Scheduling and Fairness in Multi-hop Wireless Networks

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by

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Abstract

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Multi-hop wireless networks have been a subject of research for many years now. But recently, availability of inexpensive hardware has spurred interest in a new class of applications called *mesh networks* where multi-hop wireless routing is used to avoid the cost of deploying a wired backbone. However, both simulations and deployments based on the current generation of hardware (based mostly on the 802.11 standard) show very poor fairness between competing flows. In fact, the imbalance in throughput can be so severe that some nodes are completely shut out from sending any data at all. Some of this unfairness can be attributed to the lack of flexibility in the 802.11 Medium Access Control (MAC) layer to provide control over resource allocation. Recent work also suggests that Carrier-Sense Multiple Access (CSMA) based MAC protocols suffer from a more fundamental limitation due to problems caused by hidden terminals and asymmetric links.

In this work, we address this problem by making the following three contributions. Firstly, we build an Overlay MAC Layer (OML) that implements a time-slot based scheduler that works on top of inexpensive 802.11 based hardware without

any changes to the standard. The overlay approach also provides a lot of flexibility and can easily be adapted to specific applications and networks through simple software modifications. Secondly, we have developed a distributed algorithm that can efficiently detect interference by using passive measurements. While we use the inferred interference pattern to improve scheduling under OML, our algorithm can also be used in a number of other applications such as channel assignment, resource allocation and admission control. Thirdly, we have designed a transport layer algorithm called End-to-end Fairness using Local Weights (*EFLoW*) for providing global Max-Min Fairness. *EFLoW* uses an iterative additive-increase multiplicative-decrease (AIMD) based approach using only state from within a given node's contention region in each iteration. It can automatically adapt to changes in traffic demands and network conditions.

We have evaluated our system by conducting experiments in both ns-2 and a 30-node testbed built using commodity hardware. We show that our algorithms can improve fairness by over 90% with only a small cost (typically less than 10%) in efficiency for a wide variety of traffic patterns and network conditions.

Professor Ion Stoica
Dissertation Committee Chair

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Chapter 1

Introduction

Multi-hop wireless networks have been a subject of research for many years now. Initially, the primary motivating applications for such networks revolved around emergency response systems and military operations. The high mobility of the nodes and the lack to time makes it impossible to deploy a wired network in these scenarios. However, the proliferation of the 802.11 standard and the availability of inexpensive hardware has spurred interest in a new class of applications called "mesh networks." Many research projects [7, 22] and companies [4, 3] are considering the use of multi-hop wireless networks to avoid the cost of deploying a wired backbone. Such networks are particularly useful in the context of emerging markets and developing countries where little to no infrastructure is available [13].

However, both simulations and deployments based on the current generation of hardware (mostly based on the 802.11 standard) show very poor fairness between competing flows. In fact, the fairness problem can be so severe that some flows are completely shut out from sending any data at all [53, 29]. This translates to outages in connectivity in the case of mesh networks. As a result, the connectivity of a particular node to the Internet might be affected in presence of flows from other nodes. In this

work, we first try to understand what causes this unfairness and then propose a set of solutions that can prevent it. Our main contributions can be summarized as follows.

- We identify various problems with contention based Medium Access Control (MAC) layers in general, and the 802.11 standards in particular. We propose the Overlay MAC Layer (OML) which can solve these problems without the expense of developing new hardware or standards.
- OML, as well as several other applications require knowledge of which links' transmissions interfere with each other. Using passive measurements, we have developed an efficient distributed algorithm that can detect interference quickly and accurately.
- While OML and other experimental MAC layers provide control over resource allocation in a local neighborhood, providing end-to-end fairness requires global knowledge of network topology and traffic demands. Instead of computing the allocation directly, we propose an iterative increase-decrease based algorithm (*EFLoW*) which uses a very light weight control protocol.

1.1 Fairness in Wireless Networks

Both Medium Access Control (MAC) and the Transport layers play an important role in determining the fairness achieved by a wireless network. The MAC layer controls the allocation between nodes that directly compete with each other for access to the wireless medium. In this case, we say that these nodes belong to the same *contention region* and MAC decides which nodes are allowed to transmit at any given time. Our experiments show that there are two key factors that can cause unfairness at the MAC layer.

1. Certain nodes are allowed to access the medium more often that others. In the extreme case, some nodes are not allowed to send packets at all.

2. Even when nodes are allowed fair access to the medium, transmissions from on certain links are more likely to be successful. This is because the MAC may incorrectly allow interfering links to transmit at the same time and the resulting collisions affect some links more than others.

Ideally, the MAC layer must be able to schedule transmissions without collisions or underutilization of the channel due to idle times. It should also provide the higher layers control over how the medium is shared between different nodes in a single contention region. However, the MAC layer is unaware of end-to-end flows which can only be seen from the transport layer or above. Therefore, it is the responsibility of the transport layer to use the MAC layer efficiently to achieve a good allocation to *flows*. This includes making sure that the allocation at each hop is the same and that flows aren't constrained to the same low throughput as other flows that are bottle-necked elsewhere in the network.

1.2 Fairness at the MAC Layer

The popularity of the 802.11 standard has made it the de facto choice for developing and deploying not only mesh networks, but a multitude of other new applications. Despite making deployment easier, the 802.11 protocol does pose serious limitations in addressing the different demands of these emerging applications. The 802.11 MAC protocol was carefully engineered for the wireless LAN environment [45] and many of the underlying assumptions may not hold in the new environments. Several problems have been reported in earlier research [31, 29]. Our measurements indicate that such problems are indeed very common in multi-hop and heterogeneous environments. In

particular, we show that link asymmetry or hidden terminals can either cause (a) poor fairness, sometimes even shutting off all flows through a node or (b) excessive collisions, leading to poor performance.

Two main approaches have been proposed in the literature to address these problems. The first approach is to build workarounds in the routing or transport layers to avoid the cases where the MAC layer performs badly [28, 66, 7, 22]. The second approach focuses on replacing the MAC layer with new protocols [11, 19] and standards such as 802.16 (WiMax) and 802.11e. The first approach is more easily deployable as the functionality of both the routing and transport layers is fully implemented in software, and thus relatively easy to modify. However, the 802.11 MAC layer suffers from certain limitations, such as unfairness due to asymmetric interference, that cannot be fully addressed through changes only to the higher layers. In fact recent research indicates that CSMA-based MACs in general are susceptible to certain types of *information asymmetry* which makes them ill-suited for multi-hop networks [29]. In contrast, the second approach is far more powerful since modifying the MAC layer can directly address all the 802.11 limitations, but it is much harder to implement and experiment with. The MAC layer is implemented partly in hardware, partly in firmware, and partly in the device driver of the Network Interface Card (NIC). Thus, changing the MAC layer can require hardware and firmware changes, a highly expensive proposition.

In this work, we propose a third approach that combines advantages of changing the MAC layer with the ease of deployability of modifying only the higher layers. We propose the design of an Overlay MAC layer (OML) on *top* of the 802.11 MAC layer which does not require any changes to the hardware or the 802.11 standard. Our approach is inspired by the success of the "Overlay" networks, which have been used in the past few years to study new network protocols and implement new functionality without any modifications to the underlying IP layer. In the context of the

MAC layer, using an overlay offers the following three advantages. First, it provides an immediately useful system which can be used in 802.11 networks while waiting for newer standards to become more widespread. Second, the additional flexibility of having the MAC layer in software allows better integration with routing and application requirements. Third, it allows research on new protocols to be conducted on any of the numerous, already-deployed 802.11 testbeds. However, these advantages do not come for free. Similar to the limitations of overlay networks compared to changes in the networking layer itself, OML also suffers some additional overhead compared to direct changes at the MAC layer. In addition, the design of OML is limited by the interface exposed by the 802.11 MAC layer. For example, OML cannot carrier sense the communication channel since 802.11 network cards do not typically export the channel status to higher layers. Despite such limitations, we believe that the advantages of the our overlay approach make it a valuable alternative to modifying the MAC layer.

OML uses loosely synchronized clocks to divide the time in equal size slots, and then uses a distributed algorithm to allocate these slots across the competing nodes. In addition to preventing unfavorable interaction between senders at the underlying MAC layer, OML also allows users to implement application-specific resource allocation in the same way overlay networks allow application-specific routing. The slot allocation algorithm of OML, called Weighted Slot Allocation (WSA), implements the weighted fair queuing policy [21], where each of the competing nodes that has traffic to send receives a number of slots proportional to its weight. However, the net allocation to each node may be further limited by the performance of the underlying MAC within each time-slot. But if OML has an accurate picture of which links interfere with each other, it can eliminate almost all contention at the 802.11 layer. Thus OML will have very good control over the allocation since each node will transmit continuously in every time-slot assigned to it.

The interference pattern used by OML can either be obtained using heuristics (based on the network distance between nodes) or can be dynamically inferred from passive measurements using an algorithm we develop later in this work. We have implemented OML in a simulator and two test-beds. Our results show that besides providing better flexibility than the 802.11 MAC layer, OML also improves throughput and predictability by minimizing losses due to contention.

1.3 Detecting Interference between Links

The problem of determining the interference pattern of a network can be informally stated as follows: given any subset of links, what is the the throughput on each link when there are simultaneous data transfers on all of these links? Knowing the exact interference pattern has a wide range of applications. In the case of OML, it allows us to know which links can be scheduled in the same time-slot without loss of performance at the underlying MAC layer. It is also a prerequisite for efficiently solving many other problems such as channel assignment [52], resource allocation [63], admission control [42] and so on. We can obtain the interference pattern by performing active measurements, but this can take a very long time even in small networks. Measuring all subsets of links results in a combinatorial explosion which is usually avoided by considering only pairs of links at a time [46]. However, even in this case a network with n nodes can have $O(n^2)$ links which require $O(n^4)$ measurements.

For the above mentioned reason, most previous work [11, 10] (including our initial implementation of OML) has relied on heuristics to approximate the interference pattern. Typically, these heuristics are based on either signal strength measurements or the physical or network distance between nodes. However, these heuristics do not work well across all different types of wireless networks. Using physical distance does not work well in indoor environments where the location and the type of walls can have

a huge impact on signal propagation. Using network distance (number of hops) works well only in dense networks where all nodes use omni-directional antennas. Using signal strength does not account for multi-path fading which can have a significant impact on performance while using certain types of modulation.

In this work, we propose a new method which seeks to measure the interference pattern more accurately than heuristics by using passive measurements. Since we *use existing traffic only* to determine interference, the additional overhead of our measurements is very low. We try to quickly detect interference as it occurs during normal operation of the network by taking advantage of the fact that our MAC consists of two separate layers; OML and the underlying MAC. First, the underlying MAC provides protection against collisions and inefficiencies while we try to determine if a given pair of links interfere or not. Second, once we know that the links do interfere, OML can then make use of this information to improve performance and fairness by not allowing the links to transmit at the same time. Our experiments show that our algorithm can accurately (typically less that 10% false positives or negatives) determine the interference pattern for links that are in use in less than a minute for a wide variety of traffic and network conditions.

1.4 Fairness at the Transport Layer

A number of previously proposed MAC layers, including OML, allow the higher layers control over how transmission opportunities are shared between nodes in a single contention region by assigning weights to each node. However, the MAC layer is oblivious of end-to-end flows; it is the job of the higher layers to compute these *local* weights in such a way that the desired allocation is achieved. While this is straight forward in the case of wired networks, it is known to be an NP-complete problem in multi-hop wireless networks [28] due the interaction between flow constraints (the

allocation at each hop of a flow must be the same) and interference constraints (no two competing nodes can transmit at the same time).

Previous work focusing on end-to-end fairness at the transport layer [28, 66] has typically relied on a centralized coordinator to compute the global allocation for all flows. This allocation depends on both the capacity of links in the network and the configuration and bandwidth demands of flows. In a mesh network, changes in either traffic demands or network conditions can trigger frequent updates (which require communication with the coordinator) for a number of reasons. Firstly, measurement studies [7] have shown that the capacity of links, which depends on the current Signal-to-Noise Ratio (SNR), varies on short time scales and is hard to predict. Secondly, the workload primarily consists of HTTP flows which arrive and depart frequently. Finally, in the case of multi-hop flows, we must ensure that the allocation of bandwidth at each hop is consistent. Thus, changes in capacity or demands at any intermediate hop of a flow will trigger changes at every hop in the flow. This leads to a cascading effect and can quickly lead to reconfiguring the entire network.

In this work, we develop a distributed transport layer algorithm called *End-to-end Fairness using Local Weights* (*EFLoW*), which can be used in conjunction with any MAC scheme with support for weighted-fair allocation within a contention neighborhood. We show that by using a simple additive-increase multiplicative-decrease algorithm to compute these weights, we can obviate the need for an expensive control protocol or global reconfiguration. Our control protocol is lightweight and involves only exchanging information between nodes in the same contention region. Under certain assumptions, we show that *EFLoW* converges to a Max-Min fair allocation of bandwidth to flows.

1.5 Organization

The rest of this dissertation is organized as follows. In Chapter 2, we discuss relevant previous work in various areas including fairness at the MAC and transport layers, measurement studies and test-bed deployments. Chapter 3 presents the design and evaluation of OML. Next, we show how we can obtain the interference pattern using passive measurements on top of OML in Chapter 4. Chapter 5 shows how can build on top of these two approaches to provide end-to-end fairness to flows by using *EFLoW*. Finally, we present our conclusions and discuss open issues and possible avenues for future work in Chapter 6.

Chapter 2

Previous Work

In this Chapter, we provide a brief overview of related work in various areas. We start with discussion of proposals targeted at the MAC layer in Section 2.1. Next, in Section 2.2 we take a look at various methods for determining the interference pattern of a wireless network. Section 2.3 gives a summary of work addressing the transport and higher layers. Finally, we look at other related work such as deployment studies and overlay networks in Section 2.4.

2.1 MAC Layer

There are several MAC layer proposals that address fairness amongst directly competing nodes. There are several ways to achieve this depending on whether the MAC is contention-based or time-slot based. In addition to these proposals, the 802.11 MAC protocol in particular has received a lot of attention recently; a number of projects have been aimed at understanding its performance and limitations.

Contention-based algorithms: he allocation in CSMA-based MACs is determined by how nodes choose their back-off timers. A number of authors [14, 17, 59] have

proposed back-off algorithms with support for weights. These protocols are complimentary to *EFLoW*; they can be extended to provide end-to-end fairness by using the *local weights* computed by *EFLoW*. However, the accuracy of allocation under such schemes can suffer due to collisions or excessive back-off. In fact, recent work [29] shows that all contention-based MACs are susceptible to a generic co-ordination problem in multi-hop networks. In practice, the resulting inefficiencies may be severe enough to cause flow throughputs to span several orders of magnitude.

Time-division multi-plexing(TDMA): In TDMA approaches, time is divided into slots and and only nodes that don't interfere with each other are allowed to transmit during a given slot. The schedule is pre-computed in a centralized or distributed manner depending on the approach. Examples in literature include [11, 10]. Given accurate knowledge of the interference pattern, these approaches are capable of eliminating the hidden terminal problem and other inefficiencies faced by CSMA MACs. However, the schedule has to be recomputed every time the quality of a link changes or when flows arrive or depart.

802.11 MAC Limitations: Many researchers have reported problems with the 802.11 MAC protocol. Heusse *et al.* have described how senders with heterogeneous data rates can affect the system throughput adversely [31]. The Roofnet and Grid projects [7, 1] have reported a variety of problems with 802.11 multi-hop testbeds such as low throughput and unpredictable performance. We add to the set of the problems reported in these studies, by showing that asymmetric interference at the MAC layer can cause significant unfairness.

2.2 Detecting Interference

Every wireless MAC protocol has to implicitly deal with the issue of interference in one way or another. CSMA-based approaches use carrier-sensing internally to determine if two transmissions would interfere. However, in most TDMA approaches the interference pattern has to be specified externally. More recently, there have also been some attempts at measuring and understanding interference outside the context of MAC design.

Geographic partitioning: One of the simplest and earliest methods for reducing interference in wireless networks makes use of the fact that if we partition the network into non-overlapping geographic zones, we can bound the distance between links based on which zones they belong to. Thus, by carefully choosing the size of each zone we can ensure that links can only interfere with other links in either their own zone, or in zones that are adjacent to their zone. This approach is commonly used in cellular networks where the base stations serve as "anchors" to aid in this partitioning. But in the case of mobile and ad-hoc networks, there is no straightforward way to achieve this. Also, the number of channels available in ISM[1] bands is typically much lower than in licensed bands that cellular networks use *e.g.,* there are only 3 non-overlapping channels available for 802.11b and 802.11g. This approach tends to be very conservative in terms of reusing channels and hence may not be appropriate for such networks.

Heuristics based on physical distance: Several works have suggested that the interference range can be approximated by a distance threshold *e.g.,* twice the communication range. While this can serve as a very useful assumption in modeling and simulations, it is very difficult to use in practice. Firstly, estimating the distance accurately requires additional hardware support which may not always be available.

[1]The Industrial, Scientific and Medical band allocated by the FCC for unlicensed radios.

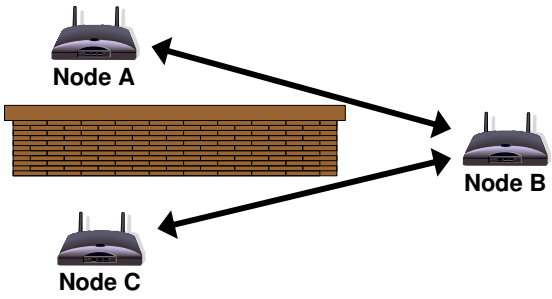


Figure 2.1. In the presence of RF-opaque obstacles, the 2-hop interference model can overestimate interference *e.g.,* between nodes A and C.

Secondly, though some distance-based models work well in open spaces, they may not be applicable to indoor and urban environments with a large number of obstacles.

Heuristics based on network distance: Network distance has often been suggested as an alternative to physical distance to bound the interference range. For example, one form of the *two-hop* heuristic states that if the senders are more than two network hops away from each other, their transmissions will not interfere. More sophisticated models might even take into account the quality of the links between the nodes in terms of other metrics like loss rate and ETX [7]. These models do not work well in networks with a low density of nodes; in such networks there may not always be a one-hop neighbor that is capable of forwarding to other interfering nodes. Another drawback with this model is that it can overestimate interference in the presence of opaque obstacles or in the presence of directional antenna as shown in Figures 2.1 and 2.2.

Heuristics based on Signal Propagation Models: This approach seeks to explain how communication and interference happen based on low-level modeling of signal propagation. Assuming a particular model, we first collect all available data (signal strengths and loss rate across all links, limited known interference data etc.)

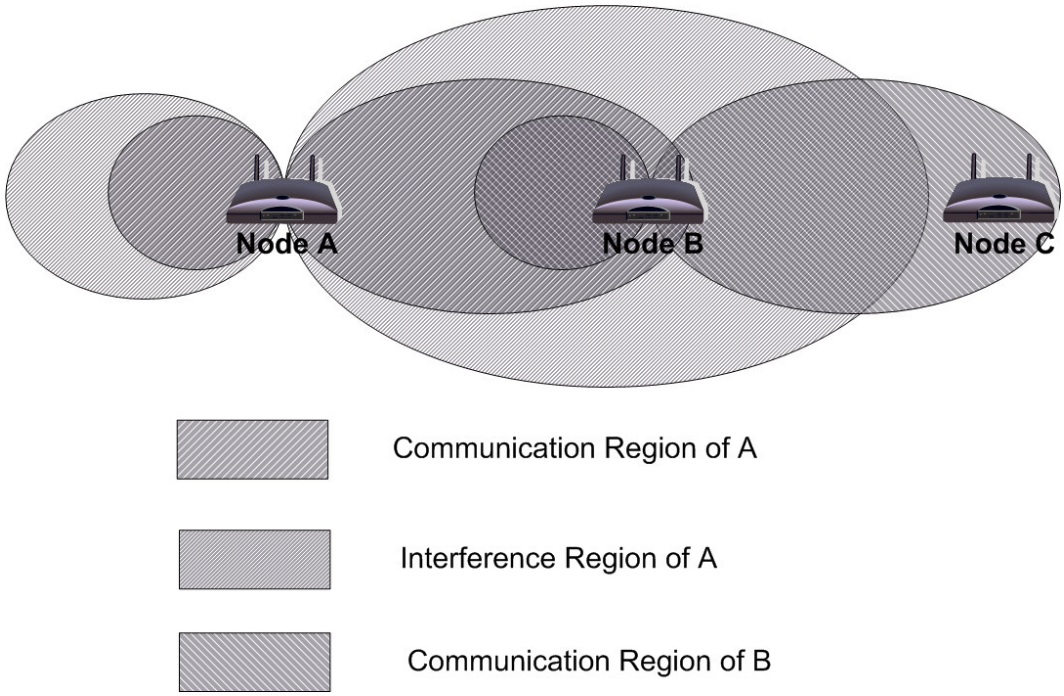


Figure 2.2. If nodes are equipped with directional antenna, the interference-range does not correlate very well with network distance. Node C is withing 2 hops of node A, but is outside its interference range.

and try to fit them to a model. The resulting model can then be used to make predictions about unknown quantities. While this is a very promising approach, the large number of models available for wireless signal propagation, and the large number of parameters these models use means that some tweaking might be needed for each individual network. This method was first suggested by Reis et. al. in [55], and further improved by Qiu et. al. in [49]. However, both theses approaches require active measurements where each sender is allowed to transmit in isolation. The results of these measurements have to be aggregated at a centralized coordinator. In this paper we do not rely on the availability of a good model for any given network. Also, we are able to seamlessly integrate the collection of measurement data into the normal operation of the network, including an application (scheduling) that uses our interference measurements.

Exhaustive Measurement: Padhye et. al. propose a measurement based ap-

proach in [46]. By using broadcast packets, they are able to reduce the number of required experiments from $O(n^4)$ to $O(n^2)$ in a network of size n. However, even $O(n^2)$ can be pretty large for certain networks, and it is not clear when we can schedule these measurements in a network that may be changing slowly over time.

Interference from External Sources: A number of authors have studied the impact of external interference on 802.11 networks [30, 37]. Solutions are typically based on frequency hopping either at sub-packet timescales at the physical layer (also known as spread spectrum techniques) [54], or at longer timescales at the higher layers [30]. Such techniques are orthogonal to our work. Since nodes that use different schedules to switch frequencies cannot communicate with each other, all nodes in a single network are usually synchronized to the same schedule. Thus, links in the same network will still interfere with each other.

2.3 Transport and Higher Layers

End-to-end flows are only visible from the transport layer and above; hence these layers are responsible for providing end-to-end fairness between competing flows. There have been several earlier approaches that address fairness at these layers. They differ in many ways including the amount of support they require from the MAC layer, the types of networks, traffic patterns and routing protocols they can accommodate, or whether they are targeted at TCP in particular or at all traffic in general. We categorize these approaches based on whether they provide end-to-end fairness, and if so, whether they use a centralized coordinator or not.

Local Fairness: In [23], the authors propose an adaptation of the fair queuing algorithm where each node adds its virtual time to the header of each packet it sends. This approach works well for the wireless-LAN environment where all nodes can hear

each other. However, for multihop networks, since a given node can be part of a large number of contention regions, it is not clear which virtual time to use at that node.

Fairness with a Centralized Coordinator: A number of authors have proposed mechanisms systems where a centralized coordinator controls allocation to all flows in the network. For example, in [66], Yi et al. propose a mechanism to limit the sending rate of TCP flows by delaying the ACKs at intermediate nodes. However, the rate for each flow has be computed in advance which requires global knowledge of network state.

Sridharan and Krishnamachari develop efficient algorithms for computing the max-min fair allocation at centralized coordinator in [60] and [61]. By formulating the problem as a linear program, they are able to compute the allocation within a few iterations for tree-based traffic patterns with only one sink. They also show how to compute a time-slot based schedule for this restricted traffic pattern once the allocation is computed [60].

Inter-TAP Fairness Algorithm (IFA) is a scheme developed by Gambiroza et al. in [28]. They propose a new definition for fairness based on *ingress aggregation* constraints, but only show how to compute the allocation for a very restricted traffic pattern; they use a parking-lot scenario with a chain of nodes where all nodes are trying to reach the gateway at one end. Also, their ingress aggregation constraints can sometimes prevent nodes the available bandwidth efficiently. When there are two flows originating at a node, one experiencing very little contention and the other a lot of contention, the definition compares the total throughput of both flows with that of other nodes. This introduces an artificial dependence between the two flows and prevents the node from sending data on the flow experiencing little contention so that its other flow is not starved by the network.

Distributed End-to-end Fairness: Recently, there have been a number of pio-

neering studies on congestion control and fairness in sensor networks [24, 51, 15]. However, all these approaches only work with a restricted traffic pattern. They also focus only on how to compute the fair allocation and do not address the scheduling problem.

Lu et al. propose a distributed mechanism to coordinate the back-off window between the nodes in a contention-based MAC to achieve end-to-end fairness[41]. However they assume that the interference range and the carrier-sense range are the same; recent measurement studies [46] show that this is not always true. Their analysis also assumes that the effect of packet losses due to collisions is negligible; thus their scheme is also susceptible to the problems of CS-based MACs mentioned earlier.

2.4 Other Related Work

Mesh Network Deployment: The GRID [1] project at MIT was one of the earliest research projects aimed at deploying 802.11-based multi-hop wireless testbeds. It started as indoor test-bed with tens of nodes distributed in a single floor of an office building. An outdoor network that spans several city blocks using rooftop antenna [7] was also deployed later. Our software environment is very similar to that used by the GRID project; we were able to use their indoor test-bed for some of our experiments. In recent years, a number of other similar test-beds have been deployed [22, 55] for research purposes.

Click Modular Router: Click is a software architecture for building flexible and configurable routers that has been under active development at MIT and UCLA since 2000. A Click router is assembled from packet processing modules called *elements*. Individual elements implement simple router functions like packet classification, queu-

ing, scheduling, and interfacing with network devices. A router configuration is a directed graph with elements at the vertices; packets flow along the edges of the graph. Our system is implemented on top of Click by adding several new elements that handle packet scheduling, clock synchronization, slot allocation etc. Click (and hence our system) can either be compiled as a Linux kernel module or as a user-level executable that interfaces with the kernel using the *TUN/TAP* device driver.

Distributed Slot Allocation: The Weighted Slot Allocation (WSA) algorithm that we propose for OML is roughly based on the Neighborhood-aware Contention Resolution (NCR) protocol, which was previously proposed by Bao and Garcia-Luna-Aceves [11, 10]. WSA extends NCR in two aspects. First, unlike OML, NCR assumes that the interference graph in a multi-hop network consists of isolated, easily identifiable cliques, *i.e.,* if node A interferes with B and B with C, then A interferes with C. Second, while NCR uses *pseudo-identities* to support integer weights, WSA can support arbitrary weights which is a key requirement for building *EFLoW* on top of OML.

Time slots above MAC: In [32] Hohlt *et al.* propose a two-level architecture for controlling the radios in a sensor network. The first level can be any MAC protocol, whereas the second level called Flexible Power Scheduling (FPS) is used to turn the radio on and off to save power. At a high level, this architecture is similar to OML, but there are key differences in the functionality provided by the second layer of FPS and OML. These differences stem mainly from the fact that our goals are quite different: while the main goal of FPS is to reduce power consumption, the primary goal of OML is to improve throughput and fairness.

The functionality implemented by the first level of FPS is to prevent collisions, while the functionality of the second level is to control power consumption. In contrast, both levels in OML aim to prevent collisions and improving throughput. The

reason for a second level in OML is to overcome the lack of flexibility and to work around the cases where the underlying MAC does not perform satisfactorily.

The algorithm used by FPS has to coordinate time slots only among nodes that communicate with each other, while OML has to take into account all nodes that interfere with a given node. FPS assumes a many to one communication pattern with each source sending periodic data, while OML is intended to work for any communication pattern. However, the generality of OML comes at a price; it may not suitable for use in sensor networks. With OML, receivers have to listen all the time and our current implementation of WSA uses floating point computations.

Overlay Networks: OML is similar in spirit to the overlay network solutions that aim to improve routing resilience and performance in IP networks [8, 9, 18, 62]. Overlay networks try to overcome the barrier of modifying the IP layer by employing a layer on top of the IP to implement the desired routing functionality. Similarly, OML runs on top of the existing 802.11 MAC layer, and its goal is to enhance the MAC functionality without changing the existing MAC protocols.

Chapter 3

Overlay MAC Layer (OML)

In this Chapter, we take a closer look at the Overlay MAC Layer. We begin by explaining the need for OML in Section 3.1. Next, we describe our design and algorithms in detail in Section 3.2. Section 3.3 gives a brief overview of our implementation. Finally, we present our results in Section 3.4.

3.1 Motivation

In this Section, we motivate our approach of building an Overlay MAC Layer. We use the 802.11 MAC as a starting point since it has emerged as the most popular choice for implementing wireless data networks. Briefly, our motivation stems from the following three factors:

1. Due to *asymmetric interaction* between flows and an *inflexible default allocation policy*, the efficiency and fairness of the 802.11 MAC suffer in a number of scenarios.

2. Solutions that only modify layers above the MAC, though applicable in certain

cases, are of limited use in addressing certain undesirable interactions at the MAC layer.

3. The additional flexibility, the low cost and the immediate deployability of an overlay solution makes it an attractive alternative for developing a new MAC layer or modifying an existing one.

Next, we substantiate the first two factors by conducting experiments on a six node wireless testbed using 802.11a radios. In Section 3.1.1, we illustrate the limitations of 802.11 and in Section 3.1.2 we argue that these limitations cannot be fully addressed at layers above the MAC layer.

3.1.1 Limitations of 802.11 MAC Layer

In this Section, we illustrate two specific limitations of the 802.11 MAC protocol using simple experiments:

1. *Asymmetric interactions:* Interference between two flows either at the senders or at the receivers can cause one flow to be effectively shut-off.

2. *Sub-optimal default allocation:* As other researchers have pointed out [31], we show that the default allocation of the transmission medium by the MAC layer fails to meet the requirements of some applications.

3.1.1.1 Effect of Asymmetric Interaction

In this Section, we take a closer look at the impact of interference on performance at the MAC layer. Interference has often been cited as being the cause for poor performance in other testbeds [7]. Here, we try to understand and quantify the effect of interference through experiments on a multi-hop testbed. To avoid any

complex interaction between the MAC, routing, and transport layers, we restrict our experiments to two simultaneous 1-hop UDP flows. Figure 3.1 shows the location of machines in our testbed in relation to the floor plan of our office building. We also show the signal strength of each link in either direction as reported by the device driver of the wireless card. The topology of the testbed resembles a chain and we study how simultaneous transmissions along two links in the chain interfere with each other.

Asymmetric Carrier Sense: We conduct our first set of experiments using broadcast packets only. This allows us to better understand the interaction between *senders*, as broadcast communication abstracts away the ACKs and packet retransmissions at the MAC layer. We make the two senders continuously send broadcast UDP packets and measure the sending rate of each sender averaged over 1 minute. The sending rate of each node depends only on whether it can carrier sense transmissions from the other node.

We conducted this experiment for each of the 15 pairs of nodes in our six node testbed. As expected, nodes that were far away (*e.g.,* nodes 1 and 5 in Figure 3.1) did not carrier sense each other and were able to simultaneously send at about 5.1 Mbps. When the nodes are close to each other (*e.g.,* nodes 2 and 3), they carrier sense each others' transmissions, and hence share the channel capacity to send at about 2.5 Mbps each. However, in *three* cases we found than one of the nodes was transmitting at more than 4.5 Mbps, while the other was transmitting at less than 800 Kbps. The only possible explanation for this asymmetric performance is that one sender can carrier sense the other, but not vice-versa. This can impact all configurations of flows where both senders are active, irrespective of who the receivers are.

Asymmetric Receiver Interaction: Now, we study the interference at the receiver. To eliminate the effects of sender (carrier sense) interference, we only consider

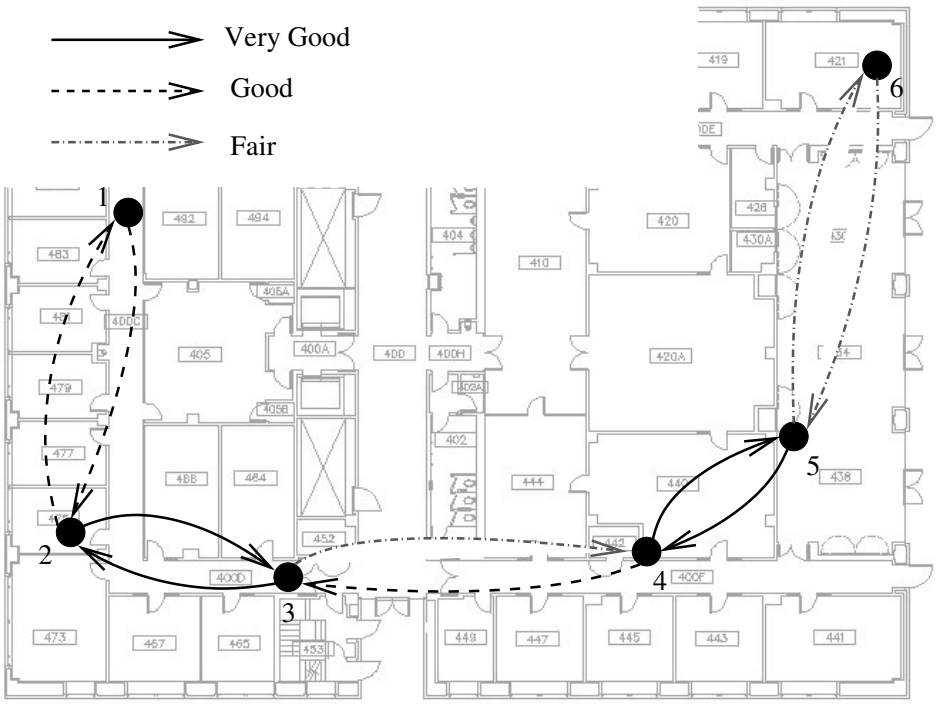


Figure 3.1. Location of the nodes and the signal strength of each link as reported by the driver in our multi-hop testbed

senders that are able to broadcast simultaneously at full rate. In these cases, we conducted experiments with each sender sending unicast UDP packets simultaneously to receivers within their communication range at the maximum rate. We found that depending on the configuration of the flows, either one or both receivers can be affected by interference. For example, when the flows from node 1 to 2 and from node 3 to 4 are simultaneously active, node 2 experiences interference from node 3, whereas the latter flow is not disrupted by transmissions from node 1. We found two such cases where the sending rate of the affected sender drops by more than 60% due to the repeated back-off at the MAC layer triggered by retransmission timeouts. Furthermore, based on the statistics gathered from the device driver, we found more than 85% of the packets sent by this sender were not received at the destination. In the second case, both flows can experience problems due to interference, *e.g.,* when both nodes 1 and 3 are sending to node 2. This case illustrates the classic hidden terminal

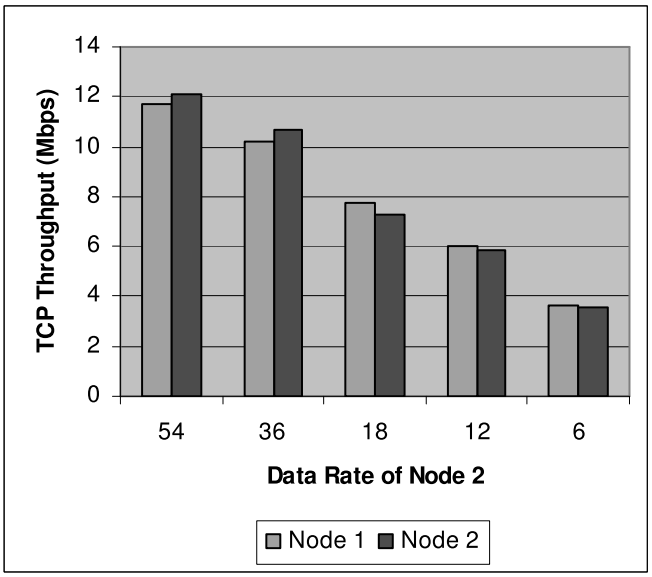


Figure 3.2. 802.11 throughput in the presence of heterogeneous data rate senders

problem, where back-off at the MAC layer causes the sending rate of both senders to drop, which in turn reduces the loss rate of both flows to 35%. However, in this case, the channel utilization and hence the total system throughput drops by about 55%.

3.1.1.2 Sub-optimal Default Allocation

In this Section, we show two examples where the default bandwidth allocation by the 802.11 MAC is far from ideal. These examples illustrate the need for a flexible allocation policy which can be controlled by applications.

Heterogeneous Transmission Rates: The 802.11 MAC allocates an equal number of *transmission opportunities* to every competing node. However, as shown in previous work [31], this fairness criterion can lead to a low throughput when nodes transmit at widely different rates.

We illustrate this behavior using a simple experiment comprising two heteroge-

neous senders connected to a single access point. We emulate heterogeneous senders by fixing the data-rate of transmissions from Node 1 to the access point at 54 Mbps and varying the data-rate of Node 2 between 6 Mbps and 54 Mbps. Figure 3.2 shows the average throughput of two TCP flows originated at the two nodes. This experiment shows that while the behavior is fair as nodes see equal performance irrespective of their sending rate, it hurts the overall system throughput. In particular, as the sending rate of Node 2 decreases from 54 Mbps to 6 Mbps, the total system throughput decreases from 24 Mbps to 7.2 Mbps. In addition, this behavior leads to poor predictability. For example, if Node 1 is the only active one, it will have a throughput of roughly 24 Mbps. However, when Node 2 starts transmitting at 6 Mbps, the throughput seen by Node 1 drops to 3.6 Mbps.

Several Flows Traversing a Node: In a multi-hop network, the fairness policy implemented by 802.11 can lead to poor fairness. This is because of the fact that the fairness policy of 802.11 does not account for the traffic forwarded by a node on the behalf of other nodes.

Consider the example in Figure 3.3 where nodes N_1, N_4, N_5 and N_6 each generate a single flow to node N_2. Assume the interference range is twice the transmission range. N_2 cannot receive when nodes N_4, N_5 or N_6 are transmitting, and N_3 cannot receive when N_1 is transmitting. According to the 802.11 fairness policy, N_1 and N_3 each get 1/3 of the bandwidth (of N_2), while the rest of the nodes share the rest. As a result, N_4, N_5, and N_6 get only 1/9 of the entire capacity. It is worth noting that a better solution, would be to allocate 3/7 of the capacity to node N_3, and 1/7 of the capacity to each of the other senders. This way, the transmission rates of nodes N_4, N_5, and N_6 will increase from 1/9 to 1/7 of the capacity.

While this is an analytical example, our experiments show that 802.11 can indeed lead to significant unfairness in a multi-hop network. Actually, in some cases, the

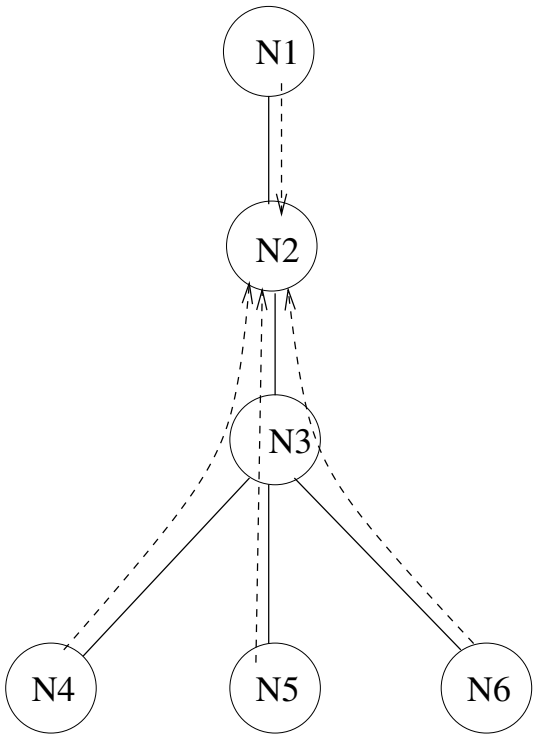


Figure 3.3. Example of interaction of multiple flows in a multi-hop network

unfairness is so pronounced that it causes flows to be shut-off. Ideally, the MAC must provide some mechanism to give higher priority to nodes that relay traffic from a lot of flows.

3.1.2 Need for a MAC Layer Solution

A natural question that arises is whether the limitations we described so far can be addressed at a layer above the MAC layer, such as the network or transport layer. Several proposals have tried to answer this question affirmatively [28, 66]. In a nutshell, these proposals estimate the capacity and interference patterns of the network, and then use this information to limit the sending rate (usually, employing token-buckets) of each node at the network layer. While these solutions may perform well in many scenarios, our experience suggests that they fall short in certain practical

situations. For instance, consider the problem of fair allocation as defined in [28]. In order to achieve fair allocation, the following two requirements should hold:

1. Every node should access the medium for only its fair share of time.

2. When a node is transmitting data, other nodes should not interfere with it.

By limiting the sending rate of each node according to its fair share, we can address the first requirement. However, this solution would work only if the underlying MAC layer satisfies the second requirement. Unfortunately, as discussed in Section 3.1.1, the 802.11 MAC fails to satisfy this requirement quite often. As an example, in the chain configuration of our testbed, we found that both the first and the third link of the chain taken in isolation had a loss rate of less than 5% and were able to support a TCP flow at close to the channel capacity of 4.6 Mbps. But when we started simultaneous flows on both links, one flow always received less than 100 Kbps whereas the other flow received in the excess of 4 Mbps. In order to mitigate this problem we tried rate limiting both TCP flows to 2.3 Mbps. In this case we found that the throughput of the first flow only improved to about 580 Kbps even though the other flow was only receiving 2.3 Mbps (as specified by the token bucket).

The inability of 802.11 to effectively address the second requirement suggests that a general solution to fully address all the 802.11 limitations requires changes to the MAC layer.

3.1.3 Advantages of an Overlay Solution

Given that the 802.11 MAC limitations cannot be fully addressed without changing the MAC layer, we propose the use of an Overlay MAC layer as an alternative to building a new MAC layer. We believe that the overlay approach offers several

advantages such as low cost, flexibility and the possibility of integration with higher layers.

First, changing the MAC layer requires the use of expensive proprietary hardware, or waiting for a new standard and hardware to become available. In contrast, OML can be deployed using existing 802.11-based hardware.

Second, the fact that OML is implemented in software makes it easier to modify OML to meet the diverse requirements of the ever increasing spectrum of wireless applications [7, 1, 50, 31]. For example, in our OML implementation, we support service differentiation both at the flow and node granularity. In the case of a rooftop network [20], one could modify OML to take advantage of the relative stationarity of the link quality and interference patterns.

Finally, the software implementation of OML also enables us to have tighter integration between the link, network and transport layers. For example, Jain *et. al.* [33] show that it is possible to significantly increase the throughput of an ad-hoc network by integrating the MAC and routing layers. Another example is that the transport layer can provide information about the traffic type (*e.g.,* voice, ftp, web), and OML can use this information to compute efficient transmission schedules.

Of course, these advantages do not come for free. Fundamentally, OML incurs a higher overhead and it is more inefficient than a hardware implementation of the same functionality. Furthermore, OML is limited to using only the interface exposed by the 802.11 MAC layer as opposed to all the primitives that the hardware supports.

3.2 Design

In this Section, we present our solution, an Overlay MAC Layer (OML), that alleviates the MAC layer issues described in Section 3.1. We first state our assumptions.

3.2.1 Assumptions

The primitives available for the design of OML are determined by the interface exposed by the device driver. For the sake of generality, OML makes minimal assumptions about this interface. In particular, we assume that

1. The card can send and receive both unicast and broadcast packets.
2. It is possible to set the wireless interface in promiscuous mode to listen to all transmissions from its 1-hop neighbors.
3. It is possible to disable the RTS-CTS handshake by correspondingly setting the RTS threshold.
4. It is possible to limit the number of packets in the card's queue to a couple of packets. This is critical for enabling OML to control packet scheduling because once the packets are in the card's queue, OML has no control over when these packets are sent out.

We note that these assumptions already hold or can be enforced, eventually by modifying the device drivers in most 802.11 cards.

While carrier-sensing is a very important primitive for the design of a MAC protocol, we do not assume that OML can use this primitive. This assumption reflects the fact that most 802.11 cards do not export the current status of the channel or the network allocation vector (NAV)[1] to the higher layers.

[1]NAV is maintained internally in the hardware to keep track of RTS, CTS and reservations in the packet header.

3.2.2 Solution

As noted in the previous section, the only control that OML can exercise over packet scheduling is *when* to send a packet to the network card. Once the packet is en-queued at the network card, OML has no control on when the packet is actually transmitted. Thus, ideally, we would like that when OML sends a packet to the network card, the network card transmits the packet immediately (or at least with a predictable delay).

To implement this idealized scenario, we propose a solution that aims to (a) limit the number of packets queued in the network card, and (b) eliminate interference from other nodes, which is the major cause of packet loss and unpredictability in wireless networks. Goal (a) can be simply achieved by reducing the buffer size of the network card.

To achieve goal (b), we synchronize clocks and we use a TDMA-like solution where we divide the time into slots of equal size l, and allocate the slots to nodes or links according to a weighted fair queuing (WFQ) policy [21]. We call this allocation algorithm the Weighted Slot Allocation (WSA) algorithm. Note that WSA can either operate on a per-node basis (a node can transmit to any destination in a given time-slot) or a per-link basis (a node can only transmit to a specific destination in a given time-slot). However, for ease of explanation, we assume the per-node model in the rest of this Section. It can easily be generalized to the per-link model by considering each node as a number of *pseudo-nodes*, where each pseudo-node transmits to only one destination.

WSA assigns a weight to each node, and in every contention region[2] allocates slots in proportion to the weights of the nodes. Thus, a node with weight two will

[2]We use the term contention region to refer to a set of nodes that compete with each other for access to the channel.

get twice as many slots as a node with weight one in the same contention region. Only nodes that have packets to send contend for time slots, and a node can transmit only during its time slots. Since a time slot is allocated to no more than one node in a contention region, no two sending nodes will interfere with each other. This can substantially increase the predictability of packet transmission, and reduce packet loss at the MAC layer. However, in practice, it is hard to totally eliminate interference. In an open environment there might be other devices out of our control (*e.g.,* phones, microwave ovens), as well as other nodes that run the baseline 802.11 protocol which can interfere with our network. Furthermore, as we will see, accurately estimating the interference region is a challenging problem.

The reason we base WSA on the WFQ policy is because WFQ is highly flexible, and it avoids starvation. WFQ has emerged as the policy of choice for providing QoS and resource management in both network and processor systems [21, 47, 65]. However, note that WSA is not the only algorithm that can be implemented in OML; one could easily implement other allocation mechanisms if needed.

There are three questions we need to answer when implementing WSA:

1. What is the length of a time-slot, l?
2. How are the starting times of the slots synchronized?
3. How are the times slots allocated among competing clients?

We answer these questions in the next three sections.

3.2.2.1 Slot size

The slot size l is dictated by the following considerations:

1. l has to be considerably larger than the clock synchronization error.

2. l should be larger than the packet transmission time, *i.e.,* the interval between the time the first bit of the packet is sent to the hardware, and the time the last bit of the packets is transmitted in the air.

3. Subject to the above two constraints, l should be as small as possible. This will decrease the time a packet has to wait in the OML layer before being transmitted and will decrease the burstiness of traffic sent by a node.

In our evaluation, we chose l to be the time it takes to transmit about 10 packets of maximum size (*i.e.,* 1500 bytes). We found this value of l to work well in both simulations and in our implementation.

3.2.2.2 Clock synchronization

Several very accurate and sophisticated algorithms have been proposed for clock synchronization in multi-hop networks [25, 56, 58]. We believe that it is possible to adapt any of these algorithms to OML. However, OML does not require very precise clock synchronization since we use a relatively large slot time. For evaluation of OML, we have implemented a straightforward algorithm that provides adequate performance within the context of our experiments on the simulator and the testbed.

We synchronize all the clocks in the network to the clock at a pre-designated *leader node*. We estimate the one-way latency of packet transmission based on the data-rate, packet size and other parameters of the 802.11 protocol. We then use this estimated latency and a timestamp in the header of the packet to compute the clock skew at the receiver.

3.2.2.3 Weighted Slot Allocation (WSA)

In this Section, we describe the design of WSA. The challenge is to design a slot allocation mechanism that (a) is fully decentralized, (b) has low control overhead, (c) and is robust in the presence of control message losses and node failures.

For ease of explanation, we present our solution in three stages. In the first two stages, we assume that any two nodes in the network interfere. Thus, only one sender can be active in the network at any given time. Furthermore, in the first stage, we assume that all nodes have unit weight. In the final stage, we relax both these assumptions.

Step 1 - Network of diameter one with unit weights: One solution to achieve fair allocation is to use pseudo-random hash functions as proposed in [10]. Each node computes a random number at the beginning of each time slot, and the node with the highest number wins the slot. The use of pseudo-random hash functions allow a node to compute not only its random number, but the random numbers of others nodes without any explicit communication with those nodes.

Let H be a pseudo-random function that takes values in the interval $(0, 1]$ uniformly at random. Consider c nodes, n_1, n_2, ..., n_c, that compete for time slot t. Then each node computes the value $H_i = H(n_i, t)$ for $1 \leq i \leq c$, where H is a pseudo-random function, and t is the index of the slot in contention. A node n_r wins slot t if it has the highest hash value, *i.e.,*

$$\arg\max_{1 \leq i \leq c} H_i = r. \tag{3.1}$$

Since H_i is drawn from a uniform distribution, it is equally likely that any node will win the slot. This results in a fair allocation of the time slots among the competing nodes.

A node n_s will incorrectly decide that it can transmit if and only if it is unaware of another node n_i such that $H_i > H_s$. While the probability that this can happen cannot be neglected (*e.g.,* when a node n_i joins the network), having more than one winner occasionally is acceptable as the underlying MAC layer will resolve the contention using CSMA/CD. As long as such events are rare, they will not significantly impact the long term allocation.

Step 2 - Network of diameter one with arbitrary weights: Let w_i denote an arbitrary weight associated to node i. Then we define $H_i = H(n_i, t)^{1/w_i}$, and again allocate slot t to node r with the highest number H_r. The next result shows that this allocation will indeed lead to a weighted fair allocation.

Theorem 1. *If nodes $n_1,\ldots,n_c$ have weights $w_1,\ldots,w_c$, and H is a pseudo-random function that takes values in the range $(0, 1]$, then*

$$P[(\arg\max_{1\leq i\leq c} H_i) = r] = \frac{w_r}{\sum_{j=1}^{c} w_j} \tag{3.2}$$

Proof. The probability distribution function for H_i is given by

$$\begin{aligned} P[H_i \leq x] &= P[H(n_i, t)^{1/w_i} \leq x] \\ &= P[H(n_i, t) \leq x_i^w] \\ &= x_i^w \end{aligned}$$

Hence the probability density function for H_i, obtained by differentiating the distribution function is given by

$$f_{H_i}(x) = w_i x^{w_i - 1} \text{ for } 0 \leq x \leq 1 \tag{3.3}$$

Next, we compute the distribution function of the $\max[H_i, H_j]$ as follows,

$$\begin{aligned} P[\max(H_i, H_j) \leq x] &= P[H_i \leq x \text{ and } H_j \leq x] \\ &= P[H_i \leq x].P[H_j \leq x] \\ &= x^{w_i + w_j} \end{aligned}$$

By induction on the above result, we get

$$P[\max(H_{i_1}, \ldots, H_{i_n}) \leq x] = x^{w_{i_1} + \ldots + w_{i_n}} \tag{3.4}$$

Now let $W = \sum_{i=1}^{k} w_i$ and let Y denote the random variable given by

$$Y = \max_{1 \leq i \leq k, i \neq r} H_i$$

From Equations 3.4 and 3.3, the distribution and the density functions of Y are given by

$$P[Y \leq y] = y^{\sum_{1 \leq i \leq k, i \neq r} w_i} = y^{W - w_r}$$

$$f_Y(y) = (W - w_r) y^{W - w_r - 1}$$

Finally, we can compute

$$
\begin{aligned}
P[(\arg\min_{1\le i\le k} H_i) = r] &= P[H_r > Y] \\
&= \int_0^1 \int_0^x f_{H_r}(x) f_Y(y) dx\, dy \\
&= \int_0^1 w_r x^{w_r - 1} y^{W - w_r} |_{y=0}^{y=x} dx \\
&= \int_0^1 w_r x^{W-1} dx \\
&= w_r / W
\end{aligned}
$$

□

Step 3 - Larger diameter network: To enable frequency reuse in networks of a larger diameter, WSA must be able to assign the same slot to multiple nodes as long as they do not interfere with each other. Ideally, the decision to allocate a new time slot should involve only nodes that interfere with each other. Therefore, a given node n should use only the hash values computed for nodes in its contention region. Note that WSA will only ensure weighted fairness among the nodes in the same contention region. Nodes in different contention regions can get very different allocations, based on the level of contention.

Computing the set of nodes that interfere with a given node is a hard problem; we develop a distributed algorithm to solve this in the next Chapter. Alternatively, for the time being, we use a heuristic based on the number of network hops between two nodes; we assume that a node can interfere with *all* nodes within k-hop distance, where k is given. When a node wants to contend for a slot it broadcasts its intention to all nodes within k hops. The set of nodes form which a node hears a broadcast message is then the set of nodes it assumes it interferes with.

There is a clear trade-off between the probability of interference and the bandwidth utilization in choosing the value of k. As the value of k increases, both the

probability of interference and the utilization decrease. The reason why the utilization decreases is because, as k increases, a k-hop region will cover more and more nodes that do not interfere with each other in the real system. In this paper, we assume two values of k, k=1, which represents an optimistic assumption as the interference range is typically greater than the transmission range, and k=2, which as we found in our experiments is a more conservative assumption.

Even if we are able to obtain the exact interference pattern, there are two other sources of inefficiencies in the WSA algorithm. First, the node which wins a slot may not have anything to send during that slot. To address this problem we use a simple timer mechanism, called *inactivity timer*. When k=1, each node n_i initializes an inactivity timer at the beginning of each time slot. Let n_j be the winner of that slot. If the timer of node n_i expires, and the node still has not heard any transmissions from n_j, it assumes that the slot is free. If n_i has the next highest hash value after n_j it starts transmitting in that slot. When k=2, nodes within one hop of n_j will announce to n_i that n_j is silent. To suppress multiple announcements from one-hop neighbors of n_j to n_i, we enforce that of all the nodes within one-hop of both n_i and n_j, only the node with the highest hash value will notify n_i. When the interference pattern is externally specified, there may be nodes which are more than two hops away that interfere; in such cases we simply designate one specific path for control messages to such neighbors. In practice, we set the inactivity timer to the time it takes to transmit three MTU-sized packets. This helps us avoid false-positives due to packet losses.

The second source of inefficiency is due to race conditions caused by overlapping regions. Consider three nodes n_i, n_j, n_k such that n_i and n_k do not interfere with each other, but both n_i and n_k both interfere with n_j. Assume the random numbers of the three nodes are such that $H_i<H_j<H_k$. Then, n_i will not transmit because n_j has a higher random number, and n_j will not transmit because n_k has a higher

random number. Thus, although n_i and n_k do not interfere, they would not be able to transmit in the same slot. Once again, the inactivity timer at n_i will detect that n_j is not actually using the slot, and thus n_i will be able to use the remainder of the slot.

3.2.2.4 Putting it all together

Given the hash function H, every node needs the following information to decide if it is allowed to transmit in a given time-slot.

1. The addresses of all other nodes it interferes with and have packets in their queue.
2. The weights of these nodes.

We keep track of this information in a data structure called the *activeList*. We will start by explaining how we maintain the *activeList* in the simplest case *i.e.,* when the interference region is determined by the one-hop neighborhood. We include the current queue length and the weight of the sending node as part of the OML header in every packet. Since all nodes are in promiscuous mode, they automatically receive up-to-date information about their immediate neighbors. However, when a new node enters the competition, other nodes many not aware of its queue length yet. Since they do not compute the hash for the new node, they may incorrectly try to send in a slot where the new node has the lowest value. Both nodes will compete using the 802.11 MAC, but as soon as the new node is successful in sending its first packet, other nodes become aware of its queue length and weight. Thus, we only use the underlying MAC to bootstrap the first packet of new nodes that enter the competition.

When $k = 2$, we also include the queue lengths and weights of all 1-hop neighbors in the OML header. Thus, snooping on packet transmissions from immediate

neighbors will give us up-to-date information about 2-hop neighbors. Finally, when the interference pattern is externally specified, some 3-hop neighbors may also be a part of the interference region of a node. But in our experience, this only happens rarely and is handled as a special case. We explicitly request a 1-hop neighbor which is 2-hops away from the target node to relay information about it.

3.3 Implementation

The testbed we use for our evaluation is Section 3.4 consists of 6 wireless nodes based on commodity hardware and software. The hardware consists of small form-factor computers equipped with a 2.4GHz Celeron processor and 256MB of RAM. Each computer is also equipped with a Netgear WAG511 [5] tri-mode PCMCIA wireless Ethernet adapter. This card is capable of operating in 802.11a,b and g modes, but we conduct all our experiments in 802.11a mode to avoid interference with another production 2.4GHz wireless network operating in same environment.

We have installed Linux (kernel 2.4.22) in all these systems along with the MadWiFi [2] driver for the wireless cards. For routing, we use the Click software router [38] because it provides an easily extensible modular framework. Also, the MIT Grid [1] project provides a readily download-able implementation of DSR [36] and AODV [48] on top of Click.

We have also implemented OML in a simulator and used our implementation on another test-bed at MIT. However, since we use these two environments mostly to study the performance of end-to-end flows in a larger network, we defer discussion about them to Chapter 5.

Next, we explain how OML is implemented in our testbed. We start with a brief overview of Click followed by a description of the new *elements* we have implemented

in Click. Finally, we show how these elements are used in a typical multi-hop wireless router configuration.

3.3.1 Overview of Click

The Click Modular Router[38] is a software architecture developed at MIT for building routers. We only cover the essentials required to understand our implementation of OML. For further details, we refer the reader to [38].

Elements are the building block of a Click router. Elements typically perform some simple packet processing task and are implemented as a C++ class. Packets flow in and out of elements through *ports*. A router is a directed graph of elements connected through their ports using *push* or *pull connections*. In a push connection, data flow is initiated by the upstream element (*e.g.,* input to a queue), whereas in a pull connection, the downstream element is the initiator (*e.g.,* output from a queue). The configuration for the router is a directed graph with elements at the vertices and packets flowing along the edges and is specified through a simple text file.

In addition to ports which deal with the data flow, elements can also implement *handlers*. Handlers are used to implement control operations outside the data flow, *e.g.,* a queue might export a handler that reports the current length of the queue. A downstream element that implements a scheduling algorithm can then use this handler to decide which input to dequeue from.

3.3.2 OML Elements in Click

The majority of OML functionality is implemented using the two elements, *TimeSlotEnforcer* and *ContentionResolver*. *TimeSlotEnforcer* is responsible for making sure that send requests are issued only in the allowed time slots. The length of a slot (l)

is specified through the configuration file. This element is included just before the packets are sent to the device in the router. It implements one *pull* input and one *pull* output. It does no packet manipulations, but the downstream request is forwarded up stream only if the node is allowed to send packets in that slot. It also implements two handler functions. *TimeSlotEnforcer::SetAllowedSlots* takes a bitmap argument to notify the element about which future slots are active. *TimeSlotEnforcer::SetTimeOffset* is used to set a clock skew relative to the system time for synchronization.

The *ContentionResolver* implements a bulk of the control signaling for OML. Every packet being sent to or received from the device flows through the *ContentionResolver*. This element manipulates packet headers as well as introduces new packets for control traffic. It has two *push* outputs, one for regular packets and one for high-priority control packets that go directly to the head of the output queue. It communicates with the *TimeSlotEnforcer* using the *SetAllowedSlots* handler.

The *EncapTimeStamp* and *DecapTimeStamp* elements handle clock synchronization. *EncapTimeStamp* implements a *pull* input and output and is connected to the output of *TimeSlotResolver*. The *DecapTimeStamp*element removes the header added by *EncapTimeStamp* and computes a new clock offset. It also implements the *DecapTimeStamp::GetTimeOffset* handler which is used by both *EncapTimeStamp* and *ContentionResolver*.

A typical router configuration that implements OML is shown in Figure 3.4. We do not show the elements of the routing protocol in this Figure. The OML elements can be used either with the DSR or AODV elements developed in the GRID project or with the *LinearIPLookup* element to implement static routing.

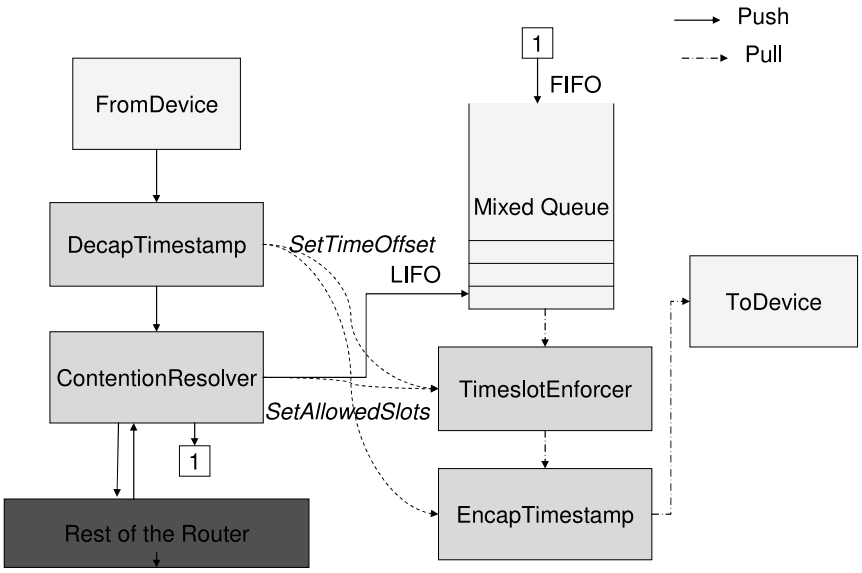


Figure 3.4. Click configuration for OML

3.4 Results

In this Section, we present the results from our evaluation of OML and WSA on our experimental testbed. Since OML only deals with fairness within a single contention region, we restrict our experiments to simple traffic patterns where the flows are only routed over one hop or multi-hop flows where all nodes are in the same contention region. We defer the discussion for more complex and general scenarios to Chapter 5 where we also take transport layer into account. We use WSA on a per-link basis in all our experiments.

Our main results can be summarized as follows. Section 3.4.1 shows that OML/WSA can increase the overall throughput of the system by fairly allocating transmission time, instead of transmission opportunities as in the 802.11 protocol. Section 3.4.2 shows that even in a simple chain topology involving one-hop flows,

802.11 can cause a significant number of TCP flows to experience starvation, while OML can avoid this. Section 3.4.3 shows that, as expected, the starvation problem under 802.11 is even worse in the case of multi-hop routing, but OML can help in this case also. In Section 3.4.4, we evaluate the allocation accuracy of WSA using a simple scenario. Finally, in Section 3.4.5, we show that OML is also effective in the presence of short flows.

To quantify the fairness of the system, we use the metric defined by Chiu and Jain in [16]. Consider a system with M flows, with weights $w_1, w_2, \ldots, w_M$, each receiving a throughput $x_1, x_2, \ldots, x_M$. The fairness index F is defined as

$$F = \frac{(\sum_i x_i/w_i)^2}{M \sum_i x_i^2/w_i^2}$$

Note that $F = 1$ when each flow's throughput is exactly in proportion to its weight, and $F = 1/M$ when only one flow receives the entire throughput.

3.4.1 Heterogeneous data rates

In this experiment, we reconsider the scenario described in Section 3.1.1.2 where two nodes are simultaneously sending one flow each to the access point (unlike the rest of our experiments, we use the infrastructure mode for this one). One node operates at 54 Mbps, and the other node operates at rates ranging from 6 to 54 Mbps. We use the WSA algorithm to allocate the bandwidth, with each node having the same weight. This leads to each node receiving an equal channel-access time, rather than an equal number of transmission opportunities as in 802.11. Note that this allocation implements the temporal-sharing policy as proposed in [57]. Figure 3.5 shows the throughputs of both flows. The two flows receive throughputs approximately propor-

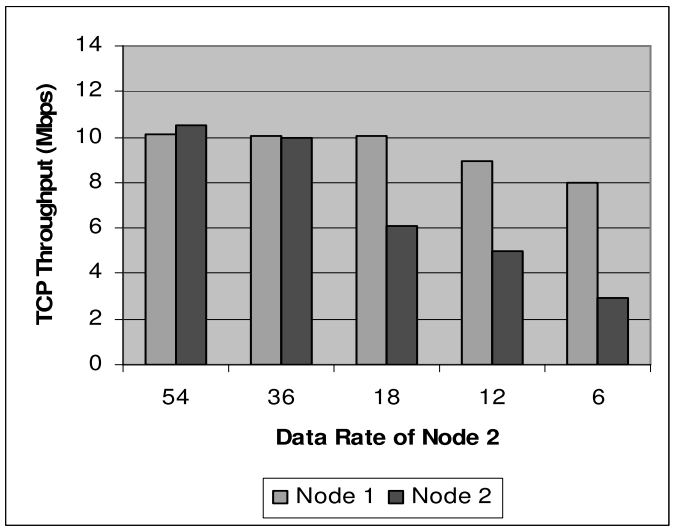


Figure 3.5. 802.11 throughput in the presence of heterogeneous data rate senders using OML

tional to the rates they are operating at, and the total system throughput does not drop by as much (compared to 802.11) when the second node is operating at 6 Mbps.

3.4.2 Chain topology

In this experiment, we configure our testbed as a five-hop chain, and pick two random links in this chain. Figure 3.6 shows the CDF of the flows' throughputs along these links over 50 trials. We consider four scenarios: (a) the flows are started one at a time (b) the flows are started at the same time (*i.e.,* simultaneous flows) using baseline 802.11 (without OML), (c) simultaneous flows using OML assuming one-hop interference ($k = 1$), and (d) simultaneous flows using OML assuming two-hop interference ($k = 2$). When using OML each flow is assigned a unit weight.

As expected, in scenario (a) when only one flow is active, the flows get very good throughput. In 96% of the cases the throughput is greater than 4Mbps. However,

in scenario (b) when the flows are simultaneously active without OML, in 24% of the cases, one of the two flows is not even able to establish a TCP connection. This is shown in Figure 3.6, where 12% of the flows have zero throughput. Furthermore, about 20% of all flows receive less than 1.5 Mbps. In contrast, when using OML with $k = 2$ only about 10% of the flows have a throughput below 1 Mbps. The results when using OML with $k = 1$ are not as good: we only eliminate 6% of the cases with starvation. On the other hand, more flows achieve a higher throughput when $k = 1$: around 35% of all flows have throughputs around 4.5 Mbps. These results illustrate the trade-off between using one-hop and two-hop interference regions. Using a two-hop interference region leads to more conservative spatial reuse of the channel. Hence, there are cases in which two nodes cannot transmit simultaneously even though they do not interfere with each other. In contrast, using a one-hop interference region leads to higher throughput, but at the cost of hurting the fairness.

Table 3.1 further illustrates the trade-off between throughput and fairness as determined by the value of k. The table shows the average aggregate throughput (*i.e.,* the sum of throughput of both flows) and the average fairness index, F, over the 50 trials. In addition to the previous experiment scenarios, we consider another one, called "OML with Actual Interference". This scenario is intended to show the best results OML can achieve when it has complete information about the interference pattern. To implement this scenario, we measure the interference between any two flows in advance and pre-load this information into each node. Recall that for $N = 2$, F ranges from 0.5 (least fair) to 1.0 (most fair). Not surprisingly, we obtain the best fairness, without significantly sacrificing the throughput when using the actual interference pattern. OML with $k = 1$ comes close to this in terms of throughput, whereas OML with $k = 2$ comes close in terms of fairness. Finally, note that native 802.11 achieves the highest throughput, but the lowest fairness. This is because when one

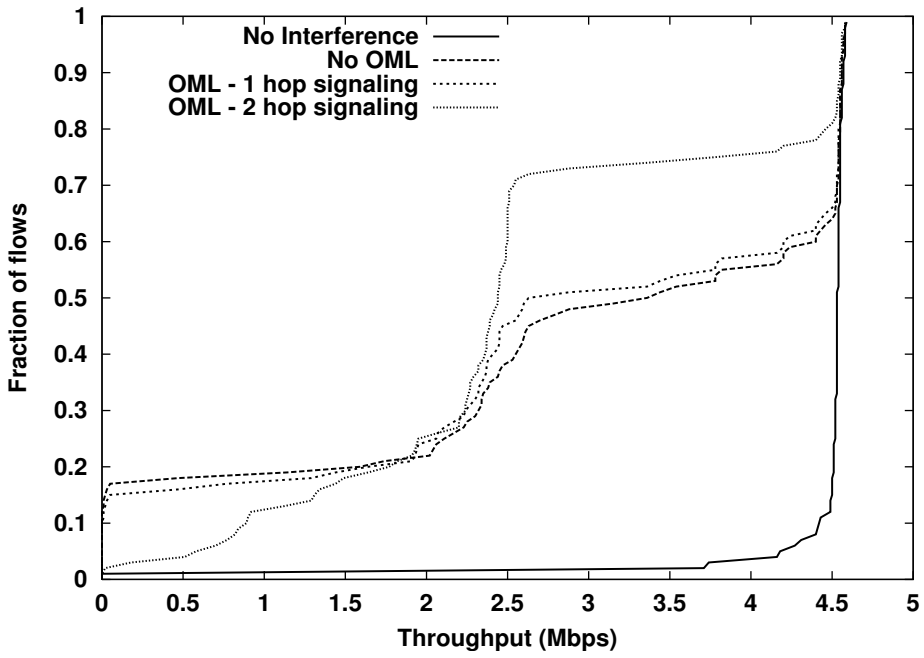


Figure 3.6. CDF of throughput of 1-hop flows in the network

	OML $k = 1$	OML $k = 2$	OML Actual Interference	No OML
Throughput	5.7742	5.1784	5.5712	5.8654
Fairness Index	0.821	0.913	0.922	0.803

Table 3.1. Average system throughput (Mbps) and fairness for two simultaneous flows

link has a better signal quality than the other, it allows the better link to transmit all the time at the expense of starving the weaker link.

3.4.3 Multi-hop routing

In this Section, we study how OML can improve the flow throughputs in a multi-hop wireless network. For this purpose, we arrange the nodes in a simple Y-shaped topology and initiate three TCP flows. Flows A and B originate at nodes 1 and 2, respectively, and terminate at node 4, after being routed via node 3. Flow C is an one-hop flow from node 5 to node 4. The weight of each node is proportional to the

Active Flows	Throughput (Mbps)		
	Flow A	Flow B	Flow C
A&C w/o OML	0.14		4.40
A&C with OML	1.69		2.36
B&C w/o OML		0.06	4.48
B&C with OML		1.71	2.08
A&B w/o OML	1.31	1.65	
A&B with OML	1.11	1.11	
A,B&C w/o OML	0.11	0.03	4.42
A,B&C with OML	0.96	0.94	1.96

Table 3.2. Interaction of flows in an ad-hoc topology

number of flows it sees: the weights of nodes 1, 2, and 5 are one, the weight of node 3 is two, and the weight of node 4 is three. In this experiment, we use OML with two-hop interference regions.

First we note that when only one of the three flows is active, flows A and B receive about 2.24 Mbps, while flow C receives 4.55 Mbps, with or without OML.

Table 3.2 shows the throughputs of each flow when more than one flow is active. Without OML, flow C effectively shuts-off the two-hop flows. In contrast, OML allocates at least 1 Mbps to each flow, when only one of the two-hop flows is active, and at least 0.94 Mbps, when all flows are active. The reason why the throughputs of the two-hop flows are not higher is because the two-hop interference assumption is violated. According to this assumption, transmissions from nodes 1 or 2 should not interfere with with transmissions from node 5. However, this assumption did not hold in our experiment. Let flow D be the one-hop flow from node 1 to 3. When flows C and D were simultaneously active, C received 4.03 Mbps whereas D received only 0.89 Mbps. This shows that transmissions from nodes 1 and 5 do indeed interfere.

Weight of A	1	1	1	1
Throughput of A	2.26	1.48	1.36	0.92
Weight of B	1	2	3	4
Throughput of B	2.20	2.97	3.06	3.64

Table 3.3. Throughput received by senders with different weights

Size of Flow	16k	32k	64	128k	256k
Time w/o OML (s)	0.06	0.12	0.23	0.46	0.93
Time with OML (s)	0.11	0.17	0.29	0.50	0.97

Table 3.4. Transfer time of a short flow in the presence of a long flow

3.4.4 Weighted allocation

In this experiment, we evaluate the accuracy of WSA, and thus its ability to provide per-node service differentiation by setting the node weights. We consider two nodes A and B that send a TCP flow each to a third node C. We set the weight of node A to 1, and assign a weight ranging from 1 to 4 to node B. Table 3.3 shows the throughput achieved by the nodes for each combination of weights. As expected, the ratio of throughputs obtained by the two nodes tracks the ratio of their weights closely.

3.4.5 Short Flows

Due to the additional delay caused by queueing packets while waiting to obtain a time-slot, OML can impose a significant overhead on short TCP flows. To quantify this overhead, we conduct the following experiment. We start two simultaneous one hop flows, both terminating at the same node; the first flow is a long-lived TCP flow, while the second is a short TCP flow of varying size. Table 3.4 shows the transfer time of the second flow as a function of its size. As we can see, the effect of OML translates to a fixed time overhead, usually less than 60ms. As the size of the flow increases, the degradation reduces. For flows larger than 100 kilobytes, the resulting

throughput degradation is less than 10%. This fixed overhead is simply the time it takes for the TCP slow start to ramp up under OML. Though we expect this overhead to be higher for longer (more than one hop) flows, the steady state throughput of such flows is also lower.[3] Thus, we expect the relative overhead to be similar.

[3]Flows that traverse multiple hops in wireless networks are limited to a lower throughput due to the contention between the different hops of the same flow.

Chapter 4

Detecting Interference using Passive Measurements

As mentioned earlier, OML, as well as a lot of other applications require information about which links interfere with each other in a wireless network. In this Chapter, we develop an algorithm that can detect interference as it happens during the normal operation of the network. We start with an overview of our approach in Section 4.1, followed by a presentation of the details of our algorithm in Section 4.2. Next, we explain our evaluation methodology by describing our simulation and testbed environments in Section 4.3. Finally, we presents the results from our evaluation in Section 4.4.

4.1 Our Approach

Our approach is motivated by the success of *passive measurement* techniques in wired networks [35, 27]. Passive measurements provide a non-intrusive way to obtain certain network-level performance metrics by studying the behavior of existing traffic.

In our system, the additional overhead is limited to a small header in each packet used to broadcast control information in a local neighborhood.

One of the advantages of using passive measurements is that even though there may be a large number of links in a network (which can take a long time to measure exhaustively), a lot of applications *e.g.,* scheduling and resource allocation only need interference information for the links that are currently in use. In practice, we found that for most traffic patterns, this subset of links tends to be much smaller for the following two reasons. Firstly, because of the limited channel capacity, we can only have a limited number of flows simultaneously sending data at any given time. Secondly, of the available links, the routing protocol tends to prefer certain links over others. However, active measurement based approaches may be forced to measure the entire network even in this case because there is no way of knowing *which* links are going to be used until a flow actually starts sending data on that link. By this time, it is too late to do start measurements using the same link without severely hurting the performance of that flow. On the other hand, since passive measurements use existing traffic, they automatically detect interference for useful links only.

Note that some applications may indeed need interference information for all links in the network. In such cases, it may be possible to supplement passive measurements with active measurements during periods of lean user traffic. However, such extensions are outside the scope of this paper. As a start, we have chosen to focus on time-slot based scheduling; this application only requires information about links that carry traffic.

To detect interference through passive measurements, we have to keep track of the following information.

- Exactly which links are sending data during any given time period.

- The throughput on each of these links during this period.

As mentioned in Chapter 3, OML divides time into equal size slots and keeps track of which nodes are transmitting in each slot; thus using OML automatically takes care of the first requirement. For the second requirement, we made a simple change to OML to log the number of successful packet transmissions during every time slot at the sender of each link.

Once we detect interference between a pair of links, we feed that information back into OML, which in turn will ensure that these links are not allowed to transmit in the same time slot subsequently. Previously (in Chapter 3) we used very simple hop-based heuristics to approximate the interference pattern used by OML. Here, we modify this in the following way. During cold start, we do not have sufficient information to confidently state which links interfere. So, when the network starts up, we trigger an exploration process by allowing OML to schedule some interfering links in the same time slot. Over time, we detect interference based on the performance of these links. Once we are sure that we can reduce interference and improve performance by preventing a pair of links from transmitting at the same time, we augment the interference pattern used by OML with this new information.

Note that though our current implementation uses OML, our algorithm can work on top any TDMA MAC that keeps track of which nodes are transmitting in each time slot. However, we believe that the two layer MAC architecture of OML provides the following additional benefits.

- We are protected by the underlying 802.11 MAC layer while the interference pattern is being probed by deliberately scheduling interfering links in the same time slot.

- Most readily available hardware (including those based on newer standards such

as 802.11n) already implement some form of a CSMA MAC. In most cases, this layer cannot be disabled without hardware modifications.

- Even if designing and building custom hardware is possible, we believe that carrier sensing before transmission is a good idea for radios operating in the ISM bands. There will always be other hardware that could potentially interfere but do not follow the rules of the particular TDMA MAC in use.

4.2 Algorithm

We now present the design of our algorithm to detect interference using passive measurements. For ease of explanation, we initially describe a centralized version of it assuming global visibility of the entire network (Section 4.2.2). We then show how it can be implemented in a distributed manner in Section 4.2.3. But before we delve into the details, we state our assumptions and present some definitions in Section 4.2.1.

4.2.1 Definitions and Assumptions

In this Section, we explain exactly what we mean by interference in the context of this algorithm. Earlier work in this area has used different ways of specifying the interference pattern of a wireless network [34, 46]. In the most general case, we have to specify exactly what happens when each subset of links is active. However, in this work, we use a **pair-wise**, **binary** model as used in [34].

The *pair-wise* assumption means that we can specify the entire interference pattern by only specifying how any given pair of links interfere. In other words, there are no three links A, B and C such that there is interference when all three of them are active, but no interference when any subset of only two links is active. To predict

interference between k links for $k \geq 3$, we only have to check if any one of the $\binom{k}{2}$ pairs of links contained in this set interfere.

Also, for simplicity, we define interference as a binary phenomenon. More formally, given two links A and B, with throughput t_A and t_B in isolation, we measure their throughput when they are simultaneously active as $t_{A/AB}$ and $t_{B/AB}$ respectively. We say that A and B interfere if and only if $t_{A/AB} < \alpha t_A$ or $t_{B/AB} < \alpha t_B$. The parameter α lies in the range $0 < \alpha < 1$ and is chosen by the application.

The pair-wise binary assumption allows us to represent the interference in a network through a simple interference graph $G = (\mathcal{L}, E)$. The set $\mathcal{L}$ contains a vertex corresponding to each *link* in the network. We have an edge between two links A and B if and only if they interfere as per our previous definition of interference.[1] Figure 4.1 shows a simple network topology with five links labeled as A through E. For clarity, we do not consider the rest of the links (shown with dotted lines). Links A, B and D interfere with C. In addition, links A and B interfere with each other, and E interferes with D. The resulting interference graph for these five links is shown to the right.

4.2.2 Centralized Algorithm

In this Section, we assume that we have information about the entire network at a centralized location. Since we are operating on top a time-slot based MAC, we assume that we have history available for all time slots between *startTime* and *endTime*. This information is stored in the array *shi* of type *SlotHistoryInfo*. Each *SlotHistoryInfo* record has information about every link in the network in a particular time slot. *shi[t].link[i].active* indicates if the link i was transmitting during time slot

[1] We use the terms *nodes* and *links* when we refer to the network topology graph, and the terms *vertex* and *edge* with respect to other graphs. For example, links are actually vertices in the interference graph.

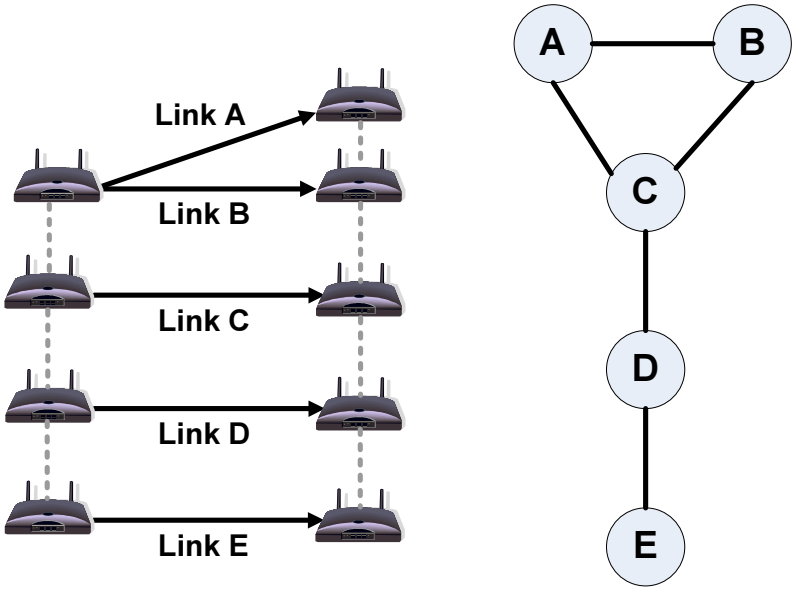


Figure 4.1. The interference graph for five links in a simple network topology

t. If so, *shi[t].link[i].rate* contains the throughput achieved on that link. Figure 4.2 shows an example *shi* for the network in Figure 4.1. For clarity, we omit the rate information and show only the boolean values of *link[i].active*; an X indicates that the link was active in that particular time slot.

The objective of our algorithm is to determine the set of links that interfere with any given link L, henceforth referred to as the *target* link. We can do this by performing the following two steps:

1. Identify the time slots in which the target link was affected by interference.

2. Come up with a set of links that can explain the interference in these time slots.

Step 1: Identifying time-slots with interference: By definition, the target link L was affected by interference in a given time slot if its rate is less than αt_L. However, to use this definition, we need an estimate of t_L, the throughput of link

Slot	A	B	C	D	E
1	X	X	X		X
2			X		
3	X	X		X	
4			X	X	
5	X	X	X	X	X
6				X	X

Figure 4.2. An example input for our algorithm. We only show which links are active in each time-slot. The throughput in each of these slots is also available in *shi*.

L in isolation. We use $r_L^{\max}$, the maximum rate achieved by link L in *shi* as this estimate. Note that $r_L^{\max} \leq t_L$ and is equal to t_L if there is at least one slot in *shi* where L was not affected by interference. For the example in Figure 4.2, the estimate is correct for, and is available from slot 2 for link C, slot 3 for link D and slot 1 for link E respectively. However, it is incorrect for links A and B which are affected by interference in every slot in which they transmit; we may underestimate interference for these links. In practice, this was not really an issue in most of our experiments (see Section 4.4) due to the following reasons.

- We do not start OML assuming an empty interference pattern. We found that the 1-hop interference model almost never overestimates interference. We use that as the baseline and hence a lot of interference is already eliminated from the input *shi*. We only to need to make inferences about 2 and 3-hop neighbors.

- We maintain history for the previous 100 time slots. The more history we have, the more likely we are to get the correct *maxRate*.

- As mentioned in Section 4.1, for most scenarios not all links are trying to transmit at the same time. For a given link, based on the number of flows and their routes, the set of *active* links that can potentially interfere is small even though the number of links close enough to interfere can be large.

- If there are multi-hop flows in the network, downstream hops can only transmit

when they have already received data from the upstream hops. Also, upstream hops depend on the TCP congestion control algorithm (window size and the reception of ACKs) to be able to send data. This means that even for links that are on the route of an active flow, they are not trying to transmit data at all times.

Note that even if this turns out to be a problem for some network scenario, a simple fix is possible. We can periodically, say every 5 minutes, schedule a slot (for each link) where we explicitly disallow all links we suspect might interfere from transmitting. We can then store the obtained throughput as the maximum rate for that link.

Step 2: Identifying the links which interfere: We now know that at least *one* interfering link was active in each time slot identified in the previous step. In order to come up with a set that satisfies this property, we first construct a bipartite graph, *intGraph* as follows. The vertex set of the graph is given by $(V_1 + V_2)$ where $V_1 = \mathcal{L}$, the set of links in the network and $V_2 = T$, the set of time slots in *shi*. We connect a link M to a time slot t if and only if:

1. The target link L was affected by interference in time slot t
2. Link M was active in time slot t.

Figure 4.3(a) shows the graph constructed with link D as the target using the *shi* in Figure 4.2.

Based on our construction of *intGraph*, we know that every vertex of non-zero degree in V_2 has to be adjacent to at least one link that interferes with L. However, there can be many subsets of V_1 that satisfy this property. Since it is easier to identify links that *do not* cause interference, we start by doing that first. For example, in slot 3, links A and B are both active and yet D was not affected by interference. Therefore,

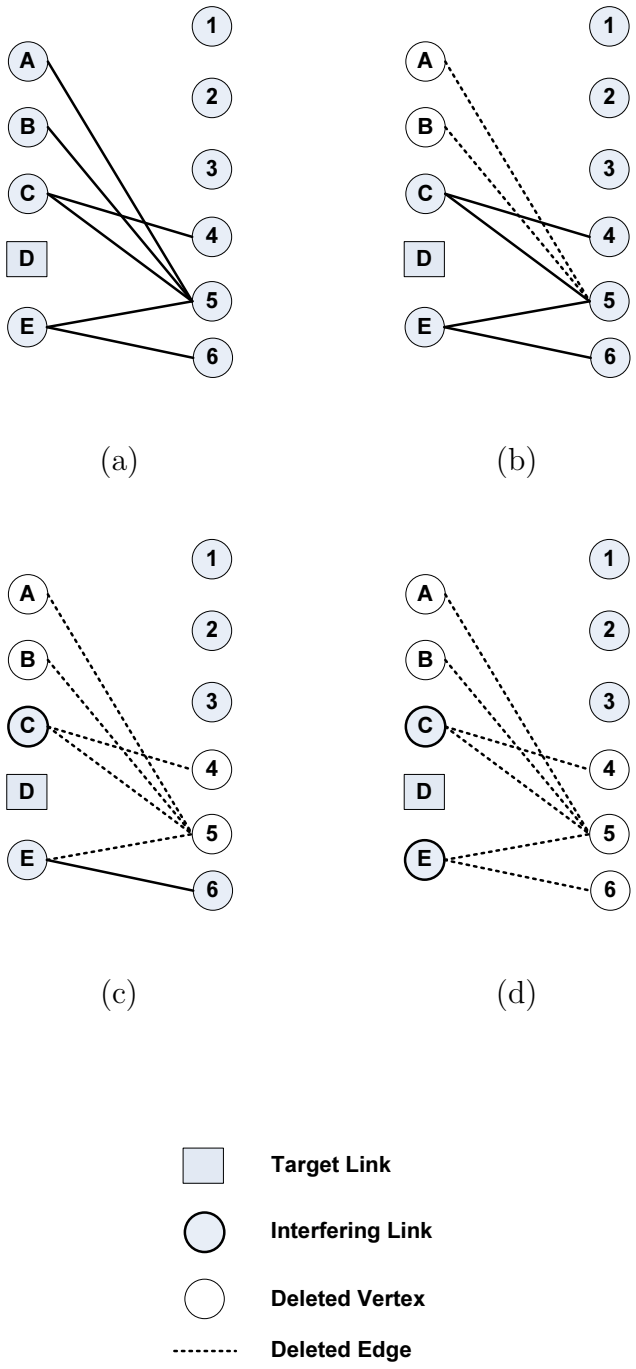


Figure 4.3. The graphs constructed by *GetInterferingLinks(D)* for the network in Figure 4.1 using the *shi* in Figure 4.2. We initially start with graph (a), which yields (b) after the clean-up phase. We then detect links C and E as interfering links over two iterations; the corresponding graphs are shown in (c) and (d).

we can *clean up* the graph by removing the vertices corresponding to A and B (see Figure 4.3(b)). More formally, we identify all links N such that (a) both L and N are active and (b) L achieves a rate of at least βt_L, and remove such links from V_1 in the *clean up* phase. The parameter β is lies in the range $\alpha < \beta < 1$. Choosing a β that is not exactly equal to α or 1 helps guard against small variations in performance of links caused by external factors.

While the clean up phase helps reduce the ambiguity in choosing an appropriate subset of V_1, there is no guarantee that all ambiguity is eliminated. At this point, we have to take a guess and pick *some* subset that helps explain all interference. We do this by repeating the following two steps until there are no more edges in *intGraph*.

1. Pick a vertex V of non-zero degree in V_1 and add it to the set of interfering links.

2. Remove all vertices that are adjacent to V. Note that the removal of a vertex from a graph also results in the removal of all edges that are incident on that vertex.

The second step ensures that we only add a new link to the set if it accounts for interference that is not already explained by links chosen earlier. Note that even if we underestimate interference, we are likely to see it again in the near future. Therefore we believe that a conservative policy which affects performance by taking slightly longer to detect interference is preferable over an aggressive policy which makes a lot of incorrect inferences. This process takes two iterations for the graph in Figure 4.3(b). Link C is chosen in the first iteration which results in the removal of nodes 4 and 5 (Figure 4.3(c)). Link E is chosen in the next step which results in a graph with no links (Part (d) of the same Figure).

We still haven't explained how we pick a vertex from V_1 in each iteration. Intu-

itively, we want to pick links that are more likely to have caused interference at target link first. A simple heuristic is to pick the node with the highest degree in V_1 at each step; this helps explain interference in the most number of time slots. However, this does not account for the fact that interference is some slots is easier to disambiguate than in others. For example, the interference in slot 5 could have been caused by either link C or E, but the interference in slots 4 and 6 can only be explained by one link. We account for this factor by using a *weighted degree heuristic*. The weight of an edge (v_1, v_2) where $v_1 \in V_1$ and $v_2 \in V_2$ is defined as the reciprocal of the degree of v_2. Thus, the edges incident on slot 5 have a degree $\frac{1}{2}$ in Figure 4.3(b). The weighted degree of a vertex in V_1 is the sum of the weights of the edges incident on it.

Figure 4.4 gives the complete pseudocode for our algorithm. More specifically, the function *GetInterferingLinks()* returns the list of links that interfere with any given link using *shi* as input. The first **for** loop (lines 5 to 7) computes *maxRate* as an estimate for t_L. The second **for** loop (lines 11 to 20) constructs the bipartite graph which is represented as an array of adjacency lists in *intGraph*. In order to the implement the weighted degree heuristic, we also create the array *numIntLinks* which keeps track of the degree of vertices in V_2. The completes **Step 1** of the algorithm. The next **for** loop (lines 22 to 31) implements the clean up phase followed by the last **while** loop (lines 33 to 59) which removes vertices and nodes from the graph according to the weighted degree heuristic.

4.2.3 Distributed Algorithm

The key idea behind our distributed implementation is that each node is responsible for deciding which other links are interfering with its transmissions. In order to do this, each node executes the following steps:

1. Gather data about all other links it thinks might interfere with its transmissions.

2. Run *GetInterferingLinks* locally with this information.

3. Notify other nodes once it concludes that they interfere with its transmissions. This is required because interference sometimes only affects the performance of one of the two links. But OML needs the information at both senders to ensure that they don't transmit at the same time.

In order to stay current in a dynamic network, we have to repeat these steps periodically to learn new information. The data gathered in step (1) is kept up-to-date by continuously flooding information about recent timeslots piggybacked on the headers of data packets. This mechanism is described in detail in Section 4.2.3.1. Step (2) can be significantly optimized by saving the constructed *intGraph* between calls to *GetInterferingLinks*. In each call, we first discard all edges and vertices where the time slot is less than the new *startTime*. To add new links, we only have to process *shi* between the previous *endTime* and the current one. Finally, in step (3) we only have to notify neighbors that haven't already been notified during previous calls to *GetInterferingLinks*.

4.2.3.1 Scoped flooding of link information

The first observation we make regarding *GetInterferingLinks* is that at any given node, we only need time-slot level information about other links that could potentially interfere with transmissions originating at that node; the algorithm does not require information about links that are "far away" in the network. In practice, we found three network hops to be a very good upper bound on the interference range of a given link. Thus at every node we need information about links that are within three network hops of this node and the time slots in which these links were active. Instead of explicitly flooding this information, we will now explain how we can achieve a much lower overhead by using packet snooping and piggybacking techniques.

Since OML enables promiscuous receptions at every node, we automatically get information about links that are one hop away. If sender is one hop away, we can snoop on all data transmissions on that link. On the other hand, if only the receiver is one-hop away, we can use the link layer ACKs to determine when data was being sent on that link. In order to gather information about 2-hop and 3-hop neighbors, we piggyback this data in the headers of data transmissions from each node. For each link, the time-slots in which it was active are represented through a simple bitmap that uses only 1-bit for each time-slot. We include information about every 1-hop neighbor in each packet; this ensures that each node has relatively up-to-date data about its 2-hop neighbors by listening to all transmissions in promiscuous mode. However, to save space in the header, we only include information about selected 2-hop neighbors (chosen at random and changed periodically). We found this to be an acceptable trade-off for two reasons

- For each 3-hop neighbor n, there is typically more than one immediate neighbor that can reach n over 2-hops. We will receive information about n as long as it is chosen by any one of these neighbors.
- Even if the node is not chosen by any of these 1-hop neighbors, it is likely to be chosen later when one of them updates their selection. Thus, it might take longer to infer interference from 3-hop neighbors. In our experiments, interference from 3-hop neighbors was very rare and did not significantly affect performance.

Our implementation typically adds 40 to 60 bytes of overhead in the header of each packet. It may be possible to further optimize this using header compression techniques [40, 64]. A lot of our header overhead consists of bitmaps that indicate when each link was active; these bitmaps tend to be sparse since only one of several interfering links can be active at any given time. We can potentially save more

space by using bloom filters [43] or other advanced techniques [44] designed for such bitmaps. The implementation of such techniques, however, is outside the scope of this work.

4.2.3.2 Removing Stale Inferences

We use a very simple time-out based mechanism to ensure that we expire stale inferences as the network changes. We assume that the inferences made by *GetInterferingLinks* are valid only for a maximum of one hour. This is a very conservative assumption; when the nodes are not mobile, the interference changed for only 8% of the link pairs when measured five days apart [46]. But it works well in practice because most inferences take less than one minute in our experiments(see Section 4.4). We randomize the expiration timer between 45 and 60 minutes to avoid synchronization where a large number inferred interferences become invalid at the same time.

The above mechanism works well when nodes are stationary and channel conditions are static. However, if the network (nodes move) or the channel conditions change (a microwave oven is turned on, obstacles such as furniture are moved etc.) we need a mechanism to detect this change and react sooner than an hour.[2] We use the throughput of a link in the absence of any interference as an indicator for its stability. This throughput is calculated as *maxRate* in *GetInterferingLinks*. If *maxRate* changes by more that 50% from the time a particular inference was made, we do not consider it to be valid any longer. We immediately expire this information by sending a explicit message to the other senders.

[2]Our scheme may not be suitable for networks where the mobility is very high. In this case, we may be unable to detect interference quickly enough for it to be useful. Even though we do detect changes pretty quickly, we require that the changes are not too frequent.

```
01  function GetInterferingLinks(Link l,
02                               SlotHistoryInfo shi[], startTime, endTime)
03
04      maxRate = 0;
05      for i = startTime to endTime
06          maxRate = max (maxRate, shi[i].l.rate)
07      endfor
08
09      intGraph[1 .. numLinks] = {}
10      numIntLinks[startTime .. endTime] = 0
11      for i = startTime to endTime
12          if shi[i].l.active = false continue
13          if shi[i].l.rate > α * maxRate continue
14
15          for j = 1 to numLinks
16              if shi[i].link[j].active and link[j] ≠ l
17                  intGraph[j].Append(i)
18                  numIntLinks[i]++
19          endfor
20      endfor
21
22      for i = startTime to endTime
23          if shi[i].l.rate < β * maxRate continue
24          for j = 1 to numLinks
25              if shi[i].link[j].active
```

(continued . . .)

```
26                      for each tSlot ∈ intGraph[j]
27                          numIntLinks[tSlot] − −
28                      endfor
29                      intGraph[j].Clear()
30              endfor
31      endfor
32
33      while (true)
34              maxLength = 0
35              maxLink = 0
36              for i = 1 to numLinks
37                      currentWt = 0
38                      for each tSlot ∈ intGraph[i]
39                          currentWt += 1/numIntLinks[tSlot]
40                      endfor
41                      if currentWt > maxLength
42                          maxLength = currentWt
43                          maxLink = i
44              endfor
45
46              if maxLength = 0 return
47
48              link[maxLink].interfere = TRUE
49              link[maxLink].inferenceTime = intGraph[maxLink].head()
50              toClear = intGraph[maxLink].Copy()
```

(continued . . .)

```
51
52              for i = 1 to numLinks
53                  for j = 1 to length(toClear)
54                      intGraph[i].Remove(toClear[j])
55                  endfor
56              endfor
57
58              intGraph[maxLink].Clear()
59          endwhile
60
61      endfunction
```

Figure 4.4. The pseudocode for the centralized algorithm to get the list of links that interfere with a given link

4.3 Evaluation Methodology

In order to evaluate our algorithm, we have implemented it in a test-bed, which we describe in Section 4.3.2, and in a simulator which we describe below.

4.3.1 Simulation

Our simulator is based on ns2 [6], an event-based packet level simulator and it's wireless extensions developed at the Carnegie Mellon University. Even though our ultimate goal was to implement our algorithms in a real test-bed, we found a simulator to be useful for a number of reasons. Firstly, unlike a test-bed, the simulator allows us to make changes and perform experiments very quickly - this was very useful while we were designing our algorithm. Secondly, the simulator allows to us change the characteristics of the antenna and the wireless medium very easily.

4.3.1.1 Interference pattern in simulation

In order to stress our algorithm, we have implemented two new signal propagation models in ns2. The first model is a purely synthetic model that generates an interference pattern based on a certain probability of interference at each distance. The second model simulates the gain pattern of a directional antenna. These models are described below.

Hop-based interference: In this model, for any two nodes A and B, they can either:

1. Communicate with each other.
2. Carrier sense each others transmissions.
3. Do not interfere.

We say that A and B interfere if they are in categories (1) or (2). To decide the relationship between a given pair of nodes, we do the following at the start of each simulation. First, we set the communication range to 250m and compute the distance between them in terms of the number of hops. By definition, two nodes can communicate and if only if they are 1-hop away. For nodes separated by two hops, there is a 60% chance that they are within carrier sense range of each other and a 40% chance they do not interfere. At three hops, there is a 40% chance that nodes can carrier sense each other, and nodes that are more than 3 hops away never interfere.

Based on this model, to determine if two links S1 to R1 and S2 to R2 interfere, we have to first check if the senders S1 and S2 interfere. If so, carrier sensing in 802.11 will prevent them from sending at the same time and the links do interfere. Next, we have to check if S1 interferes with R2, or if S2 interferes with R1. If either of these conditions is true, the packet cannot be decoded correctly at the receiver experiencing interference. Thus, the two links do not interfere if and only if the following three conditions are met

1. Senders S1 and S2 do not interfere.
2. S1 and R2 do not interfere.
3. S2 and R1 do not interfere.

Note that for the sake of simplicity, this model implicitly assumes that the carrier sense range and the interference range are the same. That is, S1 interferes with the reception at R2 if and only if R2 can carrier sense transmissions from S1.

Directional Antenna: In this model, we use a simple directional antenna along with the two-ray ground reflection model [39] for signal propagation. The gain pattern for our antenna is shown in Figure 4.5. It consists of two lobes opposing each other. The maximum gain is 3 on the major lobe and 1 on the minor lobe. The gain

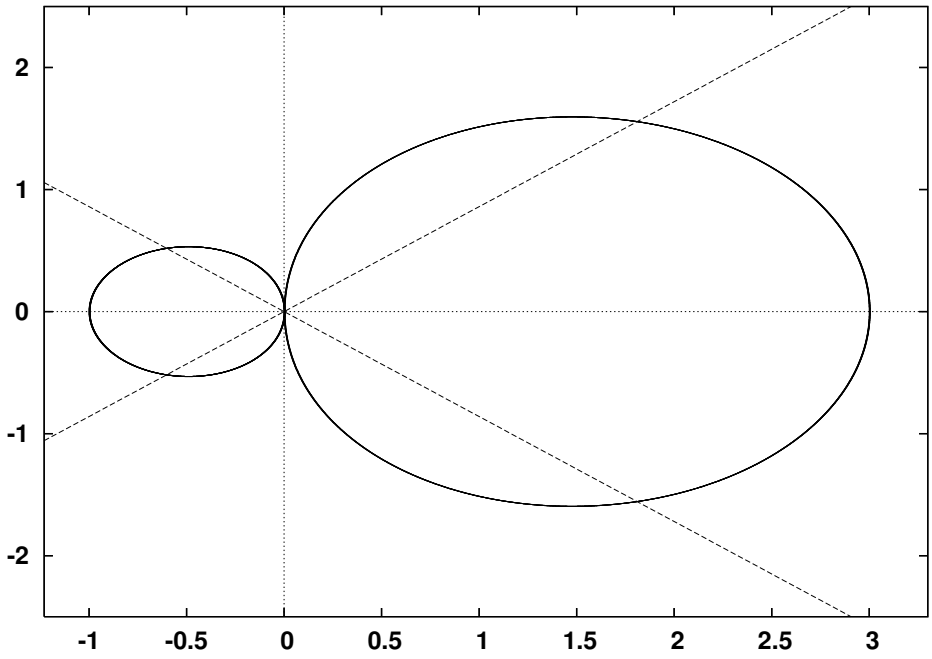


Figure 4.5. The gain pattern of our simulated directional antenna

depends quadratically on the angle and reduces to zero at a direction perpendicular from either lobe. The beam-width (defined as the angle over which the gain is at least half the maximum, 1.5 in this case) is roughly 80 degrees. For $-\pi \leq \theta \leq \pi$, the gain $g(\theta)$ is given by,

$$g(\theta) = \begin{cases} 3\left(1 - \left(\frac{2\theta}{\pi}\right)^2\right) & |\theta| \leq \frac{\pi}{2}, \text{ major lobe} \\ 1 - \left(2 - \frac{2\theta}{\pi}\right)^2 & |\theta| > \frac{\pi}{2}, \text{ minor lobe} \end{cases}$$

4.3.2 Implementation

Because of the limited size of the test-bed we used in Chapter 3, the interference pattern is too simple to evaluate our algorithm. Luckily, we were able to obtain access to a larger test-bed thanks to Prof. Robert Morris and the GRID project [1] at MIT. The test-bed consists of 30 wireless nodes based on commodity hardware and

software placed on a single floor of an office building. The hardware consists of small form-factor computers equipped with a 266 MHz Geode processor and 128MB of RAM. Each computer is also equipped with a Netgear [5] tri-mode mini-PCI wireless Ethernet adapter. This card is capable of operating in 802.11a,b and g modes, but we conduct all our experiments in 802.11a mode to avoid interference with another production 2.4GHz wireless network operating in same environment. The software platform uses Linux and the Click modular router [38] and was very similar to our previous test-bed; hence we were able to use our implementation of OML without modifications. In addition, we implemented our interference detection algorithm also on top of Click.

4.4 Results

We present the results from our evaluation in this Section. We first explain the different metrics that we use in Section 4.4.1. Next, we show the results from our simulations and our testbed in Sections 4.4.2 and 4.4.3 respectively.

4.4.1 Evaluation Metrics

In this Chapter, we are primarily interested in finding out how quickly and accurately we are able to infer interference. We also compare our performance against the 1-hop and 2-hop heuristics. We defer the discussion of performance of end-to-end flows to Chapter 5. In particular, we try to answer the following questions here:

1. Are there any interfering pairs of active links that are not detected by our algorithm? (False Negatives)

2. Do we erroneously infer interference between pairs of links that do not interfere with each other? (False Positives)

3. How long do we allow interfering links to transmit at the same time before we detect the interference?

4.4.2 Simulation

For our simulations, we create a wireless network by placing 30 nodes uniformly at random in a rectangular region. The height of the rectangle is twice the radio range and the width is changed to vary the density of node placement. We use static pre-configured routes to avoid any undesirable interactions between the execution of a routing protocol and our algorithm. Each data point is obtained by averaging over 8 simulation runs unless otherwise specified. We use $\alpha = 0.5$ and $\beta = 0.8$ in all our experiments; recall that once we infer interference between two links, OML will prevent them from sending in the same time-slot, thus limiting each to roughly half the throughput. We simulate 802.11b radios at 2Mbps in all cases.

4.4.2.1 Accuracy of Interference Inference

In this experiment, we first fix the average density of network at 16 nodes per radio range (the area of the circle covered by the transmitter of each node). We then simulate ten simultaneous flows generated between randomly chosen (source,destination) pairs while varying the signal propagation model at the physical layer. We then look at interference between links that were on the paths of any one of these 10 flows. Table 4.1 shows the false positives and negatives for various methods of determining interference. The '1-hop' and '2-hop' methods refer to heuristics based on the network distance between the senders of each link. In the case of our passive measurement

Propagation Model	1-hop		2-hop		Passive detection	
	Fal. neg.	Fal. pos.	Fal. neg.	Fal. pos.	Fal. neg.	Fal. pos.
Omni-dir. Ant.	46.1%	0	13.9%	0	3.0%	0.4%
Hop-based	46.9%	0	12.3%	4.5%	5.0%	0.9%
Dir. Ant.	60.8%	0	21.0%	2.4%	3.9%	2.0%

Table 4.1. Accuracy of interference inference for various propagation models

based algorithm, interference information was collected 30 seconds after the flows were started. The omni-directional antenna based scenario uses an antenna whose gain is unity in all directions along with the two-ray ground reflection model at the physical layer.

We can see that our algorithm works well for all types of networks with very good accuracy (at least 92% of the interfering link pairs are detected in all cases) without too many false positives. The one hop model underestimates interference by at least 40% in each case. The two hop model can predict interference correctly for about 88% of the link pairs in the best case. However, it mispredicts interference for nearly 24% of the link pairs when we use directional antennas.

4.4.2.2 Effect of Node Density

Next, we study the effect of node density on the performance of the 1-hop and 2-hop heuristics and our algorithm. We vary the average density from 8 to about 24 nodes per radio range and plot the number of false negatives in each case in Figure 4.6. The number of false positives was less than 10 in all cases and is not shown in the graph. We see that the hop-based heuristics perform very poorly at low densities. This is because of two reasons. Firstly, since the area is larger, the network diameter is larger at lower densities which means there are more link pairs separated by more than 2 hops. Secondly, in the case of the 2-hop model, there are fewer intermediate nodes that can forward data to other interfering nodes at lower densities.

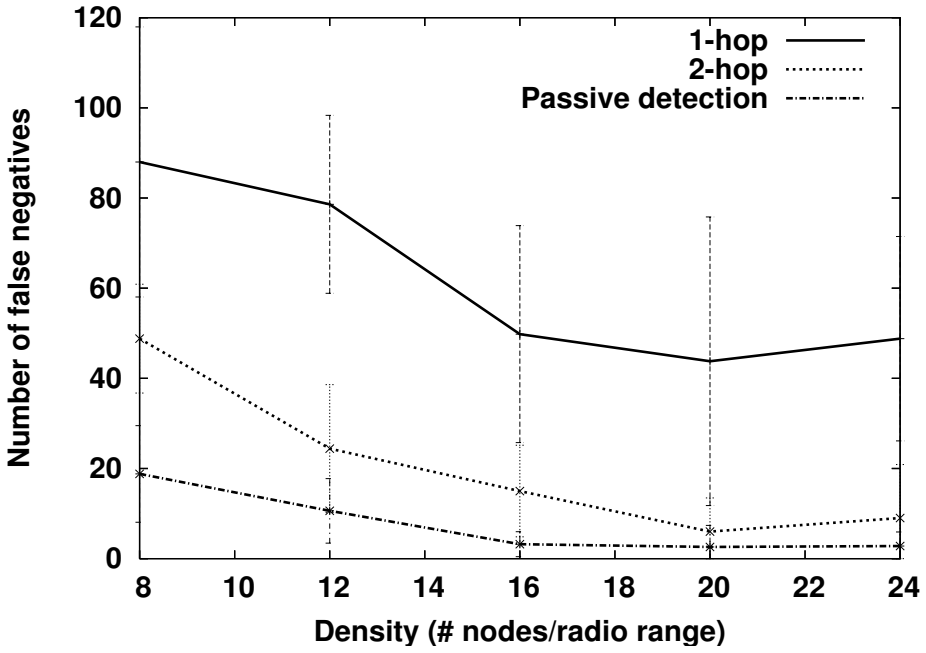


Figure 4.6. The effect of node density on the accuracy for various methods of guessing interference

4.4.2.3 Time to Detect Interference

Figure 4.7 shows the CDF of the time it takes for our algorithm to detect interference (for the correctly detected links). We repeat simulations with 10 simultaneous flows 20 times. In each case, we log the time it takes to detect interference between two links after the time we start sending traffic on both of them. We can see from the plot that we detect 80% of the interfering link pairs within 10 seconds and 90% within 20 seconds.

4.4.3 Test-bed

We will now present the experimental results from our testbed. We start with an analysis of the 1-hop and the 2-hop heuristics in Section 4.4.3.1. We then look at the performance of our algorithm in Section 4.4.3.2.

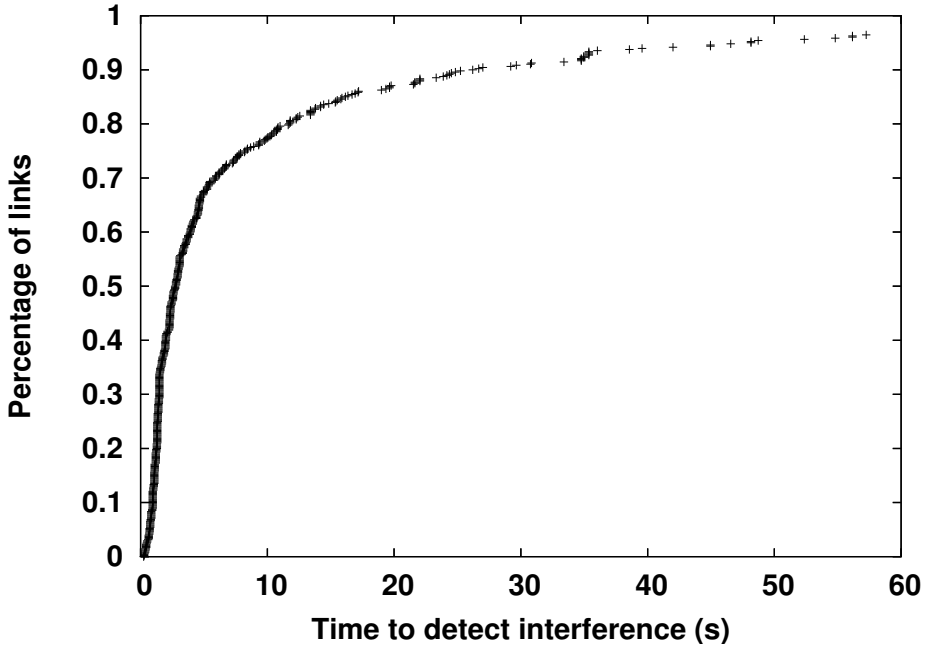


Figure 4.7. The CDF of the time taken to detect interference

4.4.3.1 Performance of 1-hop and 2-hop heuristics

In our first experiment, we measured the loss rate for broadcast packets (modulated at 6 Mbps at the physical layer) sent from each node in our network. We say that a link exists from node A to B if B can receive at least 75% of the packets broadcast by A. According to this definition, there were 191 links in our network.[3]

Next, we measured the interference between all pairs of links by using the methodology described in [46]. By using broadcast packets, we only had to conduct $\binom{30}{2} = 435$ experiments each lasting about 30 seconds. Of the 18,145 link pairs, there were 5125 pairs that did not interfere. In other words, there was a 72% chance that any given pair of links would interfere.

Table 4.2 shows the performance of the 1-hop and the 2-hop heuristics for these links. We can see that the 1-hop heuristic has few false positives whereas the 2-

[3]There were only 6 node pairs where the reception rate was between 20% and 80%. Therefore, any choice of the threshold in this range would've resulted in a similar number of links.

Heuristic	False -ve	False +ve
1-hop	18.0%	0.8%
2-hop	2.2%	15.4%

Table 4.2. Performance of the 1-hop and 2-hop heuristics in our testbed

hop heuristic as has few false negatives. Thus, the 1-hop heuristic underestimates interference whereas the 2-hop heuristic overestimates it.

4.4.3.2 Performance of Passive Detection

In this Section, we evaluate the performance of our algorithm by considering the following scenario. In each experiment, we start two simultaneous TCP flows between randomly chosen sources and destinations in our testbed. A single trial lasts 2 minutes and we repeat this 50 times, starting the network from a cold start in each trial. Since we use passive measurements, we can only detect interference between a pair of links if they were both on the path of a flow in the same trial. Based on the data from our earlier active measurements, there were 290 such link-pairs in total. Our algorithm made 296 inferences of which 282 were correct; in other words, there were 8 false negatives (2.7%) and 14 false positives (4.8%). Thus our algorithm performs better than either the 1-hop or 2-hop heuristic by simultaneously achieving both low false positives and negatives.

Figure 4.8 shows the CDF of the time it takes to detect interference in our testbed. The performance is almost as good as in the simulator; we make 85% of our inferences in less than one minute.

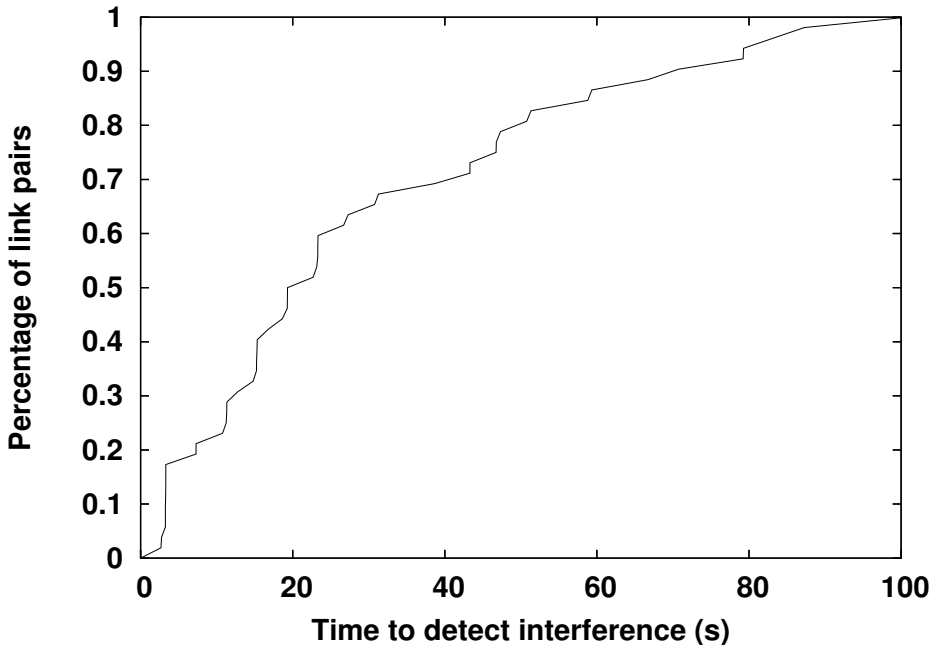


Figure 4.8. The CDF of the time taken to detect interference in our testbed

Chapter 5

End-to-end Fairness using Local Weights

In the Chapter, we present *End-to-end Fairness using Local Weights* (*EFLoW*), a distributed transport-layer algorithm that can compute the correct local weights to be used at the MAC layer. *EFLoW* provides Max-Min fairness (as defined in Section 5.1) between competing flows. We describe the design of *EFLoW* in detail in Section 5.2. Finally, we present results from our simulations and experiments in Section 5.3.

5.1 Max-Min fairness

While there are many different definitions of fairness used in wireless network literature [12], Max-Min fairness appears to be the most popular choice among researchers [60, 61, 51, 41]. The key idea behind Max-Min fairness is that we should not provide more throughput to any flow if it prevents us from giving at-least as much throughput to other flows [12]. Weighted Max-Min fairness is a generalization of this approach with support for weights, and is formally defined below.

5.1.1 Max-Min Fairness

Consider a network with n flows. Let f_i be the throughput received by the i-th flow. The allocation is said to be Max-Min fair if for each flow i, an allocation of $f_i + \epsilon_1$ is possible only by reducing the allocation of another flow j to $f_j - \epsilon_2$ where $f_j < f_i$. On the other hand, if we can find a subset H of flows $\{m_1, m_2, \ldots, m_k\}$, which all have throughput higher than f_i, and if we can increase the throughput of f_i by only decreasing the throughput of flows from H, then the allocation is not fair.

Informally, Max-Min fairness states that inequality in throughput of flows is allowed under the condition that penalizing the richer flows wouldn't provide any benefits to the poorer flows. On the other hand, if we can provide better service to any flow by only penalizing flows that receive a higher throughput, then the allocation is not Max-Min fair.

The above definition does not provide any means to differentiate service between flows. A simple method to address this limitation is to associate a weight with each flow. When comparing two flows, we always check if their throughputs are in proportion to their weights. More formally, if flow f_i is assigned a weight of w_i, we make a simple change to the above definition by modifying the condition that $f_j < f_i$ to $f_j/w_j < f_i/w_i$. Similarly, for all $t : f_t \in H$, $f_t/w_t > f_i/w_i$.

To illustrate this definition, consider the example in Figure 5.1, which shows a network with three flows (f_1, f_2, f_3). Assume an interference model where two links interfere if and only if they share an end-point. Then, (a), (b) and (c) are three possible allocations in terms of the fraction of time for which the source is active. It is easy to see that (a) is unfair because we can increase f_1 without decreasing any other flow's allocation. Similarly, (b) is not fair because we can increase f_3 by decreasing f_2 and $f_2 > f_3$. Finally, (c) is fair even though f_1 receives twice the throughput of

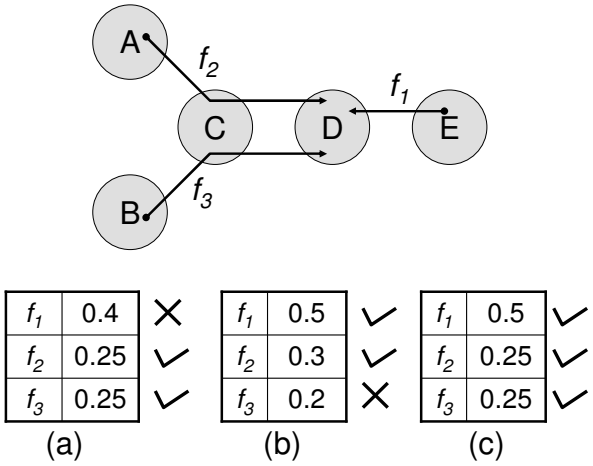


Figure 5.1. Example for Max-Min Fair allocation

the other flows. This is because node C is already saturated and we cannot increase f_2 or f_3 even if we decrease f_1.

Weighted Fair Queueing (WFQ) [21] is a popular method for providing Max-Min fairness when all flows compete for the same resource. WFQ states that throughput must be assigned to flows in proportion to their weight. If the allocation to any flow is more than its demand, the surplus is again divided among the rest of the flows in proportion of their weights. This process continues until there is either no surplus capacity or when all demands are satisfied. In wired networks, per-hop WFQ guarantees end-to-end Max-Min fairness. But because of the shared nature of the medium, this is not true for wireless networks.

5.2 Achieving End-to-end Fairness

In this Section, we describe *EFLoW* , a distributed transport layer algorithm for achieving end-to-end fairness. Though our current implementation of *EFLoW* uses WSA, it can also work on top any MAC layer that provides the higher layers control over resource allocation within a single contention neighborhood. More specifically, we assume that the MAC layer allows the transport layer to dynamically associate a weight with each node, and allocates transmission opportunities to each node in a contention region proportional to its weight by implementing a Weighted Fair Queueing (WFQ) discipline [21].

We use the term *local weight* to refer to the weights at the MAC layer. They are only used to resolve conflicts between directly competing nodes. In contrast, the weights defined in Section 5.1 are assigned to flows (not nodes or links) and have global significance. In Section 5.2.1, we detail our system model, and show why achieving end-to-end fairness is hard even when the MAC provides support for local fairness. We then present the design of *EFLoW* in Section 5.2.2.

5.2.1 System Model

For ease of explanation, we first consider a network in which all nodes are in the same contention region. We generalize this to a multi-hop network in the next step.

5.2.1.1 Stage 1: Single Contention Region

Consider a simple network consisting of three nodes (A, B and C), where there are three flows originating at each node (Figure 5.2). Scheduling happens at two levels in our system. First, within a node, we have to decide which packet will be sent to the wireless interface. Second, at the MAC layer, we have to decide which node transmits

in each time slot. Unless otherwise specified, we assume that a node uses WFQ to schedule its flows [21].

Since all nodes compete with each other, ideally we would like to achieve the functionality of a single WFQ system with six queues. Thus, to achieve max-min fairness, we have to assign to each node a local weight that is the sum of the weights of all its *backlogged* flows. Note that a flow f is said to be *backlogged* at a node N if the arrival rate of f at N is greater than its service rate. For instance, the weight of node A, W_A, should be computed as $W_A = w_{11} + w_{12} + w_{13}$, assuming that all flows of node A are backlogged. In this case, the allocation to the first flow is given by,

$$\begin{aligned} f_{11} &= \frac{w_{11}}{w_{11} + w_{12} + w_{13}} \times (\text{Alloc. to A at MAC}) \\ &= \frac{w_{11}}{w_{11} + w_{12} + w_{13}} \times \frac{W_A}{W_A + W_B + W_C} \times C \\ &= \frac{w_{11}}{W_A + W_B + W_C} \times C \end{aligned}$$

Thus, computing a schedule at the MAC layer requires knowledge of the demands of all six flows as well as the capacity C at the MAC layer.

5.2.1.2 Stage 2: Mutli-hop Network

To illustrate the additional challenges that arise in a multi-hop network, we revisit the example topology in Figure 5.1. For a multi-hop network, a *contention region* is defined as a maximal set of nodes where all pairs interfere with each other. Since no two nodes in the same region can transmit simultaneously, the total allocation in each region has to be less than the channel capacity. Assuming a one-hop interference model, Figure 5.3 shows a network in which (a) there are multiple contention regions and (b) the same node (node C) is a part of multiple contention regions. The final allocation to node C is the *minimum* of its allocation from regions 1 and 2. So if we

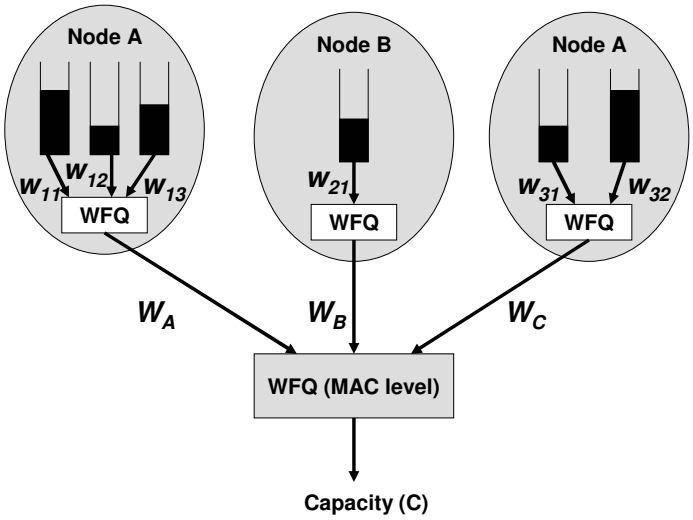


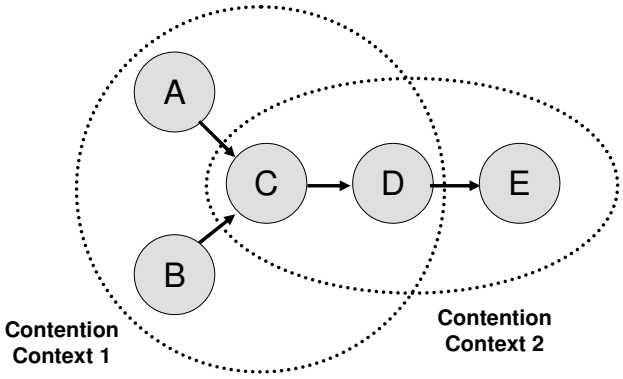
Figure 5.2. Scheduling within a single contention neighborhood

assume that region 1 is more congested, it will determine the final allocation at C. This in turn means that the queues in C will appear as backlogged in region 1, but as not backlogged in region 2. Because of this dependence, we also need the information from region 1 to compute the local weights for nodes in region 2.

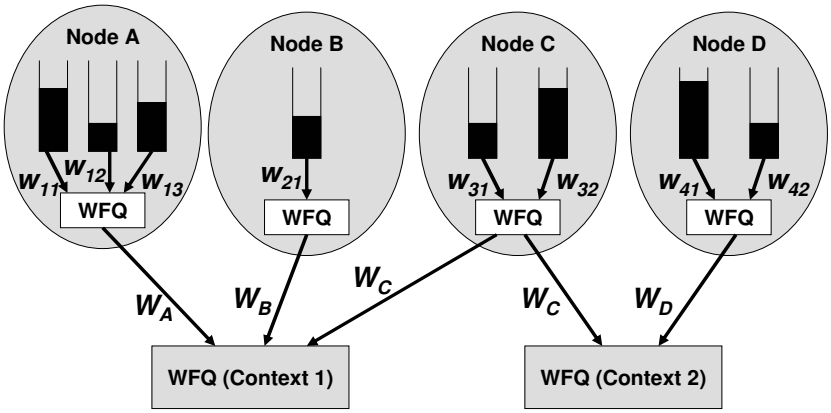
To further complicate matters, the arrival rate of a multi-hop flow at a node depends on its service rate at the previous hop. This introduces a dependence between contention contexts that may be far away from each other in the network topology. Because of the cascading effect of these dependencies we cannot determine the local weights without complete knowledge of the state of the network. This includes the topology, the interference patterns, routes and the flow demands all of which can change fairly frequently. In fact, since computing the Max-Min fair allocation itself is NP complete [28], computing the weights is also an NP complete problem.[1]

Next, we present a *decentralized* algorithm that aims to approximate the ideal

[1]Given the local weights, we can directly compute the resulting allocation in linear time.



(a)



(b)

Figure 5.3. Contention contexts in a multi-hop network

allocation without requiring global state. At any given node, our algorithm uses only information gathered from its contention region, i.e., from nodes it directly competes with. Note that this property in conjunction with using the promiscuous mode allows us to piggyback almost all control traffic on existing traffic.

5.2.2 EFLoW

The key idea behind our algorithm is that instead of trying to compute the local weights directly, we use an iterative process that brings us closer to the correct allocation in each step and oscillates around it at steady state. The next theorem provides

the theoretical building block of our design by providing an alternate characterization of Max-Min fairness.

Definition 1. *If flow f_1 is backlogged at node N_1, we define this backlog to be a* permissible backlog *in a contention region C iff for every flow f_2 passing through node $N_2 \in C$, the service rate of f_2 at N_2 is no greater than the service rate of f_1 at N_1 (in proportion to the weights of f_1 and f_2).*

Theorem 2. *The allocation is Max-Min fair if and only if every node where any flow is backlogged is a permissible in at least one contention region.*

Proof. For ease of explanation, we assume that every flow has unit weight (the proof can easily be generalized to non-unit weights). The proof is by contradiction.

$\Longrightarrow$: Assume the contrary, *i.e.,* there is a flow f backlogged at a node N which is impermissible in *every* contention region C_j that N is a part of. In other words, there is at least one flow f_j through $N_j \in C_j$ where $f_j > f$. By reducing the allocation to all f_j, there is some surplus capacity in every contention region that contains N. Thus we can increase the allocation to N and hence to f, which violates the definition in Section 5.1.

$\Longleftarrow$: We first make the observation that in order to increase the allocation f of a flow, we have to increase its allocation at at-least one node N where it is backlogged. In order to do this we have to decrease the transmit rate of at least one node N_j in every contention region C_j such that $N \in C_j$. Without loss of generality, assume that the backlog at N is permissible in C_1. Since no flow at $N_j \in C_1$ is being serviced faster than f, we cannot decrease the allocation of N_j without violating the definition of Max-Min fairness.

□

Note that the condition in Theorem 2 can be verified at each node using only in-

formation gathered from its own contention neighborhood(s). In contrast, our earlier definition required a global view of the network. In a nutshell, our approach is to identify impermissible backlogs and use an additive-increase multiplicative-decrease algorithm (similar to the TCP's congestion control algorithm) to adapt the weights to fix it. Initially, we set the local weights of all nodes to 1 and perform the following operations periodically:

1. Service nodes using their current local weights for a time window, W. W is a system-wide constant expressed in terms of the number of time slots. Within each window, each node receives a number of time slots proportional to its weight.

2. Estimate the service rate of a unit-weight backlogged flow at each node over the previous window.

3. Compare each backlogged node N's service rate with the service rate of every node it competes with. If there are nodes with a higher service in *every* contention region N is a part of, add N to the set I.

4. At the beginning of each time window, add a positive constant α to the local weight of all nodes in the set I (additive increase). Reduce the local weight of all other nodes by multiplying it by $(1 - \beta)$ where β is a constant between zero and one (multiplicative decrease).

Note that step 3 identifies all nodes with impermissible backlogs and adds them to the set I. In step 4, we increase the weights of all these nodes while decreasing the weight of nodes which are either (a) not backlogged or (b) have only permissible backlogs. This moves to system closer to its ideal state in the next time window.

To implement step 2, each node has to keep track of the service rates of its queues. Since we use a *virtual time* (VT) based implementation of WFQ for scheduling flows

within a node, the required book-keeping is already taken care of. Recall that VT measures the service (*i.e.,* the number of bits) received by a continuously backlogged flow with unit-weight. Let t_W be the start of a window period, and let N.VT(t) denote the virtual time of node N at time t. Then, N.VT($t_W + W$)−N.VT(t_W) represents the service received by a continuously backlogged flow during the window starting at t_W. Note that this service is exactly the value we need at step 2 within a multiplicative constant, *i.e.,* $1/W$. Thus, at step 3 we can directly use the progress of the virtual time in the previous window to compare the service rate of queues at different nodes.

Step 4 requires broadcasting the the service rates of queues in a two-hop neighborhood. For 1-hop neighbors, this is easily accomplished by including the current virtual-time as part of the *EFLoW* header of every packet. By listening in promiscuous mode, nodes can easily determine the service rates of their immediate neighbors. In order to disseminate this information to 2-hop neighbors, every node also includes a list of its 1-hop neighbors and their service rates in the *EFLoW* header.[2] As in Chapter 4, 3-hop neighbors which interfere are treated as a special case - we explicitly request a 1-hop neighbor to forward information about them. Due to this *lazy* approach, nodes may not always have the most up-to-date information about their two-hop neighbors. But in practice, we found that the AIMD algorithm is robust enough to converge despite this limitation.

Figure 5.4 shows the pseudocode for updating the local weights. We experimented with values of $W = 20$ and $W = 40$ time slots and found that either choice works well in practice. The function *queue.ExpWtAvg()* returns the exponentially weighted average queue size which we use to determine if the node is backlogged.

[2]The *EFLoW* header is variable in length, and was 48 bytes on average in our experiments. Further optimizations (compression, differential coding etc.) are possible, but are outside the scope of this work.

```
function Node::ProcessEndWindow()

    //first check if we are backlogged
    if queue.ExpWtAvg() < congestionThreshold
        //not congested hence multiplicative decrease
        weight ← weight × (1 − β)
        return

    myServiceRate ← VT(currTime) −
                    VT(currTime − W)

    minMaxServiceRate ← ∞
    for all C ∈ N.GetContentionRegions()
        cServiceRate = C.GetMaxServiceRate()
        if cServiceRate < minMaxServiceRate
            minMaxServiceRate = cServiceRate
    endfor
    if minMaxServiceRate > myServiceRate
        //additive increase
        weight ← weight + α
    else
        //congested, but must go into MD
        weight ← weight × (1 − β)
```

Figure 5.4. Pseudocode for the function executed by every node at the end of W time slots

5.3 Results

We have implemented *EFLoW* on top of OML in both *ns-2* and in Click. We present results from our simulations in Section 5.3.1 followed by the results from the MIT test-bed in Section 5.3.2. But first, we start with a brief overview of the metrics we use.

The two important aspects of the system we wish to quantify are its *efficiency* and *fairness*. To quantify the fairness of the outcome, we again use the fairness index as defined in Section 3.4. One possible way to quantify efficiency is to simply look at the average throughput of a flow in the system. However, a key drawback of this metric is that the resource demands for supporting a given throughput is not the same across different flows. For example, it takes three times as many wireless transmissions to provide the same throughput to a flow routed over three hops compared to a flow where the source and sink can communicate directly. Instead, we use the number of useful transmissions per second as the metric for efficiency. A packet transmission is considered to be *useful* if (a) the packet is eventually delivered to the end point of the flow and (b) the packet is not a duplicate when it is received at the end point.

If we overestimate the interference pattern, OML and *EFLoW* will prevent links from sending at the same time even though they do not interfere. This means that our efficiency will suffer compared to using the correct interference pattern. On the other hand, if we underestimate interference, links that do interfere with each other will be allowed to transmit in the same time slot. As a result, the accuracy of resource allocation in OML will suffer, and hence the fairness seen by *EFLoW* will also be suboptimal.

5.3.1 Simulations

We have implemented our algorithm in *ns2* for the following three reasons. Firstly, we can easily change the topology or the size of the network in a simulator. Secondly, the simulator allows us to implement an Oracle which uses global state to approximate the ideal allocation. We use this Oracle (see Section 5.3.1.2) for comparison with *EFLoW*. Thirdly, the simulator allows us to implement an idealized TDMA MAC so that we can evaluate *EFLoW* independent of the inefficiencies introduced by OML (see Section 5.3.1.1).

As in Chapter 4, we simulate 802.11b radios at 2Mbps in each of our experiments. RTS/CTS was disabled in all the experiments because there is no need for it when using time slots. Surprisingly we saw better fairness and throughput with RTS/CTS disabled even when not using OML or a TDMA MAC. As reported in some earlier work [22, 7], our observations confirm that the overhead of RTS/CTS is not justified in relation to the number of collisions it prevents.

5.3.1.1 TDMA MAC Protocol

Since OML is designed to run on 802.11 hardware without any changes to the standard, it has a number of limitations compared to a TDMA MAC implemented with hardware support. For example, OML uses time-slots which are long enough to send 10 packets of about 1500 bytes each. The clock synchronization and enforcement at the boundaries of slots are not perfect either. In order to evaluate *EFLoW* independent of these limitations, we have implemented an idealized TDMA MAC in ns-2 as follows. The slot duration is calculated as the time it takes to transmit an MTU-sized packet, receive an acknowledgment plus a guard time[3] of 200 microsec-

[3]The guard time is a short idle time between two slots. It is used in TDMA MACs to provide robustness against clock drift and small errors in clock synchronization.

onds. We decided to implement link-layer acknowledgments to prevent unnecessary retransmissions and time-outs at the TCP layer. We included carrier-sensing also to protect against imperfect interference information while using the 1-hop or 2-hop heuristics.

5.3.1.2 Ideal Allocation

As mentioned earlier, computing the ideal allocation is a hard problem even in the case of a simulation. Instead of trying to compute it exactly, we use the following slot allocation algorithm (called the *Oracle*) to approximate the ideal schedule. The algorithm uses global knowledge to sort all nodes based on how much service their queues are receiving. We can then prioritize assignment of slots to nodes which have received less service. The pseudocode for the Oracle is shown in Figure 5.5. Note that this is only an approximation since we break ties arbitrarily in *GetLeastServiced*.

To quantify the benefits of *EFLoW*, our first step is to look at the convergence time of our iterative algorithm (Section 5.3.1.3). Next, we look at the effect of the number of simultaneous flows in Section 5.3.1.4. After that, we study the impact of the accuracy of the interference pattern in Section 5.3.1.6 followed by the effect of the traffic pattern in Section 5.3.1.5. Finally, we study the performance of TCP flows in Section 5.3.1.7 and the scaling behavior of *EFLoW* when increasing the network size in Section 5.3.1.8.

5.3.1.3 Convergence of *EFLoW*

To get a better understanding of the convergence of *EFLoW*, we start by looking at the effect of the parameters α and β. In order to do this, we first construct a network of 30 nodes with an average density of 16 nodes per radio range. Nodes are equipped with omni-directional antennas and the signal propagation follows the

```
function ::AssignSlot(NodeList NL)

    for all N ∈ NL
        N.active = false
    endfor
    while NL ≠ φ
        N = NL.GetLeastServiced()
        N.active = true
        NL.Remove(N)
        for all M ∈ NL
            if M.Interferes(N)
                NL.Remove(M)
        endforendwhile
```

Figure 5.5. Pseudocode for the Oracle to approximate the ideal allocation

two-ray ground reflection model. We simulate 8 simultaneous flows with a duration of 30 seconds in each experiment. We also record the throughput of each flow at the end of 2 seconds and 5 seconds. The traffic pattern is a *star i.e.,* sources are chosen at random but all flows terminate at the same node.[4]

Figure 5.6 shows the fairness index as a function of α (β is fixed at 0.5). Since we are only trying to look at the convergence process, we simulate the TDMA protocol described in Section 5.3.1.1. We also assume that the exact interference pattern is available for the MAC layer; this can easily be obtained from global state in the simulator. Surprisingly, the fairness index (especially after 30 seconds) does not vary much even when we change α by nearly three orders of magnitude (from 0.001 to 0.5). This reason for this follows from the fact that the absolute values of the local weights do not matter in WSA; the allocation would be the same even if we multiply all local weights by the same constant factor. Therefore, the behavior of *EFLoW* will also not change if we multiply both α and the initial weights by the same constant. In other words, multiplying α by a constant c *has the same effect* as dividing the initial weights by c. Thus, Figure 5.6 shows that the value of α does not affect the convergence of *EFLoW*; the differences in fairness at 5 seconds (due to differences in the initial conditions) disappear after 30 seconds.

Next, we fix $\alpha = 0.5$ and vary β from 0.001 to 0.9 and plot the fairness in Figure 5.7. We can see that the value of β has a much greater effect on the fairness. As mentioned in [16], higher values of β lead to both quicker convergence and larger oscillations after convergence. $\beta = 0.5$ appears to be the best compromise; the fairness after 5s is comparable to using higher values of β. At the same time the oscillations are small also; the standard deviation in fairness after 30 seconds is the lowest for $\beta = 0.5$.

[4]We consider the star traffic pattern to be indicative of the scenario where all nodes are trying to reach the same wired gateway.

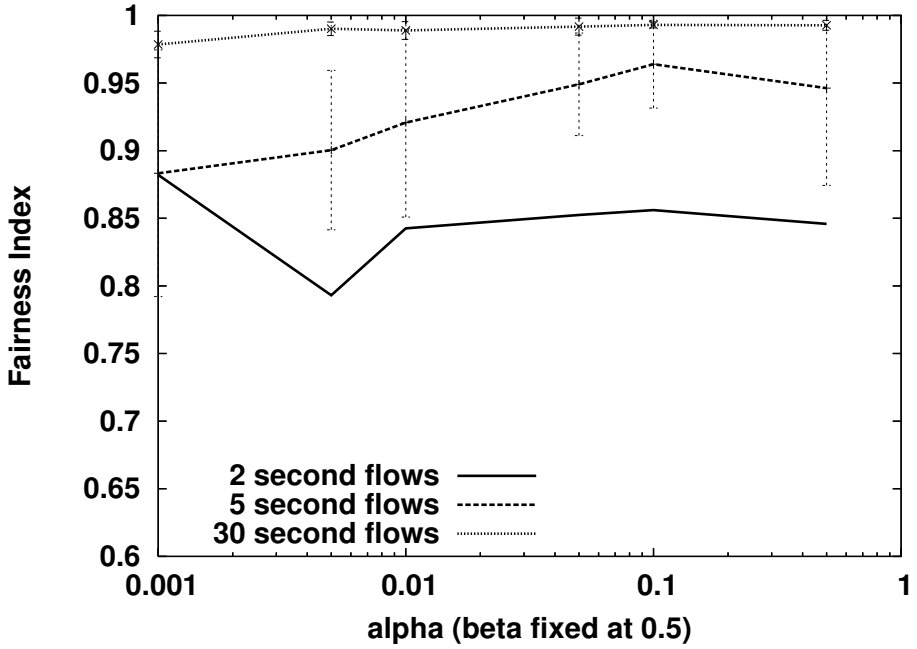


Figure 5.6. Effect of the additive increase parameter α on fairness.

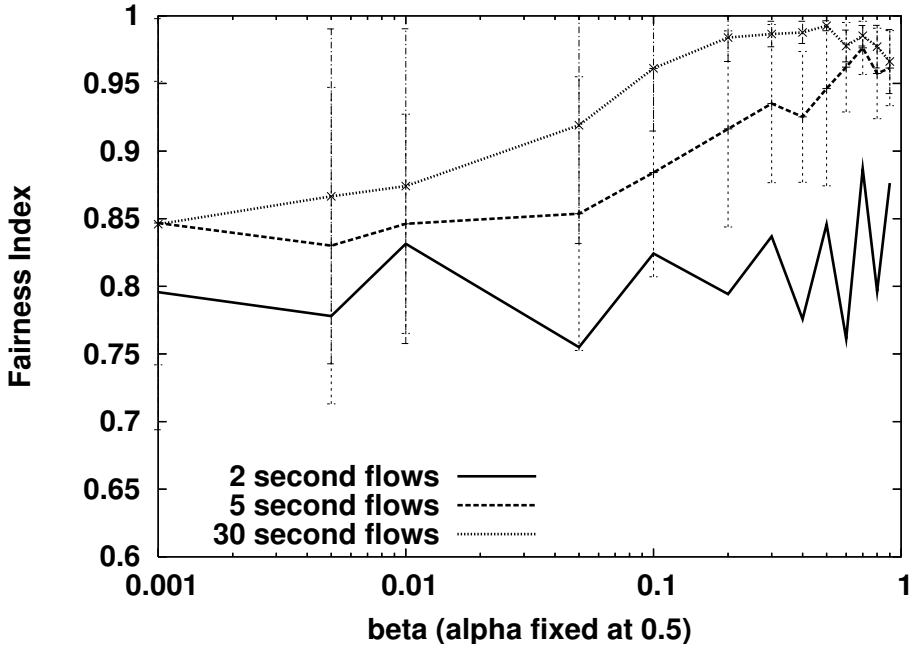


Figure 5.7. Effect of the multiplicative decrease parameter β on fairness.

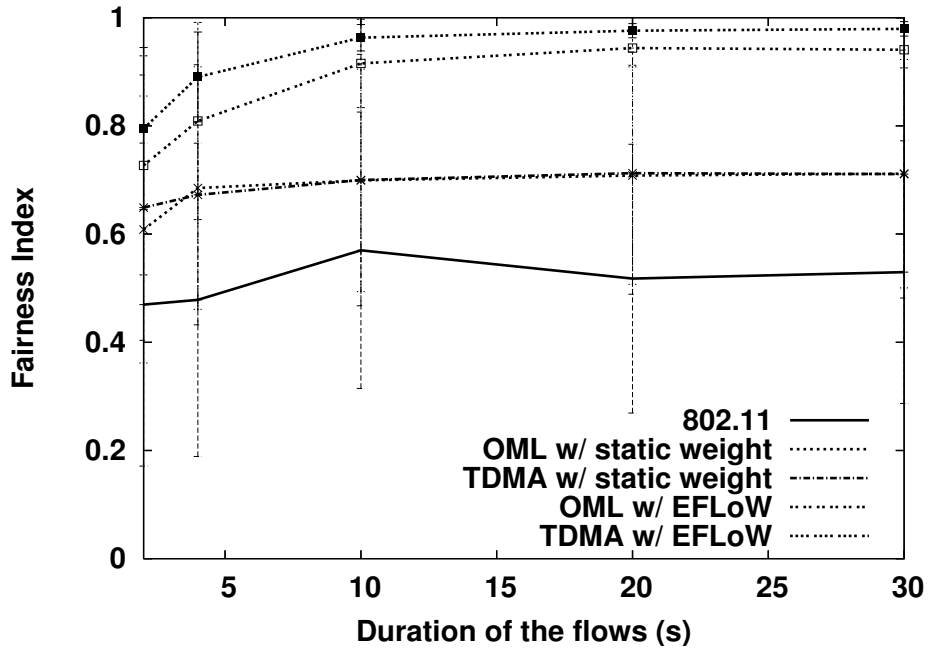


Figure 5.8. Effect of the duration of the flows on fairness with and without *EFLoW*

Finally, we compare the performance of *EFLoW* with that of (a) the standard 802.11 protocol and (b) WSA with static weights, *i.e.,* we do not use *EFLoW*, by plotting the fairness as a function of the duration of the flows in Figure 5.8. We repeat (a) and (b) for both OML and the TDMA MAC. We can see that both OML and the TDMA MAC do not perform well with static weights; the fairness index, while better than 802.11, is below 0.7 in all cases. This shows that fairness within a single contention neighborhood does not directly give us end-to-end fairness in multi-hop wireless networks. *EFLoW* converges within 10 seconds irrespective of the length of the slots; the fairness index is better than 0.9 in both cases.

5.3.1.4 Effect of the number of flows

In this experiment, we construct the network as in the previous section and simulate between 2 and 10 simultaneous flows each lasting 100 seconds. Figures 5.9 and 5.10 plot the fairness index for OML and the TDMA MAC respectively. The

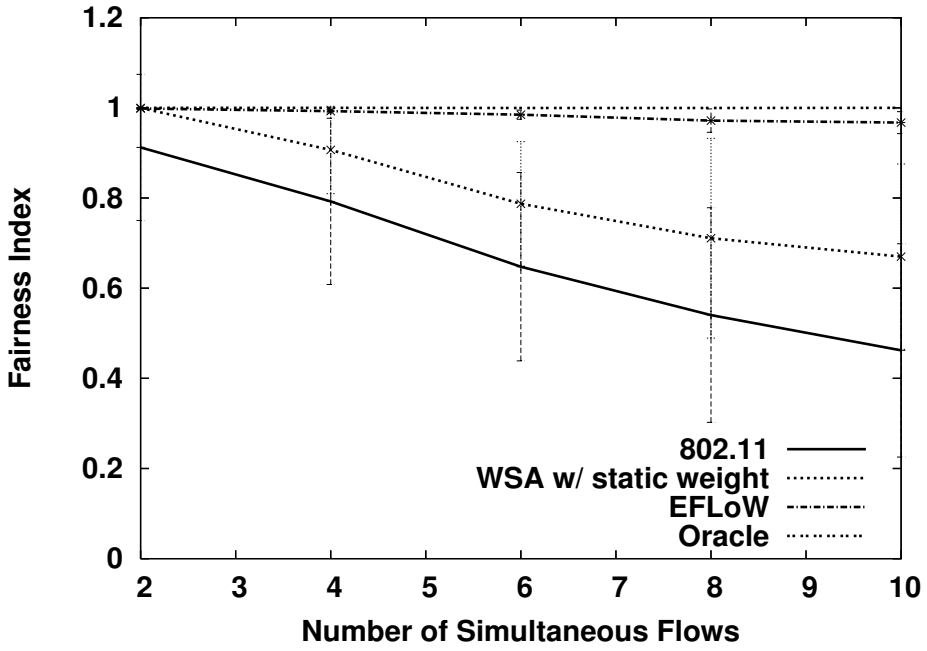


Figure 5.9. Effect of the number of simultaneous flows on fairness with and without *EFLoW* when using the TDMA MAC (star traffic)

802.11 MAC shows poor fairness (index below 0.6) when there are 6 or more flows. OML with static weights does not improve on this much. On the other hand, *EFLoW* shows very little degradation in fairness even with a large number of flows; the fairness index is always greater than 0.9 (0.95 with the TDMA MAC). The Oracle yields a fairness index which is almost exactly 1 in all cases.

Next, we compare the utilization of the transmission medium under various schemes. WSA and *EFLoW* can lead to some inefficiency due to the control overheads. However, the efficiency can also suffer under 802.11 because of the poor fairness; a lot of transmissions may not reach their destination if an intermediate hop experiences starvation. To quantify this overhead, we plot the number of *useful* transmissions per second versus the number of flows in Figures 5.11 and 5.12 (for the TDMA MAC and OML respectively). The efficiency of *EFLoW* is close to, and in most cases even better than 802.11.

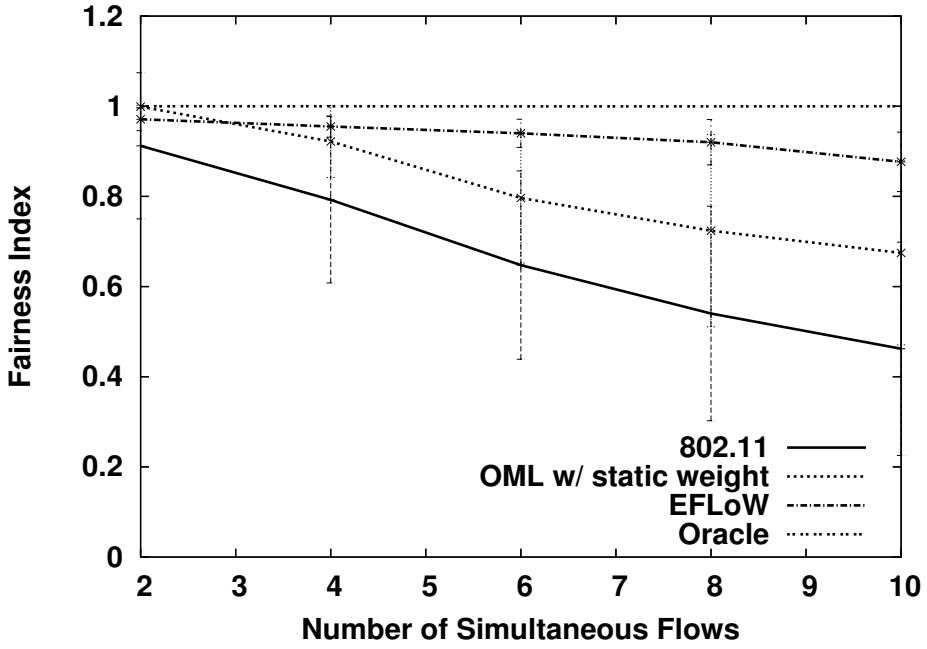


Figure 5.10. Effect of the number of simultaneous flows on fairness with and without *EFLoW* when using OML (star traffic)

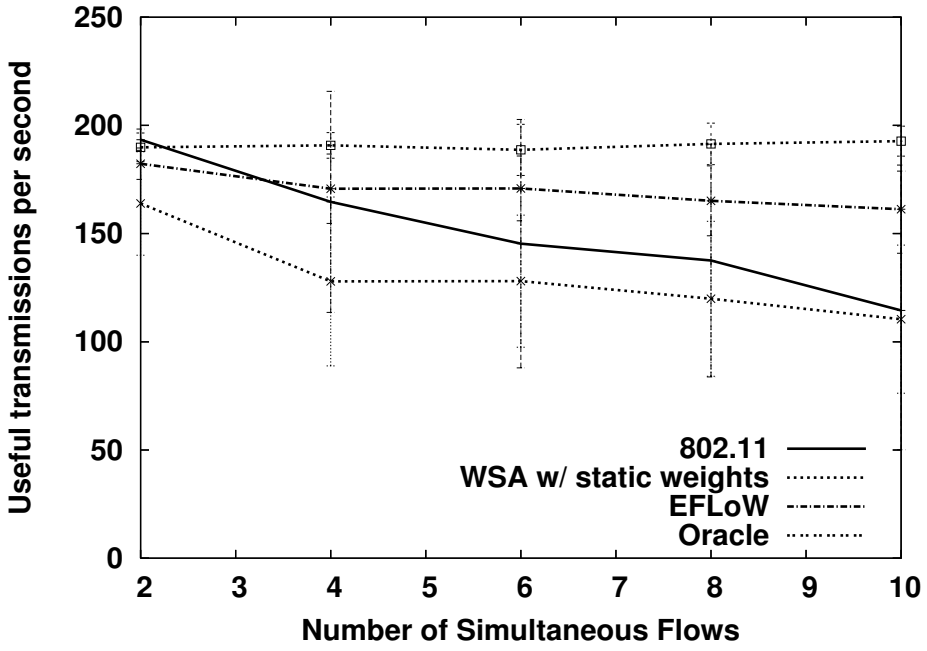


Figure 5.11. Effect of the number of simultaneous flows on utilization with and without *EFLoW* when using the TDMA MAC (star traffic)

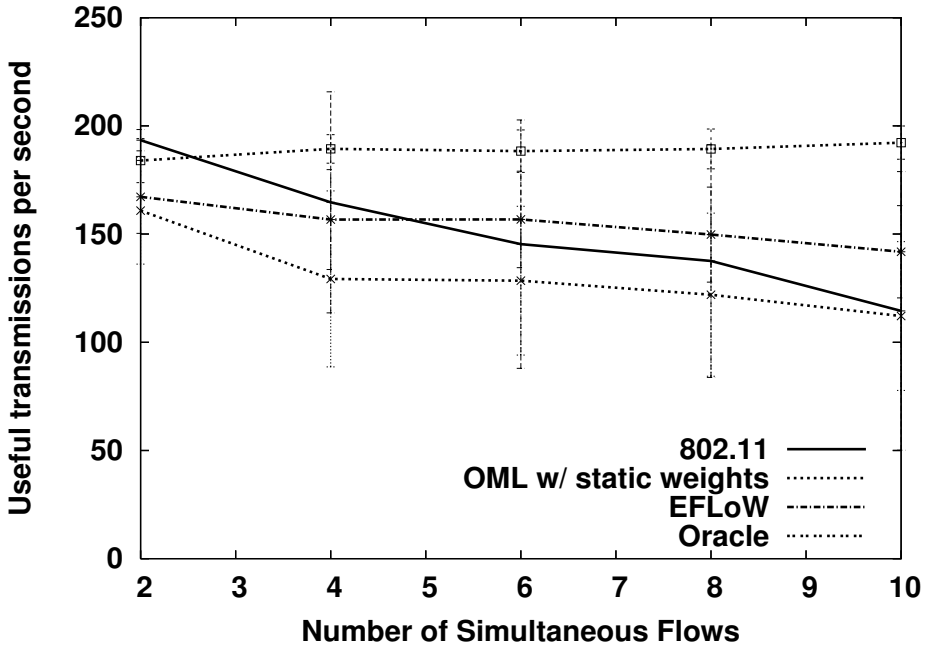


Figure 5.12. Effect of the number of simultaneous flows on utilization with and without *EFLoW* when using OML (star traffic)

5.3.1.5 Random flows vs. Star traffic pattern

We now study the effect of the traffic pattern on the performance of *EFLoW* . In addition to the star pattern, we simulate flows with both the source and destination chosen at random.[5] We reproduce the plots in Figures 5.9 and 5.11 for the random traffic pattern in Figures 5.13 and 5.14 respectively. Note that the fairness index is lower than 1 even using the Oracle; this is because not all flows have the same throughput even in the correct max-min fair allocation. Also, the difference in fairness between the various schemes is much lower in this case. This is explained by the fact that there is no single hot-spot (the common sink in the star pattern); most flows experience a lower level of contention.

Next, we take a closer look at the distribution of the flow throughputs when there are 10 simultaneous flows. For each flow, we look at the ratio of the its allocation under

[5]We call this the *random* pattern.

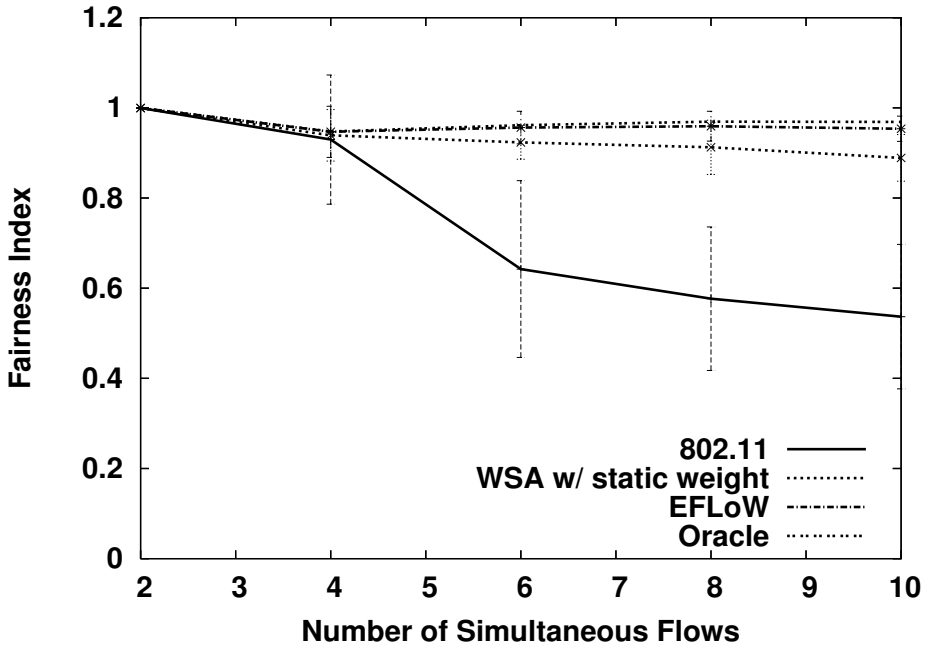


Figure 5.13. Effect of the number of simultaneous flows on fairness with and without *EFLoW* when using the TDMA MAC for the random traffic pattern

EFLoW and 802.11 to the ideal value obtained from the Oracle. If the allocation is good, we expect most of these ratios to be close to 1. We plot the CDF of these ratios for the star and random traffic pattern in Figure 5.15. We see that while *EFLoW* performs slightly better (more numbers closer to 1) for the random pattern because of the lower level of contention, the performance is very good and much better than 802.11 in both cases.

5.3.1.6 Effect of accuracy of interference information

In this Section, we look at the effect of imperfect interference information on *EFLoW* . To do this, we compare the performance of the 1-hop and 2-hop heuristics and our passive detection algorithm against using the actual interference pattern. For each method of determining interference, we simulate a varying number of simultaneous flows with both source and destination chosen at random. Figures 5.16 and 5.17 show the fairness obtained for the hop-based and the directional antenna

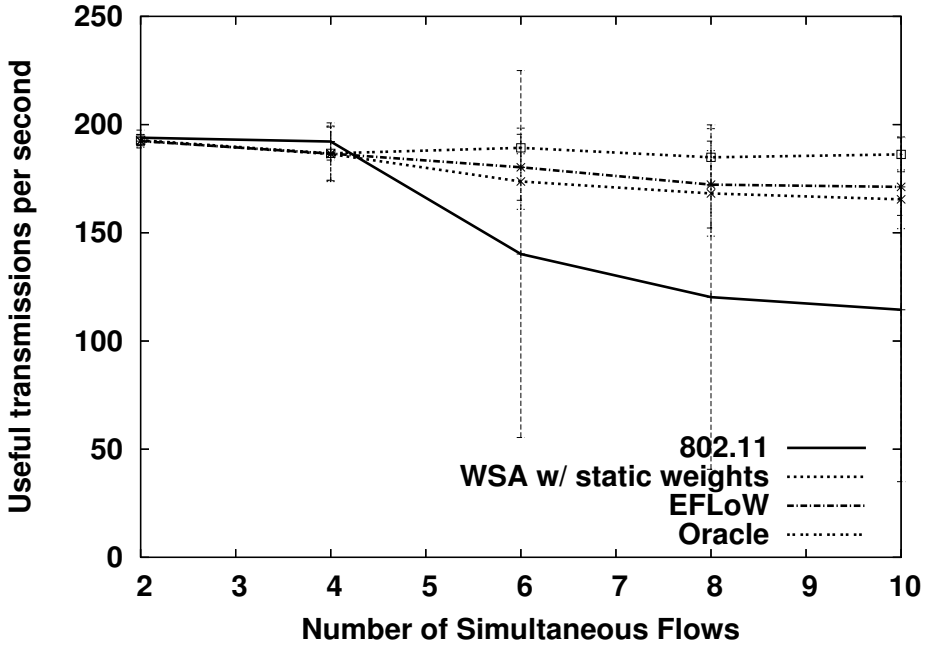


Figure 5.14. Effect of the number of simultaneous flows on utilization with and without *EFLoW* when using the TDMA MAC for the random traffic pattern

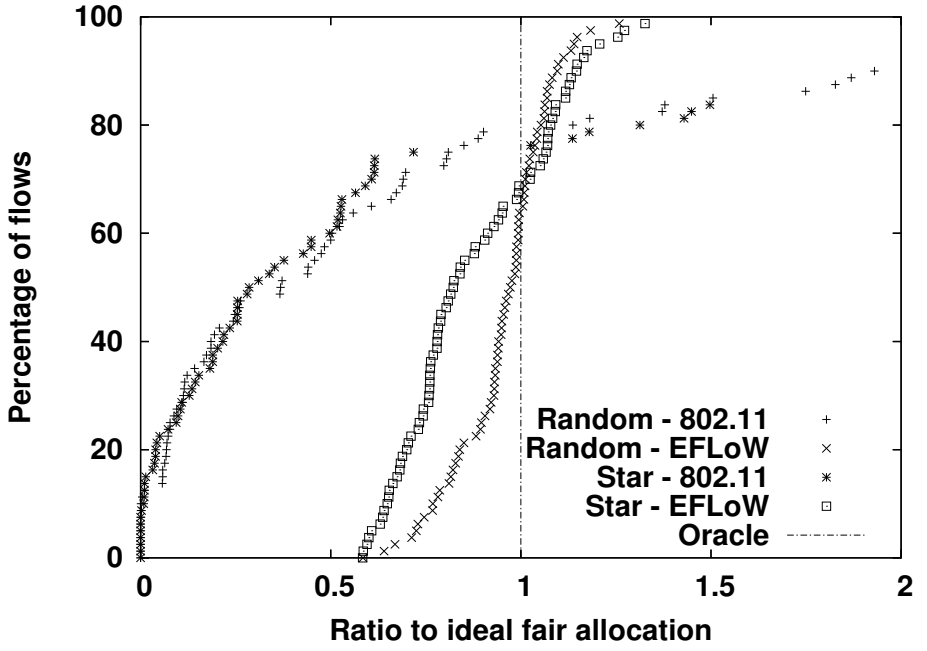


Figure 5.15. CDF of the ratio of allocation to the ideal under 802.11 and *EFLoW* (with TDMA MAC) for the star and random traffic patterns

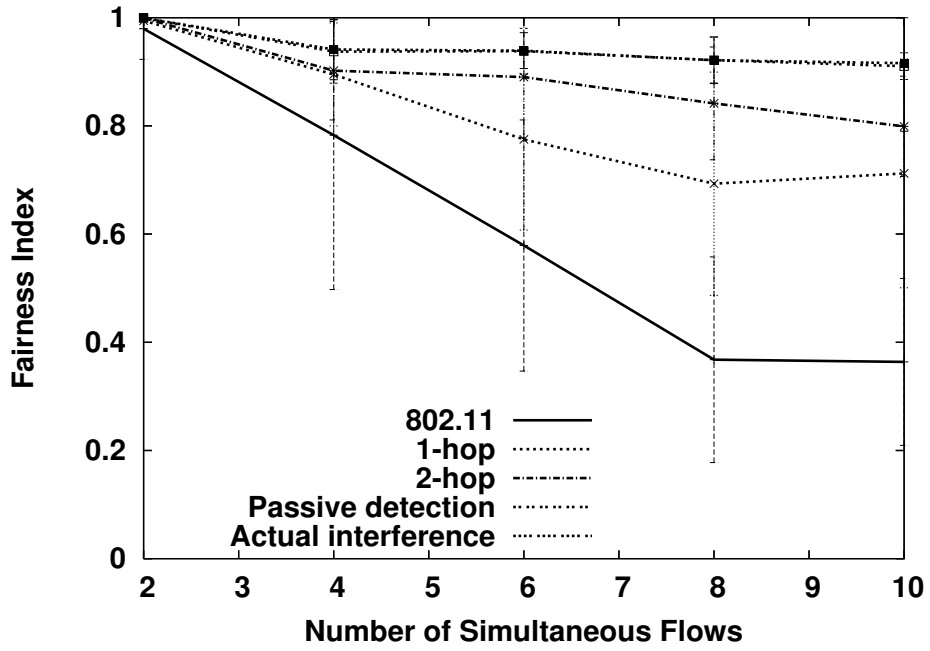


Figure 5.16. The effect of the interference information on fairness when using *EFLoW*(Hop-based interference)

based propagation models respectively (see Section 4.3 for more information about these models). We can see that though the reduced accuracy of the 1-hop and 2-hop heuristics affects the fairness of *EFLoW*, especially in the case of directional antennas, we still obtain better performance than 802.11 in all cases. On the other hand, the performance when using our detection algorithm is almost as good as having the exact interference pattern.

Figure 5.18 looks at the efficiency obtained for the hop-based interference model. We can see that the few false positives that we get with our learning algorithm do not impact system efficiency by much. However, the 2-hop model leads to about 18% loss of efficiency even though the number of false positives is very similar. This is explained by the fact that the amount of information available for detecting interference is directly proportional to the amount of traffic carried by a link. Thus our algorithm tends to have better accuracy for the more *important* links. We obtained similar results for the directional antenna based propagation model also (see Figure 5.19).

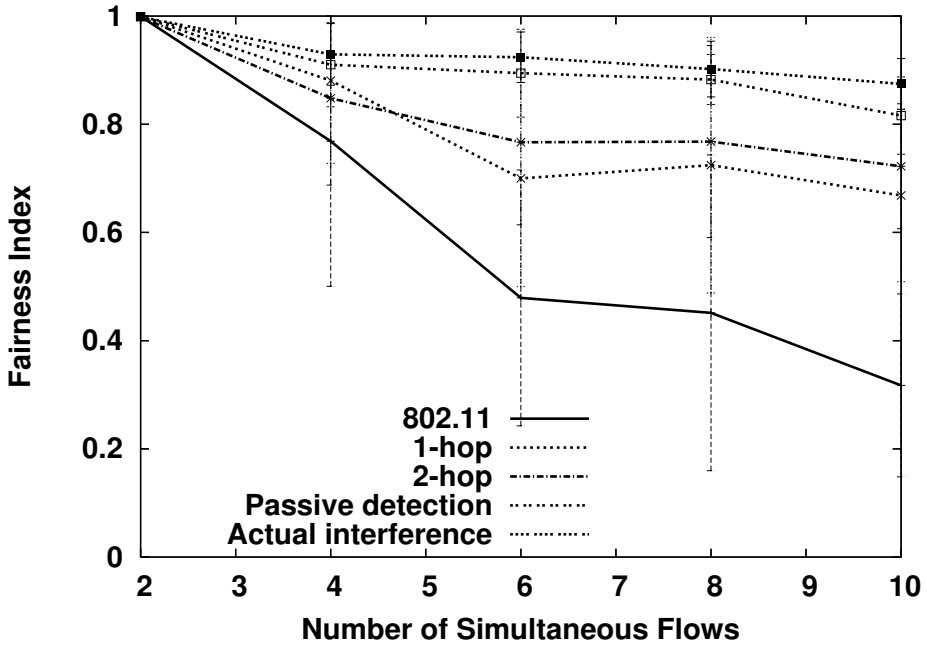


Figure 5.17. The effect of the interference information on fairness when using *EFLoW*(Directional antenna)

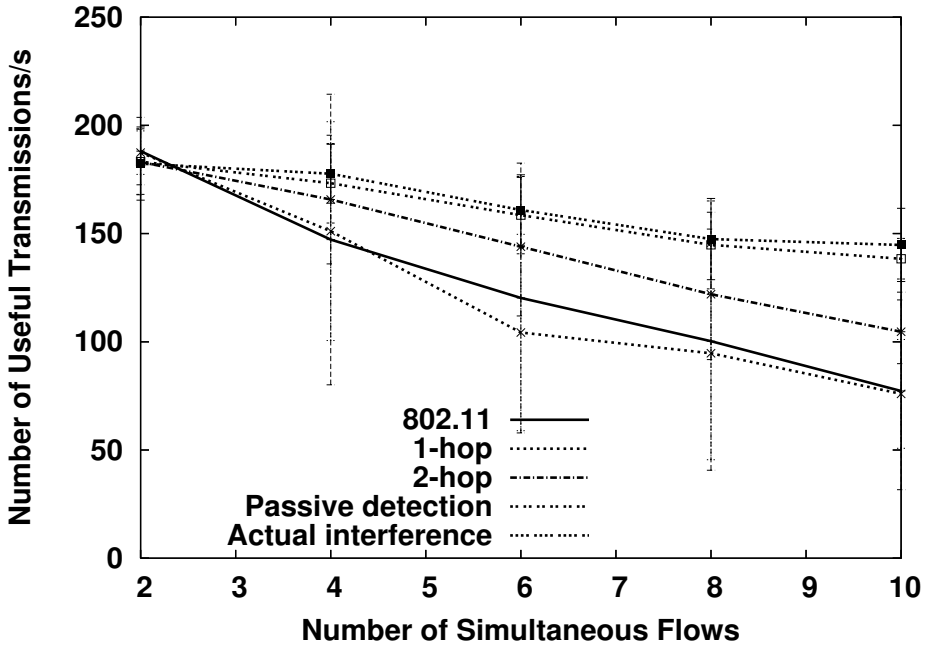


Figure 5.18. The effect of the interference information on efficiency when using *EFLoW*(Hop-based interference)

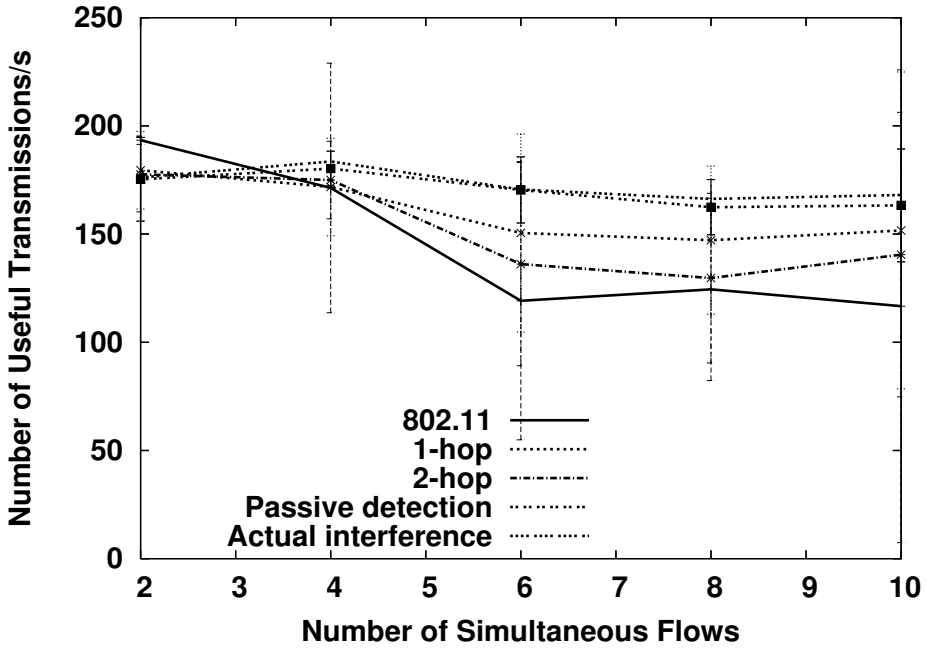


Figure 5.19. The effect of the interference information on efficiency when using *EFLoW*(Directional antenna)

5.3.1.7 TCP flows

In this Section we investigate the performance of TCP flows with and without *EFLoW*. To get a closer look at the throughput of the different flows, we use a CDF plot instead of the fairness index. For each type of flow (TCP, UDP with and without *EFLoW*), we repeat experiments with 10 simultaneous flows 10 times and thus obtain 100 flow throughputs. The resulting CDF plots are shown in Figure 5.20. The first line shows the starvation problem under 802.11; 30 flows have throughput less than 50kbps. In other words, when competing with 9 other simultaneous flows, there is a 30% chance that very little data will reach the destination. From the second line in Figure 5.20, we can see that when using the TDMA MAC, *EFLoW* performs similarly for both UDP and TCP. While the absolute throughputs are slightly lower for TCP due to the additional bandwidth used by the ACKs, the shape of the two curves, and hence the fairness is very similar. But this no longer the case when we use *EFLoW* with OML; about 10 flows receive less than 10kbps.

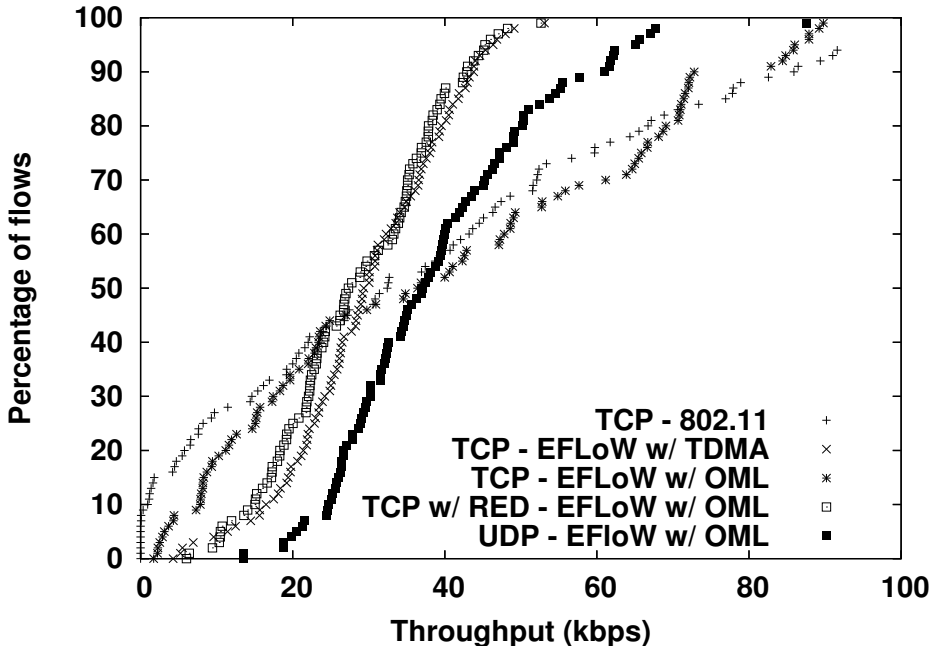


Figure 5.20. Comparison of performance of TCP and UDP flows

Upon further investigation, we found that this was due to the coarse-grained nature of time-slots in OML. When the queue at a node is full, all packet arrivals will be dropped until the node is allowed to transmit again. This translates to a huge burst of losses which can force TCP flows back into slow-start. However, we were able to avoid this by implementing a variation of Random Early Dropping (RED) that tries to prevent queues from ever getting full. The key difference between RED as described in [26] and our implementation is that instead of drop-tail, we have per-flow fair-queuing at every node. Hence, when an packet arrival triggers an early drop, we drop a packet from the longest sub-queue (as opposed to dropping the incoming packet). With this simple modification, the performance is very close to that of the TDMA MAC; 94% of the flows receive at least 15 kbps (see the fourth line in Figure 5.20).

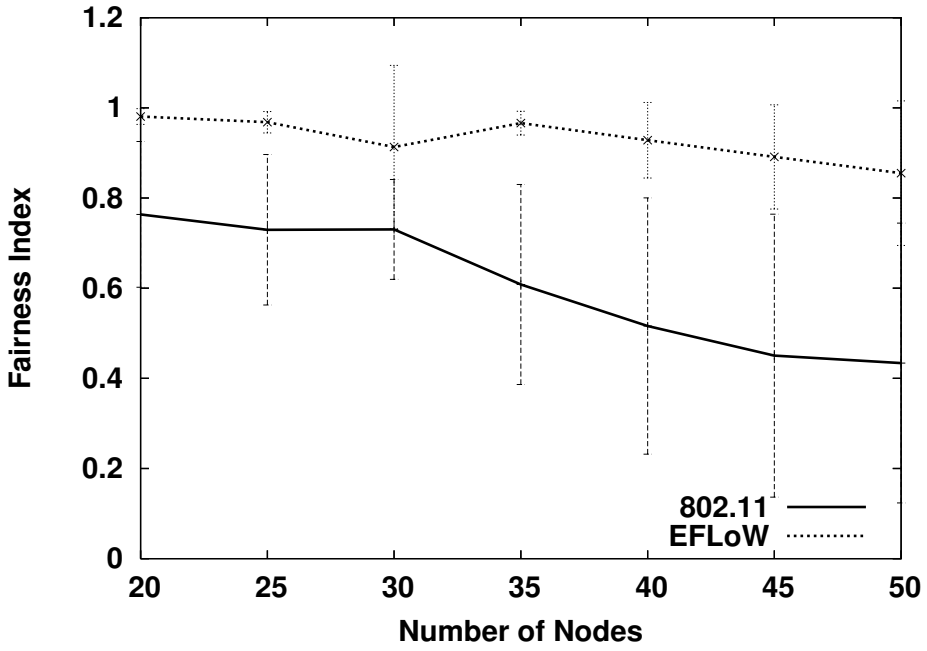


Figure 5.21. Effect of the network size on the performance of 802.11 and *EFLoW* (with OML)

5.3.1.8 Scaling Network Size

In this experiment, we simulate 5 simultaneous flows while scaling the network size from 20 to 50 nodes. In each case, the nodes are distributed at random in a square area; the dimensions of the square are varied to keep the average node density constant at 25 per radio range. The average fairness index for each case is shown in Figure 5.21. We can see that the performance of 802.11 progressively worsens as we increase the network size, whereas *EFLoW* shows only a slight degradation.

5.3.2 Test-bed

We now present the results from experiments on the MIT test-bed described in Section 4.3.2. Once again, we use static routing based on measurement of link quality in advance. We look at the performance of long-lived TCP flows in Sections 5.3.2.1 and 5.3.2.2 followed by the performance of short flows in Section 5.3.2.3.

	802.11	1-hop	2-hop	Passive detection	Actual interference
Average system throughput (Mbps)	1.618	1.485	1.382	1.387	1.418
Average no. of useful transmissions (/s)	273.5	268.2	257.0	261.2	268.0

Table 5.1. Efficiency of various schemes for 2 simultaneous flows in the test-bed.

5.3.2.1 Two simultaneous flows

In this experiment, we study the performance of two simultaneous TCP flows in our test-bed. We pick two random sources and sinks and measure the throughput of both flows over 2 minutes. We determine the interference pattern by either using heuristics, our passive detection algorithm or the actual measured interference pattern obtained in Section 4.4.3.1.

We repeat the experiment 50 times, and in each case measure F both with default 802.11 as well as with *EFLoW* and plot the CDF of F in Figure 5.22. Recall that since there are two flows, F takes values from 0.5 (least fair) to 1 (most fair). We see that without *EFLoW*, one of the two flows is starved ($F \approx 0.5$) in 38% of the cases; this never happens when using *EFLoW* with the correct interference pattern. We see good fairness with the 2-hop heuristic because it has very few false negatives in our test-bed (as mentioned earlier, false positives do not affect the fairness of *EFLoW*). The passive detection algorithm performs almost identically.

Table 5.1 shows the efficiency of our system in each case. The system throughput is defined as the sum of the throughput of both flows in each trial. However, as mentioned earlier, this metric is biased towards schemes that give higher priority to flows routed over fewer network hops (such as 802.11). The difference between the various schemes is much lower when we look at the number of useful transmissions. We

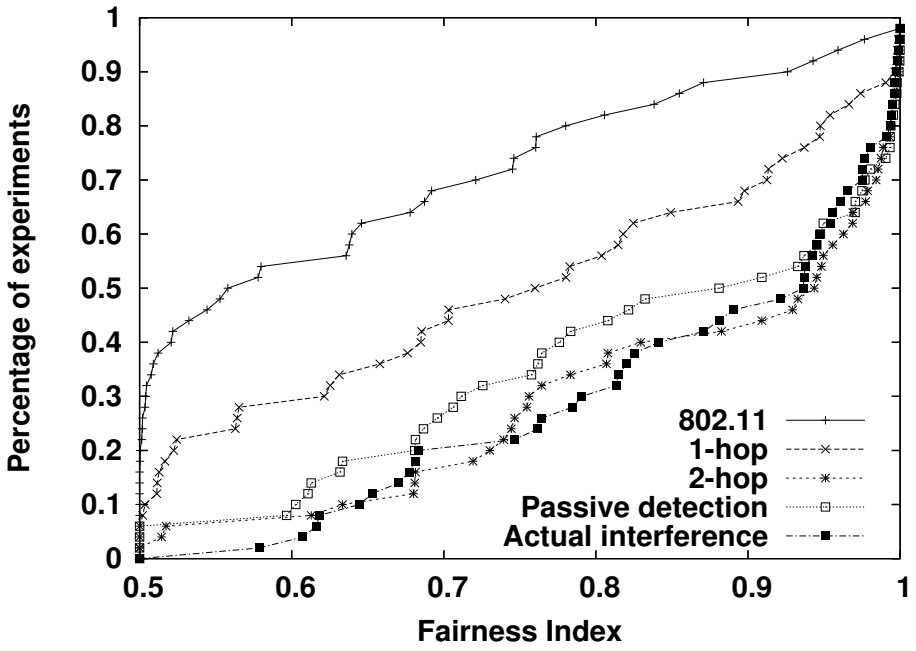


Figure 5.22. The CDF of the fairness index of two simultaneous TCP flows in the test-bed

expect the efficiency of the 2-hop heuristic to be lower due to the large number of false positives, but in practice, the difference is very small especially when compared to the passive detection algorithm. This is because of two reasons. First, the efficiency of the 2-hop heuristic is only affected when it overestimates interference for the bottleneck link of a flow, which is usually the link experiencing the most interference. Because of the high level of interference in our test-bed (recall that 72% of the link pairs interfere), the bottleneck is likely to interfere with almost all the other links and hence very unlikely to be affected by false positives. Second, the advantage of our passive detection algorithm is further diminished because of the delay in detecting interference. Therefore, we don't see as much improvement as when using the actual interference pattern.

Based on these results, one could argue that the 2-hop heuristic is sufficient for our test-bed. However, this depends on a number of factors that may not always be true. Other applications may be more sensitive to false positives than *EFLoW*. There

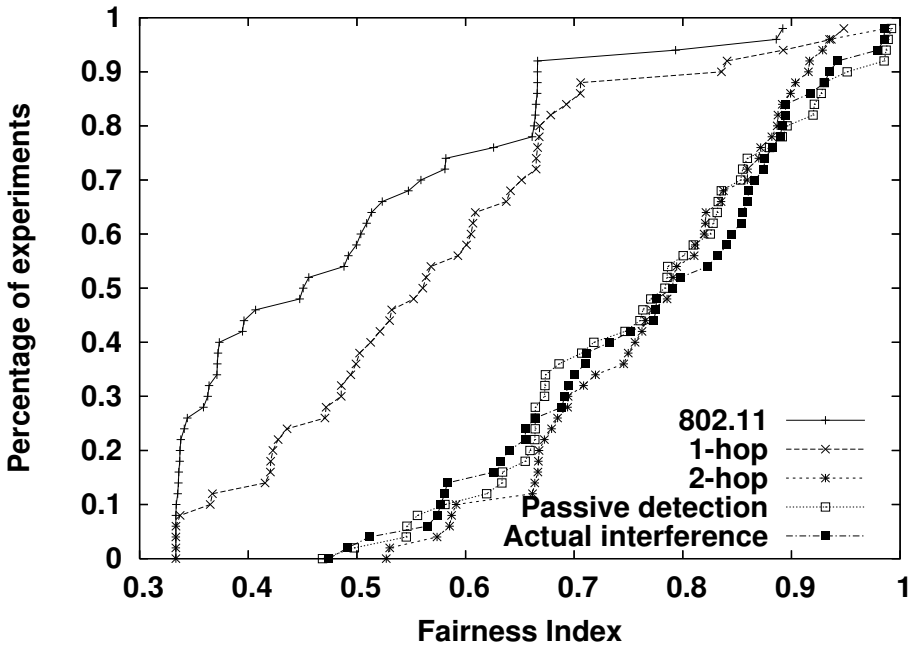


Figure 5.23. The CDF of the fairness index of three simultaneous TCP flows in the test-bed

may be more links pairs that do not interfere if the nodes are spread over a larger area. As shown in our simulations, the type of antennas is a crucial factor as well. The key advantage of passive measurements is that the algorithm can automatically correct for all these factors without any modifications.

5.3.2.2 Three simultaneous flows

In this Section, we conduct 50 experiments as before, with the change that there are three simultaneous flows in each experiment. The CDF of the resulting throughputs is shown in Figure 5.23. Note that $F = 0.333$ if two of the 3 flows are starved, which happens in roughly 25% of the cases under 802.11. If one of the flows is starved, F lies between 0.33 and 0.66 depending on the distribution between the other two flows. Finally $F > 0.66$ implies that all three flows were able to establish a connection; this happens in 80% of the cases when we use the actual interference pattern.

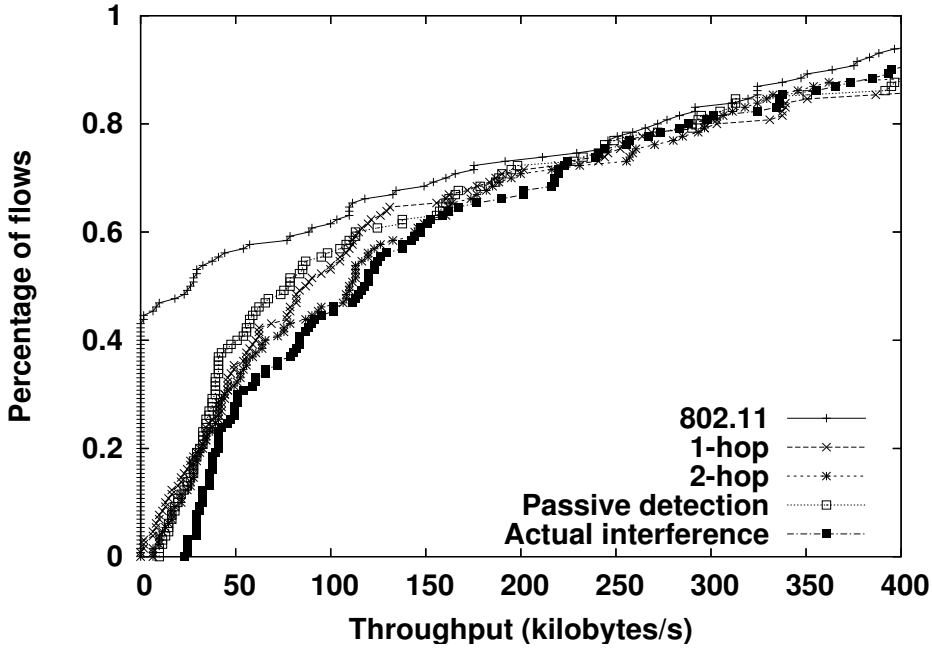


Figure 5.24. The CDF of the throughput of short flows in our test-bed

As before, both the 2-hop heuristic and the passive detection algorithm perform well also.

5.3.2.3 Short flows

There are three factors that can affect the performance of short flows in *EFLoW*. First, as mentioned in Section 3.4.5, the coarse-grained nature of time-slots in OML can delay the slow-start phase of TCP. Second, we do not have the correct allocation at each hop until the AIMD algorithm of *EFLoW* converges. Third, when using our passive interference detection algorithm, there is an additional delay in determining which links of the new flow interfere with other links. We study the impact of these factors by conducting an experiment with the following traffic pattern. There are two long flows active at any given time. The duration of these long flows is set at 2 minutes by replacing one of them every minute with a new flow. In addition we instantiate three short flows periodically at 15 second intervals. The duration of each

of these flows is chosen at random between 2 to 10 seconds. Thus there are 2 long flows and up to 3 short flows active at any given time.[6] We run this experiment for 15 minutes and plot the CDF of the throughput of the short flows in Figure 5.24. A throughput of zero indicates that a TCP connection was not established during the duration of the flow. We can see that this happens for almost 44% of the flows when we use the standard 802.11 MAC. However, with *EFLoW*, all the short flows were able to establish a connection when using the actual interference pattern. The performance is not as good while using the 2-hop heuristic or our passive detection algorithm, but is still much better than the 802.11 MAC protocol.

[6]The short flows model web browsing and the long flows model file sharing applications and downloads.

Chapter 6

Conclusions

Recently, there has been a lot of excitement about multi-hop wireless networks built using commodity hardware. Deployments have become wide spread thanks to the inexpensive and ready availability of 802.11 based hardware. However, the observed performance have been disappointing especially while trying to support multiple users at the same time. In this work, we have proposed a set of solutions that can alleviate this problem by giving all nodes fair and efficient access to the wireless medium. Though our primary target is mesh networks, we believe our approach is also applicable to other scenarios such as sensor and vehicular networks.

The first component of our solution is the Overlay MAC Layer. OML implements a time-slot based scheduler on top of any other MAC protocol. This allows us to experiment with existing 802.11 based hardware. The ability to modify the MAC without changes to standards is not only important to save costs but also to save time; standards take a very long time to emerge since they have to satisfy the needs and demands of a lot of different entities and applications. Also, since OML is based on time slots, it avoids various problems that have plagued contention-based MACs in multi-hop networks. Ultimately, the overlay approach will enable us to experiment

with new scheduling and bandwidth management algorithms, and evaluate their benefits to existing applications, before implementing them in hardware at the MAC layer; WSA and *EFLoW* are only a first step in this direction.

In order to reap the benefits of OML we need an accurate picture of which links interfere with each other. We found previously used heuristics to be inadequate and active measurements are expensive in terms of both resources and time. Motivated by the fact that OML (and many other applications) only need interference information for links that are in use, we develop an algorithm based on passive measurements as an alternative. Our approach measures interference for exactly the links we need without introducing any additional traffic in the network. We have demonstrated the power of this approach by comparing it with other heuristics and show that it can quickly and accurately determine the interference pattern for a wide variety of traffic patterns and network scenarios.

Even with a method to control allocation within a local neighborhood (OML), achieving end-to-end fairness in multi-hop networks is not an easy problem. Most previous work assumes that state about the entire network can be consolidated at a single node, which can lead to a very high control overhead. The distributed approaches either assume a restricted traffic patterns or are tied to a particular contention based MAC. Our solution, *EFLoW*, addresses the above mentioned limitations by using an additive-increase multiplicative-decrease (AIMD) based algorithm. By using *EFLoW* and WSA, we are able to both compute a fair allocation and a time-slot based schedule to achieve that allocation, without the need for a centralized coordinator. In addition, our approach does not restrict the traffic pattern in any way and automatically adapts to changes in flow demands and network topology.

6.1 Open Issues and Future Work

We now mention some open issues and discuss possible avenues for future research. Before we delve into specific components like OML or *EFLoW*, we first look at the system as a whole.

System-wide issues: So far, all of our experiments (including simulations) have been done with fixed nodes. While we have included some basic mechanisms to handle mobility in all our algorithms, we have not evaluated them yet. Mobility will definitely degrade the performance of our system, but to what extent is yet to be determined. Further modifications might be necessary, especially when the mobility is very high as in some sensor and vehicular networks.

WSA works well only when all nodes sharing the channel implement it. If there are other nodes using the standard 802.11 protocol, the aggregate bandwidth of all nodes using WSA will be limited to that of *one* node using 802.11. To address this limitation, both *EFLoW* and WSA have to be modified to make nodes more aggressive in trying to transmit data. The additional traffic will affect the accuracy of our interference detection algorithm also; more sophisticated mechanisms based on machine learning or other statistical techniques might help in factoring out the effect of external senders.

Overlay MAC Layer: The current slot length (time to transmit 10 packets) used by OML might be too long for certain real-time applications. TCP performance can also be improved by using more fine grained scheduling. We believe that more can be done to optimize this aspect of OML. First, we must analyze the implementation, identify bottlenecks, and mitigate them when possible; interrupt priority and using call-backs versus polling are two areas that can be improved. Secondly, modifications to the firmware of the Network Interface Card (NIC) could provide additional benefits. Finally, packet fragmentation might be necessary to achieve better enforcement at the

boundaries of time-slots. Currently, we simply transmit the whole packet even if we are close to the end of a slot and do not have enough time to transmit without spilling over to the next slot.

Interference Detection: While our approach based on passive measurements is indeed very promising, we have assumed that the link qualities and the interference pattern are stable enough to be tracked; more measurements are needed to confirm the types of networks where this true. Also, our algorithm currently works on top of time-slot based MACs only; it may be possible to generalize this to contention based MACs by monitoring the queue lengths, back-off windows and the number of collisions. Another avenue that is worth exploring is whether passive measurements can be combined with the modeling based approaches used in [55] and [49]. This might help improve the detection time and to move beyond a pair-wise interference model. Lastly, we have only used our interference measurements for one application - scheduling with OML. We plan to study other applications such as channel assignment and admission control in the future. Also, the routing protocol may also be able to exploit availability of interference information by picking paths that don't interfere with each other; however, this requires information about links that are not (yet) carrying any traffic. It would be interesting to see how we can satisfy this requirement by augmenting passive measurements with periodic probing of new paths and other active measurements.

End-to-end fairness: Though we have empirically shown that *EFLoW* can provide max-min fairness for a number of scenarios, a formal proof of convergence remains an open problem. Also, we have not investigated whether *EFLoW* can be adapted to other definitions of fairness; examples include proportional fairness, allocations which try to maximize a utility function and so on.

Though both TCP and *EFLoW* use AIMD based algorithms, the mechanisms that

change the local weights and the window size proceed independently of each other. This may be suboptimal because *EFLoW* might decrease the allocation while TCP is increasing the window size and vice versa. We plan to see whether performance can be improved by synchronizing the two processes; for example by dropping a packet every time *EFLoW* does a multiplicative decrease.

EFLoW does not provide any delay guarantees; we have only looked at the throughput of flows so far. We plan to look at how we can augment *EFLoW* to support real-time traffic; one possible method is to reserve certain time-slots for specific flows in advance. Lastly, more experiments are needed to understand the performance of *EFLoW* with more realistic workloads. Replaying actual traffic logs or using application level benchmark suites will be very useful in this regard.

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