University Of Crete

Department Of Computer Science

# Congestion Control in IEEE 802.11 Wireless Networks using Explicit Congestion Notification and Load-Based Marking

**Despina Triantafyllidou**

**Master's Thesis**

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UNIVERSITY OF CRETE
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# Congestion Control in IEEE 802.11 Wireless Networks using Explicit Congestion Notification and Load-Based Marking

Thesis submitted by

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in partial fulfillment of the requirements for the
Master of Science degree in Computer Science

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## Abstract

Wireless local area networks (WLANs) will continue to be a major growth factor for communication networks in the up-coming years. The IEEE 802.11 standard sees increasing public deployment and, hence, it is important to ensure that different users gain fair access to the network resources. Nevertheless, the nature of the wireless medium cannot guarantee reliable data transfer, neither long-term fair share of the resources. The resource sharing model of the underlying 802.11 MAC protocol forces stations to continuously contend to capture the channel, in order to transmit. In infrastructure networks, the unfairness problem occurs between the uplink and the downlink traffic. In multihop wireless networks, the unfairness problem is more intense due to location dependent contention, which can result in different stations obtaining a different estimate of the level of congestion. Under these circumstances, some flows can increase their throughput, while others might starve.

In the present thesis we argue that we can improve fairness in a wireless network by providing more accurate congestion information to the end systems, which should consider the traffic load within the corresponding contention area. This information, along with an appropriate packet marking algorithm, can enhance resource sharing and improve TCP fairness maintaining the TCP end-to-end behavior and semantics.

Our approach is based on two key ideas; first, it uses explicit congestion notification (ECN), as the common end-to-end signaling mechanism, for conveying congestion information across the wireless links. Second, marking is performed using a load-based

marking (LBM) algorithm, where the marking probability is a function of the aggregate utilization, measured appropriately for each network topology. For infrastructure networks, where an access point (AP) acts as the gateway for the traffic flowing in both the uplink and the downlink direction, we consider the aggregate traffic in both directions. For multihop wireless networks, the level of congestion around a station is given by the sum of the receiving rates within the station's collision domain.

We evaluated the proposed approach through NS-2 simulations. LBM demonstrated slightly higher fairness compared to drop-tail queuing (DT), in the case of infrastructure networks, and noticeably higher fairness compared to DT, in multihop networks. On the other hand, in both network topologies, the achieved utilization remained the same, as with DT. Finally, the proposed approach significantly reduced the end-to-end delay and delay jitter.

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# Έλεγχος Συμφόρησης σε Ασύρματα Δίκτυα IEEE 802.11 με χρήση του Explicit Congestion Notification και ενός Μηχανισμού Μαρκαρίσματος Βασισμένου στην Εκτίμηση Φόρτου

**Δέσποινα Τριανταφυλλίδου**

Μεταπτυχιακή Εργασία

Πανεπιστήμιο Κρήτης, Τμήμα Επιστήμης Υπολογιστών

## Εκτενής περίληψη

Λόγω της εξάπλωσης που υφίστανται τα ασύρματα τοπικά δίκτυα (WLANs) τα τελευταία έτη, η βελτίωση της απόδοσής τους ήδη αποτελεί μείζον αντικείμενο έρευνας. Τα δύο κύρια πρότυπα αυτήν τη στιγμή είναι το ETSI high performance european radio (HIPERLAN) και το IEEE 802.11 WLAN, με πιο διαδεδομένο το δεύτερο, στην έκδοση των 11 Mbps. Λόγω της ευρείας επέκτασης των ασύρματων τοπικών δικτύων τεχνολογίας IEEE 802.11 σε δημόσιους χώρους και της ενοποίησής τους με ενσύρματα δίκτυα μεγάλων εταιρειών και με το Internet, ο αριθμός των χρηστών τους αυξάνεται συνεχώς. Είναι, επομένως, σημαντικό να εξασφαλιστεί ότι η πρόσβαση των χρηστών στο δίκτυο επιτυγχάνεται με υψηλούς ρυθμούς αποστολής και λήψης, με μικρές καθυστερήσεις και ότι οι πόροι του δικτύου κατανέμονται δίκαια μεταξύ τους.

Εντούτοις, η φύση του ασύρματου μέσου μετάδοσης δεν μπορεί να εγγυηθεί ούτε την αξιόπιστη μεταφορά της πληροφορίας αλλά ούτε και το δίκαιο διαμοιρασμό των πόρων μεταξύ των χρηστών. Σύμφωνα με το μοντέλο διαμοίρασης των πόρων του υπο-επιπέδου MAC του 802.11, οι σταθμοί που πρόκειται να μεταδόσουν υπόκεινται σε ανταγωνισμό για την πρόσβαση στο κανάλι. Στα δίκτυα υποδομής, το πρόβλημα της δικαιοσύνης υφίσταται μεταξύ των ροών που ταξιδεύουν και προς τις δύο κατευθύνσεις, από και προς το σταθμό πρόσβασης (access point (AP), για το Basic Service Set (BSS) στο 802.11). Στα ασύρματα δίκτυα πολλαπλών βημάτων, το πρόβλημα είναι πιο σύνθετο. Επειδή ο ανταγωνισμός καθορίζεται από τις σχετικές θέσεις των κόμβων, διαφορετικοί χρήστες

αντιλαμβάνονται διαφορετικό επίπεδο συμφόρησης. Υπό αυτές τις συνθήκες, μερικοί αποστολείς πετυχαίνουν να αυξήσουν το ρυθμό τους, ενώ άλλοι καταλαμβάνουν πολύ μικρό ή και μηδενικό ποσοστό της χωρητικότητας, σε σχέση με τους άλλους.

Στην παρούσα εργασία επιτυγχάνουμε βελτίωση της δικαιοσύνης σε ασύρματα δίκτυα τεχνολογίας 802.11 μεταβιβάζοντας στους TCP αποστολείς ορθή πληροφορία για το επίπεδο της συμφόρησης περιοχών, στις οποίες οι ασύρματοι σταθμοί υπόκεινται σε ανταγωνισμό για την πρόσβαση στο κανάλι. Οι αποστολείς πρέπει να λάβουν υπόψη την εκτίμηση φόρτου στην περιοχή αυτή. Ένας κατάλληλος αλγόριθμος μαρκαρίσματος βασιζόμενος στην παραπάνω πληροφορία μπορεί να βελτιώσει τη δικαιοσύνη διατηρώντας, παράλληλα, την από άκρο-σε-άκρο συμπεριφορά του TCP αμετάβλητη.

Η προσέγγισή μας βασίζεται σε δύο κύριες ιδέες. Σύμφωνα με την πρώτη ιδέα, ο μηχανισμός explicit congestion notification (ECN) χρησιμοποιείται για την από άκρο-σε-άκρο σηματοδότηση της συμφόρησης στο ασύρματο δίκτυο, μεταβιβάζοντας τη σχετική πληροφορία διαμέσου των πακέτων που προωθούν οι ασύρματοι σταθμοί στους TCP αποστολείς. Σύμφωνα με τη δεύτερη ιδέα, τα πακέτα μαρκάρονται σύμφωνα με έναν αλγόριθμο εκτίμησης φόρτου (load-based marking - LBM) στο ασύρματο κανάλι. Η πιθανότητα μαρκαρίσματος σε ένα κόμβο είναι συνάρτηση του ποσοστού χρησιμοποίησης του καναλιού στην περιοχή όπου η μετάδοσή του συγκρούεται με τις μεταδόσεις άλλων σταθμών.

Η επιλογή να αποτελεί ο φόρτος του δικτύου τη μεταβλητή για τον υπολογισμό της πιθανότητας μαρκαρίσματος προκύπτει από την παρατήρηση ότι στα ασύρματα δίκτυα η συμφόρηση δεν είναι χαρακτηριστικό κάποιου συνδέσμου, στον οποίο ο τοπικός ενταμιευτής έχει υπερχειλίσει, αλλά του ασύρματου καναλιού. Σ' αυτό, οι διάφορες ροές μοιράζονται τη χωρητικότητα και προς τις δύο κατευθύνσεις, uplink και downlink. Επιπλέον, οι ασύρματοι σταθμοί διαθέτουν ενταμιευτή μόνο για τα πακέτα που πρόκειται να στείλουν (downlink) αλλά όχι για αυτά που λαμβάνουν (uplink). Η απουσία ουράς που να συγκεντρώνει πακέτα από τις διάφορες ροές καθιστά την εφαρμογή παραδοσιακών active queue management (AQM) αλγορίθμων, όπως ο random early detection (RED), αδύνατη. Αλλωστε, το επίπεδο συμφόρησης ενός ασύρματου δικτύου δεν μπορεί να αντικατοπτρίσει η πληροφορία που συγκεντρώνει τοπικά ένας και μόνο σταθμός. Αυτό, όμως, που μπορεί να υποδηλώσει συμφόρηση, είναι το ποσοστό χρησιμοποίησης του ασύρματου μέσου, στην περιοχή όπου η μετάδοση από άλλους σταθμούς θα προκαλέσει συγκρούσεις.

Η συνεισφορά της παρούσας εργασίας έγκειται στον τρόπο υπολογισμού του ποσοστού χρησιμοποίησης του ασύρματου καναλιού, που θα πρέπει να είναι διαφορετικός σε καθεμια απο τις δύο τοπολογίες που ορίζει το πρότυπο 802.11, στα δίκτυα υποδομής και στα δίκτυα αυθαίρετης δομής πολλαπλών βημάτων. Βασικό χαρακτηριστικό των δικτύων υποδομής είναι η ύπαρξη ενός AP, από το οποίο διέρχεται όλη η κίνηση που ταξιδεύει από και πρός όλους τους κόμβους του ασύρματου. Συνεπώς, ο υπολογισμός του ποσοστού χρησιμοποίησης του καναλιού γίνεται στο AP λαμβάνοντας υπόψη τη συνολική κίνηση και προς τις δύο κατευθύνσεις.

Στα δίκτυα πολλαπλών βημάτων δεν υπάρχει σημείο από το οποίο να διέρχεται όλη η κίνηση. Μάλιστα, στα δίκτυα αυτά, η περιοχή της συμφόρησης δεν είναι η ίδια για όλους τους κόμβους, όπως στην περίπτωση του BSS. Αντιθέτως, κάθε κόμβος, ανάλογα με τη θέση που βρίσκεται, αντιλαμβάνεται διαφορετικό επίπεδο φόρτου στο δίκτυο, λόγω του ότι ακούει μόνο τη μετάδοση των κόμβων που βρίσκονται στην εμβέλεια ανίχνευσης φέροντος (carrier sensing range). Οι μετάδόσεις αυτών μπορούν να παρεμβληθούν στις μεταδόσεις του κόμβου. Άρα, η πιθανότητα μαρκαρίσματος θα είναι διαφορετική, μεταξύ διαφορετικών κόμβων. Το επίπεδο συμφόρησης γύρω από ένα κόμβο δίνεται από το άθροισμα των ρυθμών λήψης στην περιοχή συγκρούσεών του.

Ο αλγόριθμος εφαρμόστηκε στις δύο βασικές τοπολογίες του προτύπου 802.11. Χρησιμοποιήθηκε ο προσομοιωτής δικτύων NS-2 για τη διεξαγωγή πειραμάτων με διάφορες τοπολογίες και για διάφορες παραμέτρους κίνησης. Οι μετρικές απόδοσης που εξετάστηκαν ήταν η διαπερατότητα (throughput), η δικαιοσύνη (fairness index), η από άκρο-σε-άκρο μέση καθυστέρηση των πακέτων στο ασύρματο και η μεταβλητότητά της (end-to-end average packet delay, delay jitter). Τα αποτελέσματα, όσον αφορά στη διαπερατότητα του δικτύου (συνολικός ρυθμός μετάδοσης όλων των χρηστών), εμφανίζονται ακριβώς τα ίδια σε σύγκριση με το μηχανισμό drop-tail queuing (DT), που συνήθως εφαρμόζεται. Ο αλγόριθμος LBM επιτυγχάνει να βελτιώσει σημαντικά τη δικαιοσύνη, στην περίπτωση των δικτύων πολλαπλών βημάτων, σε σχέση με το DT, ενώ στα δίκτυα υποδομής η βελτίωση είναι μικρότερη. Η μεγαλύτερη συνεισφορά του αλγορίθμου είναι στη μείωση της μέσης καθυστέρησης. Και στις δύο τοπολογίες η βελτίωση είναι μεγάλη, καθώς επίσης και η μεταβλητότητα της καθυστέρησης (τυπική απόκλιση) μειώνεται σημαντικά.

**Επόπτης Καθηγητής:** Βασίλειος Α. Σύρης
Επίκουρος Καθηγητής
Πανεπιστήμιο Κρήτης

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The present study is the topping of a most creative period, during which I have exploited the knowledge I gained during my studies, and raised research activity in an area of science that fascinates me. During this, my supervisor, Assistant Professor Siris Vasilios, has always led me towards the right directions, with patience and guidance, providing me insight for my every step. For this, and for the opportunities he gave me to get to the gist of the research with our publications, I express my deep gratitude to him.

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The period of my studies in this University, has been a life lesson for me, from which I have absorbed the maximum I could. Among the people who have stood by me I would like to refer first to my friends, and especially those I have been with during the last seven years; Vaso, Ioanna, George K., George S., and Panayiotis. I can't omit to refer to Mr. and Mrs. Sfakianakis, who treated me like a family, while mine was away, as well as to the Kroustalakis' family, who cared for me, as if I was a child of their own. Last, but certainly not least, I owe the maximum to Nikos, for always being my better half, believing in my efforts and potentials and spurring me with his continuous prospects "to be written in history".

Nevertheless, none of my expectations would I have fulfilled, if it hadn't been my family; my father, Steve, my mother, Domna, and my beloved brother, Akis, who stood as the cornerstone to all my expectations. For this, I dedicate this work to them.

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# Introduction

Since the mid-1980s, when the growth of wireless local area networks (WLANs) commenced, a revolution has taken place in the area of wireless communications. In late 1980s, the IEEE standardization "umbrella" 802 made the first attempt to define a standard and as a result, the 802.11 Working Group was established. This group was responsible for defining the physical and MAC sublayer standards for WLANs. In 1997, the IEEE released the 802.11, as the first international standard for WLANs, defining speeds of 1 Mbps and 2 Mbps. In September 1999, they ratified the 802.11b high rate amendment to the standard, which added two higher speeds. The IEEE 802.11 WLANs are, nowadays, widely developed.

Besides, as the Internet expands to integrate wireless and wired networks, the number of users accessing it through wireless links, and IEEE 802.11 WLANs in particular, is growing rapidly, mainly due to the proliferation of wireless hotspots and enterprise WLANs. Along with this, the amount of traffic volume of wireless users is also expected to increase. Moreover, emerging multimedia services over wireless networks will have different bandwidth and delay requirements. The effective support of such QoS requirements in wireless networks is an area of intense research.

## Problem description

Under the status described above, the efficient control and management of wireless resources is a challenge. Enhancing the TCP performance proves to be critical for the overall wireless networks' performance, for the following reasons. First, the majority of the wireless 802.11 traffic is TCP traffic. Second, the limited capacity of the wireless spectrum poses constraints to the network performance and to the service level provided to the end users. Compared to wired networks, there is a limited ability to increase the capacity of wireless networks, since this is determined by the available wireless spectrum. The typical solution in the wired domain is to overprovision the network, however, this has serious problem in the wireless counterpart. The third reason is the medium access contention among the wireless nodes. The underlying CSMA/CA protocol forces stations to compete to capture the channel, every time they desire to transmit. In the case where some connections manage to gain access to the channel leaving others starved, this may lead to significant unfairness.

Apart from the limited offered data rates and the channel contention, wireless networks have other characteristics that can lead to reduced performance. Probably the major performance factor for wireless is the significant bit error rate (BER), where frame loss of up to 1 percent is not uncommon, and errors occur in bursts, rather than being evenly spaced in the packet stream. Errors results in packet loss, that triggers TCP to decrease its congestion window, despite the fact that the loss is not due to congestion.

Additionally, wireless media exhibit longer latency delays than wired ones. This affects TCP throughput and increases the interactive delays perceived by the user. It has been shown in [14] that connections with longer round trip times (RTTs) are prone to TCP unfairness. Especially in the case of wireless networks, where nontrivial unfairness is caused by the underlying MAC protocol, this is a critical impediment for the TCP performance. In infrastructure networks the unfairness problem occurs between the uplink and the downlink traffic. In multihop wireless networks, the unfairness problem is more intense due to location dependent contention, which can result in different stations obtaining a different estimate of the level of congestion. All the above motivate the need for efficient and fair congestion control over wireless networks, containing either infrastructure or ad-hoc topologies.

## Contribution of the present work

In the present thesis we argue that we can improve fairness in a wireless network by accounting for the traffic load in the wireless area, and by conveying more accurate congestion information to the end systems. The nodes in the network signal the level of congestion, estimated as a function of the aggregate utilization, using explicit congestion notification (ECN) marking. The utilization estimation that accounts for the congestion triggers an effective marking mechanism.

Our approach is based on two key ideas; first, it uses ECN as the common end-to-end signaling mechanism, for conveying congestion information across the wireless links. Second, marking is performed using a load-based marking (LBM) algorithm, where the marking probability is a function of the aggregate utilization measured appropriately for each network topology. For infrastructure networks, where an access point (AP) acts as the gateway for the traffic flowing in both the uplink and the downlink direction, we consider the aggregate traffic in both directions. For the case of multihop wireless networks, the level of congestion around a station is given by the sum of the receiving rates within the station's collision domain. The proposed marking approach takes into account the particular characteristics and resource usage model of the wireless link, managing to maintain the TCP end-to-end behavior and semantics.

We evaluated the proposed approach against the drop-tail queuing (DT) mechanism through NS-2 simulations. LBM demonstrated slightly higher fairness in the case of infrastructure networks, and noticeably higher fairness in multihop networks. On the other hand, in both network topologies, the achieved utilization remained the same, as with DT. Finally, the proposed approach significantly reduced the end-to-end delay and delay jitter.

## Relation to previous work

It would be interesting to position our approach with respect to other paradigms and compare the findings. The cross-layer networking has emerged as an effective way for designing efficient network protocols over wireless link technologies [23]. Cross layer design departs from the strict layer separation, which has been the traditional approach for network protocol design. According to this, the physical and MAC layer knowledge of the wireless medium can be communicated to higher layers, in order to provide efficient methods of allocating network resources or improving spectral efficiency. Our approach follows the cross-layer design in the sense that the proposed ECN marking procedure takes into account the particular characteristics and the resource sharing model of IEEE 802.11 WLANs, and conveys congestion information to the upper layer.

Interestingly, our approach maintains TCP operation and end-to-end semantics, hence, adheres to the end-to-end argument stated in [22]. The reasoning of the argument is that any function within a communication system can completely be implemented at the end points, provided that this is efficient for the system performance, so the functionality inside the network remains simple. The argument provides a rationale for moving functions upward in a layered system, closer to the application that uses the function. According to our approach, the congestion control is performed at the end-systems, where the aggregate information for the level of congestion for the whole end-to-end path can invoke the appropriate congestion control algorithms.

Although the application of ECN to wireless networks is not new, e.g. see [15, 18], its application as a common signaling mechanism for conveying congestion information in a way that takes into account the particular characteristics of the underlying wireless technology was first proposed in [25] for the case of 3G networks based on wideband CDMA (WCDMA). As we argue here, IEEE 802.11 WLANs differ from 3G WCDMA based cellular networks, hence the marking procedure for each should be different.

On the other hand, the application of an LBM algorithm for wired networks, and its interaction with various end-system congestion control algorithms was initially studied in [26]. This work investigated the service differentiation achieved using weighted window-based congestion control, and the impact of the marking algorithm on the performance. More precisely, the authors investigated the ability of the end-system algorithms, working in conjunction with three different marking algorithms in routers, one of which was LBM, to offer different throughput to connections with different weights or willingness-to-pay values. The LBM algorithm, implemented upon a virtual queue, deterministically marked packets from the time the queue overflowed until it became empty again. Among the results of this work was the ability of LBM to reduce the average queuing delay and delay jitter. Our approach for marking considers a load metric other than a queue overflow, as will be explained in the next chapters.

Later in this thesis, we will discuss two studies, which are most closely related to ours, in the sense that they also trigger marking mechanisms based on early congestion exchange information. The work in [10] proposes a Link RED with adaptive pacing dropping packets' scheme, based on the observation that the number of the MAC layer retransmissions signal network overload, when they exceed some minimum number. The latter, called Neighborhood RED [29], argues that the improvement of spatial reuse can lead to better fairness, and proposes to perform marking based on the aggregate queue size of a node's neighborhood.

## Thesis outline

The rest of this thesis is structured as follows:

In chapter 1, we present some fundamentals of WLANs curriculum, as well as some basic active queue management (AQM), and TCP-with-ECN theory. We also motivate the marking algorithm we propose, based on the particular performance issues of WLANs.

In chapter 2, we describe our approach for seamless congestion control over the wireless link, based on the following two key ideas: first, we use ECN as the common end-to-end congestion signaling mechanism for conveying congestion information across the network. Secondly, marking is performed using a load-based estimation of the marking probability.

In chapter 3, we show how the proposed approach can be implemented in heterogeneous networks containing both wired and wireless IEEE 802.11 links. The marking mechanism is invoked by the AP, based on periodic estimations of the aggregate throughput of the traffic flowing in both directions (uplink and downlink). We also present the simulations conducted with NS-2 and the simulation results.

In chapter 4, we extend the proposed marking algorithm to multihop wireless networks. Due to the absence of an AP, we implement ECN marking at each sending node based on periodic estimations of the aggregate received rate within their collision domains. We also present and discuss simulations and results upon some basic multihop scenarios.

In chapter 5 we present the main contributions for improving the TCP performance in wireless networks. Generally, we classify them into three categories, namely,

- link-layer protocols, mainly employing retransmission techniques,
- split connections, which refer to heterogeneous networks, and use separate protocols for the wired and the wireless network, and
- end-to-end schemes and new protocols, for wireless networks.

In particular, we discuss two proposals which belong to the last category, which implement ECN marking in wireless, and are closely related to ours, in the sense that they also trigger marking mechanisms based on early congestion exchange information.

Finally, in chapter 6 we conclude by presenting the main findings of our work and identifying future research work.

## Publications

The work presented in chapters 2 and 3 has been included in the proceedings of the IFIP-TC6 Networking Conference Networking 2004, pages 1470-1475, May 2004, as *"Seamless Congestion Control over Wired and Wireless IEEE 802.11 Networks"*, by Vasilios A. Siris and Despina Triantafyllidou [27].

Also, the work presented in chapter 4 will appear in the proceedings of the WWIC 2005 Conference, May 2005, as *"ECN Marking for Congestion Control in Multihop Wireless Networks"*, by Vasilios A. Siris and Despina Triantafyllidou [28].

# Chapter 1

# Background theory

The basic architecture, features and services of 802.11b are defined by the original 802.11 standard, with changes made only to the physical layer. These changes result in higher data rates and more robust connectivity. The 802.11b is, now, the most widespread WLAN standard. It describes a WLAN that operates in the 2.4GHz frequency range, with a data transmission speed of up to 11Mbps, using spread spectrum technology.

## 1.1 The two 802.11 architecture modes

The 802.11 standard supports two network topologies: the Extended Service Set (ESS), and the Independent Basic Service Set (IBSS), commonly referred as infrastructure and ad-hoc networks, respectively.

### 1.1.1 Infrastructure WLANs

In infrastructure mode, the wireless network often consists of at least one AP connected to a wireline network infrastructure, and a set of wireless end stations. This configuration is called Basic Service Set (BSS). In such a topology, wireless stations access the wireless channel under the coordination of the AP. Since most corporate WLANs require access to a higher speed wired backbone LAN for services, they operate in infrastructure mode, and rely on the AP, that acts as the logical server for the single WLAN channel. Indeed, the communication between two nodes A and B, is targeted from node A to the AP (uplink direction) and then from the AP to node B (downlink

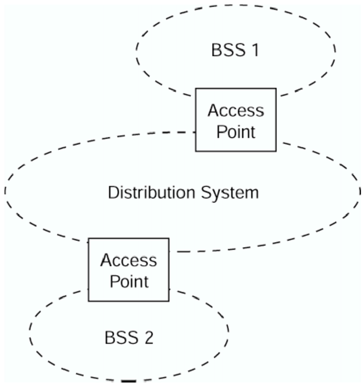



Figure 1.1: An ESS 802.11 WLAN consists of multiple cells interconnected by APs and a distribution system, such as Ethernet. A distribution system is an element that interconnects BSSs within the ESS, via APs.

direction). The AP performs a bridging function connecting multiple WLAN cells or channels, or connecting WLAN cells to an enterprise wired LAN.

For requirements exceeding the range limitations of an independent BSS, 802.11 defines the ESS as illustrated in Figure 1.1. This type of configuration satisfies the needs of large coverage networks of arbitrary size and complexity.

The standard for infrastructure WLANs not only defines the specifications, but also includes a wide range of services. The target environment of the standard includes buildings with offices, convention centers, airport gates and lounges, hospitals, as well as outdoor areas, such as parkings, campuses, and building complexes.

### 1.1.2 Ad-hoc WLANs

An IBSS is a standalone BSS that has no backbone infrastructure, and consists of at least two wireless nodes (Figure 1.2). This type of network is often referred as an ad-hoc network, because it can be constructed quickly, without much planning. An ad-hoc network is a group of wireless nodes, that communicate by relaying packets through intermediate nodes. An ad-hoc WLAN is often set up in order to serve a temporary need. Ad-hoc networks do not require the presence of preexisting infrastructure, forming instead a cooperative impromptu network, without central coordination.

Typically, nodes communicate over the same wireless channel. As a result, closely located nodes cannot transmit simultaneously. Instead, they contend for the wireless

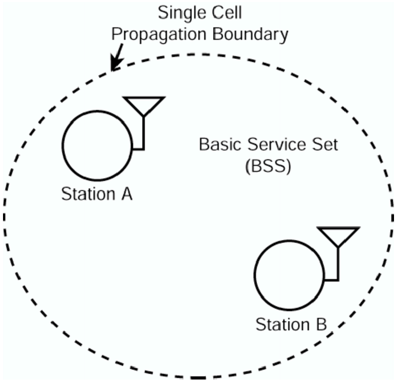



Figure 1.2: An Independent BSS (IBSS): an ad-hoc wireless network.

channel following a decentralized MAC protocol, such as CSMA/CA, which runs on every node.

## 1.2 The 802.11 MAC layer

The MAC layer consists of a set of protocols, responsible for maintaining order in the use of a shared medium. The 802.11 standard specifies a carrier sense multiple access with collision avoidance (CSMA/CA) protocol, according to which, a node willing to transmit a packet first listens to the medium, to ensure that no other node is currently transmitting. If the channel is clear, it transmits the packet. Otherwise, it chooses a random backoff time, to wait, until it is allowed to transmit its packet. During periods in which the channel is clear, the transmitting node decrements its backoff counter. When the backoff counter reaches zero, the node transmits the packet. The 802.11 collision avoidance mechanism is depicted in Figure 1.3

If the probability that two nodes will choose the same backoff factor is small, collisions between packets are minimized. Collision detection, however, as is employed in Ethernet, cannot be used for the radio frequency transmissions of IEEE 802.11. The reason for this is that when a node is transmitting, it cannot hear any other node in the system which may be also transmitting, since its own signal will drown out any others, arriving at the node. In order to minimize collisions and avoid the overhead of

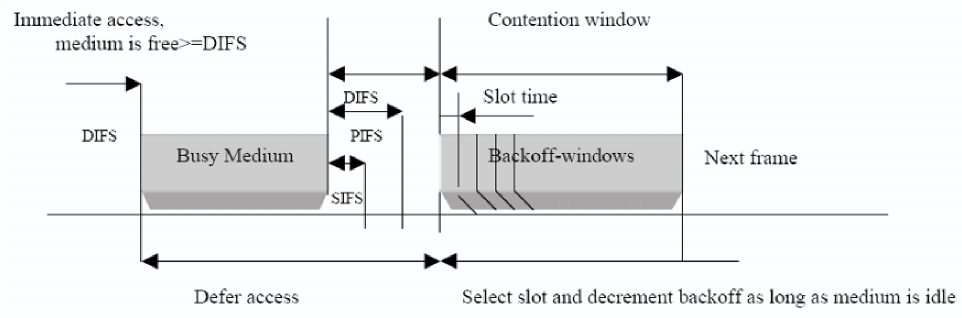



Figure 1.3: The 802.11 CSMA/CA protocol function.

retransmitting lost packets, the 802.11 MAC layer has a virtual carrier sense built-in mechanism, the RTS/CTS.

### 1.2.1 The MAC 802.11 built-in RTS/CTS mechanism

A station waiting to transmit a packet, will first transmit a short control packet, called Request-To-Send (RTS), which includes the source address, the destination address, and the duration of the following transaction (the packet and the respective ACK). If the medium is free, the destination station responds with a response control packet called Clear-To-Send (CTS), which includes the same duration information. This information is encoded within the RTS/CTS packets in the duration field. The duration field is such that the transmission can be completed within the designated time period. Stations receiving an RTS or a CTS, set their virtual carrier sense indicator, called network allocation vector (NAV), for the given duration, and use this information while sensing the medium. If a transmitter node does not receive a CTS packet, it enters into the exponential backoff mode. After this exchange, the transmitting node sends its packet. If the packet is received successfully, as determined by a cyclic redundancy check (CRC), the receiving node transmits an acknowledgement (ACK).

Due to the small size of the RTS and CTS frames, this method reduces the overhead of collisions. The mechanism, also, reduces the probability of a collision on the receiver area, caused by a station that is hidden from the transmitter, to the short duration of the RTS transmission, because the station hears the CTS and reserves the

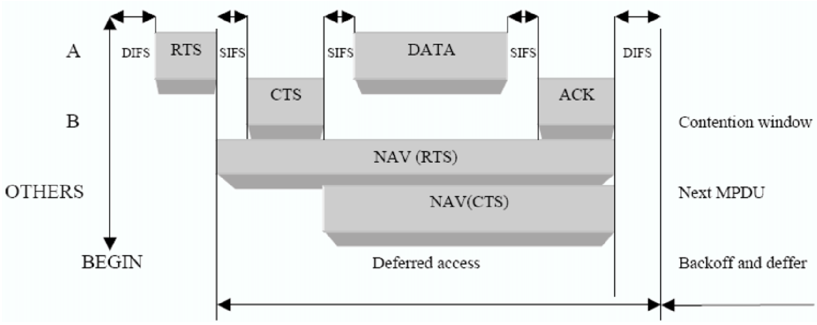



Figure 1.4: The 802.11 RTS/CTS mechanism.

medium as busy, until the end of the transmission. We will describe the hidden terminal phenomenon in the next section. The RTS/CTS mechanism is depicted in Figure 1.4.

## 1.3 TCP performance over WLANs

The increasing popularity of WLANs avers the important role that the wireless links will play in future internetworks. TCP, as a reliable transport protocol, has been tuned for networks with wired links and fixed hosts. In such networks, TCP assumes congestion in the network to be the dominant cause for packet drops, and for unexpected end-to-end delays. TCP performs well over such networks, by retransmitting lost packets and by adapting its congestion window to the perceived end-to-end delays and packet losses, after timeouts and duplicate ACKs.

Unfortunately, when losses happen due to reasons other than congestion, as is the usual case in wireless networks, then the lost packets metric does not reflect the congestion level in the network, and the TCP congestion window reduction results in unnecessary transmission rate decrease. Communication over wireless links is often characterized by significant bit-error rates, often bursting to very high rates, channel corruption, and high RTT variation. The primary reasons for the increased BER are atmospheric noise, physical obstructions found in the signal's path, multipath propagation, and interference from other systems. In all those cases, TCP is trying to recover from congestion.

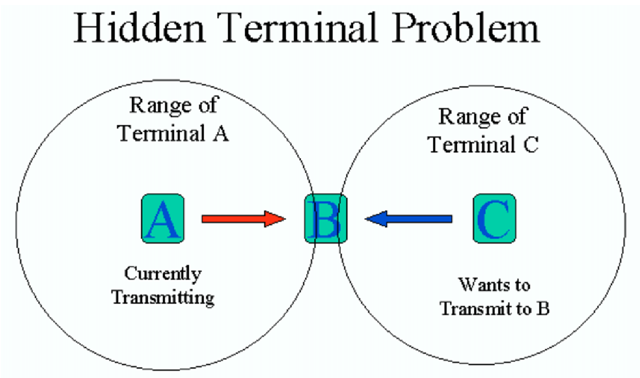



Figure 1.5: Transmission of C will collide with transmission from A to B.

### 1.3.1 Unfairness issues in wireless environments

The TCP inefficient behavior over the wireless links poses an initial motivation for alternative ways for congestion control in wireless networks. Furthermore, wireless environments have other characteristics that influence the performance of TCP, and pose limits to the service level provided to the end users. More precisely, wireless networks exhibit inherent unfairness, namely they do not manage to allocate the channel bandwidth equitably among the traffic flows. This unfairness is prominent towards connections with long round-trip times, in which case belong the wireless connections.

Yet, the native characteristics of the 802.11 MAC layer are responsible for this unfairness. According to the CSMA/CA protocol, the wireless nodes compete for the channel within a collision domain, and in most cases this contention results in some flows capturing the channel for their transmission, leaving others starved. Especially in ad-hoc networks, the principal reasons for this are the spatial reuse, and the location dependency. The former argues that the space is, also, a kind of shared resource, in the sense that TCP flows not traversing common nodes may still be competing for shared space and, thus, interfere with each other. The latter implies that given their relative positions within a network, TCP flows get different perception of the congestion level in the network, for example in terms of packet delay and loss rate. When this happens, and some flows experience more packet loss than others, they tend to reduce their congestion window more frequently. It is therefore clear, that getting correct feedback of the bottleneck is crucial, in order to achieve fairness, especially in ad hoc networks,

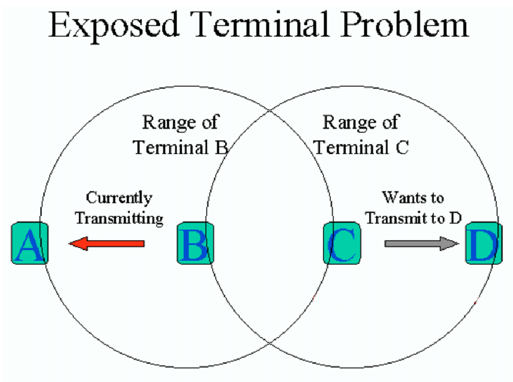



Figure 1.6: C cannot send to D, due to carrier sense.

where a centralized control of the flows -as in the case of a wireless cell with the presence of a base station or an AP- is absent.

The reasons for the unfairness mentioned above, originate from the native CSMA characteristics. Among the inefficiencies introduced by that protocol, causing significant unfairness in wireless, are the so called hidden terminal and exposed terminal scenarios, described next.

### The hidden terminal scenario

The hidden terminal scenario is depicted in Figure 1.5. Generally, the hidden terminal problem describes the situation where, even if the medium is free near the transmitter, it may not be free near the receiver. In Figure 1.5, the station C is hidden from transmissions from A to B. In the most common case where the transmission range of C is smaller than its interference range, C will not hear the transmission from A to B, and will try to transmit, therefore interfering with A's signals.

### The exposed terminal scenario

The exposed terminal scenario is depicted in Figure 1.6. Generally, the exposed terminal problem describes the situation where, even if the medium is busy near the transmitter, it may be free near the intended receiver. In Figure 1.6, the station C is exposed to transmissions from B to A. Therefore, because C does not know the position of D, and because it can hear the CTS from A (A is in its carrier sensing range), it will detect the channel as busy and won't ever send to D.

Apart from the performance issues mentioned above, there are several other critical issues in WLANs, such as the power management, security, and signal propagation. Their analysis is out of the scope of the present work.

## 1.4 AQM and random early detection (RED)

In wired networks, the bottleneck among the fixed links is a link where, either its capacity is smaller than its precedents, or flows are merged, or both. Congestion in that particular link is detected, normally, when the buffer of its output queue is full and packets are starting to be dropped (Figure 1.7). In order to avoid buffer overflows in wired networks, AQM schemes have been developed. The basic philosophy of AQM is to trigger packet dropping before buffers overflow, with drop probability being proportional to the degree of congestion. In existing AQM schemes, link congestion is estimated through queue length, input rate, events of buffer overflow and emptiness, or a combination of these factors. RED gateways use the average queue length, so the packet drop probability is piecewise linearly proportional to that [9].

The RED queue management policy monitors the average queue size for an output queue, and, using randomization, chooses flows to notify of that congestion. Transient congestion is accommodated by a temporary increase in the queue. Long-lived congestion is reflected by an increase in the calculated average queue size, and results in randomized feedback to some of the connections to decrease their windows. The probability that a flow is notified of congestion is proportional to that connection's share of the throughput through the gateway.

The RED queue management scheme is depicted in Figure 1.7. The RED gateway calculates the average queue size using a low-pass filter with an exponential weighted moving average. The average queue size is compared to two thresholds, a minimum threshold and a maximum threshold. When the average queue size is less than the minimum threshold, no packets are dropped. When the average queue size is greater than the maximum threshold, all arriving packets are dropped with a standard (maximum) probability. This ensures that the average queue size does not significantly exceed the maximum threshold.

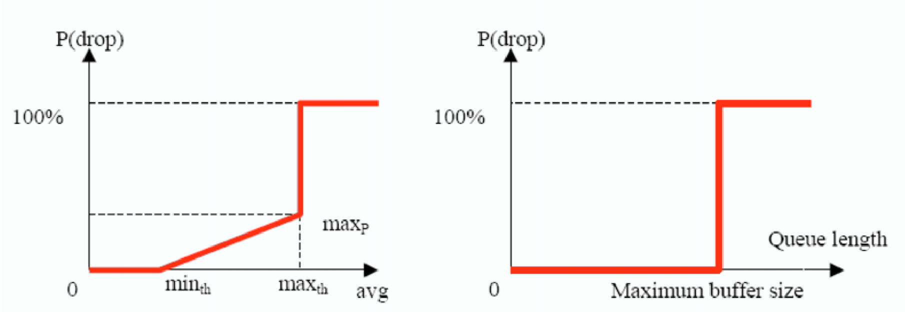



Figure 1.7: The drop function of RED and DT schemes.

When the average queue size is between the minimum and the maximum threshold, each arriving packet is dropped with probability p, where p is a function of the average queue size *avg*. Each time a packet is dropped, the probability that it is dropped from a particular connection is roughly proportional to that connection's share of the bandwidth at the gateway.

RED gateways keep the average queue size low, while allowing occasional bursts of packets in the queue. They demonstrate no bias against bursty traffic, and avoid the global synchronization of many connections decreasing their window at the same time. They can be used to control the average queue size in a network link, and are intended for networks where the transport protocol responds to congestion indications from the network. Moreover, RED is shown to improve fairness. The congestion control mechanism in RED gateways simplifies the congestion control job required of the transport protocol, and is applicable to transport-layer congestion control mechanisms other than the current versions of TCP, including protocols with rate-based rather than window-based flow control.

## 1.5 TCP with ECN

ECN has been approved as an IETF proposed standard [21]. With ECN, congestion of a network link is explicitly signaled by having routers set the congestion experienced (CE) bit located in the IP header, rather than implicitly signaled through lost packets, as is the case with TCP current operation.

End systems negotiate the ECN capability during the TCP connection setup. If both are ECN-capable, the TCP sender indicates this by setting the ECN-capable transport (ECT) bit in the IP header of each outgoing packet. ECN-capable routers are responsible for monitoring the congestion levels and mark packets of ECN-capable sources as congestion grows critical, instead of passively waiting until the buffer space runs out and resorts to drops. This can only be accomplished if the router employs a queue management algorithm, aimed at preventing bursty losses due to severe congestion incidents.

If congestion is building up, routers mark a packet by setting the CE bit in the IP packet header, before forwarding it. Upon the receipt of a packet with the CE bit set, the TCP receiver sends back a TCP ACK with the ECN-echo (ECE) bit set in its TCP header, effectively notifying its transmitter of the congestion in the network. Upon the receipt of the first ACK, the TCP sender must invoke the congestion control mechanisms for a window period, carrying the ECE bit. Currently, this translates to halving its congestion window [1]. In addition, the sender sets the congestion window reduced (CWR) bit in the TCP header of the next data segment it transmits. Because the delivery of the TCP ACKs is not guaranteed, it is important to make the ECN-echo mechanism robust against ACKs losses. For this reason, the receiver continues to set the ECE bit in the subsequent ACKs until it receives a notification from the sender via the CWR bit that the congestion window has been reduced.

Clearly, ECN relies on the ability of the routers to detect incipient congestion, a function that the DT cannot provide. Although the latest ECN specification [21] does not mandate any particular AQM mechanism, its most popular implementation is over RED, where packets are probabilistically marked when the average queue length exceeds a certain limit. Therefore, in the RED scheme of Figure 1.7, $P(drop)$ denotes the packet marking probability rather than the drop probability.

ECN can provide an early warning of incipient congestion, before packets start to be dropped. Hence, to a large extent, it can avoid the overhead of retransmitting lost packets, and unnecessary packets' delay from low-bandwidth delay-sensitive TCP connections. On the other hand, ECN does not necessarily improve throughput, while it does not seem to lead to degradation of TCP performance.

In the next chapter we will analyze our approach for seamless congestion control over WLANs using ECN for the end-to-end congestion signaling, and an LBM algorithm for computing the packet marking probability.

# Chapter 2

# Seamless congestion control over WLANs using ECN

## 2.1 Why traditional AQM algorithms cannot be applied to WLANs?

Due to the fact that ECN avoids packet drops, it is also suitable for wireless networks. ECN can be invoked in wireless environments, in order to convey congestion information to the end-points of the network and trigger the appropriate congestion control algorithms. Unfortunately, as will be explained next, although ECN cooperates with RED, traditional AQM algorithms cannot be applied to WLANs. In this section we will discuss why.

In a wireless network, the bottleneck is the wireless channel, rather than a single link. Traffic flowing in several directions share the same resource (wireless spectrum), where no single queue through which all traffic traverses, exists. Therefore, congestion cannot be traced to a single node, but in an entire area involving multiple nodes. Obviously, neither can it be detected through the overflow of a single buffer. Even in a single node, a buffer in the uplink is absent; the only queue that exists, is for the outgoing packets. Apparently, the RED-with-ECN mechanism, where the marking probability is a function of the average size of some queue, cannot be applied.

Implementing RED for marking on a specific node would not be correct, either. First, a TCP connection which is penalized in channel contention may experience a queue buildup. However, dropping packets of the penalized flow may actually increase the unfairness. Secondly, the queue at any single node cannot completely reflect the network congestion state, as explained above. Third, since multiple nodes are involved in the congestion, they should coordinate their packet drops, rather than act independently. Thus, an appropriate RED-like scheme for wireless networks should consider the aggregate traffic in the wireless area, in order to perform marking.

## 2.2 Load-based marking

In wireless, there is no static link between any two nodes. Instead, all nodes share the same space. One of our key ideas is that the channel utilization percentage can evince congestion. When the aggregate utilization within the wireless area is high, the portion of the free channel for a station willing to transmit is low. Therefore, the aggregate throughput in the wireless area should indicate the level of congestion.

Based on the above, we calculate the marking probability based on measurements of the aggregate utilization over some time interval, as a function of the aggregate traffic load over the wireless channel. The same marking algorithm is used for all the traffic travelling through the wireless area, since all traffic share the wireless capacity. It is for the same reason that we consider the aggregate traffic for computing the marking probability.

The proposed LBM scheme is depicted in Figure 2.1. With LBM, the marking probability is a piecewise linear function of the average utilization. The LBM algorithm has three parameters: the time interval $t_{avg}$, over which the aggregate utilization is measured, the minimum utilization $\rho_0$, and the slope parameter a. Note that the the marking probability is zero, when the average utilization is less than $\rho_0$. For utilization values $\rho$ larger than $\rho_0$, the marking probability is given by $\min\{\alpha(\rho - \rho_0), 1\}$. A brief description of each of the three LBM parameters follows.

### 2.2.1 Analysis of the LBM parameters

The time interval $t_{avg}$ determines how quickly the algorithm adjusts the marking probability to changes of the aggregate utilization, and the timescale over which congestion is detected. Typically, $t_{avg}$ will be at least set to some number of RTTs, in order to obtain stable measurements of the load. Also, estimations should not happen rarely; we would like the system to be able to track traffic changes and adapt the marking probability effectively.

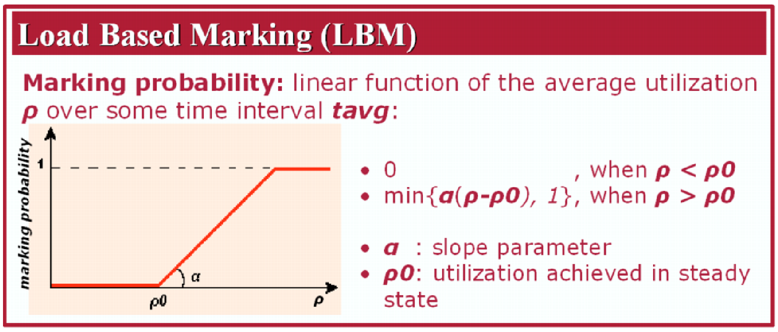



Figure 2.1: With LBM the marking probability is a piecewise linear function of the utilization. LBM has three parameters: the time interval $t_{avg}$ over which the aggregate utilization is measured, the minimum utilization threshold $\rho_0$, and the slope parameter $\alpha$.

The slope parameter $\alpha$ affects the reactivity and the stability of the marking algorithm [12]. A higher slope would yield a more reactive algorithm, since a small change of the utilization would give a large change of the marking probability.

Finally, for a fixed slope parameter $\alpha$, the minimum utilization $\rho_0$ determines the marking probability for a given aggregate utilization, hence the utilization achieved in the steady state.

## 2.3 ECN for end-to-end congestion signaling

ECN can signal congestion through marks, rather than through lost packets. In wireless networks, as explained, non-congestion related packet losses due to channel corruption can occur with a non-negligible probability. This fact as well as ECN potentials to avoid unnecessary packets' delay from low-bandwidth delay-sensitive TCP connections, as is the case in wireless connections, make it appropriate for wireless networks.

However, ECN alone cannot alleviate the problem of TCP decreasing its throughput in the case of non-congestion related losses. To address this, either TCP reaction to losses must be modified, or link-layer mechanisms should hide losses due to corruption from TCP. In the first case for example, TCP might identify and differentiate between congestion and non-congestion related losses. In the latter, a link-layer mechanism could implement forward error correction and retransmissions over the wireless link. Our proposal for using ECN goes one step further from addressing the issue of congestion and non-congestion related losses, which we assume are handled by IEEE 802.11 MAC link-layer retransmission mechanism.

In our work, ECN is used to convey congestion information across the wireless network. For wired networks, marking is performed at the output link of routers. For the wireless, marking is performed at the points of the network where information for the congestion level can be aggregated. For instance, in the case of an infrastructure network marking should be performed at the AP. For a multihop network, marking, for a traffic flow must be performed by all sending nodes, with respect to the traffic load in their collision domain.

# Chapter 3

# Marking for infrastructure wireless networks

In this chapter, we suggest how to implement ECN marking in heterogeneous networks, based on measurements of the aggregate utilization on the APs.

## 3.1 Marking at the AP

Among the basic features of an infrastructure network is the presence of one or more APs, coordinating the traffic exchange between the wireless stations within a wireless cell, or between those and the backbone wireline network. The traffic originating from the wireless stations of the wireless cell is directed to the AP, which forwards it, according to the destination, to the next hop. Based on this, we argue that the AP is the network point where all the wireless traffic can be aggregated.

The reason for this is that in the downlink of an IEEE 802.11 infrastructure WLAN there is a shared buffer located at the AP; this is not the case in the uplink direction. In the uplink, a single buffer, shared by packets originating from different hosts, is absent. Instead, each wireless station has its own local buffer. Upon the receipt of a packet from the wireless, the MAC layer of the AP hands it out to the link layer. Consequently, while a RED-like mechanism could be applied for the downlink, it cannot, for the uplink. Still, since the uplink and downlink traffic share the wireless capacity, the aggregate transmission throughput in both directions, measured at the AP, should be considered for the calculation of the utilization over the wireless.

Our approach is to use ECN to convey congestion information from both the wired

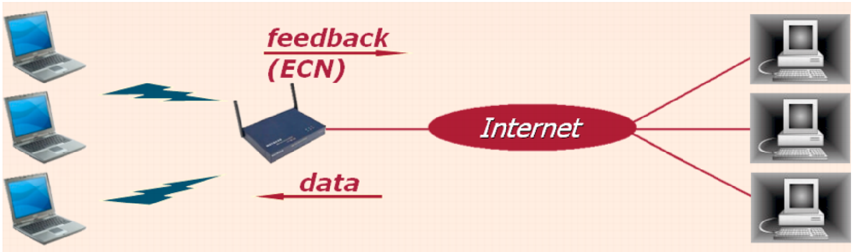



Figure 3.1: Marking performed at the AP, for both the uplink and the downlink.

and the wireless links. For a wireless link, marking is performed at the AP, Figure 3.1, and differs from the marking procedure for the wired links. In particular, the AP is responsible for packets' marking in both the uplink and the downlink, based on measurements of the aggregate utilization, taking into account the traffic flowing in both directions. Hence, the marking procedure for the wireless link takes into account its particular the resource sharing model of IEEE 802.11. The exact algorithm is depicted in Figure 2.1. For the wired links, routers are responsible for marking, based on estimations of the average queue size of their local buffers.

## 3.2 LBM adaptation

The LBM algorithm can dynamically adapt to varying traffic and load conditions, as we discuss next. The most appropriate parameter to achieve effective adaptation is the minimum utilization parameter, $\rho_0$.

Recall that, among the three LBM parameters, $\rho_0$ is the parameter expressing how tolerant is the algorithm to the utilization increase, i.e., the level of congestion that the network can tolerate, before it starts marking. When the aggregate throughput measured by the AP increases, this means that the AP has lots of packets to serve. In the uplink this can be translated to a growing number of collisions in wireless. We suppose that collisions are handled by the RTS/CTS built-in mechanism of the IEEE 802.11 MAC sublayer. For the downlink, this could lead to larger buffering rates, with a queue buildup and longer packets' queuing delay. Measuring delay from an upper layer, i.e., the interval between the time a packet is queued at the AP and the time

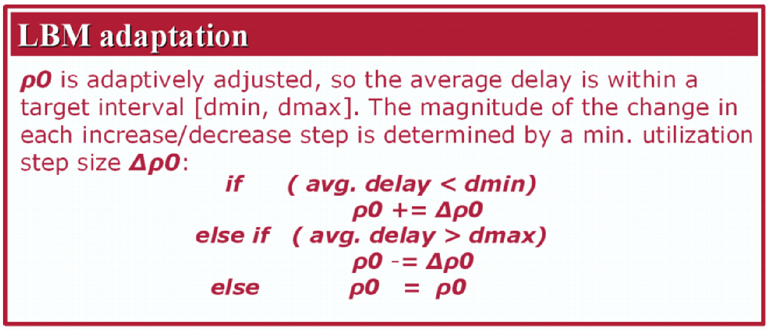



Figure 3.2: Most appropriate parameter to achieve effective adaptation is $\rho_0$.

the corresponding ACK is handed out to the AP, can illustrate the above phenomena. Therefore, given a utilization increase, the average delay over the wireless also increases.

From the above we argue that since $\rho_0$ can reduce load tolerance, it can also be set to control the average delay. Indeed, we adaptively adjust $\rho_0$, so that the average delay remains within a target interval $[d_{min}, d_{max}]$. In particular, $\rho_0$ is increased when the average delay is less than $d_{min}$, and is decreased when the average delay is greater than $d_{max}$. The magnitude of the change in each increase or decrease step is determined by the minimum utilization step size $\Delta\rho_0$. The values of $d_{min}$ and $d_{max}$ are related to the target packet delay requirements over the wireless link; indeed, the difference $d_{max} - d_{min}$ determines the allowed variation of the average delay in different traffic and load conditions. Figure 3.2 illustrates the $\rho_0$ adaptation algorithm.

## 3.3 Simulation results

The effectiveness of the LBM algorithm was tested through NS-2 [20] simulations. We compared the proposed marking approach with DT in the topology shown in Figure 3.1.

### 3.3.1 Simulation configuration

According to the scenario, traffic flows from the fixed hosts to the wireless hosts, i.e., from right to left. In the experiments, the IEEE 802.11 MAC layer performs retransmissions of corrupted packets; losses due to corruption are assumed to be independent (non-bursty). In the experiments we tested several different values of the loss rate in wireless.

We consider FTP flows that transfer files whose sizes follow a pareto distribution with average 50 KBytes and 500 KBytes. For 50 KBytes average file size, the start time of each FTP flow was randomly selected from the interval [0,0.5] seconds and for 500 KBytes average file size from the interval [0,5] seconds.

Moreover, we tested the LBM algorithm in topologies with different number of flows in wireless. The experiments also used different RTTs, in order to decide how the LBM algorithm interacts with traffic flows of different round trip latencies. Finally, in the experiments we used TCP Reno.

### 3.3.2 Selection of the LBM parameters

The experiments were conducted for a variation of values of the LBM parameters. We realize that very small values of the LBM minimum utilization threshold $\rho_0$ trigger inefficient marking, since they make the network very intolerant to the utilization increase. Therefore, $\rho_0$ is set to values over 0.1. We also consider different values of the slope parameter $\alpha$, as well as different time intervals $t_{avg}$, for the calculation of the aggregate utilization. We always consider $t_{avg}$ to be a few RTTs, for the reasons explained in section 2.2.1.

### 3.3.3 Performance metrics

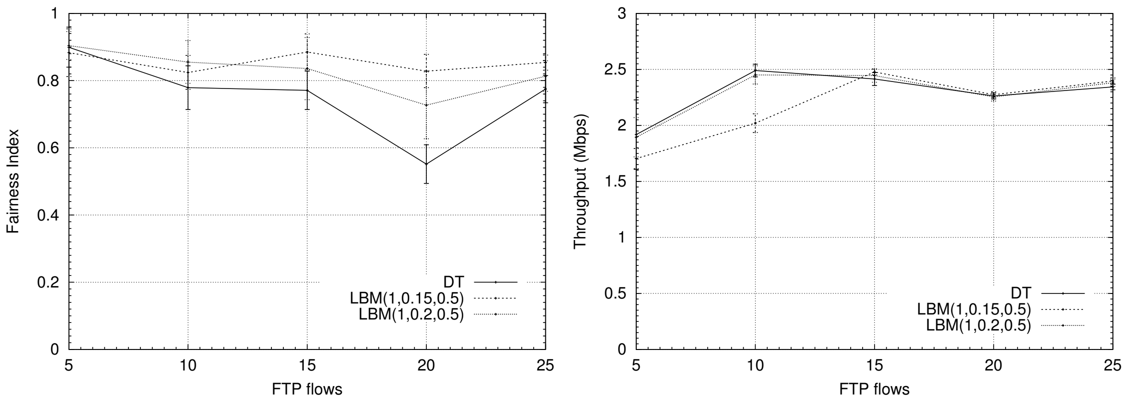
The network performance metrics we consider are throughput, fairness index, average and standard deviation of the packet delay over the wireless link.

The throughput for each flow is given by the ratio of the total transmitted traffic (data and overhead) and the duration of the file transfer, where the latter is the interval between the time the first SYN packet is sent by the sender, and the time the last ACK is received.

As a measure of fairness we consider the widely used metric given in [7]:

$$\text{Fairness Index} = \frac{\left(\sum_{i=1}^{N} x_i\right)^2}{N \sum_{i=1}^{N} x_i^2} \ ,$$

where $x_i$ is the rate of flow $i$ and $N$ is the number of flows. The fairness index takes values in the interval (0,1], with a higher value indicating higher fairness.




(a) Fairness index (b) Throughput

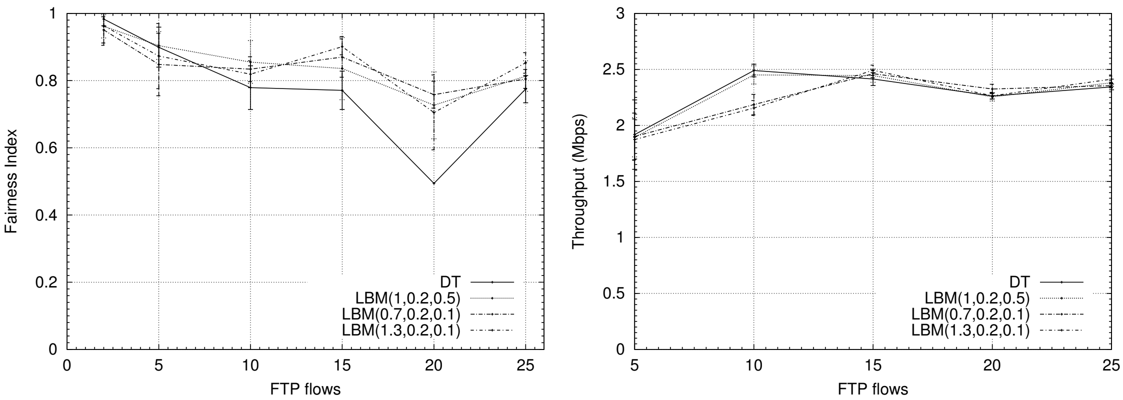
Figure 3.3: Fairness and throughput for different number of FTP flows. LBM parameters: $\alpha = 1$, $\rho_0 = 0.15, 0.2$, $t_{avg} = 500\ ms$. Average file size = 500 KBytes, RTT = 50 ms, wireless loss probability = 0.01.

The average delay for transmitting a packet over the wireless link is measured in a non-intrusive manner. For our scenario, the delay is measured as the interval between the time a packet is queued by the AP and the time the corresponding ACK is received; therefore, we also include the queuing delay.

### 3.3.4 Results and analysis

#### Fairness and throughput

Figures 3.3(a) and 3.3(b) show the fairness and throughput respectively, achieved by DT and LBM, with slope parameter $\alpha = 1$, minimum utilization threshold $\rho_0 = 0.15$ and 0.2, and averaging interval $t_{avg} = 500\ ms$. As explained in section 3.3.2, it is sufficient to set this interval to be a few times the RTT. Figure 3.3(a) shows that LBM achieves better fairness compared to DT; furthermore, the difference between the fairness achieved by LBM and DT is larger, for a larger number of FTP flows. Figure 3.3(b) shows that the utilization achieved by both DT and LBM when $\rho_0 = 0.2$ is identical. The fact that the use of ECN does not result in higher utilization compared to DT should not be that surprising, since experiment for wired networks also show that, for an appropriately dimensioned network, TCP with ECN does not achieve higher throughput compared to TCP with DT [19].



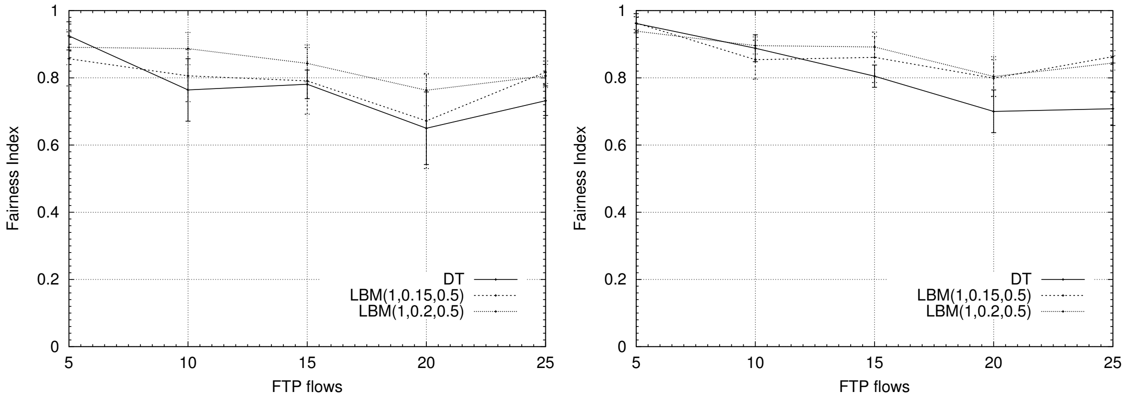

(a) Fairness index (b) Throughput

Figure 3.4: Fairness and throughput for different number of FTP flows. LBM parameters: $\alpha = 1, 0.7, 1.3$, $\rho_0 = 0.2$, $t_{avg} = 500\ ms, 100\ ms$. Average file size = 500 KBytes, RTT = 50 ms, wireless loss probability = 0.01.

Figures 3.4(a) and 3.4(b) show the fairness and throughput respectively, achieved by DT and LBM, for different values of the slope parameter $\alpha$. The conclusion from these graphs is identical to the above. Note that in the case of LBM with $\rho_0 = 0.15$, the utilization for 5 and 10 flows is smaller. This could mean that the specific value for $\rho_0$ triggers a rather early and inefficient marking, for the scenarios with the given number of flows. Figures 3.5(a) and 3.5(b), which are for RTT = 20 $ms$ and 100 $ms$ respectively, support the above conclusion regarding the improved fairness of LBM.

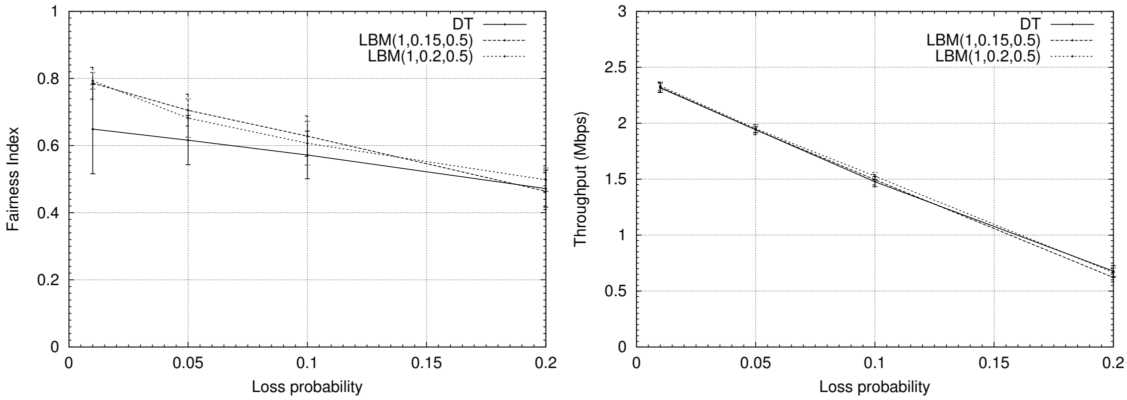
Figure 3.6(a) shows the fairness, for different packet loss probabilities over the wireless link. Observe that the difference between the fairness achieved with LBM and DT is larger for smaller loss probabilities. Figure 3.6(b) shows that the throughput achieved by LBM and DT is the same, for different values of the wireless loss probability. It is also inversely proportional to this; indeed, it decreases as the loss probability increases.

The results in Figures 3.7(a), for average file size 50 KBytes, shows that, as above, LBM achieves higher fairness, compared to DT. Figure 3.7(b), which is also for average file size 50 KBytes, shows that LBM can achieve the same throughput as DT by appropriately setting the minimum utilization threshold, $\rho_0$. Moreover, observe that, as expected, a smaller value of $\rho_0$ can lead to lower throughput.



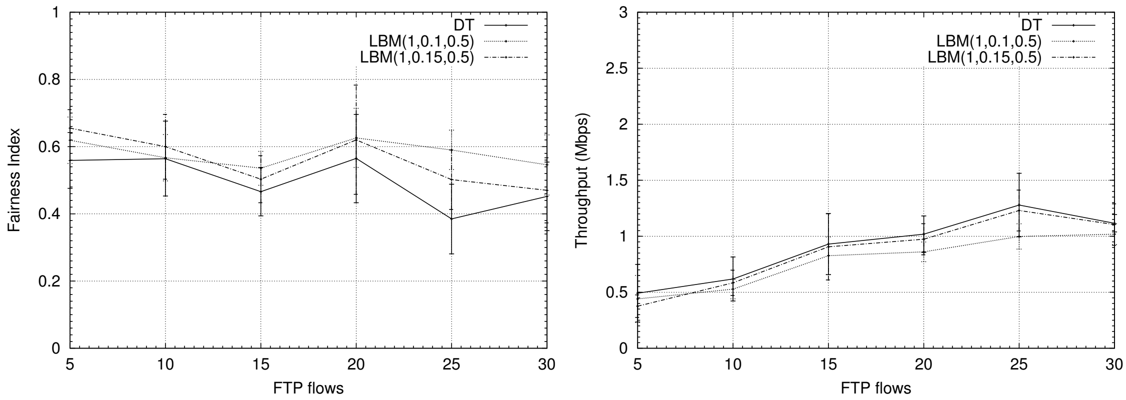

(a) $RTT = 20\ ms$ (b) $RTT = 100\ ms$

Figure 3.5: Fairness index for different RTT values. LBM parameters: $\alpha = 1$, $\rho_0 = 0.15, 0.2$, $t_{avg} = 500\ ms$. Average file size = 500 KBytes, wireless loss probability = 0.01.




(a) Fairness index (b) Throughput

Figure 3.6: Fairness index and throughput for different wireless loss probabilities. LBM parameters: $\alpha = 1$, $\rho_0 = 0.15, 0.2$, $t_{avg} = 500\ ms$. Average file size = 500 KBytes, RTT = 20 ms, 20 FTP flows.




(a) Fairness index (b) Throughput

Figure 3.7: Fairness and throughput for different number of FTP flows. LBM parameters: $\alpha = 1$, $\rho_0 = 0.15, 0.2$, $t_{avg} = 500\ ms$. Average file size = 50 KBytes, RTT = 20 ms, wireless loss probability = 0.01.

Table 3.1: Average and standard deviation of packet delay over the wireless link. File size = 500 KBytes, RTT = 50 ms, wireless loss probability = 0.01, LBM parameters: $\alpha = 1$, $\rho_0 = 0.2$, $t_{avg} = 500\ ms$.

| # of flows | DT | | LBM | |
|---|---|---|---|---|
| | avg. delay | std.dev. | avg. delay | std.dev. |
| 3 | 23.5 | 8.1 | 13.1 | 3.4 |
| 5 | 39.1 | 16.4 | 15.9 | 4.3 |
| 10 | 65.9 | 17.3 | 32.9 | 14.3 |
| 15 | 92.9 | 19.1 | 57.4 | 8.7 |
| 20 | 101.1 | 10.7 | 71.9 | 15.0 |

Table 3.2: Minimum utilization threshold values for a different number of flows, when target delay in $[10ms, 15ms]$. File size = 500 KBytes, RTT = 50 ms, wireless drop probability = 0.01, other LBM parameters: $\alpha = 1, tavg = 500\ ms$.

| # of flows | avg. delay | std.dev. | throughput (Mbps) | $\rho_0$ |
|---|---|---|---|---|
| 3 | 13.1 | 3.4 | 2.2 | 0.20 |
| 5 | 13.7 | 3.4 | 1.7 | 0.19 |
| 10 | 13.8 | 2.5 | 2.2 | 0.13 |
| 15 | 14.5 | 2.4 | 2.3 | 0.07 |
| 20 | 14.0 | 2.4 | 2.2 | 0.05 |

**Packet delay and delay jitter**

Table 3.1 shows the average standard deviation of the packet delay over the wireless link. Observe that LBM achieves a smaller average delay and delay jitter than DT, as indicated by the smaller values of the standard deviation.

Next we investigate the dynamic adaptation feature of the proposed approach. To maintain the average delay within the interval $[d_{min}, d_{max}] = [10ms, 15ms]$ milliseconds, the value of $\rho_0$ is increased when the average delay is smaller than $d_{min}$, or is decreased when the average delay is larger than $d_{max}$. Moreover, we assume that the adjustment of $\rho_0$ is performed in steps of $\Delta\rho_0 = 0.01$; this step size affects how fast the algorithm adapts to changes of the network load. The results appear in Table 3.2, and show that, by adjusting the minimum utilization threshold $\rho_0$, we can effectively control the average delay such that it remains inside the target interval.

The dynamic behavior of the LBM adaptation procedure maintains the average delay inside the target interval. The transient behavior will depend on the step parameter $\Delta\rho_0$ and the difference $d_{max} - d_{min}$.

# Chapter 4

# Marking for multihop wireless networks

In this chapter we suggest how to apply ECN marking in multihop networks, based on estimations of the congestion level within different collision domains within the network. The two key ideas of the approach are the same; first, ECN is used as the end-to-end congestion signaling mechanism, for conveying congestion information across the multihop network. Secondly, marking at a wireless node is performed using an LBM algorithm; the adjustment needed in order for the algorithm to work in multihop environments is that the aggregate utilization should be measured within the node's collision domain.

## 4.1 Why traditional AQM algorithms cannot be applied to multihop networks?

As explained in section 2.1, in a multihop network, there is no single shared buffer that is used for the packets flowing in the same area. Therefore, a RED-like marking algorithm, where the packet marking probability is a function of an average queue length, cannot be applied.

Likewise, there is no pre-defined link between any two nodes. Instead, nodes share the wireless channel and compete for it in a distributed manner under the coordination of the MAC protocol. Additionally, unlike an infrastructure network, there is no AP acting as a gateway for all the packets travelling in the wireless. Thus, congestion cannot be tracked to a specific node but to the entire wireless area around it. Moreover, due to attenuation in the wireless channel, contention is location-dependent; stations with different relative positions in the network may get different perception of the bottleneck situation. Hence, the total utilization can no longer accurately reflect the level of congestion in the wireless network. It is critical, however, to the fairness of TCP congestion control, to get correct feedback of the bottleneck.

## 4.2 Marking in the node's collision domain

Apparently, for each node we need to define this area over which the aggregate utilization should be estimated. In a multihop network, depending on the packets it sends, receives or senses within its range each node can obtain a different view of the level of congestion in its surrounding area. In fact, the node itself also contributes to congestion. The above location-dependent information must be taken into consideration to effectively and fairly control the usage of the wireless spectrum by all nodes.

### 4.2.1 Interfering nodes

For each node, the location-dependent information exists inside its collision domain. Other nodes' signals can interfere with the node's transmission in this domain, when the destination is the same.

Interference with a node may cause both one-hop and two-hop neighbors. However, the set of nodes that interfere with each other also depends on the traffic destination. For example, consider the multihop network shown in Figure 4.1, where the transmission range for each node extends up to its immediate neighbor. Assume the case where node 2 transmits to node 1, and node 3 transmits to node 4. Even though 2 and 3 are immediate neighbors, they can both simultaneously perform their transmissions, since each is two hops away from the other's destination (this is the so-called exposed terminal scenario). Similarly, which two hop neighbors interfere, also depends on the destination. Hence, if both nodes 2 and 4 transmit to node 3, they are in the same collision domain (i.e., hidden terminal scenario). On the other hand, if node 2 transmits to node 1 and node 4 transmits to node 5 on its right, then nodes 2 and 4 are in different collision domains.

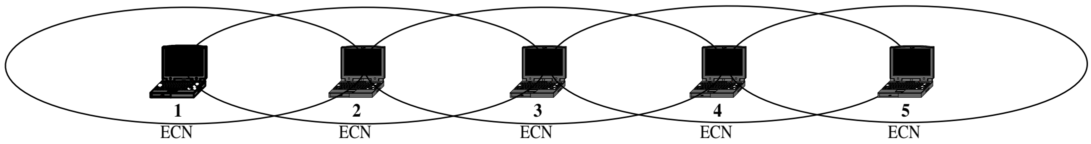



Figure 4.1: All nodes apply ECN marking based on the congestion within their collision domain.

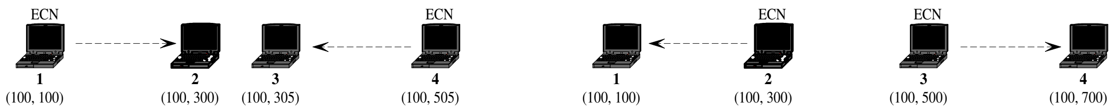
## 4.3 Calculation of the aggregate utilization

Next we discuss how a node can compute the aggregate utilization within its collision domain. First, note that summing the transmission rates of neighboring nodes is not correct, since in the exposed terminal scenario mentioned above, because nodes 2 and 3 do not interfere, node 2 should not consider node 3's transmission rate. Furthermore, considering the transmission rate would not account for contention due to two hop neighbors, for example in the case both nodes 2 and 4 transmit to node 3.

Summing the utilization measured by neighboring nodes is for the same reason incorrect; indeed, such an approach has another subtle problem since it can result in considering some transmissions twice; once considering the transmission rate and once considering the receiving rate. Another option is to sum the received rates of all neighbors. In the case where the transmission range is the same for all nodes, and the transmission range is the same with the interference range, then this approach would indeed consider the received rates of only those neighbors that a node's transmission can reach, hence can interfere with other nodes transmitting to the same destination.

Based on the sum of the received rates of all neighboring nodes within its collision domain, each node computes a marking probability using an LBM algorithm 2.1. Hence, we combine this local (collision domain specific) broadcasting of congestion information with ECN marking, which provides the end-to-end communication and accumulation of congestion information.

A node's receiving rate can be communicated to all its neighbors within its collision domain by piggy-backing this information on control packets, such as CTS and RTS messages, or data packets. Such a scheme has the goal to broadcast location-dependent congestion information within a single collision domain, which is analogous to the use of the RTS/CTS mechanism for informing nodes of the presence of other neighboring nodes with which their transmission can collide.




(a) Hidden terminal scenario

(b) Exposed terminal scenario

Figure 4.2: Simulation scenarios

## 4.4 Estimation of the marking probability

Given the congestion information from its neighbors, each wireless node should determine the probability with which it will mark packets at its output queue. Then, based on the aggregate congestion along the whole path its packets traverse, which is signaled using ECN marks, the node responds by adjusting its transmission window according to the TCP congestion control protocol.

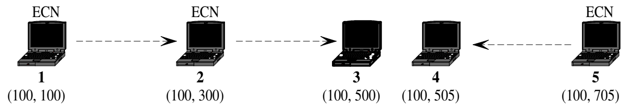
Based on the previous discussion, the marking probability is a function of the aggregate utilization over some time interval $t_{avg}$, measured by dividing the sum of the received throughput of all nodes within the same collision domain, with the wireless channel capacity. The marking probability has a piecewise linear dependence on the aggregate utilization, as illustrated in Figure 2.1. The marking probability is zero when the average utilization is less than $\rho_0$. For utilization values $\rho$ larger than $\rho_0$, the marking probability is given by $\min\{\alpha(\rho - \rho_0), 1\}$.

## 4.5 Simulation results

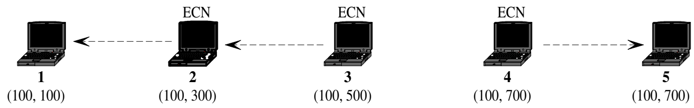
In this section we present and discuss the simulation results, comparing the proposed marking approach with DT. Simulations were conducted using, again, the NS-2 simulator.

### 4.5.1 Simulation configuration

The topologies simulated are shown in Figures 4.2, and 4.3. The transmission range is equal to the interference range, both being 250 meters. The scenario in Figure 4.2(a) implements the hidden terminal case, with traffic flowing from node 1 to node 2, and from node 4 to node 3. According to the proposed scheme, marking is performed at




(a) Hidden terminal scenario with 5 stations




(b) Exposed terminal scenario with 5 stations

Figure 4.3: Multihop scenarios

the senders, i.e., nodes 1 and 4, and the marking probability at both of these nodes considers the sum of the receiving rate at nodes 2 and 3.

The second scenario shown in Figure 4.2(b) implements the exposed terminal case, with traffic flowing from node 2 to node 1, and from node 3 to node 4. In this case, node 2 calculates a marking probability based on the receiving rate of node 1, whereas node 3 calculates a marking probability based on the receiving rate of node 4.

The scenarios in Figures 4.3(a) and 4.3(b) are multihop scenarios. In Figure 4.3(a), traffic flows from node 1 to node 3, and from node 5 to node 4. The traffic originating from node 1 is forwarded by node 2 to node 3. Marking, in this scenario, is performed by nodes 1, 2, and 5, i.e., the senders. Each of the nodes 2, and 5, computes the marking probability considering the receiving rates of both 3 and 4, while node 1 only considers the receiving rate of 2.

According to the second multihop scenario, depicted in 4.3(b), node 3 transmits to node 1 through node 2, and node 4 transmits to node5. Each of the three sending nodes, i.e., nodes 3, 2, and 4, performs marking based on the receiving rate of its one-hop neighbor. The two multihop scenarios were selected in order to test the LBM algorithm in the case of nultihop and non-symmetric traffic.

We consider FTP flows that transfer files whose sizes follow a pareto distribution with average 500 KBytes and 5 MBytes, and shape parameter 1.5. The throughput for each flow is given by the ratio of the total received traffic -data and overhead- and the duration of the file transfer, which is taken to be the interval between the time the

Table 4.1: State transition matrix (Inverted Markov matrix).

| Current state | Next state | | |
|---|---|---|---|
| | G | B | VB |
| G | 0.95 | 0.02 | 0.03 |
| B | 0.10 | 0.20 | 0.70 |
| VB | 0.20 | 0.03 | 0.77 |

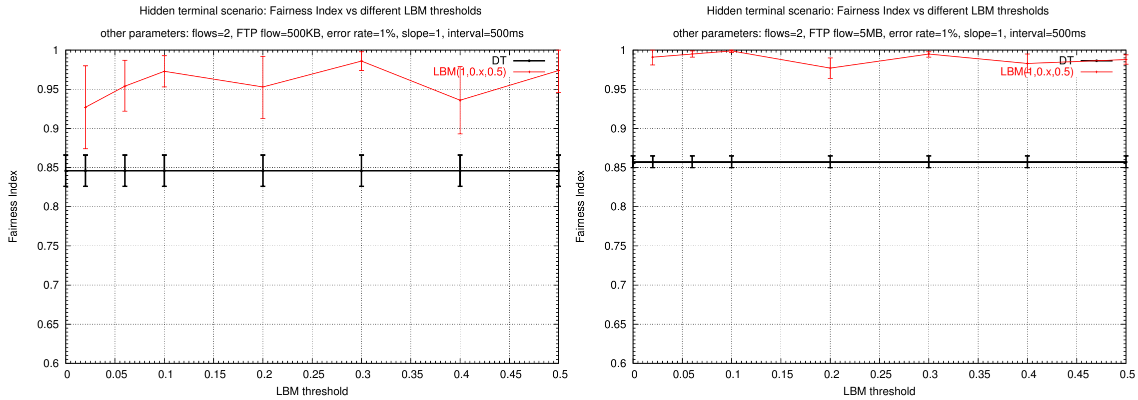
first SYN packet is sent by the sender, and the time the last ACK is received. The start time of each FTP flow was randomly selected from the interval $[0, 5]$ seconds. In the experiments we used TCP Reno. Regarding the parameters of the LBM marking scheme, in all the experiments the measurement interval parameter is 500 $ms$. On the other hand, we consider different values for the minimum utilization threshold and the slope parameters.

In the experiments, the IEEE 802.11 MAC layer performs retransmissions of corrupted packets; losses due to corruption are assumed to be bursty. To model the bursty (time correlated) errors in the wireless channel, we consider a multi-state error model similar to the one presented in [6], which consists of a three-state discrete-time Markov chain. The Markov chain time slot is equal to the slot time of the 802.11b MAC sublayer, which is $20\,\mu s$. In the good state (G), the transmission is error free. In state B errors occur uniformly with probability $p$, whereas in state VB errors occur uniformly with probability $5.5 \times p$. The probability of transition between states is shown in Table 4.1. In our experiments we consider two cases: the case of an error free transmission, and the case where errors occur according to the above model with an average error probability 1%, which is achieved for $p = 0.01$.

### 4.5.2 Performance metrics

Again, the network performance metrics we consider are throughput, fairness index, average and standard deviation of the packet delay over the wireless link. We measure the throughput and fairness index as defined in Section 3.3.3.

For the scenarios in Figures 4.2, and 4.3, the delay is measured as the interval between the time a packet is enqueued in the output queue of the TCP sender, and the




(a) 500 KB flows

(b) 5 MB flows

Figure 4.4: Fairness for the hidden terminal scenario, different LBM minimum utilization thresholds $\rho_0$, and average error probability 1%.

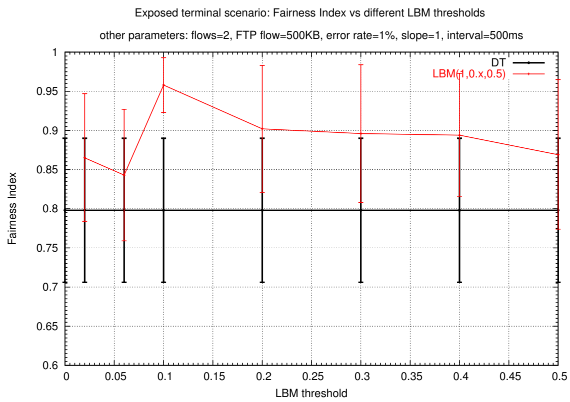
time the corresponding ACK is received; thus, we also include the queuing delay.

### 4.5.3 Results and analysis

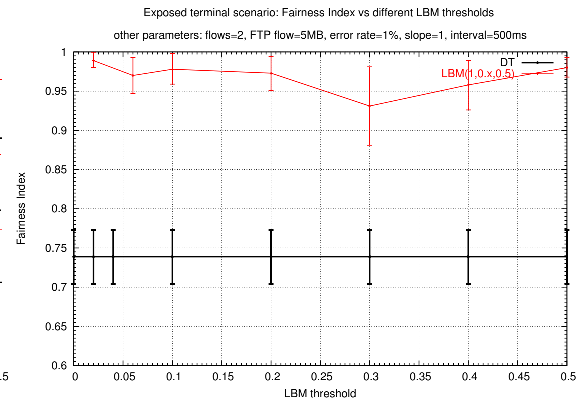
#### Fairness and throughput

Figures 4.4(a) and 4.4(b) for the hidden terminal scenario in Figure 4.2(a), and Figures 4.5(a) and 4.5(b) for the exposed terminal scenario in Figure 4.2(b), show that the LBM scheme achieves higher fairness compared to DT, for both short and long FTP flows. Indeed, an improvement is achieved for a range of values of the LBM minimum utilization threshold $\rho_0$. Also, observe that in the case of short FTP flows, the fairness improvement appears to be smaller for smaller values of the minimum utilization threshold parameter, Figures 4.4(a) and 4.5(a); this can be attributed to the fact that a smaller threshold corresponds to very early, and thus, inefficient marking. On the other hand, in the case of long FTP flows Figures 4.4(b) and 4.5(b) there does not appear to be such a dependence on the LBM minimum utilization threshold parameter.

Comparison of Figure 4.4(a) with Figure 4.4(b) for the hidden terminal scenario, and Figure 4.5(a) with Figure 4.5(b) for the exposed terminal scenario, show that the improvements are larger for longer FTP flows. Finally, the above figures show that for long FTP flows the improvement is larger in the exposed terminal scenario; on the other hand, the improvement is similar in both scenarios, in the case of small FTP flows. To illustrate what the different values of fairness imply, note that a fairness index equal




(a) 500 KB flows




(b) 5 MB flows

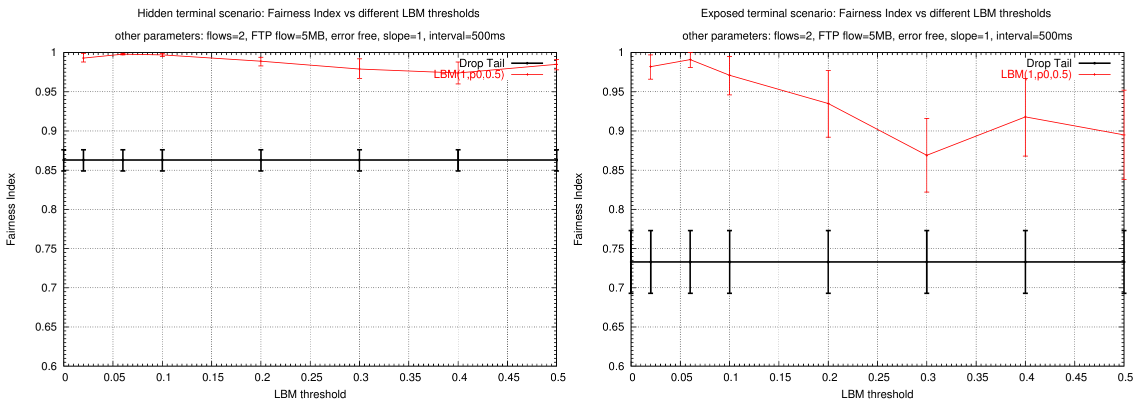
Figure 4.5: Fairness for the exposed terminal scenario, different LBM minimum utilization thresholds $\rho_0$, and average error probability 1%.

to 0.8 corresponds to the case where the two flows achieve a throughput of 2.09 Mbps and 0.75 Mbps. On the other hand, a fairness index equal to 0.97, corresponds to the case where the two flows achieves a throughput of 1.40 Mbps and 1.87 Mbps.

Figures 4.6(a) and 4.6(b) show the fairness improvement achieved by the proposed approach compared to DT, for an error free channel. As before, the gains in the exposed terminal scenario are higher than in the hidden terminal counterpart. Also observe in these graphs, that in both scenarios there is a decrease of the fairness when the LBM minimum utilization threshold increases, which is slight for the hidden terminal scenario, but substantial for the exposed terminal counterpart. This behavior must be further investigated with detailed analysis of the exact events of the simulation scenarios.

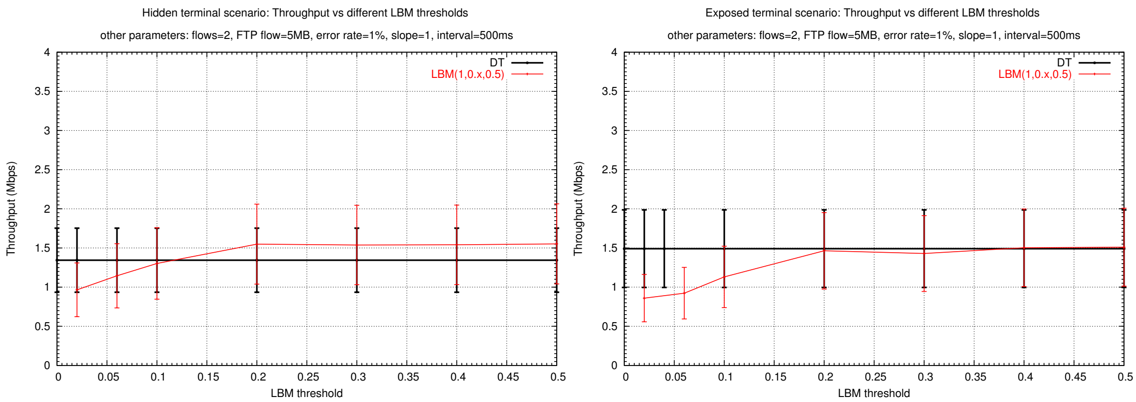
In all the above experiments, the aggregate throughput achieved by both the LBM scheme and DT is the same, as shown in Figures 4.7(a) and 4.7(b). Observe in these figures that the throughput decreases for small values of the utilization threshold, which is also the case with the fairness index, and which corresponds to a rather early and inefficient marking. Hence, improved fairness is not achieved at the expense of decreased throughput, as is the case with the neighborhood RED scheme proposed in [29].

The effect of the slope parameter to the LBM behavior was tested for both small and large values of the slope parameter $\alpha$. In Figures 4.8(a) and 4.8(b), which are for 500KB flows, and for small values of the $\alpha$, LBM shows higher fairness compared to DT, for different slope parameters. Observe that the improvement decreases for smaller



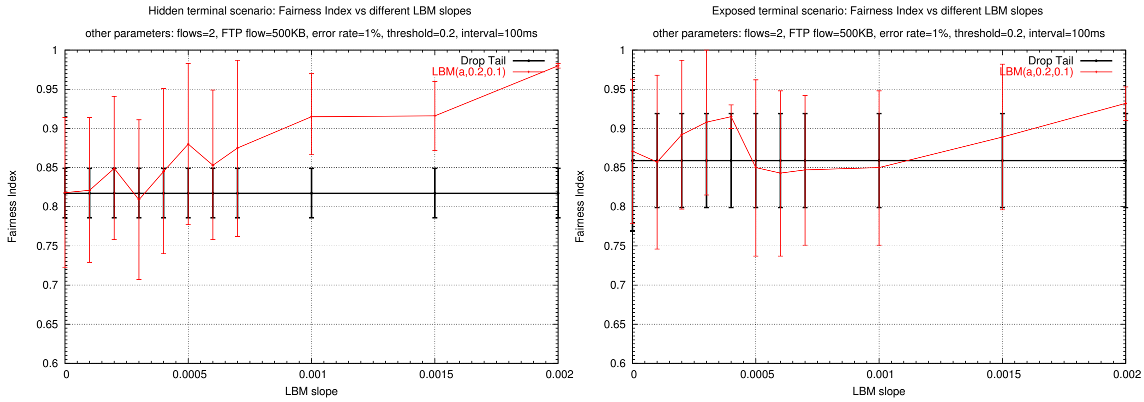

(a) Hidden terminal scenario (b) Exposed terminal scenario

Figure 4.6: Fairness for different thresholds, an error free channel, and 5 MB flows.




(a) Hidden terminal scenario (b) Exposed terminal scenario

Figure 4.7: Throughput for different LBM minimum utilization thresholds, average error probability 1%, and 5 MB flows.




(a) Hidden Terminal scenario (b) Exposed terminal scenario

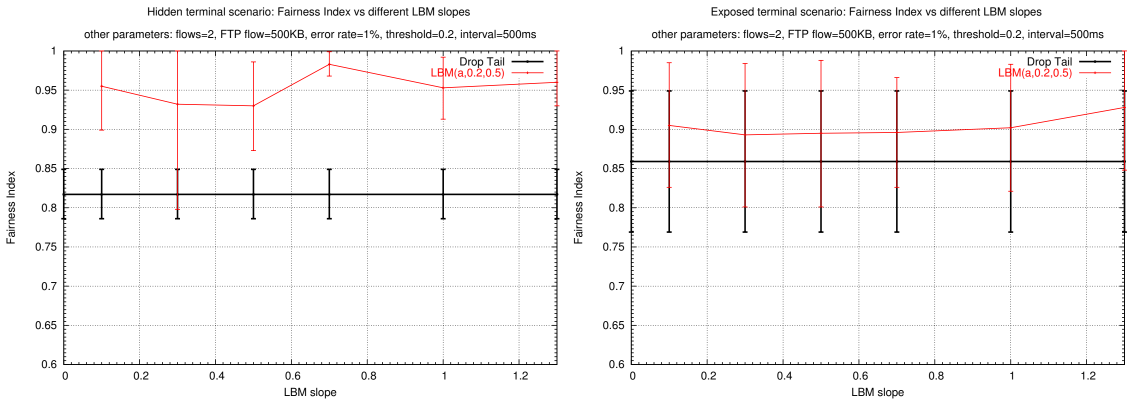
Figure 4.8: Fairness for small LBM slope parameters, 500 KB flows and average error probability 1%.

values of the slope parameter. This can be explained by the fact that a smaller slope corresponds to a less aggressive marking. Observe that for $\alpha = 0$, which corresponds to the case where the marking probability is always 0, the fairness index value equals the value for DT. This behavior confirms the correct behavior of the LBM algorithm.

Figure 4.9, which is for 500 KB flows, and for large values of the slope parameter, also shows that the LBM marking scheme achieves higher fairness, compared to DT, for different values of the $\alpha$, in both scenarios. In the case of the large values of the slope parameter, there does not seem to be a relationship of the fairness, with the increase of the value of $\alpha$. Indeed, since for large values of $\alpha$ the marking is severe, there is a possibility that LBM starts penalizing both flows with ECN marks, with the marking probability being high, even for the flows that experience unfair TCP behavior.

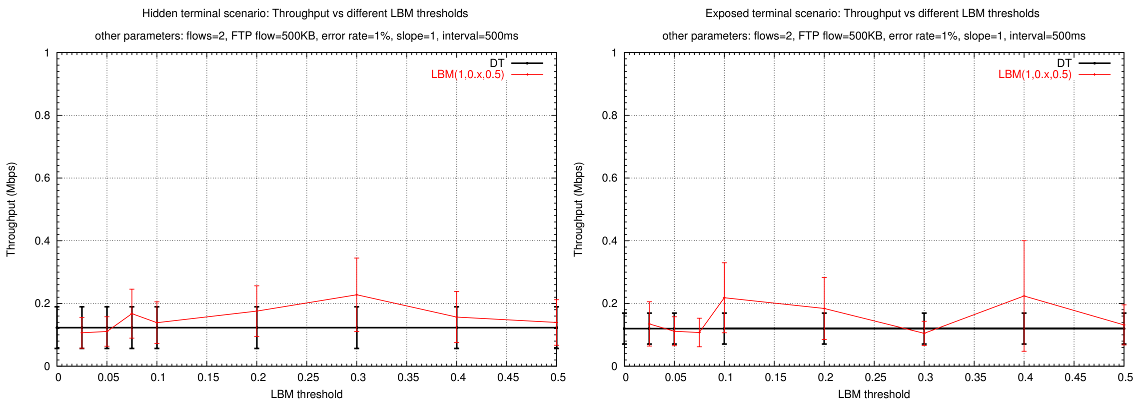
In Figures 4.10 and 4.11 results are shown for the case of the multihop scenarios of Figures 4.3 (a) and 4.3(b), respectively. Throughput seems to remain the same with that of DT, while for very small values of the minimum utilization threshold $\rho_0$ it seems to decrease. Due to the very early marking that the small values of $\rho_0$ yield, this is reasonable.

The fairness index also improves in the presence of LBM. Again, for very small values of the $\alpha$ parameter, which means that there are hardly any packets to mark, the LBM algorithm results in the same fairness with the DT. The slight difference between the fairness index values of the two algorithms (LBM and DT) in the case where $\alpha = 0$ is normal, as long as the confidence intervals are the same.



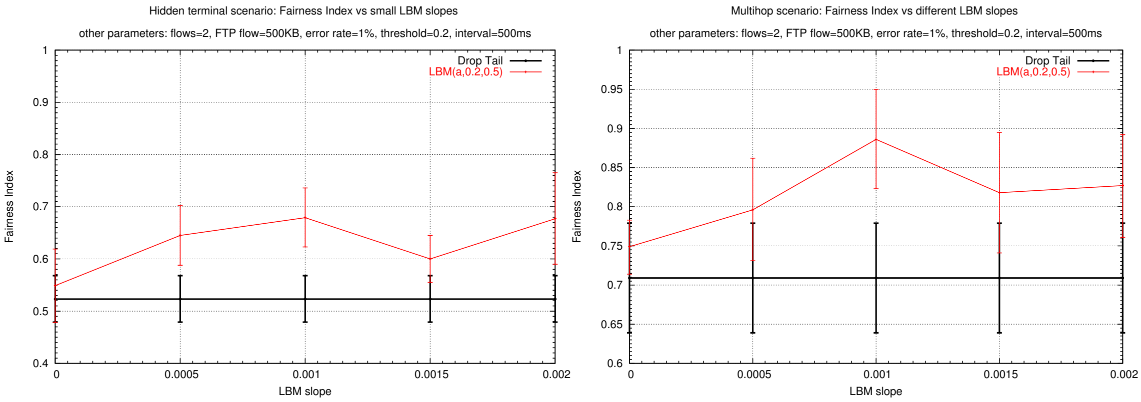

(a) Hidden Terminal scenario (b) Exposed terminal scenario

Figure 4.9: Fairness for large LBM slope parameters, 500 KB flows, and average error probability 1%.




(a) Hidden Terminal scenario (b) Exposed terminal scenario

Figure 4.10: Throughput for multihop scenarios, different minimum utilization threshold parameters, 500 KB flows, and average error probability 1%.




(a) Hidden Terminal scenario (b) Exposed terminal scenario

Figure 4.11: Fairness for multihop scenarios, small LBM slope parameters, 500 KB flows, and average error probability 1%.

### Packet delay and delay jitter

Table 4.2 shows the average and standard deviation of the packet delay over the wireless link. Observe that the LBM marking scheme achieves a smaller average delay and delay jitter compared to DT, as indicated by the smaller values of the standard deviation. In contrast to what happens for infrastructure networks, in this case the delay over the wireless is not very sensitive to slight changes of $rho_0$, so there is no point in implementing the $\rho_0$ adaptation.

Table 4.2: Average and standard deviation of packet delay over the wireless link. File size=1 MByte, loss prob.=1%, LBM parameters: $\alpha = 1$, $\rho_0 = 0.1$, $t_{avg} = 500ms$.

| Terminal scenario | Flow | DT | | LBM | |
|---|---|---|---|---|---|
| | | avg. delay(ms) | std.dev. | avg. delay(ms) | std.dev. |
| Hidden | 1 | 97.6 | 77.7 | 13.77 | 13.1 |
| Hidden | 2 | 60.9 | 57.7 | 9.24 | 26.1 |
| Exposed | 1 | 41.3 | 26.5 | 22.0 | 19.8 |
| Exposed | 2 | 49.6 | 65.9 | 13.9 | 37.9 |

In the next chapter we present the contributions for improving the TCP performance in wireless networks. Emphasis will be given to the description of two studies, which are most closely related to ours, in the sense that they also trigger marking mechanisms based on early congestion exchange information.

# Chapter 5

# Related work

In this chapter we present an overview of representative work on improving the performance of TCP over the wireless links. More specifically, we will focus on studies that apply ECN in wireless networks, identifying how it differs from the approach presented in this thesis. Also, we will describe some AQM algorithms, in order to pose our approach with respect to these paradigms.

## 5.1 Improving TCP performance over wireless links

Generally, the approaches for improving the TCP performance over wireless links fall into three categories [4], namely,

**Link-layer protocols:** There have been several proposals for reliable link-layer protocols, thereby attempting to hide the deficiencies of the wireless channel from TCP. Link-layer approaches include automatic repeat request (ARQ) schemes, that retransmit corrupted packets only over the wireless link, and forward error correction schemes. Different link layer schemes might be appropriate for different transport layer protocols (TCP and UDP) and application types. For example, in [30], a link-layer architecture is proposed, that enhances the performance of diverse applications over wireless (error prone) links. According to this work, diverse applications are best served by fundamentally different link layer schemes. Thus, they propose a multi service link layer architecture, that provides multiple Quality of Service points simultaneously, over wireless links.

Interestingly, [4] states that the performance of TCP with link-layer retransmissions can potentially be worse than in the absence of link-layer retransmissions, when the timeout of TCP and link-layer retransmissions is close, or when duplicate ACKs are transmitted by TCP receivers, even though the link layer eventually retransmits the lost packet; the latter is due to out-of-order delivery of packets over the wireless link. To avoid such negative interactions, TCP-aware approaches have been proposed, such as the SNOOP protocol [5], which combines the link-layer retransmission of corrupted packets and the suppression of duplicate ACKs.

TCP-unaware link layer protocols, such as TULIP [17] are not aware of the transport protocol. TULIP requires the network protocol (in this case IP) to indicate if a particular packet requests a reliable packet delivery service or not. TULIP attempts to prevent the TCP sender from receiving three duplicate ACKs by delivering only in-order incoming frames to IP.

**Split Connections:** Proposals in this category rely, mainly, on the existence of an AP (in the case of infrastructure networks) and note that since there are two completely different classes of subnetworks, wired and wireless, we could split each TCP connection into two connections at the point where the two subnetworks meet, i.e., at the AP. The AP keeps one TCP connection with the fixed host, while at the same time it uses another protocol designed for better performance over wireless links for the mobile host.

Proposals in this category, such as Indirect TCP (I-TCP) [3], came out very early. According to I-TCP, the AP is entitled to acknowledge the segments as soon as it receives them. However, this means that it is possible that the ACK of a particular segment arrives at the sender before the segment actually reaches the recipient. Obviously this violates TCP semantics.

**End-to-end approaches and new transport protocols:** TCP has been undergoing constant modifications and improvements, since its earliest days. In the context of the wireless networks, some new protocols have been proposed, however, these are in their infancy and have not yet been tested on a wide scale. The current trend is to leave the TCP end-to-end behavior unchanged, proposing schemes which consider the particular characteristics of wireless and exploit early congestion information given by effective metrics.

End-to-end approaches include proposals that can distinguish between losses due to packet corruption -in which case an explicit loss notification signal can be sent- and losses due to congestion. ECN also falls within this category, and has been proposed for wireless networks in [15, 18, 11]. For example, the work in [13] considers the general framework of congestion control schemes using a utility-based modelling approach. The authors propose a marking scheme which is a concave function of the traffic arriving at a link, when this rate is larger than some minimum capacity value. Moreover, this minimum capacity parameter is adjusted independently at each link, such that the incoming traffic rate is equal to some percentage of the total capacity. Our approach differs in that the marking probability is a linear function of the aggregate traffic flowing in both directions of the wireless link.

However, ECN alone, without any help from the link layer, cannot adequately address the issue of corrupted packets, since senders will still decrease their congestion window in response to packets lost due to corruption. Moreover, reducing TCP window only when ECN signals are received, poses danger in the case of high congestion, in which case ECN packets will also be lost [15, 18].

Our proposal goes one step further from addressing the issue of congestion and non-congestion related losses, and considers using ECN to convey congestion information from both the wired and wireless links. Furthermore, due to the resource sharing model of IEEE 802.11 WLANs, we propose an approach for marking packets based on the aggregate wireless link utilization and a procedure for adapting the marking algorithm to varying traffic and load conditions, in the case of single-hop networks. Other approaches consider using a RED-like mechanism [18] or some other level of congestion, such as a congestion price [2]; however, as we argue, a shared buffer does not exist in WLANs, hence such marking algorithms do not reflect the underlying resource sharing model.

For our work, we assumed that link-layer retransmissions, which are supported by the IEEE 802.11 MAC layer suppress packet loss due to corruption from the TCP layer. However, our proposal can be combined with more advanced link-layer proposals that support retransmission of corrupted packets and in-order packet delivery.

Next, we emphasize on two proposals falling into the last category of solutions, which implement ECN marking in wireless, and are closely related to ours, in the sense that they also trigger marking mechanisms based on early congestion exchange information.

The techniques described propose either dropping or marking schemes for multihop wireless networks based on measurements of the network load. Nevertheless, they suggest different metrics of the overload. The first work, named Link RED, investigates the wireless link drop behavior and notes that contention drops exhibit a load-sensitive feature. The second work expresses network load as the size of a virtual queue, of a node's neighborhood and proposes a marking mechanism called Neighborhood RED. Both studies propose RED-like mechanisms, based on the calculation of a drop/mark probability which increases, as network load also increases.

### 5.1.1 Link RED with adaptive pacing

The work in [10] studies the effect of multihop wireless links on TCP throughput and loss behavior. Through analysis and simulations the work reveals that given a specific network topology and flow patterns, there exists a TCP window size $W^*$, at which its throughput is highest, through improved spatial channel reuse. Further increasing the window size does not lead to further spatial channel reuse, but results in increased link layer contention and perceived packet losses. Secondly, the standard TCP does not operate around $W^*$, and typically grows its average window much larger. Consequently, TCP experiences throughput decrease, due to reduced spatial channel reuse. A more interesting aspect of this scheme is that the improvement of spatial reuse can lead to better fairness.

For this, the paper considers a packet drop probability scheme called Link RED, which the authors note can also be adjusted to a mark probability scheme. According to the approach, the packet drop probability, on each station, is a linearly increasing function of the number of MAC layer retransmissions, when these exceed some minimum number. The authors also propose an adaptive link-layer pacing scheme, to increase the spatial channel reuse. If a node needs to backoff, its backoff time increases by a time equal to the transmission time of the previous packet. The goal is to let TCP operate in the contention avoidance region.

#### Motivation of the work

The suboptimal throughput of TCP can be explained by its loss behavior over the multihop wireless channel. In a wired network, all incoming packets are dropped if buffer overflows at a bottleneck. It helps TCP to quickly reduce its window size to

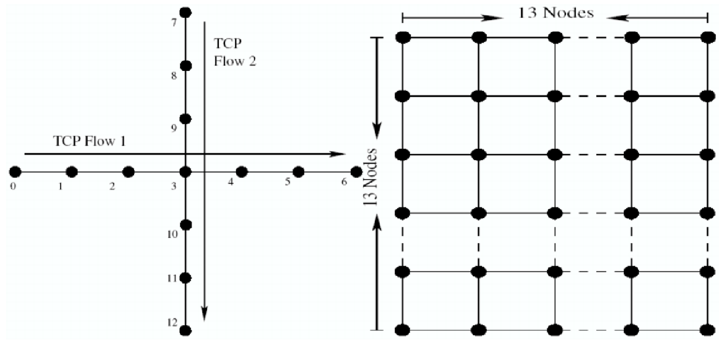



Figure 5.1: Left: cross topology with 13 nodes. 200 meter distance between two adjacent nodes with 2 TCP flows in each direction. Right: 13x13 grid topology, 200 meter distance between horizontal or vertical adjacent nodes.

release congestion. Multihop wireless networks exhibit different drop features. Unlike wired networks, where buffer overflow dominates packet losses, most packet drops experienced by TCP are due to link layer contention. Buffer overflow-induced packet loss is rare, and the contention-induced packet loss offers the first sign of network overload. The simulations show that contention drops exhibit a load-sensitive feature: as the offered TCP packets exceed $W*$ and increase further, link drop probability becomes non-negligible and increases accordingly. After the offered TCP packets exceed another threshold $W**$, the link drop probability saturates and flattens out. It turns out, however, that the link-layer drops are not significant, to stabilize the average TCP window size around $W*$. It, therefore, leads to suboptimal TCP throughput.

The scheme was verified using the NS-2 simulator, and showed to increase the performance in the case of multihop wireless networks, in terms of fairness and throughput. More precisely, for chain topologies of various number of hops, the throughput is increased up to 30%, compared to the default MAC layer retransmissions scheme. The fairness index is only investigated for the cross and grid topologies shown in Figure 5.1. For a cross 13-node topology, with 2 crossing flows, the fairness index raises from 0.5 to 0.9983 and the aggregate throughput of both flows, from 244 Kbps to 319 Kbps. For the grid topology, with four flows, the fairness improved from 0.51 to 0.95 and the aggregate throughput increased from 243 Kbps to 294 Kbps.

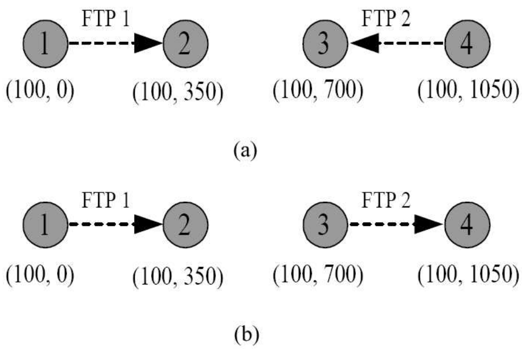



Figure 5.2: Neighborhood RED simulation topologies.

As we argue, the above study considers the network load to be represented by the retransmissions of a single node; however, in the case that a node has been subjected to severe contention, penalizing its outgoing packets may result in further unfairness.

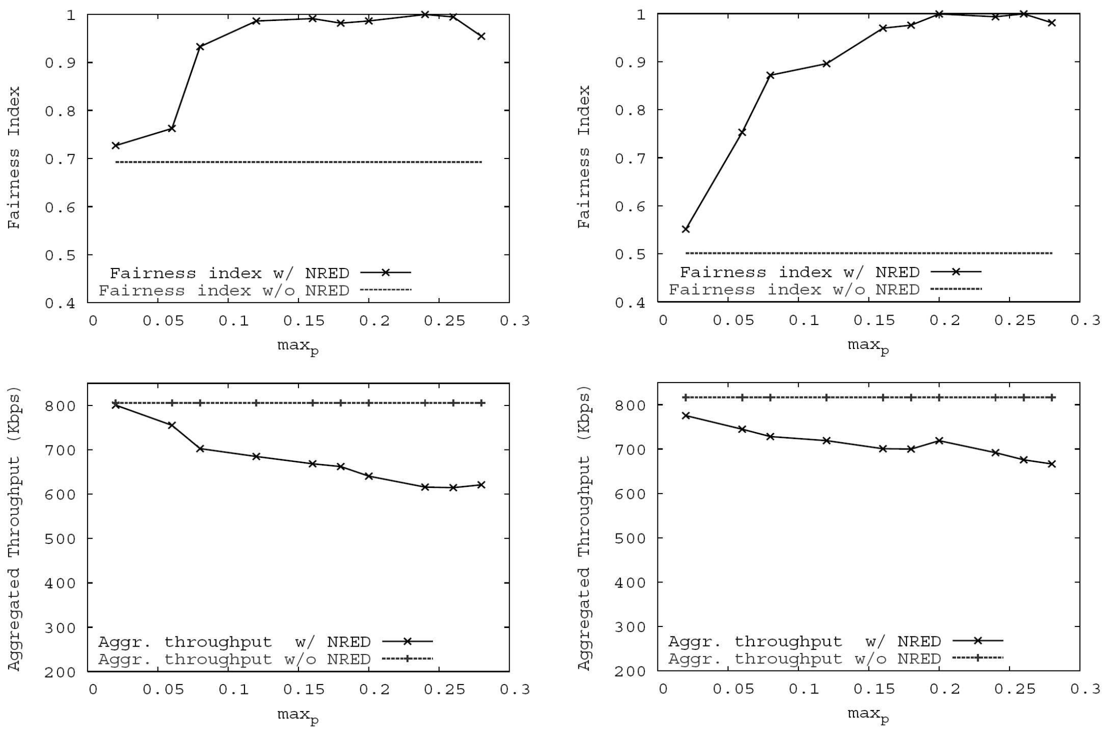
### 5.1.2 Neighborhood RED

The authors of this paper explore the relationship between TCP unfairness and early network congestion, extending a RED scheme for marking based on the aggregate (incoming and outgoing) queue size of a node's neighborhood. They argue that two unique features of ad hoc wireless networks are the key to understand unfair TCP behaviors. One is the spatial reuse constraint; the other is the location dependency. The underlying idea of this approach is that improvement of spatial reuse can lead to better fairness.

#### Motivation of the work

If we view a node and its interfering neighbors to form a neighborhood, the local queues at these nodes can be considered to form a distributed queue for this neighborhood. This distributed queue is not a FIFO queue. Flows sharing this queue have different and dynamic priorities determined by the topology and traffic patterns due to channel capture, or hidden and exposed terminal situations. Thus, they get different feedback in terms of packet loss rate and packet delay when congestion happens. The uneven feedback makes TCP congestion control diverge from the fair share.

Like the RED does, each node keeps estimating the size of its neighborhood queue.




(a) (b)

Figure 5.3: Results for Neighborhood RED, for different marking probabilities.

Once the queue size exceeds a certain threshold, a drop probability is computed by using the algorithm from the original RED scheme. Since a neighborhood queue is the aggregate of local queues at neighboring nodes, this drop probability is then propagated to neighboring nodes for cooperative packet drops. Each neighbor node computes its local drop probability based on its channel bandwidth usage and drops packets accordingly. The overall drop probability will realize the calculated drop probability on the whole neighborhood queue. Thus, the Neighborhood RED scheme is basically a distributed RED, suitable for ad hoc wireless networks.

The simulation platform used for the experiments was the QualNet simulator [24]. Experiments were conducted upon the topologies shown in Figure 5.2 and the results for the two scenarios are depicted in Figures 5.3(a) and 5.3(b). Unfortunately, fairness is achieved at the expense of decreased throughput. This is not the case with the approach proposed in this paper as our simulation results demonstrate.

Another observation upon the Neighborhood RED algorithm is that it involves a very complex set of steps that a node should follow, in order to compute its dropping probability.

## 5.2 Other AQM approaches

In this section we discuss some AQM algorithms, proposed to be implemented to wired networks. However, the ideas presented in this section have been also tuned in wireless networks. The main characteristic of these approaches is that the queue management policies adapt to certain network situations and improve the stability and performance of RED.

The work in [8], for example, is motivated by the observation that the RED queue length is very sensitive to the level of congestion and to the parameters' settings, and therefore is not predictable in advance. Consequently, the average queuing delay of RED is sensitive to the traffic load and not predictable. Take, for example, the case where a link is lightly congested and the maximum drop probability $\max_p$ is high, the average queue size is near $\min_{th}$; when the link is more heavily congested and the $\max_p$ is low, the average queue size is near $\max_{th}$. This work proposes to adaptively adjust the $\max_p$ over times scales greater than a round-trip time, in order to keep the average queue size within a target range half between $\min_{th}$ and $\max_{th}$. However, as the authors argue, the Adaptive RED algorithm has a tradeoff between throughput and delay.

Another adaptive AQM scheme, named Adaptive RIO [16], originates from the RIO proposal, and the above Adaptive RED. According to the first, when the buffer starts to build up, a RIO router always drops the *out* packets before the *in* ones. In this way, the *in* packets will receive a better service, than the *out* packets. The Adaptive RIO intends to achieve service differentiation between two classes of packets (*in*-*out*), stabilizing the queue occupation. The RED parameters are calculated differently for each class; $\max_p$ is adjusted with the above criteria, while $\min_{th}$ is calculated based on a target delay.

Other RED-like approaches follow different marking/drop probability curves i.e., exponential, such as Random Exponential Marking (REM) [2], that uses a different definition of congestion measure and a different marking probability curve. The approach tries to overcome the tradeoff between high utilization and negligible delay, decoupling the congestion measure from performance measure. Instead, the study considers the sum of some *link prices* along a path, as a measure of congestion in the path, and embeds it into the end-to-end exponential marking probability.

# Chapter 6

# Conclusions and issues for further research

WLANs will continue to be a major growth factor for communication networks in the up-coming years. The IEEE 802.11 standard sees increasing public deployment and, hence, it is important to ensure that different users gain fair access to the network resources. Nevertheless, the nature of the wireless medium cannot guarantee reliable data transfer, neither fair share of the resources. The resource sharing model of the underlying 802.11 MAC protocol requires stations to continuously contend to capture the channel, in order to transmit. In infrastructure networks the unfairness problem occurs between the uplink and the downlink traffic. In multihop wireless networks, the unfairness problem is more intense due to location dependent contention, which can result in different stations obtaining a different estimate of the level of congestion. Under these circumstances, some flows can increase their throughput, while others might starve.

In the present thesis we argue that we can control congestion within a wireless network by correctly accounting for the traffic load in the wireless area, and marking packets appropriately. The utilization estimation triggers an effective marking mechanism for conveying congestion level information to the end systems, using ECN for the end-to-end congestion signaling.

In the case of a single-hop BSS, where the wireless stations communicate through an AP, measurements of the aggregate traffic in wireless must be made on the AP taking into account both the uplink and the downlink traffic, since all traffic that originates from, or is destined to the wireless, travels through it. In the case of multihop networks, the bottleneck for a station is its collision domain, since only transmissions within this area can interfere with the transmissions of the station and, therefore, cause congestion. On the other hand, the two approaches are not different at all; in a BSS the wireless channel is a single collision domain. Only in this case, the existence of the AP provides the point of the network where the aggregate traffic can be measured while for a multihop network the utilization is given by the aggregate receiving rate within the node's collision domain.

In summary, the two key ideas employed by our work are the use of ECN as an end-to-end congestion signaling mechanism conveying congestion information from the wireless links and, second, the computation of the marking probability as a function of the aggregate utilization. So, we introduced an LBM algorithm, that correctly accounts for the aggregate utilization and decides on a packet marking probability.

The proposed approach was evaluated through NS-2 simulations. The simulations' results demonstrate that the proposed approach, for both types of networks, operating with TCP congestion control, achieves slightly higher fairness in the case of infrastructure networks, and noticeably higher fairness in multihop networks. On the other hand, in both network topologies, the achieved utilization remained the same, as with DT. Moreover, the approach yields smaller packet delay and delay jitter over the wireless. Also, for heterogeneous networks, containing both wired and wireless 802.11 links, we found that LBM can adapt to changing traffic and load conditions, to control the end-to-end delay perceived by the end users. For multihop networks, this delay is not very sensitive to small changes of the traffic conditions.

Ongoing work seeks to investigate different shapes of the marking probability curve (convex and concave, rather than piecewise-linear that we have consider up to now), and different measures of congestion of the wireless medium, such as the delay to access the wireless medium and the throughput measured in packets per time unit. It is also an interesting topic to examine how marking can be used to improve the aggregate throughput. An issue that is interesting for detailed study of the proposed scheme, is to decide on some optimum values for its parameters, under certain basic scenarios. Other interesting areas are the combination of the proposed approach with more advanced link-layer retransmission mechanisms. Finally, a related research area that can be investigates is service differentiation over IEEE 802.11 WLANs.

As far as the simulations are concerned, further experiments can be conducted with more complex topologies and traffic patterns. In the case of infrastructure networks, it would be interesting to consider simultaneous uplink and downlink traffic. For multihop networks, topologies that can be considered are cross and grid, perhaps with more than two flows. Such complex scenarios can guide the appropriate selection of the LBM parameters, and provide directions on how their value can be dynamically adjusted. Finally, another thought would be, for the LBM algorithm, to be implemented in a real multihop wireless network, in cooperation with a positioning algorithm.

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