the receiver and the transmitter can also decode these packets. However, some nodes (including nodes within radio range of the sender but outside the radio range of the receiver) may be unable to hear the CTS packet and are unable to detect whether a decision to skip frequency channels was made or not. Moreover, even though nodes within radio range of the receiver can correctly decode a CTS packet and infer that a decision to skip has been made, they are unable to set a correct defer time since it is not known a priori how many times the sender-receiver pair may skip in search of a better quality channel. This can lead to problems similar to the hidden terminal problem [17].

To solve the problem mentioned above all MOAR nodes upon reception of an RTS/CTS packet, defer (via the Network Allocation vector, NAV) for a fixed amount of time corresponding to a maximum time, $D_{\text{skip}}$, necessary for the transmitter and receiver to skip (multiple times, if required) to a better quality channel and finish the DATA/ACK transmission. $D_{\text{skip}}$ is given by

$$D_{\text{skip}} = N_{\text{max-skip}} \cdot T_D,$$  \hspace{1cm} (3.10)

where, $N_{\text{max-skip}}$ represents the maximum number of allowed channel skips and $T_D$ represents the time for the entire RTS/CTS/DATA/ACK exchange (at the base rate) including all the defer timers (EIFS, SIFS, DIFS etc) as mandated by the IEEE 802.11 standard. $N_{\text{max-skip}}$ is equal to the number of frequency channels available, which in case of IEEE 802.11b standard is equal to 11 as shown in Table 3.1.

I refer to $D_{\text{skip}}$ as a temporary reservation, to denote the fact that the reservation is not an actual reservation but represents a maximal amount of reservation time. A temporary reservation serves to inform the neighboring nodes that a reservation
has been requested but the duration of the reservation is not known. Any node that receives the temporary reservation is required to treat it the same as an actual reservation with regard to later transmission requests; that is if a node overhears a temporary reservation it must update its NAV so that any later requests it receives that would conflict with the temporary reservation must be denied. Thus the temporary reservation serves as a placeholder until either a new reservation is received or is canceled. If the sender-receiver pair decide not to skip channels then they can proceed with the DATA/ACK exchange on the home channel as dictated by OAR [12, 13] in which case other nodes can replace the temporary reservation with the exact reservation, as carried in the DATA/ACK packets.

Once the transmitter and the receiver conclude the DATA/ACK transmission by skipping to one or more frequency channels, they return to the home channel. After the final DATA/ACK transmission (recall that the sender/receiver send multiple back-to-back packets as required by the OAR protocol [12, 13]), the sender and receiver exchange another RTS/CTS packet on the home channel so that all nodes within range of either the sender and/or the receiver can correctly infer the end of channel skipping and cancel the temporary reservation timer. The RTS/CTS packets signalling the end of channel skipping have a special bit set to differentiate them from a normal RTS/CTS packet. In case a node is unable to hear either the updated reservation or the RTS/CTS transmission signalling the end of the temporary reservation, it would be able to contend for the channel again after the temporary reservation has expired. I explore the performance of the MOAR protocol via extensive ns-2 simulations in Section 3.7.

As described above, MOAR incurs a constant overhead of channel measurement via RTS/CTS exchange per skip. Thus there is a need to balance the tradeoff
between the throughput gain available via opportunistic skipping and the overhead of channel skipping/measurement. In the next section I formulate this tradeoff as an optimal stopping time problem and devise an optimal skipping rule for MOAR to limit the number of times a MOAR node skips in search of better quality channels.

3.5 Optimal Skipping Rule for MOAR

The problem of deciding the optimal number of times a MOAR node should skip in search of a better quality frequency channel can be formulated as an *optimal stopping time* problem. The theory of optimal stopping time is concerned with the problem of choosing a time to take a given action based on sequentially observed random variables in order to maximize an expected payoff or to minimize an expected cost [63, 64].

3.5.1 The Definition of Optimal Stopping Time Problem

Stopping rule problems are defined by two objects,

- a sequence of random variables, $X_1, X_2, \ldots$, whose distribution is assumed known, and

- a sequence of real-valued reward functions,

$$y_1(x_1), y_2(x_1, x_2), \ldots, y_n(x_1, x_2, \ldots).$$

Given these two objects, associated stopping rule problem may be described as follows [63]. The sequence of random variables $X_1, X_2, \ldots$ may be observed for as long as one wishes. For each $n = 1, 2, \ldots$, after observing $X_1 = x_1, X_2 = x_2, \ldots, X_n = x_n$, one may stop and receive the known reward $y_n(x_1, \ldots, x_n)$ (possibly negative), or one may continue to observe $X_{n+1}$. If one never stops, one receives $y_\infty(x_1, x_2, \ldots)$. The problem
is to decide a stopping rule which chooses a stopping time to maximize the expected reward.

A stopping rule problem has a finite horizon if there is a known upper bound on the number of stages at which one may stop. If stopping is required after observing $X_1, X_2...X_T$, then the problem has a horizon $T$. A finite horizon problem is a special case of the general stopping rule problem with $y_{T+1} = ... = y_{\infty} = -\infty$. Finite horizon stopping rule problems can be solved by the method of backward induction[63]. Since stopping is required at stage $T$, first the optimal rule at stage $T - 1$ is found. Thus, knowing the optimal rule at stage $T - 1$ the optimal stopping rule at stage $T - 2$ can be found and so on back to the initial stage. I Define

$$G^{(T)}_T = y_T(x_1, x_2.., x_T),$$  \hspace{1cm} (3.11)

and then inductively for $i = T - 1$, backwards to $i = 0$

$$G^{(T)}_j(x_1.., x_j) =$$

$$\max\{y_j(x_1.., x_j), E(G^{(T)}_{j+1}(x_1,.., x_j, X_{j+1})|X_1 = x_1,.., X_j = x_j)\}. \hspace{1cm} (3.12)$$

3.5.2 Existence of Optimal Stopping Rules

Consider the general stopping rule problem with observations $X_1, X_2...$ and rewards $Y_1...$ where $Y_n = y_n(X_1,..,X_n)$. It is proven in [63] that an optimal stopping rule exists if the following two conditions are satisfied:

**Condition 1** $E[\sup_n Y_n] < \infty$

**Condition 2** $\limsup_{n \to \infty} Y_n \leq Y_{\infty}$ a.s.

In this case (for the class of finite horizon problems) the optimal rule is given by the principle of optimality [63] as

$$N^* = \min\{n \geq 1 : X_n \geq G^*\}, \hspace{1cm} (3.13)$$
where $G^*$ denotes the expected return from an optimal stopping rule.

### 3.5.3 Optimal Number of Channel Skips

The problem of deciding the optimal number of times to skip can be formulated as an optimal stopping time problem as follows. Let $X_n$ denote the expected payoff of transmitting after skipping $n$ times. $X_n$ is a function of channel quality at that time. Suppose that $X_1, X_2, \ldots$ are iid with known distribution $F(x)$. Each additional skip involves paying the cost, $c$, of channel measurement via an RTS/CTS exchange. The problem is for a flow to decide the optimal number of times to skip in order to maximize the expected payoff.

The above problem is an optimal stopping rule problem and is similar to the house selling problem without recall [63] with observations $X_1, X_2, \ldots$ and reward function

$$Y_n = X_n - nc$$

$$Y_\infty = -\infty.$$  \hfill (3.14)

The following theorem (proof in [64]) states that Condition 1 and Condition 2 are satisfied and an optimal stopping rule exists if $X$ has a finite first and second moment.

**Theorem 2** Let $X_1, X_2, \ldots$ be identically distributed and let $c > 0$ and $Y_n = X_n - nc$

If $E[X_i^+] < \infty$, then $\sup Y_n < \infty$ a.s. and $Y_n \to -\infty$ a.s.

If $E[X_i^+]^2 < \infty$ then $E[\sup Y_n] < \infty$

Suppose $c$ is paid to observe $X_1 = x_1$. Continuing from this point on then $x_1$ is lost and the cost $c$ has already been paid, so it is just like starting the problem over again; that is the problem is invariant in time. So, continuing from this point
can obtain an expected return of $G^*$ and no more. Thus if $x_1 < G^*$ then one should continue, and if $x_1 > G^*$ one should stop. For $x_1 = G^*$ it is immaterial what one does, but let us say one stops. This argument can be made at any stage, so the optimal stopping rule is as given by Equation (3.13) and $G^*$ can be computed as

$$G^* = E[\max\{X, G^*\}] - c.$$  \hspace{1cm} (3.15)

For the case of channel skipping within the IEEE 802.11b standard, I define $c$ as the time (in $\mu$sec) for an RTS/CTS exchange at the base rate of $R_{\text{base}}$\textsuperscript{13} and is given by

$$c = 2 \cdot \left\{ \frac{L_{RTS} + L_{CTS}}{R_{\text{base}}} + SIFS \right\}.$$  \hspace{1cm} (3.16)

where $L_{RTS}$ and $L_{CTS}$ denote the length of the RTS and CTS packet (in bits) respectively and $SIFS$ denotes the Short InterFrame Spacing [7]. Since the MOAR protocol incurs an additional RTS/CTS transmission on the home channel after the final DATA/ACK transmission, the cost of skipping is two times the RTS/CTS time.

The payoff, $X_i$, in $\mu$sec after skipping $i$ times is given by

$$X_i(R) = T_{\text{data}} \cdot \frac{R_i}{R_{\text{base}}},$$  \hspace{1cm} (3.17)

where $R_i$ is a random variable denoting the achievable data rate (in Mb/sec) after $i$ skips, $R_i/R_{\text{base}}$ denotes the number of packets sent in time $T_{\text{data}}$ at rate $R_i$ by the OAR protocol[12, 13] and $T_{\text{data}}$ is the time to send a data packet at the base rate, and is given by

$$T_{\text{data}} = \frac{L_{\text{data}}}{R_{\text{base}}},$$  \hspace{1cm} (3.18)

where $L_{\text{data}}$ is the length of a data packet (in bits). From Equation (3.17) and (3.18),

\textsuperscript{13}The base rate for IEEE 802.11b is 2 Mb/sec.
the payoff $X(R)^{14}$ is given by

$$X(R) = \frac{L_{\text{data}}}{R_{\text{base}}} \cdot R \text{ with probability } p_R,$$  

(3.19)

where $p_R$ denotes the probability that the achievable data rate is equal to $R$. The achievable data rate, $R$, is a function of received SNR, which is given by

$$\text{SNR} = \frac{\bar{P}_r(d)p(\rho)}{\sigma^2},$$  

(3.20)

where $p(\rho)$ represents the fast-fading characterized by the Ricean distribution as given by

$$p(\rho) = \frac{\rho}{\sigma^2} e^{-\left(\frac{\rho^2}{2\sigma^2}+K\right)} I_0(2K\rho).$$  

(3.21)

$K$ is the distribution parameter representing the strength of the line of the sight component of the received signal and $I_0(\cdot)$ is the modified Bessel function of the first kind and zero-order [4]. For $K = 0$, the Ricean distribution reduces to the Rayleigh distribution, in which there is no-line-of-sight component. In Equation (3.20) $\sigma^2$ is the variance of the background noise and $\bar{P}_r(d)$ denotes the average received power at distance $d$.

Given the channel model of Equation (3.20) and its distribution (Equation (3.21)), the distribution of achievable data rates can be calculated as follows. Let $SNR_R$ and $SNR_{R+1}$ denote the minimum and the maximum required threshold SNR to support a transmission rate of $R$ Mb/sec. Then the probability that rate $R$ is feasible is given by $p_R$ and is calculated as

$$p_R = p(SNR_R \leq \text{SNR} < SNR_{R+1}),$$  

(3.22)

where $p(\text{SNR}) = p(\rho\sigma^2\bar{P}_r^{-1})$ is the distribution of received $SNR$ (given by Equation (3.21)).

---

14Since I assume that $X_1, X_2, \ldots$ are i.i.d, I drop the subscript $i$ for convenience.
For example, in case of IEEE 802.11b let $SNR_2$, $SNR_{5.5}$ and $SNR_{11}$ denote the minimum required threshold SNR to support transmission rates of 2, 5.5 and 11 Mb/sec respectively. Then the probability that rate $R$ is feasible, $p_R$, is calculated as follows

\[
p_2 = p(SNR_2 \leq SNR < SNR_{5.5})
\]
\[
p_{5.5} = p(SNR_{5.5} \leq SNR < SNR_{11})
\]
\[
p_{11} = p(SNR_{11} \leq SNR).
\]

Using Equation (3.19) the distribution of the payoff $X(R)$, is obtained as follows:

\[
X = \begin{cases} 
L_{data} \cdot \frac{2}{4} & \text{with probability } p_2 \\
L_{data} \cdot \frac{5.5}{4} & \text{with probability } p_{5.5} \\
L_{data} \cdot \frac{11}{4} & \text{with probability } p_{11},
\end{cases}
\]

(3.24)

where $L_{data}$ is the length of data packet in bits, $X$ is in $\mu$sec and I have used the fact that for IEEE 802.11b, $R_{base}$ is equal to 2 Mb/sec. The distribution of the payoff (as given by Equation (3.24)) is a function of the distribution of the achievable data rates as given by Equation (3.23) which in turn is a function of the channel conditions as given by $SNR$ (Equation (3.20)). Thus under the assumption that the distribution of the channel fading is known, the distribution of payoff $X$ is known too. An accurate and commonly used distribution for fast fading is the Ricean distribution [4], given by Equation (3.21).

It can be seen from Equation (3.24) that $X$ has finite first and second moments (for a finite sized data packet). Thus it follows from Theorem 1 that an optimal stopping rule exists and is given by Equation (3.13).
From Equation (3.15) the optimal payoff, \( G^* \) is a solution of

\[
g = E[\max \{X, g\}] - c
\]

\[
= g \cdot p(g > X(R)) + E[X(R)] \cdot p(g \leq X(R)) - c. \tag{3.25}
\]

I use the following method of discrete optimization to find the value of \( G^* \). Since \( g \in [0, \infty) \), I can divide the range of \( g \) in the four mutually exclusive sub-intervals. The boundaries of the four sub-intervals are defined by \( X(R) \) (Equation (3.24)) as below:

**Case 1** \( g \in (L_{\text{data}} \cdot \frac{11}{4}, \infty) \)

From Equation (3.24), since \( X \) is bounded from above by \( L_{\text{data}} \cdot \frac{11}{4} \), I have

\[
\max [X, g] = g, \forall g \in (L_{\text{data}} \cdot \frac{11}{4}, \infty)
\]

Using in Equation (3.25) to solve for \( G^* \)

\[
g = g - c, \tag{3.26}
\]

which is not possible for a non-zero value of \( c \). Thus \( g \notin [L_{\text{data}} \cdot \frac{11}{4}, \infty) \).

**Case 2** \( g \in (L_{\text{data}} \cdot \frac{5.5}{4}, L_{\text{data}} \cdot \frac{11}{4}] \)
From Equation (3.24) and the boundary conditions for this case, I have

\[
p(X < g) = p(X < L_{\text{data}} \cdot \frac{11}{4})
\]
\[
= p(R < 11 \text{ Mb/sec})
\]
\[
= 1 - p(R = 11 \text{ Mb/sec})
\]
\[
= 1 - p_{11}
\]  

(3.27)

\[
p(X \geq g) = p(X \geq L_{\text{data}} \cdot \frac{11}{4})
\]
\[
= p(R \geq 11 \text{ Mb/sec})
\]
\[
= p(R = 11 \text{ Mb/sec})
\]
\[
= p_{11},
\]  

(3.28)

where I have used the fact that \( R \) is upper bounded by 11 Mb/sec to compute \( p(R \geq 11 \text{ Mb/sec}) = p_{11} \). Using Equation (3.25)

\[
g = g \cdot [1 - p_{11}] + E[X] \cdot p_{11} - c
\]
\[
= E[X] - \frac{c}{p_{11}}
\]  

(3.29)

Note that \( g \) as given by Equation (3.29) is a function of the constant \( c \) (the cost of channel measurement via RTS/CTS) and the parameters of the distribution of the payoff \( X \) (given by Equation (3.24)). Thus, under the assumption that the distribution of channel fading is known, the value of \( g \) given by Equation (3.29) is also known.

The value of \( g \) as given by Equation (3.29) is a candidate value for the value of the optimal payoff via channel skipping, \( G^* \). For this value to be a valid value of the optimal payoff value, the boundary conditions for this case (namely that \( g \in (L_{\text{data}} \cdot \frac{55}{4}, L_{\text{data}} \cdot \frac{11}{4}) \)) need to be satisfied. Given, the distribution of payoff, \( X \), (Equation (3.24)), its parameters (namely \( E[X] \) and \( p_{11} \), can be substituted in
Equation (3.29) to obtain a candidate value of $g$ which is then compared to the boundary conditions to determine whether it indeed is the valid optimal value, $G^*$. If the boundary conditions are not met this value of $g$ is rejected.

**Case 3** $g \in (L_{data} \cdot \frac{2}{4}, L_{data} \cdot \frac{5.5}{4}]$

Using arguments similar to those used in Case 2, I have

$$p(X < g) = p(X < L_{data} \cdot \frac{5.5}{4})$$
$$= p(R < 5.5 \text{ Mb/sec})$$
$$= 1 - [p(R = 5.5 \text{ Mb/sec}) + p(R = 11 \text{ Mb/sec})]$$
$$= 1 - [p_{5.5} + p_{11}]$$

$$p(X \geq g) = p(X \geq L_{data} \cdot \frac{5.5}{4})$$
$$= p(R \geq 5.5 \text{ Mb/sec})$$
$$= p(R = 5.5 \text{ Mb/sec}) + p(R = 11 \text{ Mb/sec})$$
$$= p_{5.5} + p_{11}.$$  \hfill (3.30)

Using Equation (3.25)

$$g = E[X] - \frac{c}{p_{5.5} + p_{11}}.$$  \hfill (3.32)

As in Case 2 the value of $E[X], p_{5.5}$ and $p_{11}$ can be used to calculate the candidate value of $g$ which is then compared to the boundary conditions for this case to determine whether it indeed is the valid optimal value of $G^*$. If the boundary conditions are not met this value of $g$ is rejected.

**Case 4** $g \in (0, L_{data} \cdot \frac{2}{4}]$
Using arguments similar to Case 2 and Case 3 I have

\[ p(X < g) = p(X < 0) \]
\[ = p(R < 0) \]
\[ = 0 \] \hspace{1cm} (3.33)\]
\[ p(X \geq g) = p(X \geq 0) \]
\[ = p(R \geq 0) \]
\[ = 1. \] \hspace{1cm} (3.34)\]

Using in Equation (3.25)

\[ g = E[X] - c. \] \hspace{1cm} (3.35)\]

Again, as in Case 2 the value of \( E[X] \) can be used to calculate the candidate value of \( g \) which is then compared to the boundary conditions for this case to determine whether it indeed is the valid optimal value \( G^* \). If the boundary conditions are not met this value of \( g \) is rejected.

Thus, given the distribution of achievable data rates, one or more of the values as given by Cases 1-4 will yield a \textit{valid} value of the expected return from an optimal stopping rule \( G^* \) (which meets the requirements of the boundary conditions for that case too). Since the expected return from an optimal stopping rule is being maximized, the maximum value from the set of valid \( G^* \)s is selected as the expected return from an optimal stopping rule, \( G^* \). The optimal stopping rule is then given by Equation (3.13) as

\[ N^* = \min\{n \geq 1 : X_n \geq G^*\}. \]

In particular, since the payoff for a particular transmission data rate \( R \) is given by Equation (3.19), the expected return from an optimal stopping rule, \( G^* \), can be
extrapolated to the expected value of the optimal transmission data rate, \( R^\ast \) as

\[
R^\ast = \frac{R_{\text{base}}^2}{L_{\text{data}}} \cdot G^\ast
\]

where \( R_{\text{base}} \) is the base data rate of 2 Mb/sec and \( L_{\text{data}} \) is the size of a data packet in bits.

For IEEE 802.11 based wireless networks the set of feasible data rates is a finite sized set and it is not possible to select an arbitrary data rate. For example, in case of IEEE 802.11b standard the set of feasible data rates consists of 2, 5.5 and 11 Mb/sec and any other value of data rate (say 7.5 Mb/sec) is not feasible. Typically achievable data rates is a function of received \( SNR \) as given in Equation (3.9). Approximating that the achievable data rates and received \( SNR \) (in dB) are linearly related, given \( R^\ast \) the optimal threshold value of received \( SNR \), \( SNR_{\text{optimal}} \), is derived using Equation (3.9). For example if the optimal stopping rule gives the optimal transmission data rate as 7.5 Mb/sec the threshold signal to noise ratio, \( SNR_{\text{optimal}} \) is derived as

\[
SNR_{\text{optimal}} = \frac{11 - 5.5}{SNR_{11} - SNR_{5.5}} \cdot 7.5,
\]

where \( SNR_{\text{optimal}} \) denotes the threshold signal-to-noise ratio dictating whether a node should continue skipping or not. The optimal stopping rule, given received \( SNR \) becomes as below

If \( SNR < SNR_{\text{optimal}} \) \( \Rightarrow \) Keep Skipping

else Stop Skipping.

**Numerical Example of Optimal Skipping Rule**

Here I illustrate the application of the optimal skipping rule via a numerical example. Assume, that the channel conditions are such that the probability of data transmis-
sion at rate 11 Mb/s, 5.5 Mb/s and 2 Mb/s is given by 0, .5 and .5 respectively. Thus in Equation (3.24), \( p_{11} = 0, p_{5.5} = .5, p_2 = .5 \) and \( p_0 = 0 \). Let the length of data packet \( (L_{data}) \) be 1000 bytes and the length of RTS and CTS packet be 20 bytes each. Thus, Equation (3.24), becomes

\[
X = \begin{cases} 
4000\mu sec & \text{with probability } .5 \\
11000\mu sec & \text{with probability } .5 \\
22000\mu sec & \text{with probability } 0.
\end{cases}
\] (3.37)

and \( E[X] = 7500\mu sec \). The cost of channel measurement via RTS/CTS, \( c \), is

\[
c = 2 \cdot \frac{20 \cdot 8 + 20 \cdot 8}{2} + 10 = 340\mu sec.
\] (3.38)

Applying the optimal skipping rule:

**Case 1** \( g \in (L_{data} \cdot \frac{11}{4} = 22000, \infty) \)

From the optimal stopping it is deduced that \( g \notin (22000, \infty) \).

**Case 2** \( g \in (L_{data} \cdot \frac{5.5}{4} = 11000, L_{data} \cdot \frac{11}{4} = 22000] \)

From the optimal rule

\[
p(X < g) = p(X < L_{data} \cdot \frac{11}{4})
= p(R < 11 \text{ Mb/sec})
= 1 - p(R = 11 \text{ Mb/sec}) = 1
\]

\[
p(X \geq g) = p(X \geq L_{data} \cdot \frac{11}{4})
= p(R \geq 11 \text{ Mb/sec})
= p(R = 11 \text{ Mb/sec}) = 0.
\]
Using Equation (3.25) to solve for $G^*$

$$
g = g \cdot [1 - p(R = 11 \text{ Mb/sec})] \\
+ E[X] \cdot p(R = 11 \text{ Mb/sec}) - c
\implies g = g - c,
$$

which is not possible for non-zero $c$. Thus $g \notin (11000, 22000]$

**Case 3** $g \in (L_{data} \cdot \frac{2}{4} = 4000, L_{data} \cdot \frac{5.5}{4} = 11000]$

From Equation (3.32)

$$
g = E[X] - \frac{c}{p(R = 5.5 \text{ Mb/sec}) + p(R = 11 \text{ Mb/sec})} \\
= 7500 - \frac{340}{5} = 6820.
$$

This value is within the boundary region of this case, namely $(4000, 11000]$, thus this value of $g$ is a valid value for $G^*$.

**Case 4** $g \in (0, L_{data} \cdot \frac{2}{4} = 4000]$

From Equation (3.35)

$$
g = E[X] - c \\
= 7500 - 340 = 7160.
$$

This value is *not* within the boundary region of this case, namely $(0, 4000]$. Thus this value of $g$ is not a valid value for $G^*$.

Thus the optimal skipping rule gives the valid value of $G^*$ as $6820 \mu\text{sec}$. Using this in Equation (3.36), the optimal data rate is obtained as

$$
R^* = \frac{R_{base}^2}{L_{data}} \cdot G^* \\
= 3.41 \text{ Mb/sec}.
$$
Using this value of $R^*$ the value of $SNR_{optimal}$ is derived from Equation (3.9) as

$$SNR_{optimal} = \frac{5.5 - 2}{SNR_{5.5} - SNR_2} \cdot 3.41.$$  

The optimal stopping rule, given received $SNR$ is given as below

If $SNR < SNR_{optimal} \implies$ Keep Skipping

else Stop Skipping.

In the next section I discuss how the optimal skipping rule for MOAR can be implemented in practical wireless networks.

### 3.6 Implementation Issues for Optimal Stopping Rule for MOAR

The optimal stopping rule for MOAR which maximizes the expected payoff achievable via opportunistically skipping channels in search of a better quality channel is derived in Section 3.5.3. Equations (3.29), (3.32) and (3.35) give a set of candidate values of $G^*$. Among these candidate values the maximum value of $G^*$ satisfying the boundary conditions of the respective case is selected as the value of $G^*$ which maximizes the expected return from an optimal stopping rule. The selected value of $G^*$ is a function of the cost of channel measurement via RTS/CTS, $c$, and the distribution of the achievable data rates (Equation (3.23)) which in turn is a function of channel fading given by Equation (3.21). Given these two parameters, the optimal stopping rule is given by Equation (3.13). For a MOAR node to be able to infer the optimal stopping time in practical systems it is necessary that the node has knowledge of $c$ and the distribution of achievable data rates as given by Table 3.2, where $SNR$ is the signal to noise ratio given by Equation (3.20) and Equation (3.21). $SNR_2, SNR_{5.5}$
and $SNR_{11}$ denote the minimum threshold SNR to support transmission rates of 2, 5.5 and 11 Mb/sec respectively.

<table>
<thead>
<tr>
<th>$c$</th>
<th>The cost of channel measurement via RTS/CTS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Distribution of achievable data rates</td>
<td>$p_2$ probability that data Rate is 2 Mb/sec</td>
</tr>
<tr>
<td></td>
<td>= probability $SNR \mid SNR_{2}$</td>
</tr>
<tr>
<td></td>
<td>$p_{5.5}$ probability that data rate is 5.5 Mb/sec</td>
</tr>
<tr>
<td></td>
<td>= probability $SNR_{2} \mid SNR \mid SNR_{5.5}$</td>
</tr>
<tr>
<td></td>
<td>$p_{11}$ probability that data rate is 11 Mb/sec</td>
</tr>
<tr>
<td></td>
<td>= probability $SNR_{5.5} \mid SNR \mid SNR_{11}$</td>
</tr>
</tbody>
</table>

Table 3.2: Parameters needed by a wireless node to implement the optimal skipping rule for MOAR in practical systems

The cost of channel measurement via RTS/CTS, $c$, is a constant and for a fixed RTS/CTS packet size can be computed as in Equation (3.16). The other parameter required to implement the optimal skipping rule in practical systems is the distribution of achievable data rates. In particular, for IEEE 802.11b systems a MOAR node needs $p_2, p_{5.5}$ and $p_{11}$, the probability that the data rate is 2, 5.5 or 11 Mb/sec respectively. Alternatively, if the underlying distribution of signal to noise ratio, SNR and its parameters (mean, variance etc) are known, the nodes can compute $p_2, p_{5.5}$ and $p_{11}$ indirectly rather than requiring these to be provided explicitly. However, in practice, the parameters of the SNR distribution or the distribution itself may not be known a priori. Moreover, for mobile nodes, the parameters of channel fading distribution (and hence the distribution of data rates) may also change with time as the distance between the sender and the receiver changes. In such cases, in order to
make a skipping decision in accordance with the optimal skipping rule (Section 3.5.3) a node may need to estimate either the parameters of the underlying distribution of channel fading or the distribution of data rates.

In case the underlying distribution of the channel fading is known but the exact parameters of the distribution are unknown, a MOAR node can choose to estimate the unknown parameters. For example if it is known that the underlying channel fading has the Rayleigh or the Ricean distribution, but the mean and the variance (also the value of parameter $K$ for the Ricean distribution) are unknown, a MOAR node can sample the received SNR values during the first several DATA (and accompanying control) packets to estimate the unknown parameters. The problem of estimating unknown parameters of a known distribution from finite samples of that distribution occurs in a wide array of disciplines. Various point estimation techniques like the method of moments and maximum likelihood estimation (among others) [65, 66] have been proposed and well studied in literature. In particular, [67] compares the efficiency of different estimation techniques in estimating the unknown parameters for a Rayleigh distribution. However, estimating the unknown parameters of a Ricean distribution is computationally expensive [68]. Moreover, in certain scenarios the exact distribution of the received SNR may be unknown which makes estimating $p_2, p_{5.5}$ and $p_{11}$ infeasible.

Thus rather than estimating the underlying distribution of the received SNR I choose to directly estimate the distribution of achievable data rates by measuring $p_2, p_{5.5}$ and $p_{11}$ from samples of received SNR.

I propose a measurement based approach to estimate online the distribution of the transmission data rates required to make a correct optimal skipping decision. Each MOAR node transmits the first $N_{est}$ packets without channel skipping in an
effort to estimate \( p_2, p_{5.5} \) and \( p_{11} \). I denote \( N_{est} \) as the estimation window. Each transmitted data packet (and the accompanying control packets RTS/CTS/ACK) contribute towards the samples needed to estimate the needed parameters. The probability \( \hat{p}_R \), that the feasible data rate is \( R \) is estimated by

\[
\hat{p}_R = \frac{\sum_{i=1}^{i=N_{est}} 1(SNR_{R-1} < SNR_i < SNR_R)}{N_{est}},
\]

where, \( N_{est} \) denotes the size of the estimation windows over which \( p_R \) is being estimated, \( 1(\cdot) \) is the indicator function, \( SNR_i \) denotes the received signal to noise ratio for sample \( i \) and \( (SNR_{R-1}, SNR_R) \) denotes the SNR thresholds between which rate \( R \) is feasible.

After enough samples have been collected to estimate the distribution of the transmission rates within certain confidence, the MOAR nodes may start opportunistic channel skipping. Since the distribution of data rates may change over time due to user mobility, MOAR nodes continuously update the estimated values of \( p_2, p_{5.5} \) and \( p_{11} \) by using only the last \( N_{est} \) samples of the received SNR. In this way, MOAR is still able to perform well for scenarios where the channel conditions change (for example, due to mobility) at a time scale greater than the time required to accurately estimate the distribution of data rates. The accuracy of the estimation scheme described above depends on the size of the estimation window, \( N_{est} \). If the size of the estimation window is large then \( p_R \) can be estimated with greater confidence which in turns increases the accuracy of the optimal skipping rule for MOAR. On the other hand, a small estimation window can lead to an inaccurate estimate of \( p_R \) which in turn could reduce the throughput gains of MOAR. Thus there is an inherent tradeoff between the size of the estimation window and the throughput gains that MOAR offers.
In Section 3.7.1 I investigate the effect of estimation window size on the throughput performance of MOAR via simulations and suggest a suitable value of the estimation window size for which MOAR is able to extract maximal throughput gains available from opportunistic skipping.

3.7 Performance Analysis of MOAR

In this section, I use *ns*-2 simulations to evaluate the performance of MOAR as compared to OAR. Our methodology is to isolate the impact of each performance factor to the largest extent possible and then consider more complex scenarios to study the joint effects of numerous factors. I begin with a fully connected topology (where all nodes are within radio range of each other) and study the effects of node location, channel conditions, error in channel measurement and the effect of estimating the distribution of achievable data rates on the performance of MOAR. I then consider more general topologies consisting of a simple asymmetric topology and more complex random topologies. Our key performance metric is aggregate throughput while maintaining the same time share as IEEE 802.11.

All experiments use the fast fading model of Equation (3.2). In particular, I use the Ricean probability density (Equation (3.4)) implemented in the *ns*-2 extension [69]. In [69], a packet level simulation is used to model the short time-scale fading phenomenon using the procedure suggested in [4]. A pre-computed lookup table containing the components of a time-sequence fading envelope are modulated in frequency domain using the Doppler spectrum in Equation (3.5). Although the *ns*-2 extensions implemented in [69] result in an accurate simulation of the wireless channel for each individual flow, the fading components of channels for different flows are *identical*, a scenario not encountered in practice. This arises due to the
fact that the index into the pre-computed channel table is chosen based on the simulator's time instant, which is identical for all flows. Thus, to realistically model the wireless channel for multiple users in a manner consistent with [4], I modified the extensions of [69] such that channel lookup indexes are a function of the flow, time, and IEEE 802.11 channel. This allows us to accurately model independent fading suffered by the different frequency channels. As in [69], background noise is modeled with $\sigma = 1$.

The available rates for both MOAR and OAR, based on IEEE 802.11b, are set to 2 Mb/sec, 5.5 Mb/sec, and 11 Mb/sec, so that with OAR, nodes can respectively transmit 1, 3, or 5 consecutive packets depending on their channel condition. The values for received power thresholds for different data rates were chosen based on the distance ranges specified in the Orinoco™802.11b card data sheet. For the path loss component of the received power, the distance thresholds for 11 Mb/sec, 5.5 Mb/sec, and 2 Mb/sec are 100 m, 200 m, and 250 m respectively. As specified by the IEEE 802.11 standard, I set the rate for sending physical-layer headers to 1 Mb/sec for all packets. Each transmitter generates constant-rate traffic such that all nodes are continuously backlogged. Moreover, packet sizes are set to 1000 bytes and all reported results are averages over multiple 50-second simulations.

To accurately evaluate the performance of MOAR for complex random topologies it is necessary to model the effect of co-channel interference for the IEEE 802.11 standard so that the impact of simultaneous transmissions on different frequency channels by neighboring nodes can be incorporated. In Appendix A I model interference between different IEEE 802.11b channels using the characteristics of the transmitted signal and received filter response function and also modify the ns-2 simulator to incorporate the additional co-channel interference.
3.7.1 Fully-connected Topologies

Here, I study the various performance factors that impact the performance of MOAR in fully connected topologies in which all nodes are within radio range of each other. Such topologies are representative of a wireless LAN scenario. The performance factors I study are location distribution, Ricean parameter $K$, error in channel measurement and the impact of estimating channel distribution while employing the optimal skipping rule within MOAR. Finally, I combine all these factors to explore the performance of MOAR for random fully connected topologies.

Location Distribution

The opportunistic gain that can be achieved by skipping channels is dependent upon the temporal channel quality, which has two components, a random fading component and a constant line of sight propagation loss component. In this experiment, I study the impact of the node location distribution by considering a scenario where there is a single flow and the distance (and hence strength of the line of sight component) between the sender and the receiver is varied. The random channel fading is kept constant by setting the Ricean fading parameter, $K = 4$.

Figure 3.4 depicts the average throughput gain of MOAR over OAR as the distance between the sender and the receiver of a flow is varied. The throughput gain has two peaks corresponding to distance between the sender and the receiver of 100 m and 225 m respectively. This is due to the fact that the path loss component of the received power has distance thresholds for 11 Mb/sec, 5.5 Mb/sec, and 2 Mb/sec of 100 m, 200 m and 250 m respectively. Thus for distances less than 100 m, the average channel condition corresponds to a data rate of 11 Mb/sec, distances between 100 m and 200 m correspond to a data rate of 5.5 Mb/sec and distances between 200 m
Figure 3.4: Throughput gain of MOAR over OAR as a function of distance between the sender and the receiver node.

and 250 m correspond to a data rate of 2 Mb/sec. Whenever the two mobile nodes are close to each other, the line of sight component dominates resulting in minimal available channel diversity gains over and above what OAR can achieve. However, as the distance between the two mobile nodes approaches the thresholds where the average data rate is often switched, random channel variations become comparable with the line of sight component. This is the regime where MOAR is able to extract additional throughput gains. Finally, the relative heights of the peaks is due to the ratio of the constant overhead in switching channels to the difference in channel qualities found (2 to 5.5 or 11 Mb/sec vs. 5.5 to 11 Mb/sec) resulting in a larger peak for higher distances.
Impact of Ricean Parameter K

In this section I explore the effect of the Ricean parameter $K$ on the throughput performance of MOAR relative to OAR. For lower values of $K$ the contribution of the line of sight component to the received SNR is weaker, and hence overall channel quality is poor. With increasing $K$, the line of sight component is stronger such that the overall SNR increases (see Equation (3.4)) and a higher transmission rate is feasible. I study the effect of $K$ on the throughput gain of MOAR relative to OAR. To isolate the effect of $K$, I simulate one flow with the distance between the source and the destination fixed thereby keeping the line of sight component constant.

![Graph showing the throughput gain of MOAR over OAR as a function of the Ricean parameter K.

Figure 3.5: Throughput gain of MOAR over OAR as a function of the Ricean parameter K

Figure 3.5 depicts the average percentage throughput gain of MOAR over OAR.
versus the Ricean fading parameter $K$ for distance between the sender and the receiver fixed to 220 m, 150 m and 100 m respectively. 95% confidence intervals for 5 random simulation runs (each 50 seconds long) are also shown. Observe that MOAR outperforms OAR by 40% to 60% when the distance between the sender and the receiver is 220 m indicating that significant throughput gains can be obtained by opportunistically exploiting the temporal variations among the IEEE 802.11b channels. However, the throughput gain with increasing $K$ is dependent on the distance between the sender and the receiver. In particular, when the distance between the sender and the receiver is 100 m or 150 m, the throughput gain of MOAR over OAR decreases with increasing $K$. This is due to the fact that a larger value of $K$ represents a smaller variation in channel quality which reduces the probability that the channel conditions on one of the other IEEE 802.11 channels is better than the channel conditions on the home channel. Thus the opportunity to skip channels opportunistically decreases leading to a decrease in throughput gain of MOAR over OAR with increasing $K$.

On the other hand, when the distance between the sender and the receiver is 220 m, the throughput gain of MOAR over OAR increases with an increasing value of $K$. MOAR can skip channels opportunistically only after the initial RTS/CTS on the home channel takes place successfully. When the distance between the sender and the receiver is 220 m the line of sight component is already very weak and low values of $K$ (denoting high channel variance) makes the transmission of RTS/CTS on the home channel sometime impossible as the received power is below the threshold required to correctly decode packets. As $K$ increases, channel variance decreases and RTS/CTS on the home channel have a higher probability of being correctly received which allows MOAR greater opportunity to skip channels. Thus the throughput of
both OAR and MOAR increases with increasing $K$. Lower values of $K$ means that MOAR has lower probability of finding good channels. However, higher average channel quality provides increased opportunity to skip poor channels and find a higher data rate channel which dominates the fact that there is a lower probability of finding better quality channels. Thus the gain of MOAR over OAR increases for increasing $K$ rather than showing a decrease as one would intuitively expect and as is shown when the distance between sender and receiver is 100 m or 150m.

**Channel Measurement Error**

![Graph showing throughput loss of MOAR due to channel measurement error](image)

Figure 3.6 : Throughput loss of MOAR due to channel measurement error

I next study the impact of error in channel quality measurement on the performance of MOAR (I previously considered perfect channel measurement). I consider
the case that the measured channel SNR is the true SNR plus a Gaussian error process. Figure 3.6 depicts the performance impact of standard deviation of the measurement error. In particular, the figure shows throughput loss for MOAR with channel measurement error (as compared to MOAR with no measurement error) vs. the error’s standard deviation scaled to the mean SNR. The throughput loss is not significant (less than 7%) for standard deviations less than the mean SNR. In particular, MOAR still outperforms OAR for standard deviation of channel measurement error less than 1.5 times the mean SNR. However, as the severity of error increases, so does the loss in throughput, indicating that it is important in practice to develop techniques that can measure channel quality within reasonable error margins to fully exploit opportunistic throughput gains.

**Optimal Skipping Rule: Effects of Estimation**

I discussed the challenges involved in implementing an optimal skipping rule in actual systems in Section 3.6. In particular I proposed a measurement based scheme to estimate $p_R$, the probability that data rate $R$ is feasible. In this section, I study the impact of the size of estimation window ($N_{est}$, see Section 3.6) on the performance of MOAR and suggest a suitable value of the estimation window size in order to extract maximal throughput gain from MOAR. I consider a single flow with the distance between the sender and the receiver fixed to $d$. The random channel fading is kept constant with a Ricean parameter value of 3. The metric that I use to measure the estimation window size is number of packets. However, in realistic scenarios this metric is feasible only under the constraint that the packets constituting an estimation window are transmitted within a certain maximum time period representing the time under which the distribution of the achievable channel rates does not change.
Figure 3.7 : Effect of estimation window size on throughput gain of MOAR over OAR

For the results presented in this section it is assumed that this constraint is satisfied.

Figure 3.7 plots the average throughput gain (over 5 runs of 50 sec each) of MOAR over OAR versus the estimation window size (in packets), $N_{est}$, for different values of $d$, the distance between the sender and the receiver. For each value of $d$, for a small value of the estimation window, MOAR is not able to extract significant throughput gain due to opportunistic channel skipping. However, for the estimation window size greater than a critical value (for each value of $d$), MOAR outperforms OAR by 5%-30% depending on the distance between the sender and the receiver. The reason for this behavior is that for a smaller estimation window size, the proposed measurement based scheme to estimate the distribution of feasible data rates does not have enough number of samples to accurately estimate the
distribution correctly. Thus, in this regime the optimal skipping rule results in a conservative value of optimal skipping threshold which in effect causes MOAR to be conservative in channel skipping. Thus the throughput gain of MOAR over OAR is very small. However, for a larger estimation window size, the measurement based estimation scheme is able to estimate the channel rate distribution quite accurately which in turn implies that MOAR is able to aggressively skip frequency channels as dictated by the optimal skipping rule and hence MOAR is able to extract the maximal throughput gains available via opportunistic channel skipping.

Another observation that can be made from Figure 3.7 is that the critical value of the estimation window is dependent on the distance between the sender and the receiver. In particular, the minimum size of the estimation window for which MOAR outperforms OAR is 50 packets when the distance between the sender and the receiver is 100 m, 100 packets when the distance between the sender and the receiver is 225 m and less than 20 packets for all other distances between the sender and the receiver. This is due to the fact that the path loss component of the received power has distance thresholds for 11 Mb/sec, 5.5 Mb/sec, and 2 Mb/sec of 100 m, 200 m and 250 m respectively.\textsuperscript{15} Thus, when the distance between the sender and the receiver is either 100 m or 225 m the measurement based estimation scheme requires a larger sample size to accurately estimate the channel rate distribution. On the other hand, when the distance between the sender and the receiver is different from the threshold distances of 100 m and 225 m, there is less variability in the channel rate distribution and an accurate estimation of the distribution can be made in as

\textsuperscript{15}For distances less than 100 m, the average channel condition corresponds to a data rate of 11 Mb/sec, distances between 100 m and 200 m correspond to a data rate of 5.5 Mb/sec and distances between 200 m and 250 m correspond to a data rate of 2 Mb/sec.
few as 20 packets.

In practical systems, the distance between the sender and the receiver is either unknown a priori or can change due to node mobility. Thus it is important to set the value of the estimation window size such that MOAR is able to extract maximal gains from opportunistic channel skipping independent of the distance between the sender and the receiver. It can be seen from Figure 3.7 that for the estimation window size equal to 100 packets MOAR is able to achieve maximal throughput gain over OAR irrespective of the distance between the sender and the receiver. Thus I recommend that the minimum estimation window size be set to 100 packets to enable the optimal skipping rule for MOAR to extract maximal throughput gains via opportunistic channel skipping.

Random Fully-connected Topologies

Here I consider random topologies representative of a wireless LAN and consider a scenario where the mobile subscribers are uniformly distributed in a circular area with diameter 250 m. I fix the Ricean fading parameter to 4 and also set the size of the estimation window to 100 packets, as discussed in the previous section. Figure 3.8 shows the average percentage throughput gain of MOAR over OAR as well as the maximum and minimum values of the percentage gain for each number of flows. The curve labeled “Look-ahead” assumes that the channel state information for all the 11 channels is known a priori and thus flows need to skip at a maximum of one time to the channel with known higher rate than the present channel. This serves as an upper bound to the gain that MOAR can extract over OAR. I also implement the optimal skipping rule (as derived in Section 3.5.3) and plot the throughput gains of MOAR with optimal skipping over OAR.
Figure 3.8: Throughput gain of MOAR over OAR for random fully connected topologies

As discussed in Section 3.7.1, the opportunistic gain that MOAR can extract is dependent upon the distance between the sender and receiver of a flow. For a given random topology, some of the flows are located in a region where the opportunistic gain obtained by skipping channels is not significant. These nodes, besides contributing little to the net overall gain that MOAR can obtain, actually reduce the opportunistic gain for better located nodes. The reason for this can be attributed to the random nature of the MAC. Whenever the nodes with lower opportunistic gain access the medium, the nodes which are better located to exploit the opportunistic gain through channel skipping defer medium access. Thus the net opportunistic gain that can be obtained by exploiting channel diversity is reduced. However, on average MOAR still outperforms OAR by 14-24%. Also, the gain of MOAR with optimal
skipping is very close to the maximum gain achievable if the channel condition on all the 11 channels is known *a priori*. Thus, in realistic systems where channel state information on other channels may be unavailable, the optimal skipping rule can still enable MOAR to capture most of the performance gains available via opportunistic skipping.

### 3.7.2 Complex Topologies

In this section I study the performance of MOAR for more complex topologies where all nodes are not within radio range of each other. Unlike the Topologies studied in Section 3.7.1, in this section I study topologies which are representative of ad hoc networks. First I study the throughput gains offered by MOAR for Asymmetric Topology (Figure 3.9). Finally I study random complex topologies.

**Asymmetric Topology**

In systems with topologies that are not fully connected, i.e., all nodes are *not* within range of each other, nodes can have different probability of channel capture due to one node hearing an RTS or CTS that another node does not hear. This unequal channel access probability can result in large differences in throughput shares among nodes. This behavior is due to asymmetry in information available to each flow and is well documented in the context of IEEE 802.11 [17, 30] and was also discussed in Chapter 2.

An illustrative example of asymmetric information among nodes is depicted in Figure 3.9, in which the receiver of Flow A (node 2) is in direct radio range of Flow B, whereas the sender (node 1) has no knowledge of Flow B. As discussed in Chapter 2 and shown in Figure 3.10, Flow B obtains a significantly higher share of
the channel access time as compared to Flow A, namely 80% vs. 20% when using IEEE 802.11. This disparity in total share is attributed to the fact that Flow B can hear packets from the receiver of Flow A, and hence knows exactly when to contend for the channel. On the other hand, the transmitter of Flow A does not hear any packets from Flow B, and thus has to discover an available time-slot randomly; hence Flow A continually attempts to gain access to the channel via repeated RTS requests which in most cases result in doubling of Flow A’s contention window. As a result, the probability of Flow A capturing the channel is significantly less than that of Flow B.

In this section I show that in general topologies, even with asymmetric information, MOAR will still have throughput gain over OAR and at the same time complies with pure IEEE 802.11 in the sense that the relative throughput shares of Flow A and Flow B are still approximately same as in IEEE 802.11.

To isolate the effect of information asymmetry on the performance of MOAR in the experiment for Figure 3.9, I fix the distance between the transmitter and the receiver to 100 m for both Flow A and Flow B. Thus I ensure that the average
channel conditions for both Flow A and Flow B are kept to be the same. Further I set the Ricean parameter, $K$, to 3 and also set the value of the estimation window size to 100 packets (as discussed in Section 3.7.1).

![Throughput Graph]

Figure 3.10: Throughput of IEEE 802.11, OAR and MOAR for the asymmetric topology

Figure 3.10 plots the average throughput (in Mb/sec) (over 5 random simulation runs of 50 sec each) for Flow A and Flow B as also the total throughput for single rate IEEE 802.11, OAR and MOAR. The throughput share for Flow A is 23.14% for single rate IEEE 802.11, 15.85% for OAR and 22.88% for MOAR. Thus MOAR preserves the relative throughput share of IEEE 802.11 and OAR. However, the total throughput for MOAR is higher than that achieved by OAR which in turn is higher than that achieved by single-rate IEEE 802.11. In particular, MOAR achieves a throughput gain of 16.6% over OAR while still maintaining the same
relative throughput shares for the individual flows as OAR. Thus, both Flow A and Flow B benefit from opportunistic channel skipping and MOAR is able to provide a net throughput gain while maintaining similar time shares as IEEE 802.11 even in topologies which are not fully connected.

Random Complex Topologies

Here I consider random topologies representative of a wireless ad hoc network. In particular I consider a scenario in which nodes are uniform-randomly distributed in rectangular 1500 m by 1500 m which is greater than the transmission range of 250 m. To isolate the performance gains achievable via MOAR I disable multi-hop routing and all the flows are single hop flows. I fix the Ricean fading parameter to 3 and also set the size of the channel rate estimation window to 100 packets.

![Graph showing Throughput Gain of MOAR over OAR for random complex topologies.](image)

Figure 3.11: Throughput gain of MOAR over OAR for random complex topologies
Figure 3.11 shows the average percentage throughput gain of MOAR over OAR as well as the maximum and minimum values of the percentage gain (over 10 runs of 50 seconds each) for different number of flows. On average MOAR outperforms OAR by 18% to 28%, even in scenarios where not all nodes are within radio range of each other. Thus even in complex topologies representative of ad hoc networks, MOAR is able to exploit the throughput gain available via opportunistic skipping.

3.8 Integration of DWOP and MOAR

The goal of joint design of scheduling and medium access (Chapter 2) is to decide when a node gets to access the medium. In contrast, the goal of multi-channel opportunistic medium access is to efficiently utilize the medium, once access is granted, via skipping frequency channels in search of a better quality channel. Thus the two major contributions of this thesis, namely joint design of scheduling and MAC to achieve QoS and design of multi-channel opportunistic MAC, are complementary and can be implemented together and/or separately and together they form a unified framework for high performance wireless ad hoc networks.

In this section, I implement a subset of DWOP and MOAR together and explore the joint performance of the two protocols via simulations. In particular my goal is two-fold. First, I demonstrate that MOAR is still able to extract significant throughput gains when used in conjunction with a scheduling policy at the MAC layer. Second, I show that DWOP’s information sharing mechanism is consistent with MOAR’s opportunistic channel skipping. In particular, DWOP piggybacks priority indexes of queued packets to share information among mobile nodes. Consistent sharing of this information is critical to the packet ordering property of DWOP. Our goal is to demonstrate that opportunistic channel skipping via MOAR does not hin-
der information exchange among nodes and hence the long time scale packet ordering property of DWOP is not violated.

In the next section I explore the joint performance of MOAR and a subset of DWOP mechanisms.

### 3.9 Performance of Joint Implementation of DWOP-SCFQ and MOAR

In Chapter 2, I evaluated the ordering properties of DWOP by choosing FIFO as the reference scheduler mainly because short time scale ordering property is better illustrated via a FIFO scheduler. In this section, I focus on the average ordering property of DWOP when used in conjunction with MOAR. In particular, I integrate opportunistic multi-channel scheduling with the DWOP-Self Clocked Fair Scheduling (DWOP-SCFQ). DWOP-SCFQ seeks to emulate Self Clocked Fair Queuing (SCFQ) [34] in a distributed manner while utilizing the IEEE 802.11 MAC protocol and modulating each node’s backoff interval in proportion to the priority of its next packet to be transmitted. The priority of a packet is calculated as in [34], and depends on the weight of the node, the packet size and the system virtual time. Each node maintains a local virtual time which is updated whenever the node hears a transmitted packet on the medium by using the packet’s finish tag. Each transmitted packet carries its finish tag piggy-backed on the data packet and control packets.

I evaluate the joint performance of DWOP-SCFQ and multi-channel opportunistic scheduling for the asymmetric topology (Figure 3.9). I fix the distance between the transmitter and the receiver to 100 m for both flow A and Flow B. Thus I ensure that the *average* channel conditions for both Flow A and Flow B are kept to be the
same. Further I set the Ricean parameter, $K$, to 3 and also set the value of the estimation window size to 100 packets in case of MOAR (as discussed in Section 3.7.1).

![Throughput Graph](image)

**Figure 3.12**: Throughput of MOAR, DWOP-SCFQ and MOAR with DWOP-SCFQ for the asymmetric topology

Figure 3.12 plots the throughput for Flow A, Flow B, and the total throughput for the case when MOAR operates without DWOP-SCFQ, DWOP-SCFQ operates without MOAR and for the case when MOAR and DWOP-SCFQ operate together. MOAR without DWOP-SCFQ inherits the fairness property of IEEE 802.11 and the relative throughput shares of Flow A and Flow B are 24% and 76% respectively. On the other hand the ordering property of DWOP-SCFQ forces a more equitable share of throughput between Flow A and Flow B. However, as discussed in Chapter 2, the ordering property of DWOP-SCFQ comes at the cost of reduced net throughput.
MOAR operating in conjunction with DWOP-SCFQ inherits the fairness property of DWOP-SCFQ while obtaining a net higher throughput share for both flow A and Flow B which causes the total throughput of MOAR with DWOP-SCFQ to be 11% higher than the total throughput of DWOP-SCFQ without MOAR.

Thus, MOAR is still able to extract significant throughput gains when used in conjunction with a scheduling policy at the MAC layer and DWOP's information sharing mechanism is consistent with MOAR's opportunistic channel skipping.

3.10 Summary

In this chapter, I devised the Multi-channel Opportunistic Auto Rate (MOAR) protocol for wireless ad hoc networks. MOAR allows nodes to opportunistically skip frequency channels in search of better quality channels. Since the spacing between various IEEE 802.11 channels is greater than the coherence bandwidth, the channel quality on one of the other frequency channels may be better than on the current channel. Thus MOAR nodes are able to achieve a higher throughput by transmitting at a higher rate on better quality channels. To balance the tradeoff between the time and resource cost of channel measurement/channel skipping and the throughput gain available via transmitting on a better channel I also devised an optimal stopping rule for MOAR. I explored the performance of MOAR via extensive simulations and showed that MOAR achieves a consistent gain in throughput of 20% to 25% over current state-of-the-art multi-rate MAC protocols. Finally I evaluated the joint performance of MOAR and DWOP and show that MOAR is still able to extract significant throughput gains when used in conjunction with a scheduling policy at the MAC layer.
Chapter 4

Contributions and Future Work

In this Chapter, I conclude the dissertation by summarizing our contributions and presenting some of the interesting directions for future research to build on the research in this dissertation.

4.1 Contributions

The goal of this thesis is the design of medium-access mechanisms to enable future high performance wireless ad hoc networks. In particular this thesis has two main contributions.

First, I propose the Distributed Wireless Ordering Protocol (DWOP), a joint media access and distributed scheduling scheme designed to achieve a reference scheduling service order in wireless ad hoc networks. DWOP ensures that packets are serviced in the order as defined by a centralized reference scheduler. The key idea behind DWOP is that nodes share information regarding the priority index of queued packets by piggybacking the index on control and DATA packets. DWOP exploits this shared information within the IEEE 802.11 medium access framework with the goal of achieving a perfect service order. In contrast to related work on providing QoS for ad hoc networks, DWOP serves as a general framework to apply the wealth of packet scheduling service disciplines developed for wireline networks to wireless ad hoc networks thereby making it possible to achieve the desired goals of
fairness, throughput and delay targets and service differentiation in such networks.

Second, I propose the Multi-channel Opportunistic Auto Rate (MOAR) protocol to efficiently and fully utilization the scarce and variable wireless spectrum. MOAR exploits the presence of multiple frequency channels at the PHY layer to opportunistically skip frequency channels in search of a better quality channel if the current frequency channel is not of good quality. Since MOAR nodes transmit packets at a higher rate (on better quality channels), MOAR is able to achieve a net higher throughput as compared to state-of-art MAC protocols which exploit only the multi-rate capabilities of the PHY layer. MOAR is a completely distributed protocol which allows the sender and the receiver of a flow to mutually negotiate a skip decision via transmission of control packets. I also design an optimal skipping rule for MOAR which limits the number of time MOAR nodes skip in search of a better quality channel. The optimal skipping rule for MOAR is designed to balancing the time and resource costs of channel measurement/skipping for realistic systems with the potentially additional throughput gains available via continued skipping.

4.2 Future Work

In this section I identify several interesting research directions for future work.

- *Comparison of MOAR with other spectrum reuse protocols.*

The MOAR protocol aims to efficiently utilize the entire spectrum allocated to the IEEE 802.11 standard by skipping frequency channels opportunistically. However, MOAR by no means is the only possible method to utilize the available spectrum. For example, Multi-channel Medium Access Control (MMAC) [56] is a protocol that seeks to simultaneously utilize all the avail-
able mutually orthogonal frequency channels by load balancing users among
the three orthogonal channels in IEEE 802.11b wireless local area networks.
Similar protocols that use the entire available spectrum simultaneously by al-
locating different users to different parts of the spectrum can be devised for ad
hoc networks. Such schemes achieve a net throughput gain by enabling more
than one flow to transmit at the same time, irrespective of channel quality.
In contrast, MOAR is constrained so that only one active flow can use the
entire spectrum at any given time. In this way MOAR is able to achieve a net
throughput gain by ensuring that flows transmit only on good quality channels.

Both of the above described approaches to spectrum reuse achieve a net through-
put gain. However, without any detailed investigation it is hard to declare that
one scheme is better than the other. The fact that the gain achieved by both
the schemes is strongly correlated with the topology of the network under inves-
tigation further complicates any comparison. For example, protocols that reuse
the spectrum by allowing multiple flows to simultaneously transmit data need
an additional MAC layer mechanism to ensure that nodes serving multi-hop
flows are able to rendezvous with each other on the correct frequency channel
at the correct time. This additional mechanism could potentially negate the
throughput gains available from multiple simultaneous transmissions. More-
over, node distribution is a major factor which affects the gain achieved by
either of the two schemes.

Similarly there are other spectrum reuse techniques that can be designed to
increase the throughput of ad hoc networks by utilizing the entire available
spectrum. For example, mobile nodes could use a longer spreading sequence
(spread across the entire available spectrum) to obtain an increase in the net
processing gain of the spreading sequence. Thus network designers need to compare all approaches to spectrum reuse prior to deployment. However, in the absence of any protocol for ad hoc networks which enables spectrum reuse by enabling multiple simultaneous transmission, the comparison is currently infeasible. Therefore future work comparing different approaches to spectrum reuse is required in order to choose the best available spectrum reuse scheme for a given network.

- The performance of proposed DWOP and MOAR protocols in multi-hop topologies and interaction with routing layer.

Routing for multi hop flows in ad hoc networks has received critical attention in recent years and a number of routing protocols have been proposed within this context (e.g., DSR [70, 71, 72], DSDV [73], AODV [74], and TORA [75, 76]). In this thesis, I have focused on network topologies where all flows are single hop flows (which may or may not be within radio range of other similar flows) while evaluating the performance of the DWOP and MOAR protocols. Focusing on single hop flows allows us to isolate the performance improvements available via the proposed MAC enhancements from the effects of routing for multi-hop flows. This enables us to study the behavior of the proposed protocols in greater depth and also identify the various protocol parameters needed to extract optimal performance. However, it is important to study the interaction between various routing algorithms proposed for ad hoc networks and the protocols proposed in this thesis.

In particular, it would be interesting to study routing layer interaction with MOAR. The traditional technique used by most existing ad hoc routing algo-
rithms is to select minimum hop paths. These paths tend to contain long-distance links which have low effective throughput and reduced reliability. A number of QoS routing approaches for ad hoc networks have been proposed [77, 78, 79, 80] in the literature which seek to find a route from source to destination satisfying the end-to-end QoS requirements in terms of bandwidth or delay. However, none of the proposed approaches consider the multi-rate or multi-channel properties of the PHY layer. Thus there is a need to introduce routing layer protocols and/or mechanisms which take into account the multi-rate nature of wireless links while discovering route(s) between the sender and the receiver. A possible interesting direction for future work is to develop routing techniques that will take advantage of the multi-rate/multi-channel enhancements proposed in this thesis, by probing for routes that satisfy certain quality characteristics, such as highest capacity rather than just select minimum path hops.

Similarly the issue of priority based ordered scheduling raises some interesting questions within the context of routing in multi hop ad hoc networks. For example, assigning the correct priority for routing packets is an open problem.

- *Application of MOAR and DWOP to wireless networks other than ad hoc networks.*

Many hybrid networks aimed at marrying the ease of deployment and flexibility offered by ad hoc networks with the greater control offered by cellular networks have been proposed in the literature [81, 82, 83]. Further, [84] makes the case for *Transit Access Points (TAPs)* that form a multi-hopping wireless backbone with a limited number of *wired* ingress/egress points as a building block for
providing high-performance, scalable and widely deployed wireless Internet.

Most of the above proposed architectures for hybrid wireless networks have a special topology and/or the presence of a "local" centralized entity (e.g., the fixed TAPs in [84]) and other unique properties (e.g., the presence of multiple air interfaces). An interesting direction for future research would be to adapt the enhanced MAC layer protocols proposed in this thesis for the above proposed and other specialized wireless networks.

In particular, the key idea behind MOAR, namely opportunistically skipping channels in search of better quality channels is not limited to ad hoc networks and can be employed in a wide array of wireless networks. However, addressing the MAC layer mechanisms to capture the additional throughput gains in networks other than IEEE 802.11 enabled ad hoc networks remains a largely unexplored issue.

Scheduling in hybrid wireless networks is another promising area for future research. In particular, applying DWOP's information sharing approach for ordered packet scheduling can serve as a framework for providing QoS in a wide range of hybrid wireless network architectures.

- **Evaluation of the proposed MAC mechanisms within an experimental hardware platform.**

The ultimate test for any protocol is to deploy it in one of the environments for which it was designed and evaluate its performance to determine if its operation was successful. However, the proposed protocols in this thesis require the ability to not only control various MAC layer parameters but also the need to change the medium access mechanism. Unfortunately, due to unavailability
of a programmable implementation of the IEEE 802.11 MAC standards and the lack of such control in readily available IEEE 802.11 compliant cards makes the evaluation of an actual implementation of the proposed protocol currently difficult.
Appendix A

Modeling the Impact of Co-channel Interference in IEEE 802.11 Enabled Wireless Networks

In this section I model the interference on a particular frequency channel due to simultaneous transmissions on different frequency channels for IEEE 802.11b systems. Modeling co-channel interference between IEEE 802.11b frequency channels is necessary to accurately evaluate the performance of MOAR for complex random topologies (see Section 3.7.2) where neighboring nodes may be transmitting on different IEEE 802.11b channels at the same time.

A.1 Isolated Channel Response

The allocated spectrum in the 2.4 GHz band is from 2400 to 2483 MHz [7, 9, 8]. For North America, there are 11 channels, starting at 2412 MHz and spaced at an interval of 5 MHz each. Since each channel has an approximate bandwidth of 22 MHz, the channels are not orthogonal, e.g., channels 1-5 interfere with each other whereas channels 1 and 6 are practically non-overlapping since they are more than 25 MHz apart.

Each mobile node in a wireless ad hoc network modulates its signal via the same direct sequence spreading code known as the Barker sequence, given by:

\[ S = \sum_{l=0}^{10} s_l \psi(m - lT) \]  (A.1)

where \( S \) denotes the Barker sequence, \( \psi(t) \) is a simple square pulse, \( s_l \) represent the
chips in the Barker sequence, and $T$ is the symbol period in seconds. The modulated signal in (discrete) time domain is given by $m[n] = \sum_k b[k]S$, where $b[k]$ denotes the information bits. Assuming the information bits are independent and identically distributed (0’s and 1’s occur with equal probability), then the power spectral density of $m[n]$, $M(f)$ is given by

$$M(x(f)) = \left| \sum_{l=0}^{10} s_l \Psi(x) e^{-j\pi T x} \right|^2$$ (A.2)\]

$$\Psi(x) = \begin{cases} \frac{\sin(2\pi x)}{2\pi x} & x \neq 0 \\ 1 & x = 0 \end{cases}$$ (A.3)

$$x(k, f) = \frac{f - (2412 + 5(k - 1))}{BW}$$

Here $\Psi(x)$ is the Fourier transform of $\psi(t)$, $f = [2400, 2483]$ MHz denotes the ISM band frequencies, $BW = 22$ MHz denotes the null-to-null channel bandwidth, and $k = 1, 2, \ldots, 11$ denotes the channel number.

The modulated signal is then filtered at the IF stage of the upconverter. The IF filter for the prism chipset [85] has a 3 dB bandwidth of 17 MHz and a stopband 50 dB down at +/- 22 MHz. Based on the above characteristics, the magnitude of the filter frequency domain response is well approximated by

$$F(x) = \frac{1}{1 + (2.6x)^6}$$ (A.4)

Figure A.1 shows the power spectral density of the Barker code modulated signal before IF filtering and the filter response $F(x)$ at the IF stage. From the figure, it is clear that the IF filtering limits the effective bandwidth of the transmitted signal to approximately 22 MHz.
Figure A.1: Power spectrum of the received DSSS signal and the IF filter characteristic for channel 6

A.2 Co-channel Interference

Current IEEE 802.11b receivers are designed to detect and decode signals in their channel of operation, while considering signals from nodes on other frequency channels to be interference, much like that from other devices operating in the unlicensed band (cordless phones, Bluetooth, etc.). The level of perceived interference depends on two main factors: the extent of channel overlap and the channel conditions between the receiver and the interfering transmissions. For a receiver of channel $a$, which treats all interference as background noise, the total interference power can be approximated as

$$\eta = \text{background noise} + \sum_{i=0, i \neq a}^{10} P_i |h_{ia}|^2 O_{ia},$$  \hspace{1cm} (A.5)
where $P_i$ is the transmitted power by a node in channel $i$, $h_{ia}$ is the channel attenuation experienced by the signal, and $O_{ia}$ is the channel overlap between channels $i$ and $a$. There can be more than one node from channel $i$ which transmits at any given time (which leads to collision in that channel), in which case the total interference increases accordingly. The channel overlap $O_{ia}$ is obtained simply by calculating the total frequency overlap between channels $i$ and $a$. Assuming that the IF frequency response while receiving is the same as that while transmitting, the power spectrum of the received signal from channel $i$ in channel $a$ is given by

$$o_{ia}(x) = F(x(i))M(x(i))F(x(a))$$  \hspace{1cm} (A.6)
Integrating the overlap over the entire frequency response,

$$O_{ix} = \frac{1}{co} \int_{2200}^{2700} o_{ia}(x)dx,$$  \hspace{1cm} (A.7)

where \(co\) is the total power with complete overlap \((i = a)\) and is given by:

$$co = \int_{2200}^{2700} o_{aa}(x)dx$$  \hspace{1cm} (A.8)

Figure A.2 shows the total interference power received as a function of absolute channel difference \(|i - a|\). Even though channels with spacing 3, 4 and 5 are not necessarily orthogonal, their overlap is rather small due to the sharp roll-off of the filter \(F(x)\). Yet they can still contribute a large amount of interference if they are close to the receiver in channel \(a\) (known as the near-far effect [5]).

I incorporate the interference from transmission on other frequency channel (as calculated above) within the ns-2 simulator. In this way I am able to accurately account for interference between two simultaneous transmissions on neighboring channels.
Bibliography


